3GPP TR 23.806 V7.0.0 (2005-12)

Technical Report

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Voice Call Continuity between CS and IMS Study
(Release 7)

GLOBAL SYSTEM FOR
MOBILE COMMUNICATIONS

Keywords

Voice, Circuit Switched, IMS, LAN, Radio

3GPP

Postal address

3GPP support office address

650 Route des Lucioles - Sophia Antipolis
Valbonne - FRANCE
Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Internet

http://www.3gpp.org

Contents

Foreword

This Technical Report 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.

Introduction

During the course of Release 6, TS 23.234 [2] (3GPP system to Wireless Local Area Network (WLAN) interworking: System description) was developed that provides the possibility to offer VoIP over WLAN interworking with IMS. Thus there is the possibility to support the most prevalent GSM service (voice calls) over I-WLAN when there is coverage. By developing the capability to support seamless voice call continuity between the CS Domain and an I-WLAN, or other IP-CANs, an operator would be able to provide relief to the GSM/UMTS radio resources and increase service revenue. In addition, wireline operators with VoIP offerings should be able to use the 3GPP IMS architecture to offer converged services. This TR documents alternatives for how to provide such seamless voice call continuity between the CS Domain and IP-CANs.

During the course of this study, a number of alternative architectural solutions that enable Voice Call Continuity between CS and IMS domains have been proposed and documented in this report. As a result of analysis, some options have been removed or placed in annexes of this document (thereby ensuring that the work is not completely lost). The outcome is that two alternatives remained: Original Domain Controlled (clause 6.4) and IMS Controlled Static Anchoring (clause 6.3).

Work has been undertaken to identify and separately document any common aspects of the two alternatives. In working on the study, it was determined that the VCC solution impacts Registration, Origination, Termination, Network Domain Selection as well as mid-call supplementary services. When these areas were deemed common or only had minor differences, they were placed in common sections. The common aspects have been captured in clauses 6.2 and 6.2a. These clauses retain some solution-specific details where necessary.

1 Scope

This document contains the results of the feasibility study into the architectural requirements and alternatives for the active voice call continuity between Circuit Switched (CS) domain and the IP Multimedia Subsystem (IMS). Considerations include overall requirements, architectural requirements, evaluation of potential architectural solutions and alternative architectures.

The Feasibility Study considers different solutions for offering real-time voice call continuity when users move between the GSM/UMTS CS Domain and the IP Connectivity Access Network (e.g., WLAN interworking) with home IMS functionality. Voice call related functionality, including the need for Regulatory issues (e.g. Text Telephone (TTY as defined in TS 26.226)), Emergency Call and support for supplementary services are taken into consideration.

The objective is to identify an architectural solution that allows completely automatic connectivity from the end-user point of view, while minimizing the additional complexity and impacts to the existing system. The feasibility study shall also investigate mechanisms for selecting the most appropriate network domain to serve the user.

Existing solutions developed by the 3GPP (e.g. 3GPP system to Wireless Local Area Network Interworking (I-WLAN)) should be reused as much as possible.

The impact to, and support of service continuity for sessions/calls established following the principles outlined in the combining of CS and IMS sessions (CSI) will also be considered in this study.

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 22.090: "Unstructured Supplementary Service Data (USSD); Stage 1".

[3] 3GPP TS 23.002: "Network Architecture".

[4] 3GPP TS 23.009: "Handover procedures".

[5] 3GPP TS 23.018: "Basic Call Handling; Technical realization".

[6] 3GPP TS 23.066: "Support of Mobile Number Portability (MNP)".

[7] 3GPP TS 23.218: "IP Multimedia (IM) session handling; IM call model".

[8] 3GPP TS 23.221: "Architectural requirements".

[9] 3GPP TS 23.228: "IP Multimedia Subsystem (IMS); Stage 2".

[10] 3GPP TS 23.234: "3GPP system to Wireless Local Area Network (WLAN) interworking".

[11] 3GPP TS 23.278: "Customised Applications for Mobile network Enhanced Logic (CAMEL) Phase 4; Stage 2; IM CN Interworking".

[12] 3GPP TS 23.279: "Combining Circuit Switched (CS) and IP Multimedia Subsystem (IMS) services".

[13] 3GPP TS 24.008: "Mobile radio interface Layer 3 specification; Core network protocols; Stage 3".

[14] 3GPP TS 24.087: "User-to-User Signalling (UUS) Supplementary Service – Stage 3".

[15] 3GPP TS 24.147: "Conferencing using the IP Multimedia (IM) Core Network (CN) subsystem; stage 3".

[16] 3GPP TS 26.226: "Cellular text telephone modem; General description".

[17] 3GPP TS 29.163: "Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks".

[18] 3GPP TS 33.102: "3G security; Security architecture".

[19] 3GPP TS 33.203: "Access security for IP-based services".

[20] 3GPP TS 33.234: "Wireless Local Area Network (WLAN) interworking security".

[21] DTS TISPAN-01002-NGN: "Requirements for PSTN/ISDN simulation services".

[22] RFC 3261: "SIP: Session Initiation Protocol".[23] 3GPP TR 23.899: " Combining CS Bearers with IMS ".

[24] 3GPP TR 29.010: " Information element mapping between Mobile Station - Base Station System (MS - BSS) and Base Station System".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the definitions given in TR 21.905 [1] and the following definitions apply:

Domain Transfer: Transfer of a call or session from the CS to IMS domain, or visa versa.

Native network: The network where all incoming calls are delivered. All supplementary services are handled completely in the native network.

Foreign network: The network where the calls are NOT delivered - to be only used to deliver a "simple" call to the subscriber. The foreign network is used to extend a basic point-to-point call to the subscriber, and offer no service other than connectivity, that is, the foreign network does not offer any supplementary services.

3.2 Symbols

For the purposes of the present document, the symbols given in TR 21.905 [1] apply.

3.3 Abbreviations

For the purposes of the present document, the abbreviations given in TR 21.905 [1] and the following abbreviations apply:

B2BUA Back to Back User Agent
BGCF Breakout Gateway Control Function

CAMEL	Customised Application for Mobile network Enhanced Logic
CAP	CAMEL Application Part
CCCF	Call Continuity Control Function
CLIP	Calling Line Identification Presentation
COLP	COnnected Line identification Presentation
CReDT	Call Reestablishment on Domain Transfer
CS	Circuit Switched
CSCF	Call Session Control Function
CSI	Combining of CS and IMS session
DACCI	Dynamic Anchoring of CS Calls in IMS
DTAP	Direct Transfer Application Part
ECT	Explicit Call Transfer
ENUM	E.164 Number
GMSC	Gateway MSC
HO	Handover
I-CSCF	Interrogating-CSCF
IETF	Internet Engineering Task Force
IMPU	IP Multimedia PUblic identity
IMS	IP Multimedia Subsystem
IMSI	International Mobile Subscriber Identity
IP-CAN	IP-Connectivity Access Network
ISUP	ISDN User Part
I-WLAN	Interworking WLAN
MAP	Mobile Application Part
MEP	Mobility Event Package
MGCF	Media Gateway Control Function
MSC	Mobile Switching Centre
NeDS	Network Domain Selection
PS	Packet Switched
PLMN	Public Land Mobile Network
PSI	Public Service Identity
PSM	Protocol State Machine
PSTN	Public Switched Telephone Network
QoS	Quality of Service
S-CSCF	Serving-CSCF
UE	User Equipment
UUS	User to User Signalling
USSD	Unstructured Supplementary Service Data
VCC	Voice Call Continuity
VCC-SRF	VCC Signally Relay Function
VoIP	Voice Over IP
WLAN	Wireless Local Area Network

4 Overall Requirements

- The study shall identify the impacts to the current 3GPP specifications to support real-time voice continuity when moving between the GSM/UMTS CS Domain and IMS domain using an IP Connectivity Access Network (e.g. 3GPP IP access over I-WLAN and PS domain).

- The study does not introduce new requirements for ISIM and USIM.

- The study should minimize impacts on existing 3GPP specifications.

- The study shall not require changes to radio systems (e.g., UTRAN/GERAN or 802.xx, etc.).

5 Architectural Requirements and Considerations

5.1 Basic Assumptions

- Although the scope is mainly targeted at CS services over a UTRAN, GERAN access and IP multimedia services over a IP Connectivity Access Network with WLAN, the solution is (at least technically) assumed to be applicable to IP Multi-media services over GERAN/UTRAN, and should not be dependent on any functionality from the IP Connectivity Access Network.

Editor Note: The IP CAN for IMS access and the IMS Core Network may belong to separate services providers.

- The selection of access network should allow automatic connectivity from the end-user's point of view.
- UEs that do not support the functionality described in this TR will not be impacted.
- The radio layer protocols for xRAN, NAS in TS 24.008 [13], and PS core shall not be impacted.
- CS core impacts shall be minimized. Changes should be restricted to the IMS elements and the UEs that support IP Connectivity Access Network.
- Protocols connecting the IMS to the CS domain, to the PSTN and to other SIP networks, including other IMS networks should remain unchanged.
- The existing CS security aspects, IP Connectivity Access Network security aspects and IMS security aspects defined by 3GPP specifications (TS 33.203 [19], TS 33.234 [20]) shall be reused.
- The UE may be capable of transmitting and receiving simultaneously in the CS domain on GERAN/UTRAN, and on the IP Connectivity Access Network. However, in cases where the simultaneous transmit and receive on both the IMS and CS legs are not possible, e.g., due to restrictions specific to cellular access or due to high interference level in the receiver of one RAT caused by the transmitter of the other RAT, etc; the domain transfer may require different mechanisms to those described for the solution that supports simultaneous reception and transmission.

  The call continuity solution shall not rely on radio layer capabilities that are not supported in current 3GPP specifications as listed below: (note: the "Handovers" term is according to the definition defined in TR 21.905 [1]).

  - Handovers between 3G CS and 3G PS domains are not supported.
  - Handovers between 2G CS and 2G PS domains are not supported.
  - Handovers between 3G CS and 2G PS domains are not supported.
  - Handovers between 2G CS and 3G PS domains are not supported.
  - Simultaneous transmit and receive of 2G and 3G radio is not supported.
- The UE can be registered in either CS or IMS domains or both domains.
- The user can be reached via the same identity (i.e. MSISDN) in both IMS and GSM/UMTS CS Network. This may be on either the same device, or on different devices.
- CS services will be available when the CS domain is being utilized, and IMS services will be available when the IMS domain is being utilized. Service delivery during voice call continuity procedure will be provided across domains, subject to the constraints of each domain.
- It shall be possible to support Emergency call call continuity between domains if the target domain supports it. Call continuity procedures shall only be performed for Emergency calls if radio coverage is lost in the current domain.
- Use of available QoS mechanisms need to be considered, however, the Impact on the QoS mechanism is out of the scope.

5.2 Architectural Requirements

- It shall provide voice call continuity when the user is moving between GSM/UMTS CS Domain and IMS, even in the case that the VMSC is not in the HPLMN.
- It shall be possible to perform correlation of charging that is performed in GSM/UMTS CS Domain and for the IMS session when service continuity between the domains is performed.
  This shall ensure consistent end-user charging.
- While not in CS or IMS voice call, the UE shall be able to detect and automatically connect to the available access Network (such as GSM/UMTS radio and/or IP Connectivity Access Network). UE shall select either CS domain or IMS domain for real-time voice service based, e.g., on operator policy for and user's preference subject to service consistent constraints, e.g., if the UE have an on going call in one domain, the UE should use the same domain for additional call.
- When the UE is attached to both the CS and IMS domains, the network has the responsibility for selecting the terminating service domain for an incoming call, depending on operator policy and possibly user preference subject to service consistent constraints, e.g., if the UE have an on going call in one domain, the same domain should be used to terminating the additional incoming call.
- The architectural solution shall support a mechanism for selecting how to route the terminating voice to the UE; since it is possible for multiple devices to be registered to the IMS, terminating handling should allow for routing to multiple devices; including the CS device.
- It shall be possible for a user to be reached via the same identity (i.e., MSISDN) in both IMS and GSM/UMTS CS Network and the use of additional IMS public identifies such as Tel URI and SIP URL shall not be precluded.
- It shall be possible for UEs connected to the IMS to initiate or receive IMS session requests while a CS voice call is ongoing to a UE with the related MSISDN.
- It shall be possible for a UE to initiate/receive CS voice calls while a UE using a related Public User ID has IMS session(s) is ongoing.
- Voice call continuity should be provided such that from the end user's perspective minimal service disruption is perceived. Voice call continuity procedure latency should be minimized.
- In a CS voice call (respectively Voice call supported over the IP Connectivity Access), the UE shall be able to monitor IP Connectivity Access (respectively GERAN/UTRAN cells) for the purpose of radio mobility.
- User preferences and operator preferences shall be taking into account when making decision for requesting a CS to IMS or IMS to CS transition for an ongoing voice call.
- Initiation of the CS to IMS call continuity procedure for an on going voice call may be based on radio condition; initiation of the IMS to CS call continuity procedure for an on going voice call may be based on radio condition and IP connectivity quality to IMS domain.
- Signalling load as a result of voice call continuity should be minimized.
- Voice call quality should be maintained. The number of transcoding stages introduced by the architecture should be considered.
- Existing services not candidate for continuity provided in the CS or PS domain (e.g. LCS, SMS) shall continue to work if CS or PS attached.
- The solution shall be able to work for the radio assumptions described in section 5.1. That is, the solution shall support both a UE that is capable of simultaneous communication with the IMS leg and the CS leg; and a UE that is not capable of simultaneous communication with the IMS leg and the CS leg. The capabilities of a UE for simultaneous radio layer communication on both the CS leg and the IMS leg may be dependant upon the IP-CAN utilised for the IMS leg.

5.2.1 Operator Control Requirements

Editor's Note: NSP work from SA1 should be taken into account. The terminologies, definitions and requirements here should not be conflicted with NSP specifications.

5.2.1.1 Classification of Operator Control

Operator control is classified into two kinds as follows:

1) Pre-defined control.

   Pre-defined control is that criterion is configured before a user attempts to select access systems between CS Domain and IPCANs for voice services. For example, pre-defined control criterion be downloaded and updated over air interfaces. Pre-defined control can be permanent or temporary. Permanent pre-defined control are always in effect. On the other hand, temporary pre-defined control only takes effect for a limit time period, e.g. holidays, festivals or busy time and overrides the permanent pre-defined control.

2) Real-time control.

   Real-time control is that an operator controls the UE's selection of access systems between CS Domain and IPCANs dynamically according to network conditions or other aspects based on operators' policies. This real-time control overrides any pre-defined control.

5.2.1.2 Requirements of Operator Control

Editor's Note: Detail requirements of operator control for selecting access systems between CS Domain and IPCANs that should be supported is TBD.

5.3 Session Scenarios

5.3.1 Overview

To guide the design of solutions and determine their feasibility the following scenarios or subset of these scenarios shall be used to evaluate architecture alternatives.

5.3.2 Two party UE to PSTN calls

1) UE(A) is in a stable voice call to PSTN User B via GSM/UMTS CS Domain. After voice call continuity procedures are completed, UE(A) is in a stable voice call to PSTN User B via IMS Domain.

2) UE(A) is in a stable voice call to PSTN User B through IMS via IP-CAN. After voice call continuity procedures are completed, UE(A) is in a stable voice call to PSTN User B via GSM/UMTS CS Domain.

3) Voice call continuity from IMS via IP-CAN when UE(A) moves back to GSM/UMTS CS Domain.

4) Voice call continuity from GSM/UMTS CS Domain when UE(A) moves back to IMS via IP-CAN.

5.3.3 Two party UE(A) to UE(B) calls

1) UE(A) is in a stable voice call to UE(B) through GSM/UMTS CS domain. After voice call continuity procedures are completed, UE(A) is in a stable voice call to UE(B) via IMS Domain.

2) UE(A) is in a stable voice call to UE(B) through IMS via IP-CAN (all IP call). After voice call continuity procedures are completed, UE(A) is in a stable voice call to UE(B) via GSM/UMTS CS Domain.

3) Voice call continuity from IMS via IP-CAN when UE(A) moves back to GSM/UMTS CS Domain.

4) Voice call continuity from GSM/UMTS CS Domain when UE(A) moves back to IMS via IP-CAN.

5.3.4 Supplementary services are active when call continuity procedures occur

1) GSM/UMTS CS domain 2 way call on-hold by UE(A) when voice call continuity procedures are initiated to IMS via IP-CAN. After the voice call continuity procedures are completed, the other party remains on hold and UE(A) can remove the call hold when requested by the user.

2) IMS via IP-CAN 2 way call on-hold (UE(A) owner) when voice call continuity procedures are initiated to GSM/UMTS CS domain. After the voice call continuity procedures are completed, the other party remains on hold and UE(A) can remove the call hold when requested by the user.

3) GSM/UMTS CS domain 3 way call active (UE(A) owner) when voice call continuity procedures are initiated to IMS via IP-CAN. After the voice call continuity procedures are completed, UE(A) is still the active owner of the 3 way call and standard 3 way call control rules and procedures will be followed (e.g., UE(A) can drop the last added party).

4) IMS via IP-CAN 3 way call active (UE(A) owner) when voice call continuity procedures are initiated to GSM/UMTS CS domain. After the voice call continuity procedures are completed, UE(A) is still the active owner of the 3 way call and standard 3 way call control rules and procedures will be followed (e.g., UE(A) can drop the last added party).

5) GSM 2 way call with call-waiting active (UE(A) owner) when voice call continuity procedures are initiated to IMS via IP-CAN. After the voice call continuity procedures are completed, the other party is still in call waiting mode and UE(A) can perform standard call waiting actions (e.g. toggle between calls).

6) IMS via IP-CAN 2 way call with call-waiting active (UE(A) owner) when voice call continuity procedures are initiated to GSM/UMTS CS domain. After the voice call continuity procedures are completed, the other party is still in call waiting mode and UE(A) can perform standard call waiting actions (e.g. toggle between calls).

5.3.5 Supplementary Services are Activated After Call Continuity Procedures Have Completed

1) After GSM/UMTS CS domain to IMS via IP-CAN voice call continuity procedures have completed, UE(A) performs a subsequent add 3rd party (3 way call) or call hold.

2) After IMS via IP-CAN to GSM/UMTS CS domain voice call continuity procedures have completed, UE(A) performs a subsequent add 3rd party (3 way call) or call hold.

3) After GSM/UMTS CS domain to IMS via IP-CAN voice call continuity procedures have completed, a subsequent incoming call to UE(A) invokes call waiting.

4) After IMS via IP-CAN to GSM/UMTS CS domain voice call continuity procedures have completed, a subsequent incoming call to UE(A) invokes call waiting.

5.4 Traffic Scenarios

The following traffic scenarios are considered in the study when evaluating the different proposed solutions.

- **Mostly CS network traffic**
  This is an traffic scenario where the network supports predominantly CS traffic, and is in the phase of introducing IMS capabilities into the network.

- **Mostly IMS network traffic**
  This is an traffic scenario where the network supports predominately IMS based traffic.

- **Mixed CS – IMS network traffic**
  This is a traffic scenario where the network supports a roughly even mix of IMS and CS based traffic.

Consideration is provided for the migration in traffic growth.

6 Architecture Alternatives

6.1 General

This clause documents the set of proposed solution architectures.

6.2 Architecture Reference Model

NOTE 1: This section illustrates the reference model to support service continuity.

NOTE 2: For VCC TR, CCCF/NeDS is depicted as one logical functional entity and be label as CCCF/NeDS in all relevant diagrams and procedures.

Two logical functions Call Continuity Control Function (CCCF) and Network Domain Selection (NeDS), described below, are added to the architecture. Where appropriate, the architecture diagrams and call flows may show a CCCF/NeDS function which represents a combined CCCF and NeDS function.

6.2.1 Call Continuity Control Function (CCCF)

CCCF provides functions for call continuity between the GSM/UMTS Circuit Switch domain and IMS domain using an IP Connectivity Access Network.

- The CCCF is a logical functional entity, which must exist for each voice continuity call.

The CCCF provides the following functions for call continuity:

- Reception and processing of call continuity requests caused by radio related events, e.g. availability or loss of radio coverage.
- Establishment, catenation and release of call legs needed to transfer a voice call from CS domain to IMS domain, or visa versa.

Further specifics of the CCCF functionality and interfaces with the IMS and CS domains are for FFS and should be included in the various architecture alternatives.

6.2.2 Network Domain Selection (NeDS)

6.2.2.1 Description of Network Domain Selection Functionality

NeDS function is the control point for selecting which domain to use for terminating a call.

Normally it may be expected that a CS terminating call will terminate on the CS side of a multi-mode terminal, and an IMS terminating call will terminate on the IMS side of a multi-mode terminal, there are situations where the selection of the other domain is appropriate (e.g. in the case of a CS terminating call when the terminal is not CS-attached, but is IMS registered). In addition to technical considerations, user preferences and service availability considerations may need to be considered and are implemented in the NeDS function.

The decision of the domain in which to terminate the call could be generalised as the "Terminating network domain selection" (Terminating NeDS).functionality. Note that NDS implies Network Domain Security, so NeDS has been suggested as an alternative abbreviation.

Below are some of the factors which could influence the Terminating Network Domain Selection.

- Registration status (CS attached; IMS registered (for multimedia telephony), or both);
- IPCAN capabilities (in case of IMS registered);
- Service/subscription/operator preferences.

Domain selection shall be in accordance to TS 23.221 [8], clause 7.2. In order to generalise the discussion and understand the requirements the following is a general approach to the problem.

The Network Domain Selection (NeDS) function can be characterised as follows:

- The NeDS function is aware of whether the terminal is registered on IMS from a device that is Multimedia telephony (with IMS voice) capable, and on an access that is capable to support IMS voice.
- The NeDS function is aware of whether the terminal is attached to the CS domain.
- The NeDS function is aware of or can obtain the ongoing voice call in the IMS and the CS domain.

The NeDS function controls the decision as to the appropriate terminating domain, taking into account the operator, user and service preferences.

Editor Note: The splitting of the NeDS from the CCCF may be reconsidered, when the TR conclusion on a solution is reached.

6.2.2.2 Routing Selection Decision

This section describes decision logic for the NeDS function. Different logic may be used depending on which domain triggers this function.

Examples of the decision logic are shown in Annex B.

6.2.3 Domain selection for originating call

The UE can be registered to both domains and may choose to originate a voice call (or session) via either CS or IMS domain. The following outlines principles on UE origination.

1. If the UE is idle, and subject to operator and user preferences and radio condition, the UE may select either domain to originate a voice call.
2. If the UE has an ongoing CS domain voice call, and the user wants to originate an additional voice call, then the UE should use normal CS domain procedure for that additional voice call.
3. If the UE has an ongoing IMS domain voice session, and the user wants to originate an additional voice call, then the UE should use the IMS domain for the additional voice call.

6.2a Common procedures

This clause in general documents procedures that are common to all of the architectural alternatives documented in clause 6. Where a particular architectural alternative requires procedures that are departures from commonality those are also documented in this clause.

6.2a.1 Registration

6.2a.1.1 CS Domain Registration

The normal location update procedure (ref. TS 24.008 [13]) is used by the UE to register to the CS Domain. If the UE is also registered to the IMS domain, the UE indicates to the CCCF/NeDS Function about its CS status (i.e. IMSI attach state).

6.2a.1.2 IMS Domain Registration

6.2a.1.2.1 IMS registration with CCCF/NeDS

Registration with the CCCF/NeDS follows the same procedure as defined in TS 23.228 [9] for Application Server (AS). The filter criteria contains a condition that a 3rd party registration should be performed via the ISC interface.

After registration is performed, UE indicates to CCCF/NeDS function of its current CS status (detach/attached-idle/attached-active).

IMS registration is done independently of the UE's CS state.

6.2a.1.3 Information exchange between UE and CCCF/NeDS

When UE is registered to the IMS Domain, the following information is exchanged between CCCF/NeDS and UE.

Collect certain CS Domain related information from the UE: This includes CS status = detached, attached-idle/attached-active.

Communicate the CCCF PSI and CS number associated with the CCCF to the UE.

Communicate Operator policy and user's preference between the UE and CCCF/NeDS.

6.2a.1.3.1 Use of Mobility Event Package Subscription

NOTE: The term "Mobility Event Package or MEP" is used throughout this TR to describe the process where information are exchanged between UE and CCCF/NeDS. Another terminology e.g, Mobility Event Information, maybe adopted for this TR.

Mobility Event Package is used to by the CCCF/NeDS and UE to update each other the needed information.

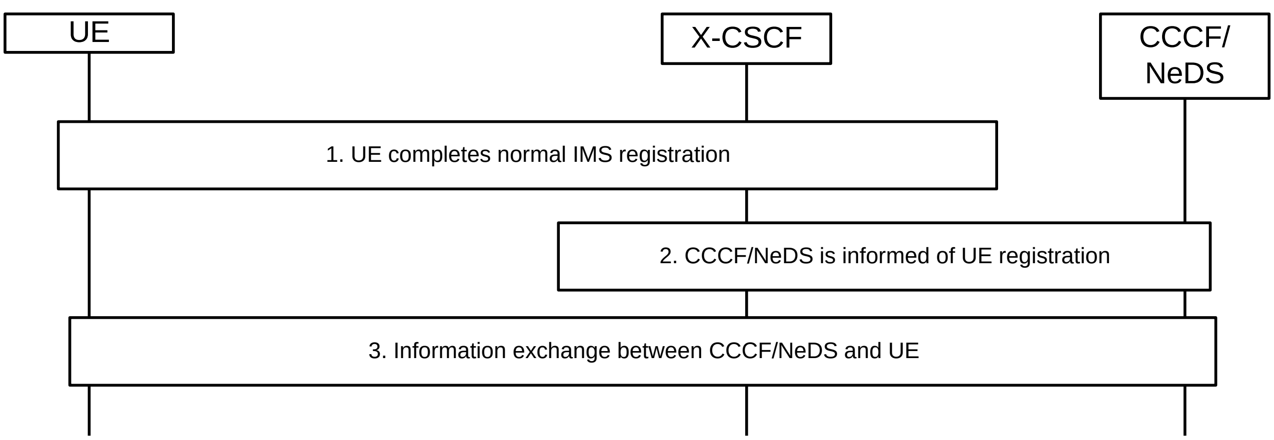


Figure 6.2a.1.3-1: Mobility Event Package call flow

1. UE is completing the IMS registration.

2. CCCF/NeDS function is informed of UE registration.

3. CCCF/NeDS and UE subscribe to each other's mobility event package. As part of the subscription, the UE indicates its CS status (ie., idle, active, detached, attached) and can also include user's preference to the CCCF/NeDS. The CCCF/NeDS can also download the CCCF PSI, CS number associated with the CCCF and operator's policy to the UE.

NOTE: Mechanism/protocols for information exchange needs to taken into account the possibility that there could be multiple IMS devices registered to IMS using the same IMPU (IP multimedia Public User Identity). However, some of the devices may not be VCC capable. The CCCF/NeDS will need to be able to distinguish the VCC device.

6.2a.2 Origination

6.2a.2.1 UE Origination from IMS Domain

This origination procedure applies to a user who is originating a voice session via their home IMS.

The CCCF/NeDS aspects of this procedure also apply when a user connects to their home IMS network via a 3GPP Visited Network.

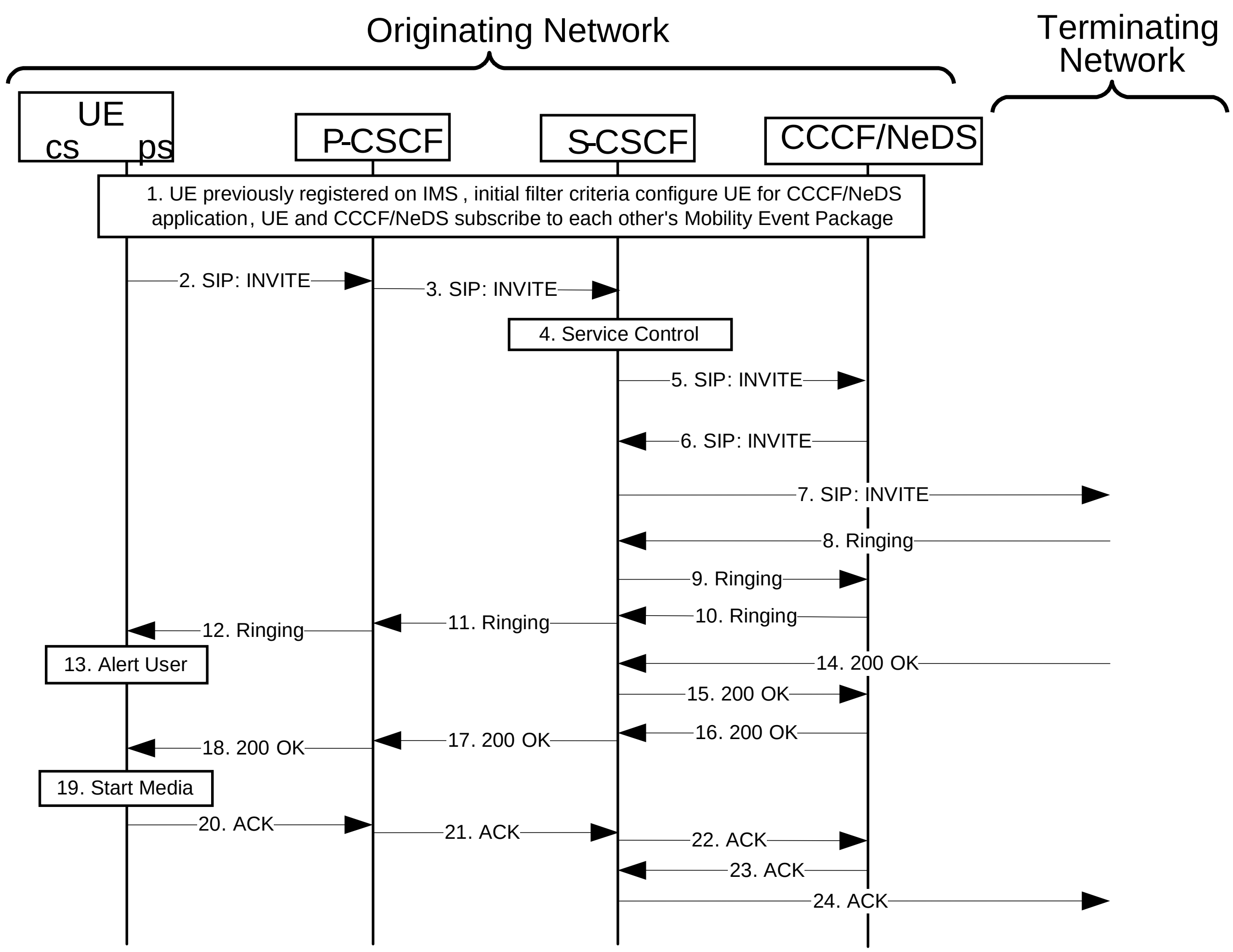


Figure 6.2a.2.1-1: UE Origination from Home IMS

Procedure Mobile Origination from home IMS is as follows:

1. UE previously registered on IMS. The UE's HSS profile includes CCCF/NeDS service. As part of CCCF/NeDS service, the UE and CCCF/NeDS subscribe to each other's Mobility Event Package.

2. UE sends the SIP INVITE request, containing an initial SDP, to the P-CSCF determined via the P-CSCF discovery mechanism. The initial SDP may represent one or more media for a multi-media session.

3. P-CSCF generates and stores an Authorization-Token. The P-CSCF remembers (from the registration procedure) the next hop CSCF for this UE. In this case it forwards the INVITE to the S-CSCF in the home network.

4. S-CSCF validates the service profile, and invokes any appropriate service logic required for this user. This includes authorization of the requested SDP based on the user's subscription for multi-media services.

5. S-CSCF forwards the request to the CCCF/NeDS according to the service origination logic defined by initial filter criteria in the UE profile. The CCCF/NeDS shall assign a call reference to uniquely identify the session.

NOTE: When and how to delivery the call reference to the UE and how to correlate the call/session in the UE is FFS.

6. CCCF/NeDS notes the call event, "record-routes" to stay in the signalling path and then forwards the request to the S-CSCF for termination handling.

7. S-CSCF forwards the request, as specified by the S-S "Serving to Serving" procedures as defined in TS 23.228 [9].

8-24. Standard IMS call progress as defined in TS 23.228 [9].

6.2a.2.2 UE Origination from CS Domain – Original Domain Control Model

There is no impact in the CS domain for the UE to originate a CS based voice call for the Original Domain Control.

If the UE is also IMS registered, the UE signals to the CCCF/NeDS (via Mobility Event Package) that it has transitioned to active mode in CS domain.

6.2a.2.3 UE Origination from CS Domain – IMS Controlled Model

6.2a.2.3.1 General

Special techniques are established at the UE and the Visited MSC in areas with overlapping coverage and with borders between the two domains so that CS originations for VCC users are routed via CCCF in user's home IMS network. The CS originating calls are routed to CCCF for enablement of a Routing B2BUA 3pcc function to control the bearer path of the call. The original called number along with other information required to complete the call is passed to CCCF so that it can originate a call to the remote party on behalf of the user.

This routing function is realized by using CAMEL to steer the calls made by VCC subscribers via the user's home IMS network. The UE establishes the call using standard call origination procedures; CAMEL origination triggers at the VMSC invoke signalling to the gsmSCF function of CCCF/NeDS. If the UE is in a location where VCC is possible and desirable, CCCF/NeDS provides a CCCF PSI to the VMSC to reroute the call toward the CCCF function in IMS. The CCCF uses the information received in the CAMEL query to originate a call to the remote party on behalf of the user. If the UE is in a location where VCC is not possible and/or desirable, the gsmSCF directs the VMSC to continue with normal call origination procedures.

NOTE: The solution may be enhanced by adding a capability in the UE and the network that allows the UE to place a call to the CCCF using a CCCF PSI when use of CAMEL in the visited network is not possible. With this method, a VCC information exchange is performed upon initial dialogue between the UE and the CS network, i.e. IMSI attach, and subsequently whenever there is a change in the information that was exchanged, e.g. when registering with a visited network with a different PLMN-id. The CCCF/NeDS uses the ATSI to interact with the HSS to determine CAMEL availability in the visited CS network. The CCCF/NeDS directs the UE to place a call to the CCCF/NeDS by using a CCCF PSI. The UE provides the information needed at the CCCF/NeDS (e.g. the original called party number and call reference) to originate a call to the remote party on behalf of the user using VCC Information Exchange or via existing called party number information element in the Call Control Setup message.

The preferred UE origination mechanism may be controlled as a network option and is communicated to the UE via the VCC information exchange as described above.

The CCCF PSI is dynamically assigned to the user upon registration with the network to allow for a scalable solution that uses geographical considerations for assignment of CCCF for the user.

Editor's note: Clarification required for emergency calls impact for the case when the user dials an emergency call number but the terminal treats this number as a normal B-party number when CAMEL is not used in the visited network.

6.2a.2.3.2 CS Origination static anchoring via origination triggers at VMSC; call walk-through

Figure 6.2a.2.3-1 describes how signalling and bearer paths are established for originations from CS-IMS users at Visited MSCs in the areas of overlapping coverage or border between the CS and PS domains.

Editor's note: It is FFS how CAMEL triggered CS originated VCC calls will work in conjunction with Dialled CAMEL services such a Number Translation.

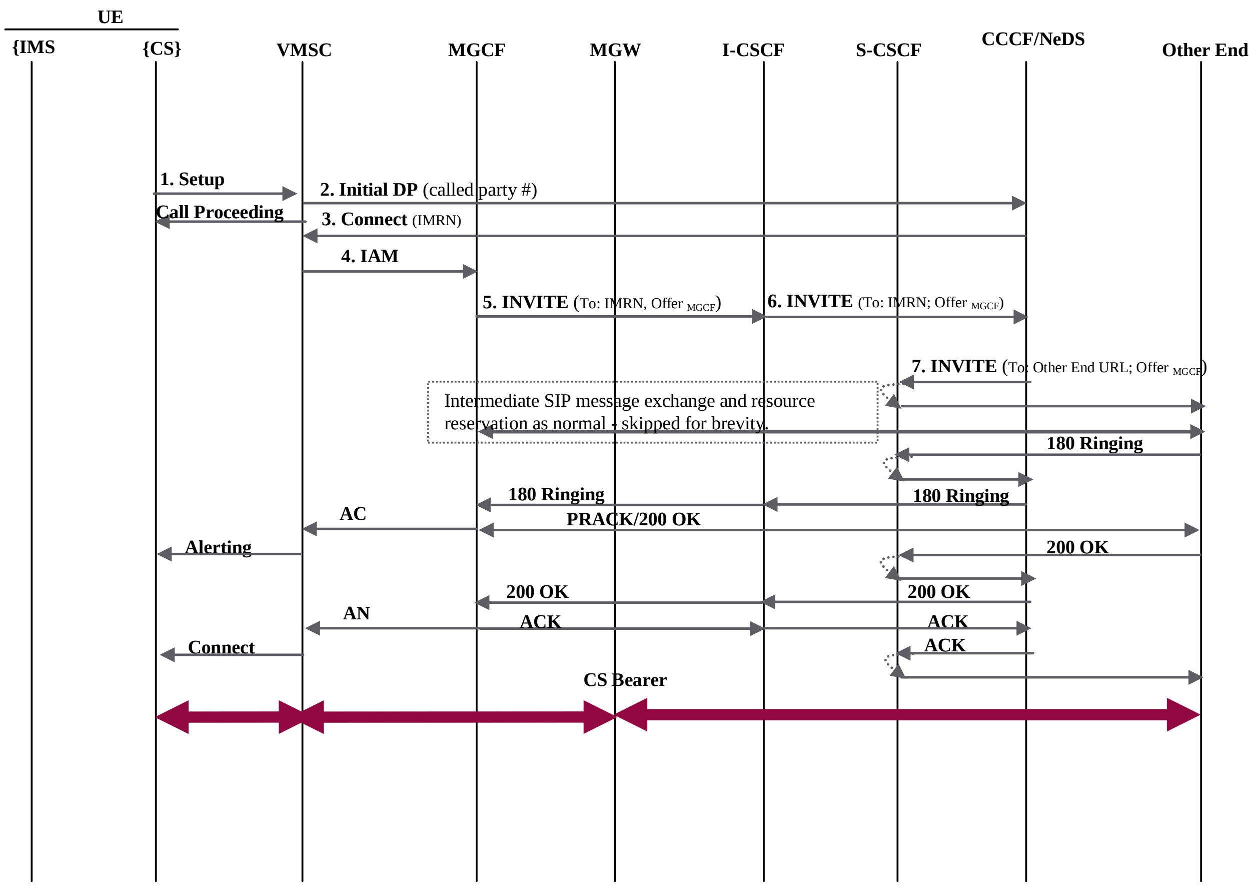


Figure 6.2a.2.3-1: CS Origination Anchoring via Origination Triggers walk-through

1. The user originates a call after registering with the VMSC.
2. Origination triggers at the VMSC result in invocation of CS origination static anchoring service at the CCCF/NeDS via a CAMEL Initial DP message toward the gsmSCF function of CCCF/NeDS.
3. The CCCF/NeDS creates an IMRN (IP Multimedia Routing Number) by appending Call Reference digits to the CCCF PSI and passes it to VMSC via the gsmSCF function in CAMEL Connect message for routing of the CS origination to CCCF function in IMS. NeDS also updates the CCCF with the respective Call Reference, the called party number, the called party sub address and any other information required by CCCF to complete the call toward the called party on behalf of the user.
4. The VMSC routes the call toward the MGCF in user's home IMS network using the IMRN.
5. The MGCF initiates an INVITE by setting the Request-URI to Tel URI format using the IMRN received in called party number in incoming IAM and sends it to the I-CSCF.
6. I-CSCF performs location query to the HSS to retrieve application server name associated with the IMRN. The Application Server hosting the CCCF is configured in the HSS against a range of PSIs wild carded to the CCCF PSI.
7. CCCF terminates the incoming leg addressed to CCCF PSI and invokes a Routing B2BUA function to originate a session to the original called party destination on behalf of the user. CCCF retrieves the S-CSCF address from the HSS over the Sh interface to route the originating session to the original called destination.

NOTE: HSS returns the address of the S-CSCF assigned to the user upon IMS registration if the user is currently registered in IMS or the S-CSCF capabilities required for S-CSCF selection at CCCF if the user is not registered in IMS as described in (AS-O) Origination at Application Server procedure described in TS 23.228 [9]; the latter requires AS behavior enhancements to enable AS to initiate calls on behalf of an unregistered user.

An ENUM dip for original called party number to SIP URI translations is required at the S-CSCF for routing to the original called destination. An IMS termination is assumed for this call walk-through. The BGCF and MGCF functions are involved in setting up of the terminating leg when terminating to the PSTN or CS Domain. CCCF maintains session states for the originating and terminating legs of the call via a third party call control (3pcc) function in order to control bearer upon Handover requests from the UE.

6.2a.2.3.3 CS Origination static anchoring using CCCF PSI

This section provides details of CS origination static anchoring technique using CCCF PSI for direct routing to the CCCF function in IMS. Three options are presented for transport of VCC information required by the CCCF to complete the other half of the call:

1. Use of USSD;
2. Use of SIP;
3. Use of called party number in Call Control Setup message.

6.2a.2.3.3.1 CS Origination static anchoring via USSD; call walk-through

Figure 6.2a.2.3-2 describes how signalling and bearer paths are established for originations from VCC users at Visited MSCs in the areas of overlapping coverage or border between the CS and PS domains. The solution assumes CS anchoring decision criteria logic at the UE and uses a USSD operation in application mode to invoke a CS Origination static anchoring service at CCCF.

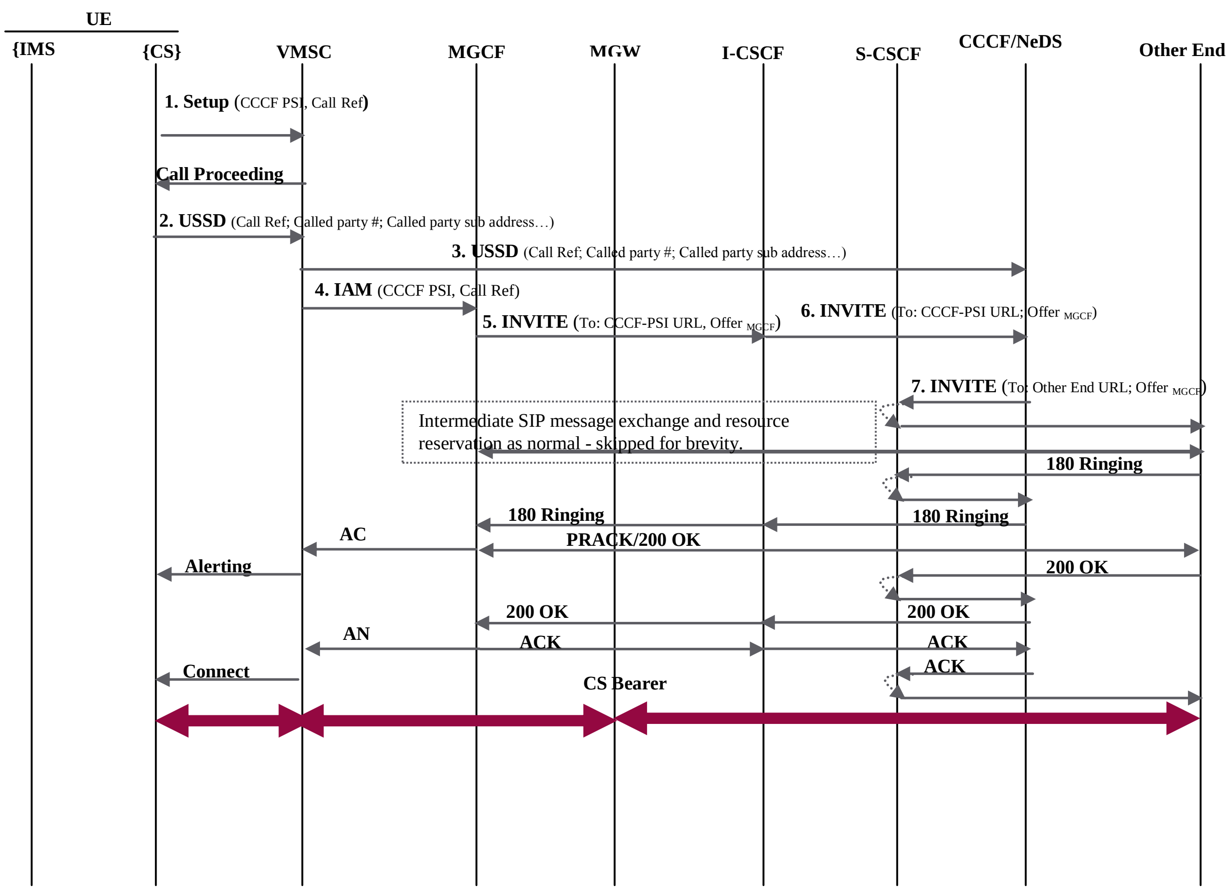


Figure 6.2a.2.3-2: CS Origination Anchored via USSD walk-through

1. When the user dials the desired destination number, the terminal originates a call to an E.164 number associated with CCCF PSI communicating the call reference of the call being established. The call reference uniquely identifies the user's session at the UE and at CCCF. The configuration of call reference in the UE and CCCF may be modelled on configuration and management of DTAP CC Transaction ID in the UE and the network respectively. The call reference may be communicated in called party address as digits appended to the CCCF PSI or in called party sub address.

2. Upon receipt of Call Proceeding from the MSC, the UE initiates a USSD operation in an application mode, requesting a CS origination static anchoring service at the CCCF. The called party number, the called party sub address and any other information required to complete the call toward the called party is sent in the USSD message.

3. The Visited MSC forwards the USSD message to the subscriber's HLR which forwards it to the gsmSCF function in the NeDS which updates CCCF with the information received in the USSD message.

4. Upon establishment of radio resources for the originating leg, the VMSC completes the call leg towards CCCF via the MGCF in subscriber's IMS network.

NOTE 1: Radio resource allocation for the originating leg in step 4 takes place in parallel to the exchange of USSD message between the UE and CCCF in step 2 and 3.

5. The MGCF initiates an INVITE by setting the Request-URI to Tel URI format using the CCCF PSI received in called party number in incoming IAM and sends it to the I-CSCF.

6. I-CSCF forwards the INVITE directly to an AS hosting the CCCF-PSI according to the standard procedure of "PSIs on the terminating side" described in TS 23.228 [9].

NOTE 2: CCCF can be configured on multiple application servers such that one application server serves multiple CS-IMS users but a particular user is served by only one application server at all times. This configuration enables a scalable solution that dynamically adjusts itself with the scope of the service subscription. The HSS maintains the CCCF instance identity associated with a particular CS-IMS user as part of the user's service subscription profile. The CCCF PSI is provisioned in the UE to enable CCCF invocation as discussed in subsequent sections. If more than one CCCF AS is used for the purpose of load sharing, multiple PSIs may be required to address individual Application Servers. Configuration of Initial Filter Criteria in the HSS and the CCCF PSI in the UE ensures that all sessions from a particular user are handled in the same AS.

NOTE 3: TS 23.228 [9] provides two ways to route towards the AS hosting a PSI; one is directly from I-CSCF to AS based on a HSS query; the other is via an S-CSCF assigned for the "PSI user". The former is used to route the incoming session directly to the CCCF AS.

NOTE 4: A race condition may arise if the USSD message in step 3 arrives at the CCCF after step 6.

7. CCCF terminates the incoming leg addressed to CCCF PSI and invokes a Routing B2BUA function to originate a session to the original called party destination on behalf of the user. CCCF retrieves the S-CSCF address from the HSS over the Sh interface to route the originating session to the original called destination.

   An ENUM dip for original called party number to SIP URI translations is required at the S-CSCF for routing to the original called destination. An IMS termination is assumed for this call walk-through. The BGCF and MGCF functions are involved in setting up of the terminating leg when terminating to the PSTN or CS Domain. CCCF maintains session states for the originating and terminating legs of the call via a third party call control (3pcc) function in order to control bearer upon Handover requests from the UE.

6.2a.2.3.3.2 CS Origination static anchoring via SIP Notify; call walk-through

Figure 6.2a.2.3-3 describes how signalling and bearer paths are established for originations from CS-IMS users at Visited MSCs in the areas of overlapping coverage or border between the CS and PS domains. The solution assumes CS anchoring decision criteria logic at the UE and uses a SIP Notify to invoke a CS Origination static anchoring service at CCCF. Prior subscription to the mobility event package was performed during the IMS registration procedure of a user accessing IMS identifying VoIP capability.

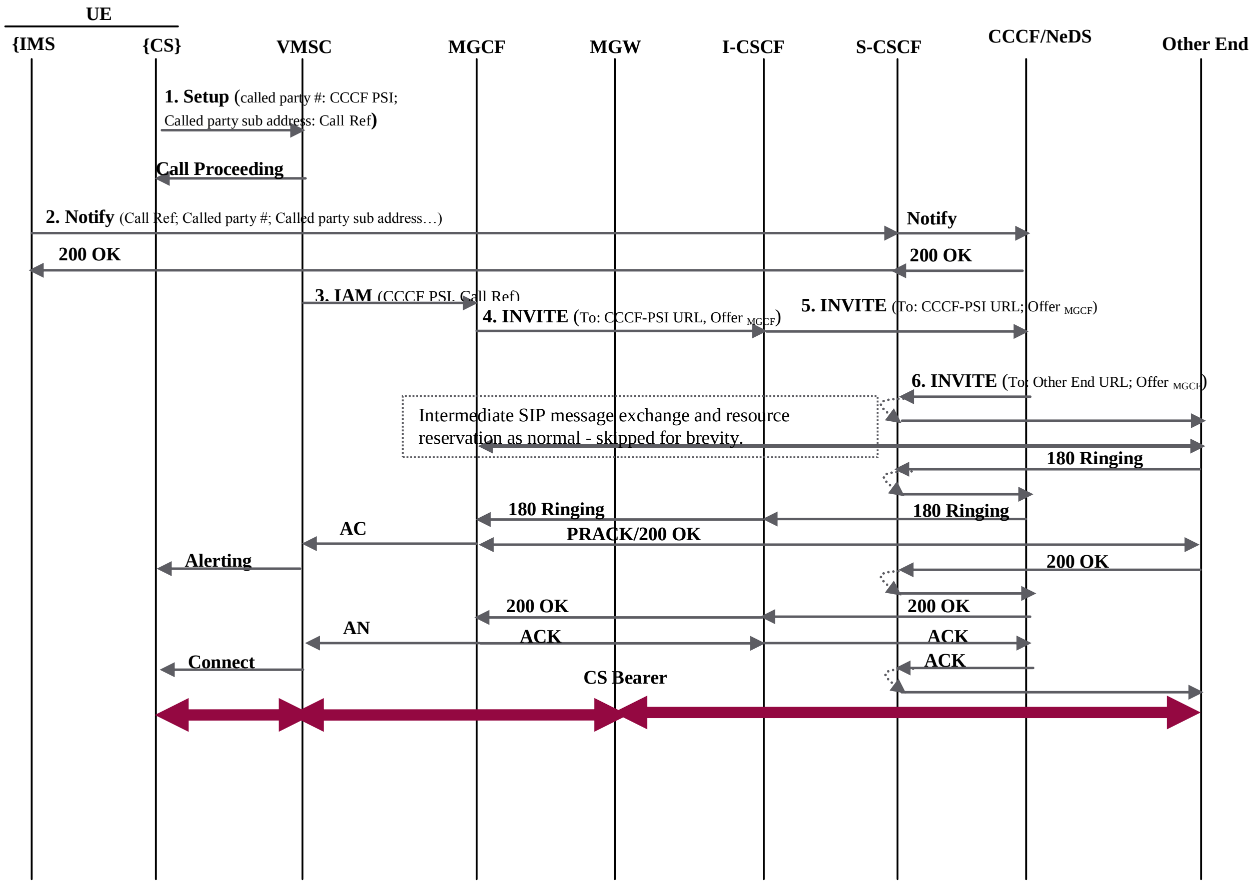


Figure 6.2a.2.3-3: CS Origination Anchored via SIP Notify walk-through

1. When the user dials the desired destination number, the terminal originates a call to a E.164 number associated with CCCF PSI communicating the call reference of the call being established. The call reference uniquely identifies the user's session at the UE and at CCCF. The configuration of call reference in the UE and CCCF is modelled on configuration and management of DTAP CC Transaction ID in the UE and the network respectively. The call reference can be communicated in called party address as digits appended to the CCCF PSI or in called party sub address.

2. Upon receipt of Call Proceeding from the MSC, the UE initiates a SIP Notify toward CCCF, requesting a CS origination static anchoring service at the CCCF. The called party number, the called party sub address and any other information required to complete the call toward the called party is sent in the Notify message.

3. Upon establishment of radio resources for the originating leg, the VMSC completes the call leg towards CCCF via the MGCF in subscriber's IMS network.

NOTE: Radio resource allocation for the originating leg in step 3 takes place in parallel to the exchange of SIP Notify message between the UE and CCCF in step 2.

The rest of the procedure is the same as described in Section 6.2a.2.3.3.1.

6.2a.2.3.3.3 CS Origination static anchoring using the CCCF PSI as called party number in SETUP message; call walk-through

Figure 6.2a.2.3-4 describes how signalling and bearer paths are established for CS originated calls from VCC users in areas of overlapping CS and I-WLAN/PS coverage or at the border between CS and I-WLAN/PS domain. The solution assumes CS anchoring decision criteria logic at the UE and uses the SETUP message only to invoke a static anchoring CS Origination service at the CCCF.

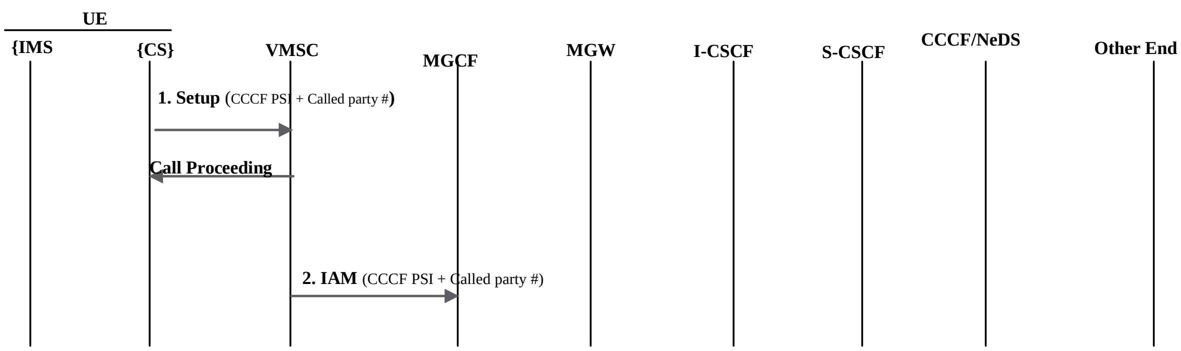


Figure 6.2a.2.3-4: CS Origination Anchored using the pre-fix solution walk-through

1. The user originates a call to a E.164 number, which is in fact a routing number pointing to the CCCF. The CCCF PSI + called party # can be transferred in the called party address field.

NOTE: This may not be possible in all scenarios due to limitations on number of digits that can be transmitted.

2. After establishing radio resources for the originating leg, the VMSC completes the call leg towards CCCF via the MGCF in subscriber's IMS network.

The rest of the procedure is the same as described in Section 6.2a.2.3.3.1.

6.2a.3 Termination

6.2a.3.1 IMS termination

6.2a.3.1.1 IMS termination - General

The handling of incoming calls routed to IMS depends on which domain the call is directed to in the first place, i.e. whether the incoming call is received via IMS or via CS.

6.2a.3.1.2 IMS termination - Incoming call received via IMS

The CS or IMS incoming call is directed to IMS. The CCCF/NeDS determines that the call should be routed to IMS.

An incoming call from IMS is routed to the S-CSCF via an I-CSCF. An incoming call from the CS domain is routed to the S-CSCF via an MGCF and the I-CSCF.

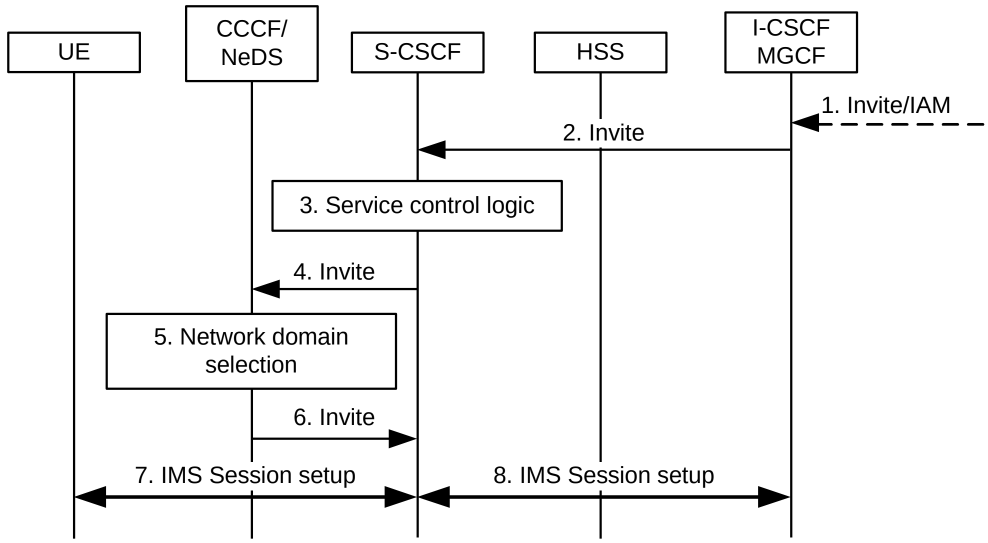


Figure 6.2a.3.1-1: IMS or CS incoming call received via IMS and routed to IMS

1-2. The originating network sends an Invite request to the I-CSCF (IMS incoming call) or IAM to the MGCF (CS incoming call). The Invite message is sent to the S-CSCF for the terminating user according to the Serving to Serving-CSCF procedures as defined in TS 23.228 [9].

3-4. The S-CSCF validates the service profile, and invokes the CCCF/NeDS as part of a termination service logic required for this user.

5-6. The CCCF/NeDS determines that the call is to be routed to IMS (Network domain selection function), and sends the Invite back to the S-CSCF. The CCCF is kept in the call chain.

7-8. A normal IMS session setup is performed towards the UE and back to the originating network. The signalling between the S-CSCF and the UE goes through the P-CSCF (not shown).

6.2a.3.1.3 IMS termination - Incoming call received via CS

The CS or IMS incoming call is directed to the CS domain. The CCCF/NeDS determines that the call should be routed to IMS.

6.2a.3.1.3.1 GMSC-controlled routing

When the GMSC has received the incoming call request, it contacts the CCCF/NeDS, which determines that the call shall be routed to IMS and provides the GMSC with a Routing Number towards IMS.

NOTE: The interface between the CCCF/NeDS and the GMSC may be implemented via the CAMEL protocol, and is described as such in the following diagrams. However, other implementations are possible (*e.g.* via an ISUP loop, or left as an internal interface, for example when the CCCF/NeDS is integrated with the GMSC). An example of an ISUP loop implementation is such that the GMSC first route the incoming call to CCCF. CCCF routes the call back to GMSC via another IAM with the ISUP Called Party parameter set as either Routing Number for routing to IMS, or as a prefix+MSISDN such that the prefix is treated by the GMSC as an indication to launch a SRI to complete the CS domain termination.

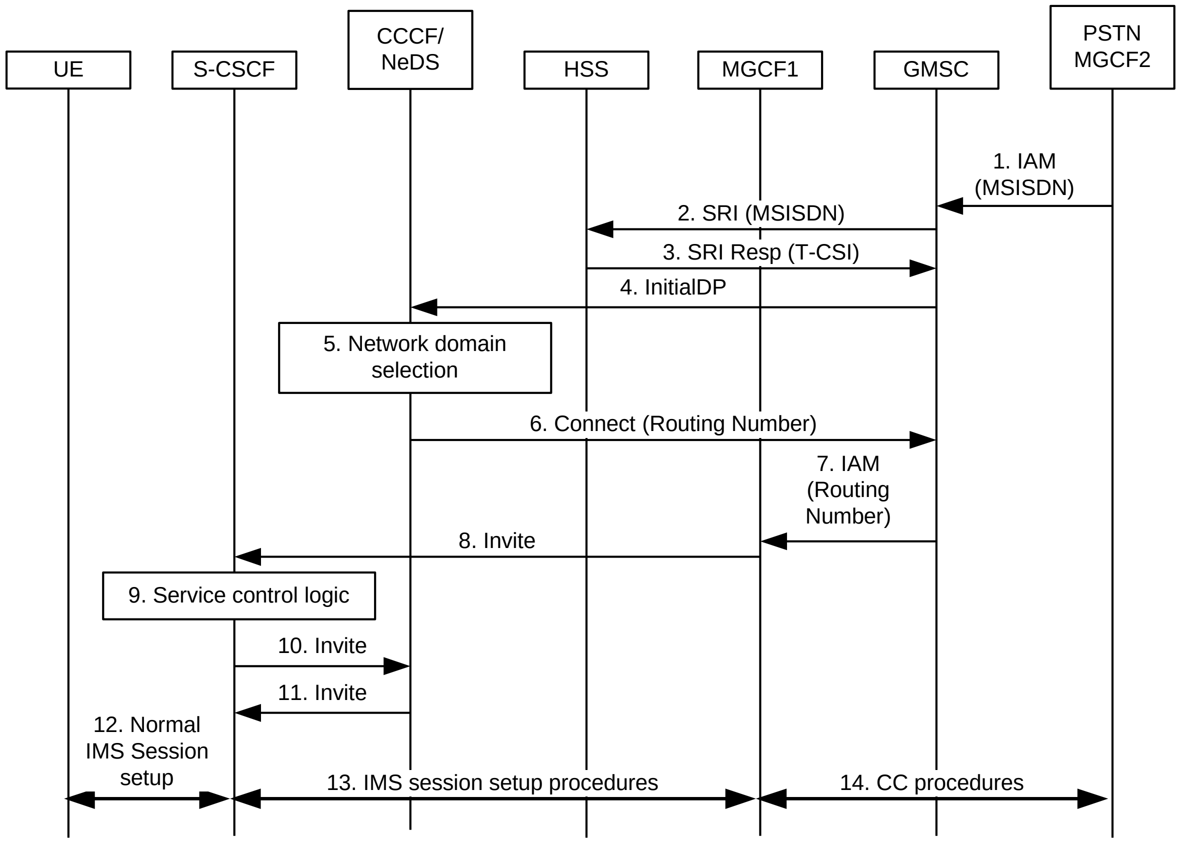


Figure 6.2a.3.1-2: IMS or CS incoming call received via CS and routed to IMS

1. The originating network sends an IAM to the CS domain GMSC (from an MGCF for an IMS-originating call, or from another CS network).

2. The GMSC requests a Send Routing Information to the HSS.

3-6. The HSS responds with a T-CSI to trigger the CAMEL dialog between the GMSC and the CCCF/NeDS. The relation between the MSISDN and the CCCF/NeDS is configured in the HSS. The CCCF/NeDS determines that the call shall be routed to the UE via IMS (Network domain selection) and returns a Routing Number towards IMS using the Connect operation to route the call to IMS.

NOTE 1: As an option, if a static routing to IMS is provisioned in the HSS, there is no need to query the CCCF/NeDS at this point. The Routing Number may then already be returned in the SRI Ack (step 3) instead of the T-CSI, and steps 4 to 6 would then be skipped.

7. The GMSC then forwards the IAM to the MGCF.

8. The MGCF resolves the Routing Number to the URI for the UE, and sends an Invite to the S-CSCF.

NOTE 2: As an option, the Routing Number may be resolved to the URI by the CCCF/NeDS, i.e. in step 8, the Routing number is treated as a PSI hosted by the CCCF/NeDS, and MGCF forwards the INVITE to the CCCF/NeDS via I-CSCF, the CCCF resolves the Routing Number to the URI for the UE, and acts as a B2BUA and directly re-issues the INVITE to S-CSCF with the URI.

NOTE 3: Alternatively, also as an option, the CCCF/NeDS may act as a redirect server, and upon receiving the INVITE from the MGCF via the I-CSCF, may resolve the Routing Number to a URI, and send back 302 Moved Temporarily to the MGCF, which would then re-route the INVITE to the S-CSCF serving the URI.

9-11. The S-CSCF validates the service profile, and invokes the CCCF/NeDS as part of a termination service logic required for this user.

NOTE 4: In case the Routing Number is resolved to the URI by CCCF/NeDS which acts in a Routing B2BUA mode, the iFC should be configured to avoid that the S-CSCF invokes the CCCF/NeDS again.

12-14. A normal IMS session setup is performed towards the UE via the S-CSCF and the P-CSCF (not shown), and the MGCF translates the result to CS signalling, as per the usual procedures.

6.2a.3.1.3.2 VCC-SRF Based Dynamic Routing

Figure 6.2a.3.1-3 below provides an example of termination routing policy execution in CS domain, while an incoming call arriving at the GMSC for a CS IMS user who is currently being served in IMS, when the call is initially directed to the CS domain.

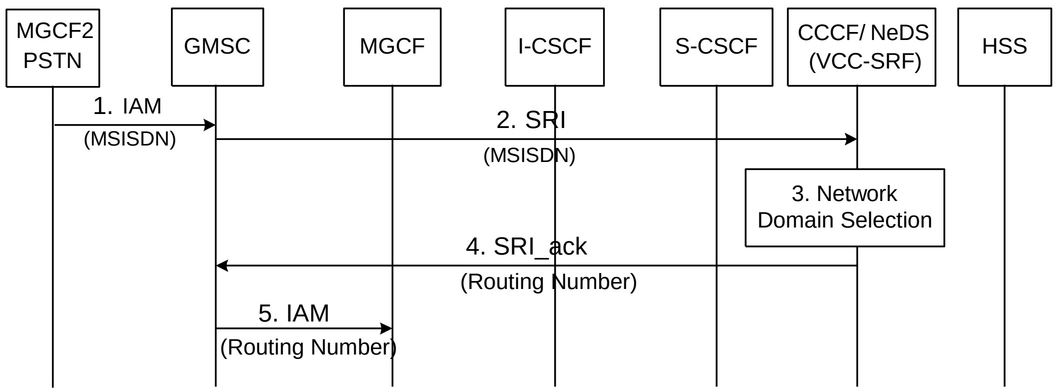


Figure 6.2a.3.1.3.2-3: IMS or CS incoming call received via CS and routed to IMS

1. IAM message addressing the MSISDN comes into the GMSC.
2. GMSC sends SRI to HSS for location information. The CCCF/NeDS (VCC-SRF) head off the SRI, trigger the embedded service logic of NeDS and treat the SRI as Routing decision Query message.

3-4. The CCCF/NeDS determines that the call shall be routed to the UE via IMS (Network domain selection), returns a Routing Number towards IMS using the SRI_Ack message to route the call to IMS.

5. GMSC sends an IAM, addressing the Routing Number, to the MGCF.

The rest of the sequence is identical to the one in chapter 6.2a.3.1.3.1, starting at step 8.

6.2a.3.2 CS termination

6.2a.3.2.1 CS termination - General

The handling of incoming calls routed to the CS domain depends on which domain the call is directed to in the first place, i.e. whether the incoming call is received via IMS or via CS.

6.2a.3.2.2 CS termination – Incoming Call received via IMS

The CS or IMS incoming call is directed to IMS. The CCCF/NeDS determines that the call should be routed to the CS domain.

An incoming call from IMS is routed to the S-CSCF via an I-CSCF. An incoming call from the CS domain is routed to the S-CSCF via an MGCF and then I-CSCF.

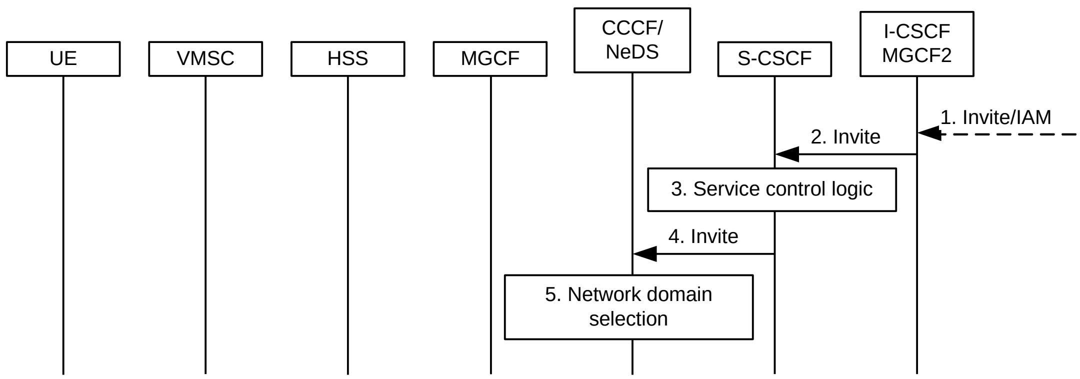


Figure 6.2a.3.2-1: IMS or CS incoming call received via IMS and routed to CS - initial routing

1-2. The originating network sends an Invite to the I-CSCF (IMS incoming call) or an IAM to the MGCF2 (CS incoming call). The Invite message is sent to the S-CSCF for the terminating user according to the Serving to Serving-CSCF procedures as defined in 3GPP TS 23.228 [9].

3-4. The S-CSCF validates the service profile, and invokes the CCCF/NeDS as part of a termination service logic required for this user.

5. The CCCF/NeDS determines that the call should be set up in the CS domain, via Network domain selection.

Different alternatives are presented for the routing of the call after Network domain selection, depending on whether re-routing through the GMSC in the CS domain is wanted.

6.2a.3.2.2.1 Direct routing to VMSC

In this alternative, the CCCF/NeDS sends a Send Routing Information to the HSS to query the MSRN. The MGCF directly sends the IAM to the VMSC.

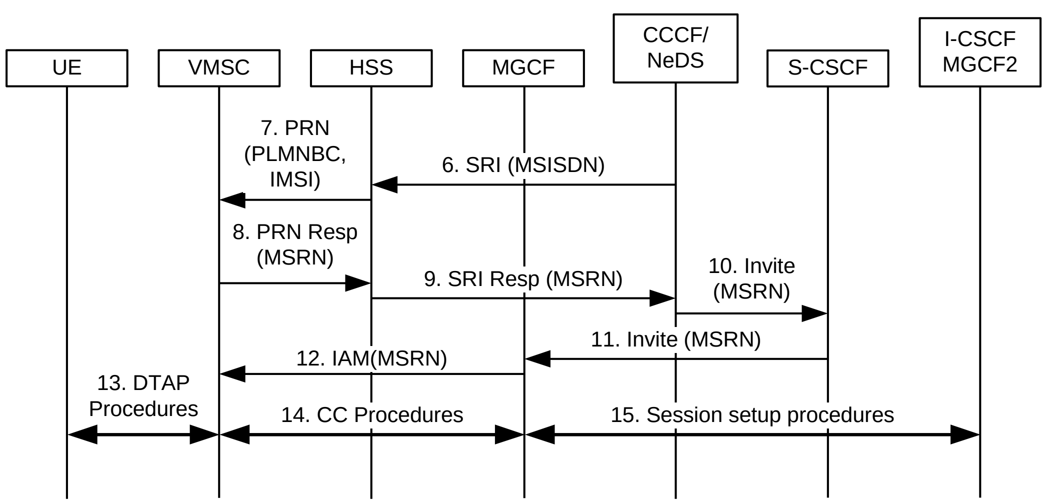


Figure 6.2a.3.2-2: IMS or CS incoming call received via IMS and routed to CS - post Network domain selection

6-9. Once the CCCF/NeDS has determined that the call should be set up in the CS domain (Network domain selection), it queries the HSS for an MSRN for the UE. The HSS in turn queries the VMSC/VLR for the MSRN, which is then returned to the CCCF/NeDS.

10-11. The CCCF/NeDS returns the Invite to the S-CSCF, which is in turn routed to the proper MGCF for entering the CS domain.

12-15. The MGCF initiates a normal CS call setup towards the visited MSC, according to the usual procedures, and the call setup proceeds as usual.

6.2a.3.2.2.2 Re-routing through the GMSC

6.2a.3.2.2.2.1 CAMEL based solution for GSM/UMTS CS termination

Figure 6.2a.3.2-3 below describes how signalling and bearer paths are established for calls originating in IMS or CS domain toward CS-IMS user when the user is roaming in CS Domain.

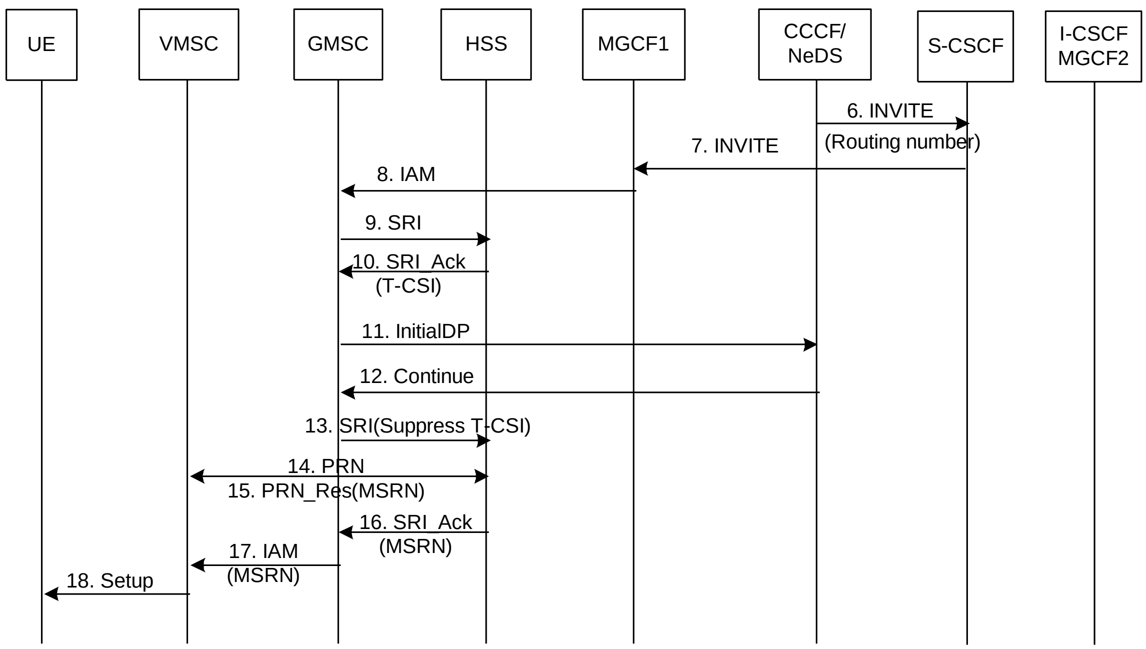


Figure 6.2a.3.2-3: CS Termination Anchored at CCCF; IMS or CS origination walkthrough - post Network Domain Selection

6. Once the CCCF/NeDS has determined that the call shall be set up in the CS domain via Network domain selection, it allocates a Routing Number towards CS domain and invokes the CCCF function. CCCF enables a B2BUA for execution of 3pcc function to maintain session states for the originating and terminating legs of the call in order to control bearer upon Domain Transfer requests from the UE and passes the control of the incoming session back to the S-CSCF.

7-8. The S-CSCF forwards the session to the MGCF which in turn forwards it to the GMSC for routing within the CS domain.

9. GMSC uses the Routing number in the incoming IAM message to discover user's MSISDN via an Intelligent Network service or translation techniques prior to performing SRI Query toward the HSS.

10-11. The HSS returns user's T-CSI in SRI Response which results in a CAMEL Initial DP message to the gsmSCF function of the NeDS for execution of appropriate service logic for the incoming session.

12-18. The gsmSCF function of NeDS returns a CAMEL Continue message to the GMSC to revert to normal processing to route the call to the user via the Visited MSC that the user is currently registered at.

6.2a.3.2.2.2.2 VCC-SRF based solution for GSM/UMTS CS termination

Figure 6.2a.3.2-4 below describes how signalling and bearer paths are established for calls originating in IMS or CS domain toward CS-IMS user when the user is roaming in CS Domain.

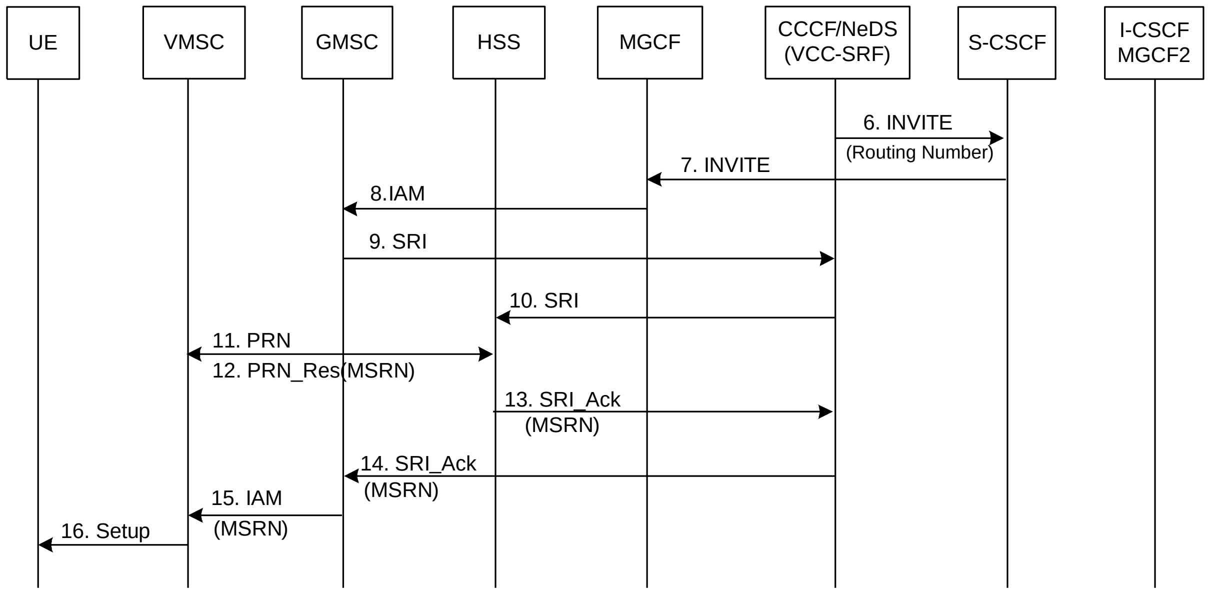


Figure 6.2a.3.2-4: CS Termination Anchored at CCCF; IMS or CS origination walkthrough - post Network Domain Selection

6-8. Once the CCCF/NeDS has determined that the call shall be set up in the CS domain via Network domain selection, the CCCF allocates a Routing Number towards CS domain and invokes the CCCF function, then it passes the control of the session back to the S-CSCF, which subsequently forwards the session to GMSC via MGCF. The procedure of these steps is the same as step 6-8 described in chapter 6.2a.3.2.2.2.1.

9-10. Based on the configuration of operator, the GMSC treats the Routing number as the MSISDN and sends SRI to HSS for routing information. The CCCF/NeDS intercepts the SRI, examines the Routing number, maps it to the CS MSISDN and sends it to HSS for location information.

11-16. HSS sends PRN to current VMSC for MSRN, and then sends SRI_ack to GMSC with MSRN received in PRN_ack. The GMSC routes the call to VMSC based on the MSRN.

6.2a.3.2.3 CS termination - Incoming call received via CS

The CS or IMS incoming call is directed to the CS domain. The CCCF/NeDS determines that the call should be routed via the CS domain.

The sequence for this call case depends on whether Voice Call Continuity procedures require the call to always be routed to IMS.

6.2a.3.2.3.1 Call maintained in CS

When the GMSC has received the incoming call request, it contacts the CCCF/NeDS, which determines that the call shall be routed to the CS domain and informs the GMSC to continue the call setup in the CS domain.

NOTE: The interface between the CCCF/NeDS and the GMSC may be implemented via the CAMEL protocol, and is described as such in the following diagram. However, other implementations are possible (*e.g.* via an ISUP loop, or left as an internal interface, for example when the CCCF/NeDS is integrated with the GMSC). An example of an ISUP loop implementation is such that the GMSC first route the incoming call to CCCF. CCCF routes the call back to GMSC via another IAM with the ISUP Called Party parameter set as either MSRN for routing to IMS, or as a prefix+MSISDN such that the prefix is treated by the GMSC as an indication to launch a SRI to complete the CS domain termination.

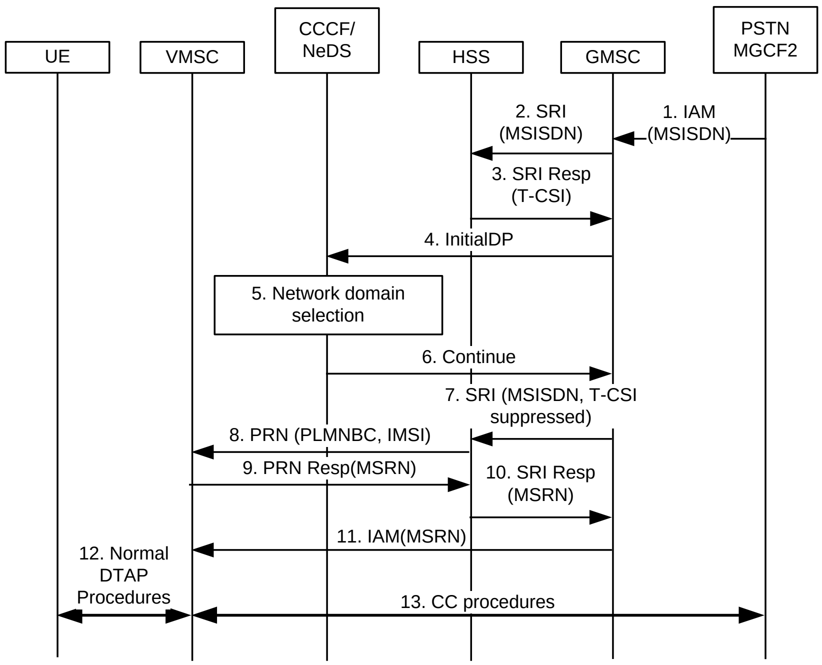


Figure 6.2a.3.2-5: IMS or CS incoming call received via CS and routed to CS

1. The originating network sends an IAM to the CS domain GMSC (from an MGCF for an IMS-originating call, or from another CS network).

2. The GMSC requests a Send Routing Information to the HSS.

3-6. The HSS responds with a T-CSI to trigger the CAMEL dialog between the GMSC and the CCCF/NeDS. The relation between the MSISDN and the CCCF/NeDS is configured in the HSS. The CCCF/NeDS determines that the call shall be routed via the CS domain (Network domain selection) and returns a Continue indication to the GMSC.

7-13. The GMSC queries the HSS again, this time suppressing the T-CSI, and from now on, usual procedures for CS call setup are performed.

6.2a.3.2.3.2 Static Anchoring: Terminating call anchoring at CCCF for users roaming in CS Domain

6.2a.3.2.3.2.1 General

It is recommended that the operator policies for routing of incoming calls are established in a manner that facilitates anchoring of CS-IMS user's incoming calls at CCCF. Incoming calls originated in the CS domain network, PSTN or other IMS networks, which are destined for CS-IMS users can be anchored at CCCF by setting up routing functions such that the incoming calls for CS IMS users are delivered to the user's home IMS network. CCCF enables a Routing B2BUA 3pcc function and routes the call to CS Domain, if the user is roaming in the CS Domain at the arrival of the call see 6.2a.3.2.2, CS Termination, Call received via IMS.

Routing identities used in this scheme are listed below:

MSISDN: Published DN associated with the user.

NeDS PSI: Published DN associated with NeDS.

IMRN: Dynamic routing number used for routing into IM Subsystem.

IMRN Composition: Call Reference digits appended to NeDS PSI.

CSRN: Dynamic routing number used for routing into CS Domain.

The routing scheme as used for processing of incoming call requests in different interworking scenarios is described in the following example call walkthroughs. These examples are based on the assumption that the NeDS and CCCF functions are co-located in the same Application Server. However, the two functions can be located in different Application Servers as a configuration option. The order of execution of NeDS and CCCF in such a configuration may be driven by iFC or sFC provisioning with appropriate priority assignments.

6.2a.3.2.3.2.2 CAMEL based solution for GSM/UMTS CS termination

Figure 6.2a.3.2-6 below describes how signalling and bearer paths are established for calls originating in CS or PSTN network, toward CS-IMS user when the user is roaming in CS Domain and the call is directed to CS.

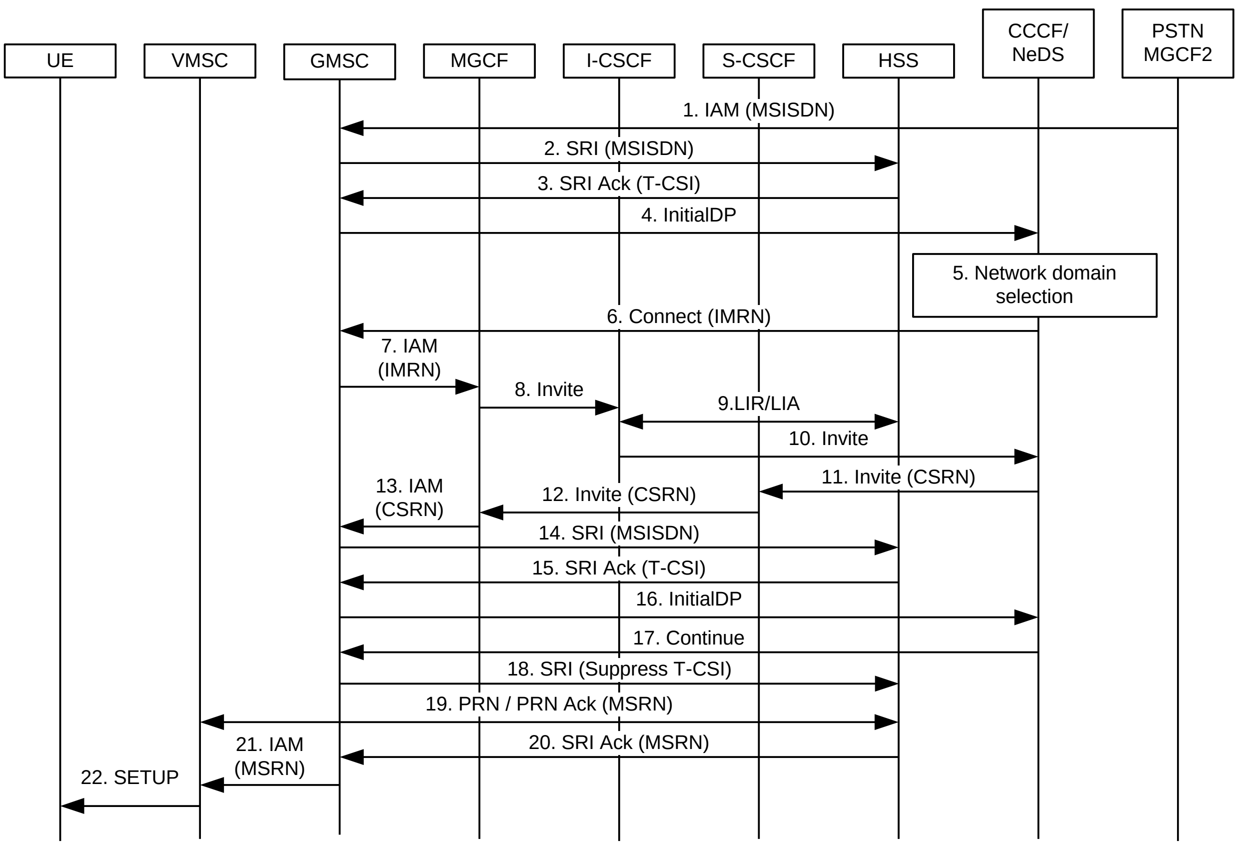


Figure 6.2a.3.2-6: CS Termination Anchored at CCCF; CS origination, Call Directed toCS walkthrough

1. A call originated in CS domain or PSTN network is delivered at the user's home GMSC function as the call is directed to CS.

2-3. The GMSC sends an SRI query to the HLR to retrieve the routing information. The HLR returns subscriber's T-CSI (Terminating CAMEL Subscription Information) indicating CS-IMS subscription.

4. The GMSC continues the CAMEL processing by sending an Initial DP to the NeDS for routing instructions.

5-6. The CCCF/NeDS determines that the call shall be routed to the CS domain, allocates a call reference for the incoming call and creates an IMRN by appending the call reference to the DN associated with the CCCF/NeDS PSI for routing to IMS, and sends a CAMEL Connect message to cause the GMSC to route the call to the user's IMS network using the IMRN.

7. GMSC sends an IAM, addressing the IMRN, to the MGCF.

8. The MGCF processes the IAM and initiates an INVITE to the I-CSCF with Tel URI set to the IMRN.

9. I-CSCF performs location query to the HSS to retrieve application server name associated with the IMRN. The Application Server hosting the CCCF/NeDS is configured in the HSS against a range of PSIs wild carded to the CCCF/NeDS PSI.

10. I-CSCF forwards the INVITE to the Application Server hosting the NeDS.

11. The CCCF/NeDS frees the IMRN, allocates a CSRN for routing of call back to CS domain and invokes the CCCF function.

    CCCF enables a B2BUA for execution of 3pcc function to maintain session states for the originating and terminating legs of the call in order to control bearer upon Voice call continuity procedure requests from the UE and passes the control of the incoming session back to the S-CSCF.

12-13. The S-CSCF forwards the session to the MGCF which in turn forwards it to the GMSC for routing within the CS domain.

NOTE: The GMSC function used for processing of incoming call arriving from CS domain/PSTN network may be different from the GMSC function used for processing of incoming call routed from the IMS.

14. GMSC uses the CSRN in the incoming IAM message to discover user's MSISDN via an Intelligent Network service or translation techniques prior to performing SRI Query toward the HSS.

15-16. The HLR returns user's T-CSI in SRI Response which results in a CAMEL Initial DP message to the gsmSCF function of the CCCF/NeDS for execution of appropriate service logic for the incoming session.

18-22. The gsmSCF function of NeDS returns a CAMEL Continue message to the GMSC to revert to normal processing to route the call to the user via the Visited MSC that the user is currently registered at.

6.2a.3.2.3.2.3 VCC-SRF based solution for GSM/UMTS CS termination

Figure 6.2a.3.2-7 below describes how signalling and bearer paths are established for calls originating in CS or PSTN network, toward CS-IMS user when the user is roaming in CS Domain and the call is directed to CS.

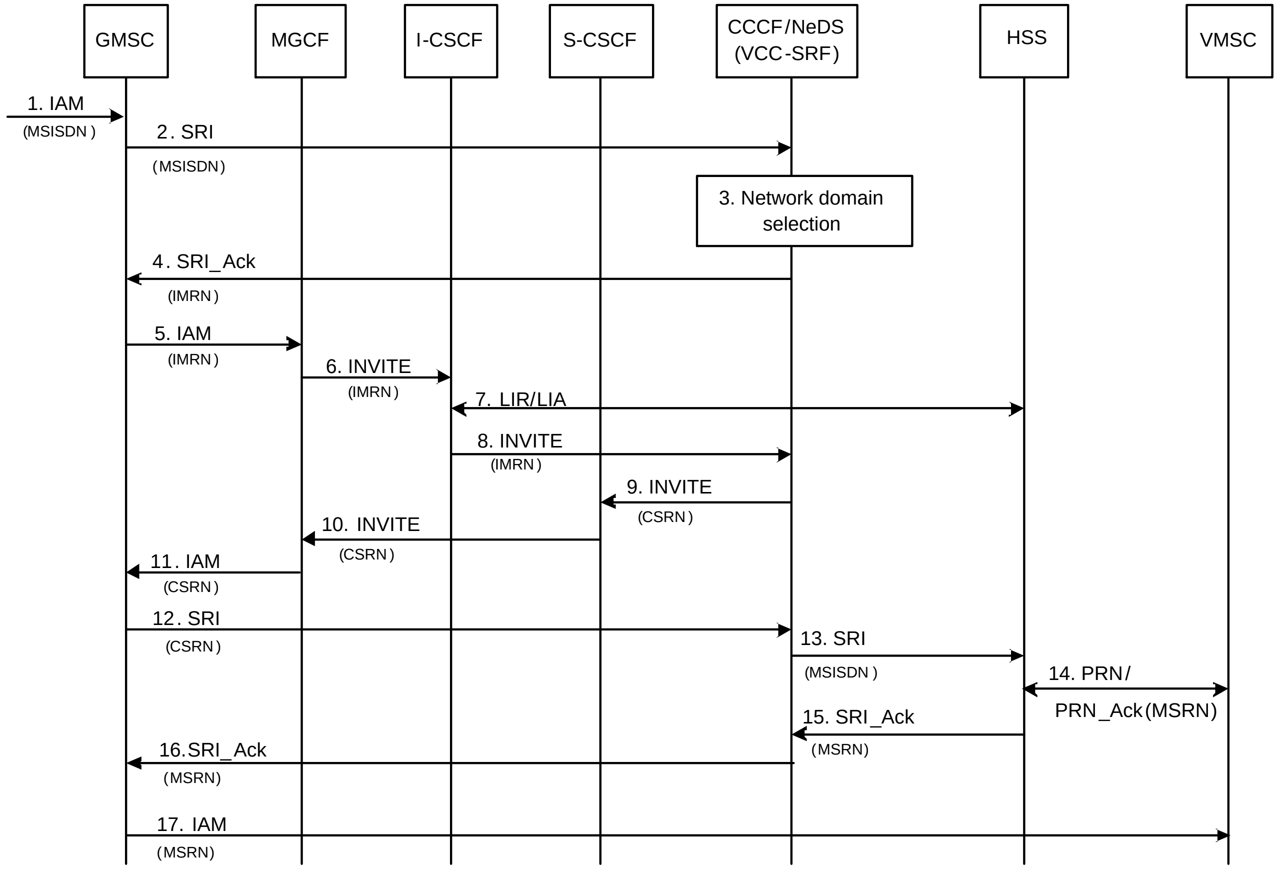


Figure 6.2a.3.2-7: CS Termination Anchored at CCCF; CS origination, Call Directed to CS walkthrough

1-2. IAM message addressing the MSISDN comes into the GMSC. GMSC sends SRI to HSS for location information. The CCCF/NeDS(VCC-SRF) intercepts the SRI and triggers the embedded service logic of CCCF/NeDS.

3-4. Based on current routing policy, The CCCF/NeDS(VCC-SRF) decides that the call shall be set up in the CS domain, and routes the incoming call to IMS domain for anchoring, allocates a call reference for the incoming call and creates an IMRN by appending the call reference to the DN associated with the CCCF PSI for routing to IMS, and sends a SRI_ack message with the IMRN as a MSRN in order to cause the GMSC to route the call to the user's IMS network.

5. GMSC sends an IAM, addressing the IMRN, to the MGCF.

6-11. The following steps performed in IMS is the same as step 8-13 described in section 6.2a.3.2.3.2.2, the session is forwarded to the GMSC for routing within the CS domain, while the called party number in IAM is CSRN..

12-13. Basing on the configuration of operator, the GMSC treats the CSRN in the incoming IAM message as the MSISDN and sends SRI to HSS for location information. The CCCF-SRF intercepts the SRI, examines the CSRN, maps it to the CS MSISDN and sends it to HSS for location information.

14-17. HSS sends PRN to current VMSC for MSRN, and then sends SRI_ack to GMSC with MSRN received in PRN_ack. The GMSC finally routes the call to VMSC based on the MSRN.

6.2a.3.3 HSS based Network Domain Selection for termination

6.2a.3.3.1 General

This section provides the details of an alternative architecture and interactions between the logical components of HSS and NeDS for fulfilling the requirement of network domain selection. It is expected that only termination mechanism will be specified.

The HSS based network domain selection does not require a new functional entity that could be collocated with the CCCF as shown in clause 6.2a.3.1.

Editor's Note: Detailed message flows for all the different scenarios need to be added.

6.2a.3.3.2 Scenarios and Possible Routing Policy

CS not reachable/IMS not reachable:

- Directly reject the call or forward to Voicemail.

CS reachable/IMS not reachable, or CS not reachable/IMS reachable:

- When a user is reachable only in one domain, e.g. CS domain or IMS domain, all terminating session are routed to the user through the domain in which the user is registered;

CS reachable/IMS reachable:

- When a user is reachable in both domain simultaneously, a terminating session may be routed to the user through:

  a) the same domain the terminating session comes from, or

  b) according to many factors, including the user's configuration, a operator's configuration, time and so on, a preferred domain is selected without considering the network the terminating call comes from;

NOTE 1: The term "Reachable" is used to designate that the user has registered in the domain, and no corresponding incoming call barring service been activated, but does not cover temporary loss of coverage.

The control can be summarised as follows:

- Routing a terminating call coming from CS domain (e.g. a call via GMSC) to the user through the CS domain. (traditional CS call);
- Routing a terminating call coming from CS domain to the user through the IMS;
- Routing a terminating call coming from IMS domain to the user through the IMS domain (standard IMS call);
- Routing a terminating call coming from IMS domain to the user through the CS domain.

NOTE 2: The text above may be treating as a general scenarios and policy description for terminating network domain selection solutions.

6.2a.3.3.3 Architecture

According to the above discussion, different scenarios have different input parameters or different output results, and a user's preferences and an operator's configurations are also needed to be taken into account. Then, a new functional module is introduced, named Network Domain Selection (NeDS), to perform routing policy decision function based on the combination of information from different aspects. Accordingly, the HSS should be enhanced with some necessary improvements to support decision information provisioning and interaction with a NeDS. A NeDS is shown in the figure below.

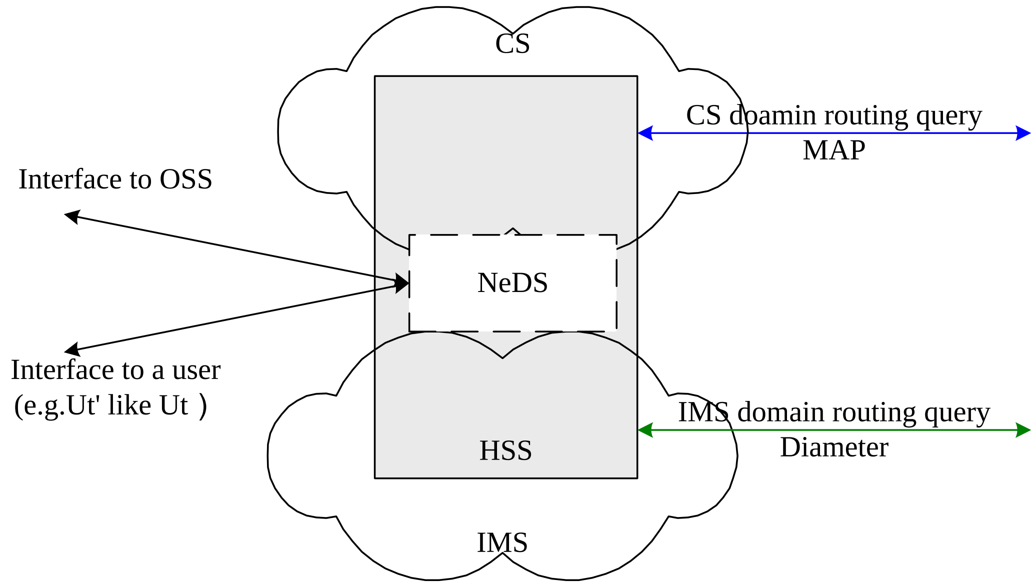


Figure 6.2a.3.3-1: architecture of HSS based network domain selection

The Network Domain Selection (NeDS) is a logical entity embedded in the HSS. A NeDS stores routing policy of a user and provides routing policy decision to the entity that initiated the routing decision query. In addition, A NeDS gets a user's status information (e.g. reachable/non-reachable) in CS and IMS that will be used to make the routing policy decision. If the HSS can make routing decision based on the information stored, the NeDS function will not be required for the routing policy.

Editor's Note: If the HSS based network domain selection is implemented for VCC, the NeDS may be split between HSS and CCCF, then the interface and information flow between the NeDS components in the HSS and in the CCCF need to be defined during specification work.

6.3 Service Continuity Model: IMS Controlled Alternative

6.3.1 General Description

Editor's Note: The TR should clarify how it is possible to have MGCF and MGW in two separate networks, globally.

Editor's Note: TR should explain how the call content interception is handled with this type of HO approach.

Editor's Note: TR should explain how CCBS is handled with this type of HO approach.

6.3.1.1 Techniques for enabling static anchoring for CS calls and IMS sessions at CCCF

CCCF controls the bearer path for all CS calls and IMS sessions of CS-IMS users that are subscribed to the CS-IMS Voice Continuity service. All CS calls and IMS sessions of such users may be anchored at CCCF to facilitate control of the bearer path upon call continuity in the initial phase of migration from CS to IMS.

As the population of CS-IMS users grows, some additional criteria may be used to refine the subscription based anchoring selection criterion. The use of location based criteria such as Global Cell Identifiers and user's current geographical coordinates is for further study.

6.3.4 Origination

VCC is realized by using IETF Third Party Call Control (3pcc) function which controls the bearer path for all CS and IMS originations from CS-IMS users that are subscribed to the CS-IMS Voice Continuity service. All CS and IMS originations of such users may be anchored at CCCF to facilitate control of the bearer path upon VCC in the initial phase of migration from CS to IMS.

As the population of CS-IMS users grows, some additional criteria may be used to refine the subscription based anchoring selection criterion. The use of location based criteria such as Global Cell Identifiers and user's current geographical coordinates is for further study.

6.3.5 Termination

VCC is realized by using IETF Third Party Call Control (3pcc) function which controls the bearer path for all CS and IMS terminations of CS-IMS users that are subscribed to the CS-IMS Voice Continuity service. All CS and IMS terminations of such users may be anchored at CCCF to facilitate control of the bearer path upon Handover in the initial phase of migration from CS to IMS.

6.3.6 Call Continuity Scenarios

6.3.6.1 General

VCC is a service in a CS-IMS user's home IMS network that anchors user's active CS calls and IMS sessions to enable active mode roaming across CS Domain and IM Subsystem.

NOTE: This is a change from 3GPP Handover procedures defined for active mode roaming within GSM/UMTS CS, wherein, calls are anchored at the system used for initial call setup, with the Handover Target node relaying the Call Control messages between the Anchor node and the UE post Handover. Although SIP extensions can be suggested for encapsulation of 24.008[13] call control protocols in SIP for VCC transitions from GSM/UMTS CS to IMS over I-WLAN, significant changes to GSM/UMTS CS protocols are required for SIP encapsulation in BSSAP for VCC transitions from IMS to GSM CS, and SIP encapsulation in RANAP for VCC transitions from IMS to UMTS CS; it is therefore not feasible to maintain the same anchor control model with active mode roaming across CS and IMS.

CCCF provides functions for CS-IMS Voice Call Continuity. All VCC transitions (i.e. initial and subsequent) associated with a particular user session are executed and controlled by CCCF upon UE's request.

Since VCC transitions are executed across CS Domain and IM Subsystem with different call control protocols, TS 24.008 [13] call control protocol is used in CS Domain whereas SIP is used in IMS for call control procedures; the VCC procedure is executed at the call control level. The call control Protocol State Machine is released in the handing-out domain and re-established in the handing-in domain.

CCCF provides cohesive billing with a complete VCC transition history for the duration of a voice session. Details of accounting and charging implications are for further study, however, it should be noted that the call/session established to enable VCC transitions are captured as call continuity legs of the call/session being transferred and therefore do not impact the direction initially used to establish the call/session for the purpose of charging.

CCCF is globally routable using Public Service Identities, a service DN is used for routing within CS Domain and PSTN networks and a SIP URI is used for routing within IMS network. CCCF PSI associated with a CS-IMS user is dynamically assigned and communicated to the UE upon registration with IMS.

Simultaneous CS Domain and IM Subsystem registration is not required at the time of CS call or IMS session establishment; the user is required only to be registered in the domain from which it is currently receiving services. Simultaneous registration is required for initiation of the CS-IMS VCC procedures.

6.3.6.2 Procedures for CS to IMS Voice Call Continuity

Figure 6.3.6.2-1 describes how signalling and bearer paths are established for execution of CS to IMS VCC procedures. IMS termination is assumed in this walk-through, whereas an MGCF function is involved in the control path for the termination in case of CS and PSTN terminations.

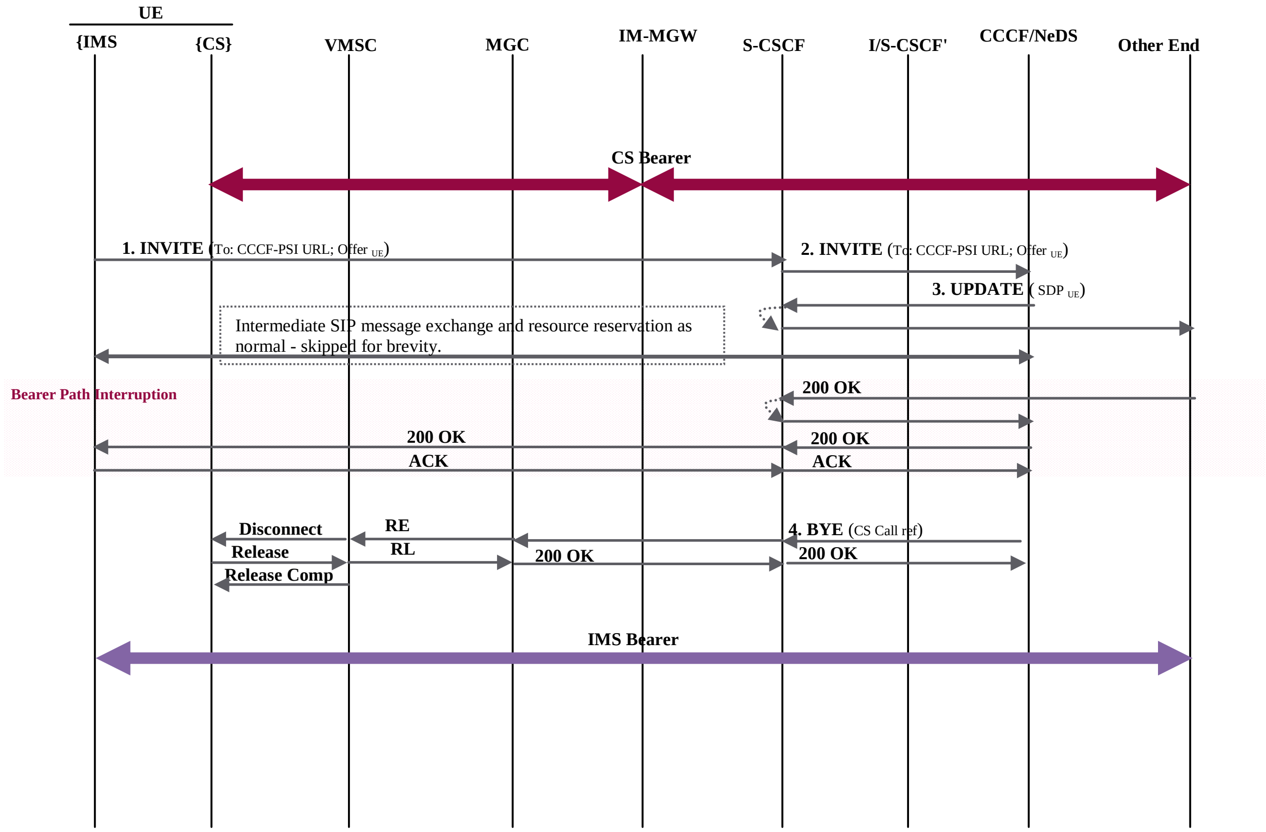


Figure 6.3.6.2-1: CS to IMS Voice Call Continuity walk-through

1. If the user is not registered with IMS at the time when the UE determines a need for VCC transition to IMS, the UE initiates Registration with IMS. It subsequently sends an INVITE including original session information to CCCF using CCCF PSI as a VCC indication requesting it to perform a VCC transition of the active CS call to IM Subsystem.

2. User's S-CSCF routes the INVITE to CCCF application server assigned to the user upon execution of filter criteria.

3. CCCF performs the transfer of the user's CS leg to IMS by using SIP Session Transfer procedures. It is an implementation option as to how the SIP Session Transfer is executed. Use of an UPDATE consisting of the SDP of the IMS leg is illustrated here; however, other options such as a ReINVITE can also be used to implement Session Transfer. Minor bearer path interruption, estimated to be about 100-200 milliseconds, is expected due to the switchover.

4. The CS bearer and signalling legs are released upon successful execution of SIP Transfer.

NOTE: CCCF initiates the release of signalling and bearer in the handing-out domain as release from the UE cannot always be guaranteed due to possibility of loss of coverage in the handing-out domain during the VCC procedure. The UE may also initiate the release of the bearer and signalling in the handing-out domain, in which case, CCCF processes the release appropriately.

6.3.6.2.1 Subsequent VCC Transition Back to CS

Figure 6.3.6.2-2 describes how signalling and bearer paths are established for execution of subsequent VCC transition to CS Domain.

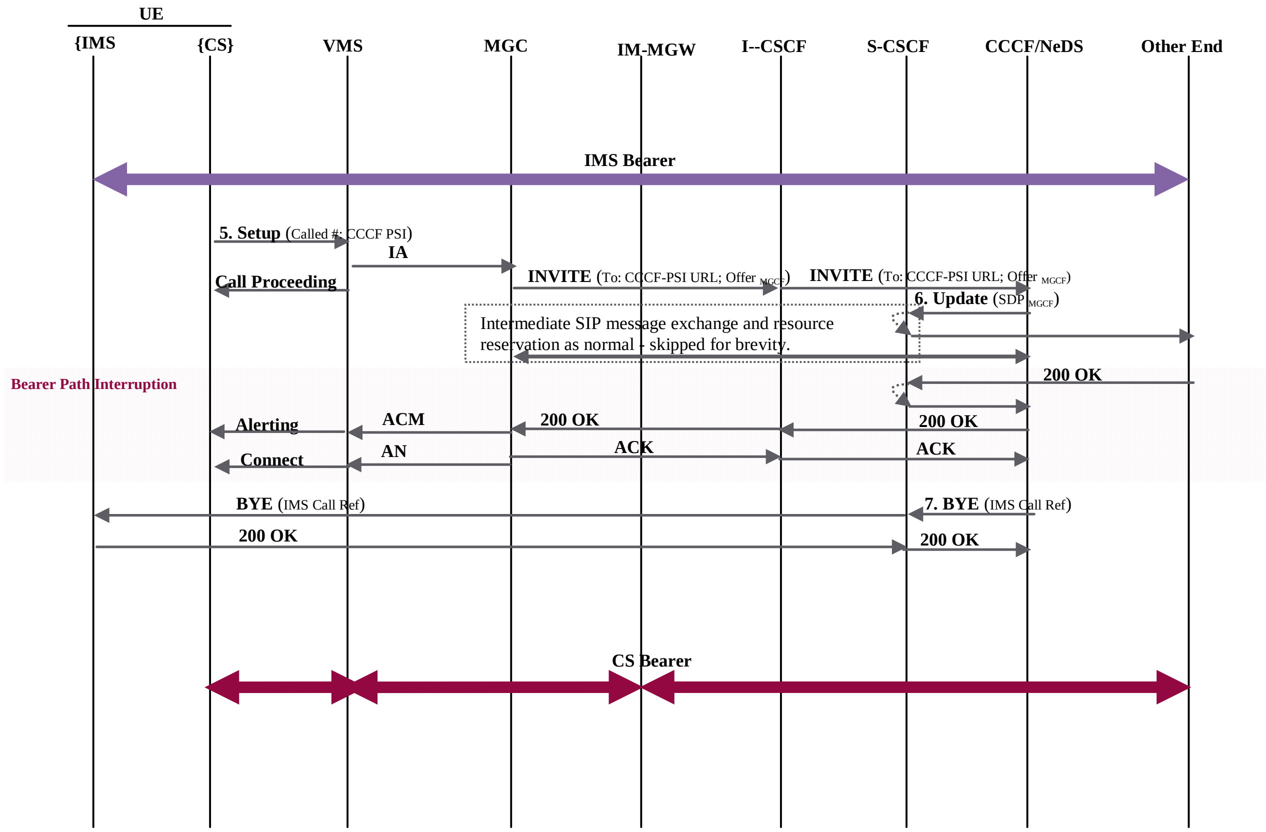


Figure 6.3.6.2-2: Subsequent VCC transition to CS walk-through

5. The UE registers with the Visited MSC when it determines a need for VCC transitions to CS. It subsequently initiates a CS call to CCCF using CCCF PSI requesting it to perform VCC transition of the active CS call to CS Domain. The CS call is routed via the MGCF and I/S-CSCF to CCCF application server.

NOTE: See TS 29.163 [17] for MGCF behavior.

6. CCCF performs the transfer of the user's IMS leg to the CS Domain by using SIP Session Transfer procedures as described in the CS to IMS Voice Call Continuity walk-through.

7. The IMS bearer and signalling legs are released upon successful execution of SIP Transfer.

6.3.6.3 Procedures for IMS to CS call continuity

The call continuity procedures are the same as the call continuity procedures shown in subclause 6.3.6.2 as the procedures between CS Domain and IMS are agnostic of the previous call continuity history.

6.3.7 Impact on Supplementary Services

IMS Controlled model facilitates flexibility in providing service control for VCC users. Two service control configurations are provided in this section, namely Distributed Service Control model and Centralized Service Control model. With distributed service control, the CCCF does not influence supplementary service execution in the serving network node prior to or post VCC. User receives services from the domain it is active in a voice call, that is, CS Supplementary services are available to the user when it is in the CS Domain, whereas, richer IMS service set is made available to user as it moves into IMS coverage, within the context of the same call/session. With centralized service control, the service control for a VCC user is centralized in IMS. Sections 6.3.7.1 through 6.3.7.3 describe the Distributed Service Control model and section 6.3.7.4 describes the Centralized Service Control model.

Information capability exchange between the UE and the CCCF/NeDS prior to the establishment of the first call for the subscriber may be used to establish the method used for service control.

6.3.7.1 Voice Call Continuity for Multi-Session calls

CCCF uses the Mobility Event package to communicate with the UE, session specific information that is required to perform VCC of a CS-IMS user when the user is involved in multiple sessions.

All CS calls and IMS sessions for a CS-IMS user are anchored at CCCF via 3pcc Routing B2BUA function using static anchoring techniques discussed in Sections 6.2a. Upon successful allocation of a B2BUA function for a particular user session, CCCF assigns it a unique identifier along with and a SIP URI that is used to uniquely identify the session when requesting its VCC to IMS or a unique CS VCC ID that can be used to uniquely identify the session when requesting its VCC to CS. Notify's with Mobility Event package are used to communicate this information to the UE upon session anchoring.

In the event that the CS IMS user is not registered with IMS when making a CS call, the session identifiers cannot be communicated to the UE upon CS call anchoring at CCCF. Connected party address is used to enable VCC for such calls. It should be noted that CLIP Override and COLP override subscription is required to ensure that the connected party address is available at the UE to enable VCC in these conditions.

The fundamental principle of transferring the call control protocol state machine from the source ("handing-out") domain to the target ("handing-in") domain discussed for single sessions Section 6.3.6 is applied to transfer multiple session between CS and IMS to provide Voice Continuity across CS and IMS with multiple sessions.

6.3.7.1.1 Anchoring of IMS Held and Active sessions at CCCF

Figures 6.3.7.1-1 and 6.3.7.1-2 below provides a walkthrough of a scenario in which the UE originates an IMS session to the other end A, holds the session toward other end A, and originates a session toward the other end B.

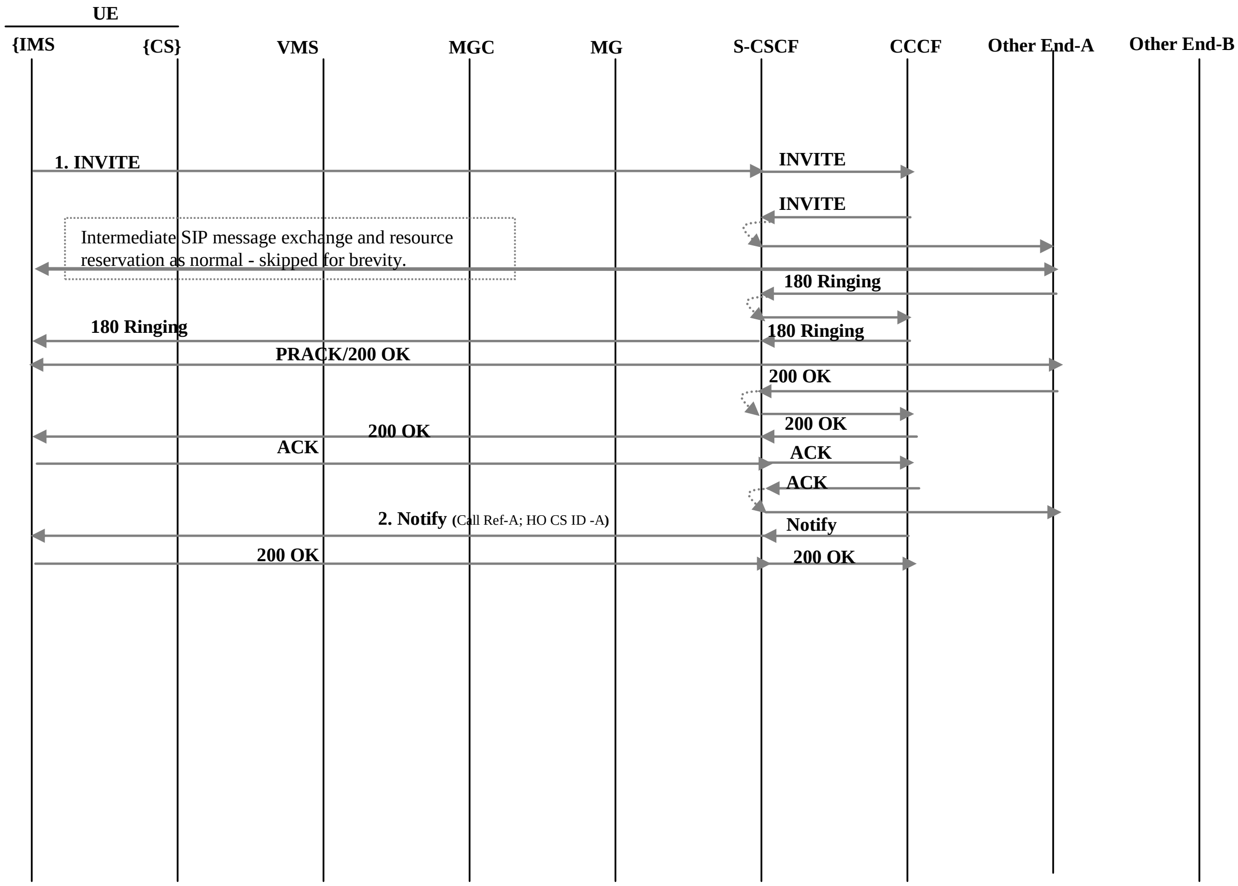


Figure 6.3.7.1-1: IMS session toward other end – A

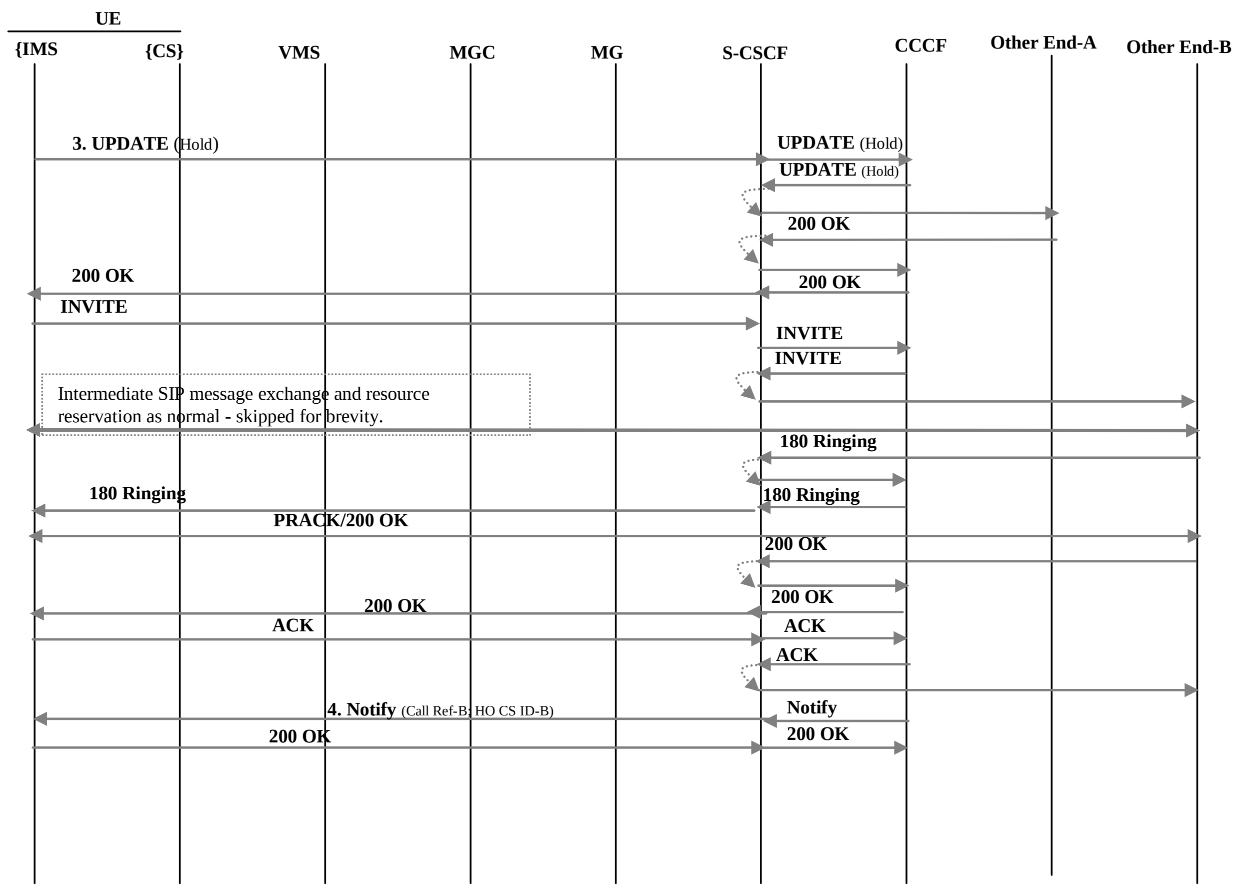


Figure 6.3.7.1-2: Hold A and originate an IMS session toward other end - B

1. As part of IMS registration, CCCF and the UE subscribe to each other for the Mobility Event package. The UE subsequently initiates an IMS session.

2. Upon successful execution of a Routing B2BUA function for IMS session to the other end A at CCCF, CCCF assigns a unique call reference identifier to the session for identification of the session between the UE and CCCF in subsequent dialogues. It also assigns a unique identifier which can be used for VCC of this session to CS Domain. The CS VCC identifier can be created by either assigning a unique routing number for the session or assigning a string of digits that can be appended to CCCF DN when requesting VCC, the latter is recommended due to the operational overhead associated with assignment of unique routing numbers for each active session. These new identifiers are communicated to the UE via a Notify with Mobility Event package.

3. The user puts session toward the other end A on hold and originates a new session towards the other end B.

4. Upon successful execution of a Routing B2BUA function for IMS session to the other end B at CCCF, CCCF assigns a unique call reference identifier to the session for identification of the session between the UE and CCCF in subsequent dialogue. It also assigns a unique identifier which can be used for VCC of this session to CS Domain. These new identifiers are communicated to the UE via a Notify with Mobility Event package.

6.3.7.1.2 IMS to CS VCC of IMS Held and Active sessions

Figure 6.3.7.1-3 below provides a walkthrough of VCC of IMS Held and Active sessions established as described in previous section.

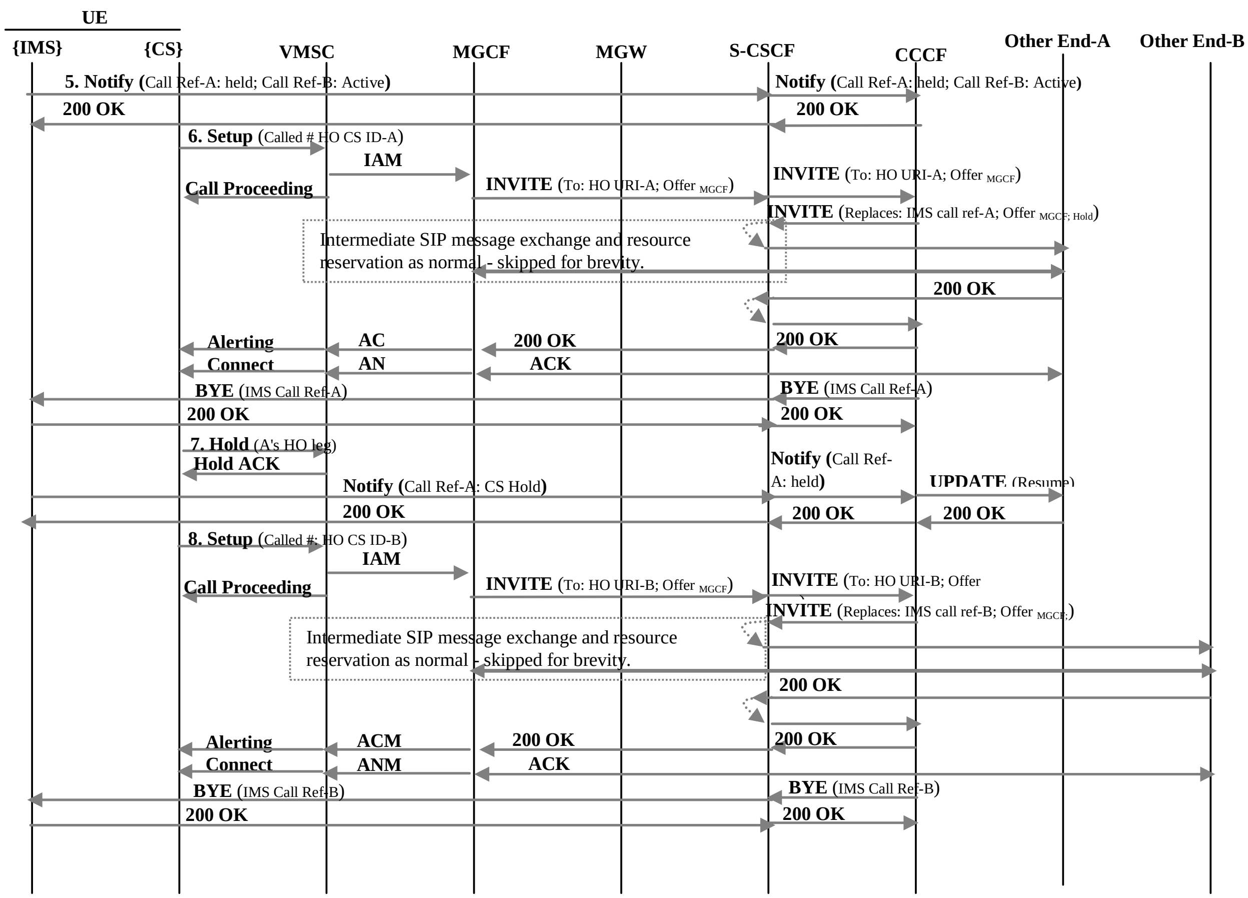


Figure 6.3.7.1-3: VCC of IMS Held and Active sessions

5. Upon detection of border conditions, the UE updates CCCF with the current session state information to be used during the VCC procedure. Session Hold, Active states are passed to CCCF in a Notify with Mobility Event package. The UE uses the session call reference identifiers exchanged during anchoring of IMS sessions to identify individual IMS sessions to CCCF.

6. It should be noted that this message exchange can be avoided if CCCF maintains the session states for the all anchored sessions as it acts a B2BUA agent. This will also eliminate certain race conditions associated with information transfer via additional messaging.

7. The UE performs VCC of the held IMS leg to CS using IMS to CS VCC procedures described in previous sections. CCCF ensures that the held status is maintained at the other end A when transferring the CS IMS user from IMS to CS.

8. The UE holds the CS leg for A's session at the MSC to re-establish the protocol state machine at the MSC. The UE sends a Notify to CCCF informing it of the execution of Hold service in CS Domain so that the media for the other end A can be resumed in IMS. It should be noted that the WLAN coverage may drop anytime after invocation of VCC procedures. CCCF ensures that the other end A's held status is resumed after release of associated IMS leg for the "handing-out" user in the event of a timer expiry for the Notify indicating CS Hold.

The UE subsequently performs VCC of the Active session to the other B to CS using IMS to CS VCC procedures described in previous sections.

The bearer path interruption caused by transfer of Held/Active sessions is the same as the bearer path interruption of transfer of a single session as the media path is affected only when transferring the Active session.

The order of Held and Active sessions needs to be maintained when transitioning between domains to ensure replication of the original service state machine in the target ("handing-in") domain.

6.3.7.1.3 Anchoring of CS Held and Active sessions at CCCF with active IMS Registration

Figure 6.3.7.1-4 below provides a walkthrough of a scenario in which the UE originates an IMS session to the other end A, holds the session toward other end A, and originates a session toward the other end B. It is assumed that the CS-IMS user has active IMS Registration at the time it establishes the CS calls for this walkthrough.

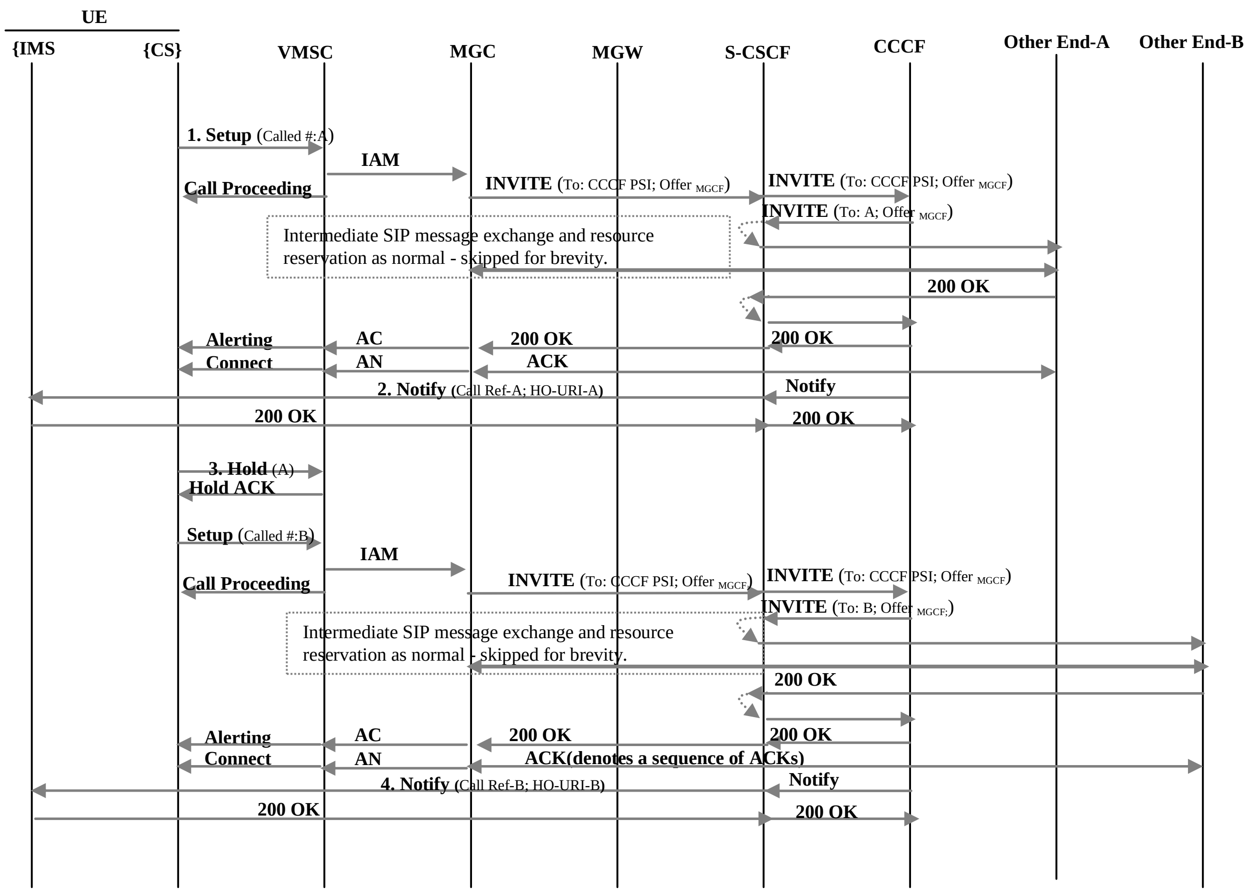


Figure 6.3.7.1-4: Anchoring of CS Held and Active sessions at CCCF

1. As part of IMS registration, CCCF and the UE subscribe to each other for the Mobility Event package. The UE subsequently initiates a CS session.

2. Upon successful execution of a Routing B2BUA function for CS session to the other end A at CCCF, CCCF assigns a unique call reference identifier to the session for identification of the session between the UE and CCCF in subsequent dialogue. It also assigns a SIP URI which can be used for VCC of this session to IMS. These new identifiers are communicated to the UE via a Notify with Mobility Event package.

3. The UE puts session toward the other end A on hold and originates a new CS session towards the other end B.

4. Upon successful execution of a Routing B2BUA function for session to the other end B at CCCF, CCCF assigns a unique call reference identifier to the session for identification of the session between the UE and CCCF in subsequent dialogue. It also assigns a SIP URI which can be used for VCC of this session to CS Domain. These new identifiers are communicated to the UE via a Notify with Mobility Event package.

6.3.7.1.4 CS to IMS VCC of CS Held and Active sessions; IMS active at the time of CS Anchoring

Figure 6.3.7.1-5 below provides a walkthrough of VCC of CS Held and Active sessions established as described in previous section.

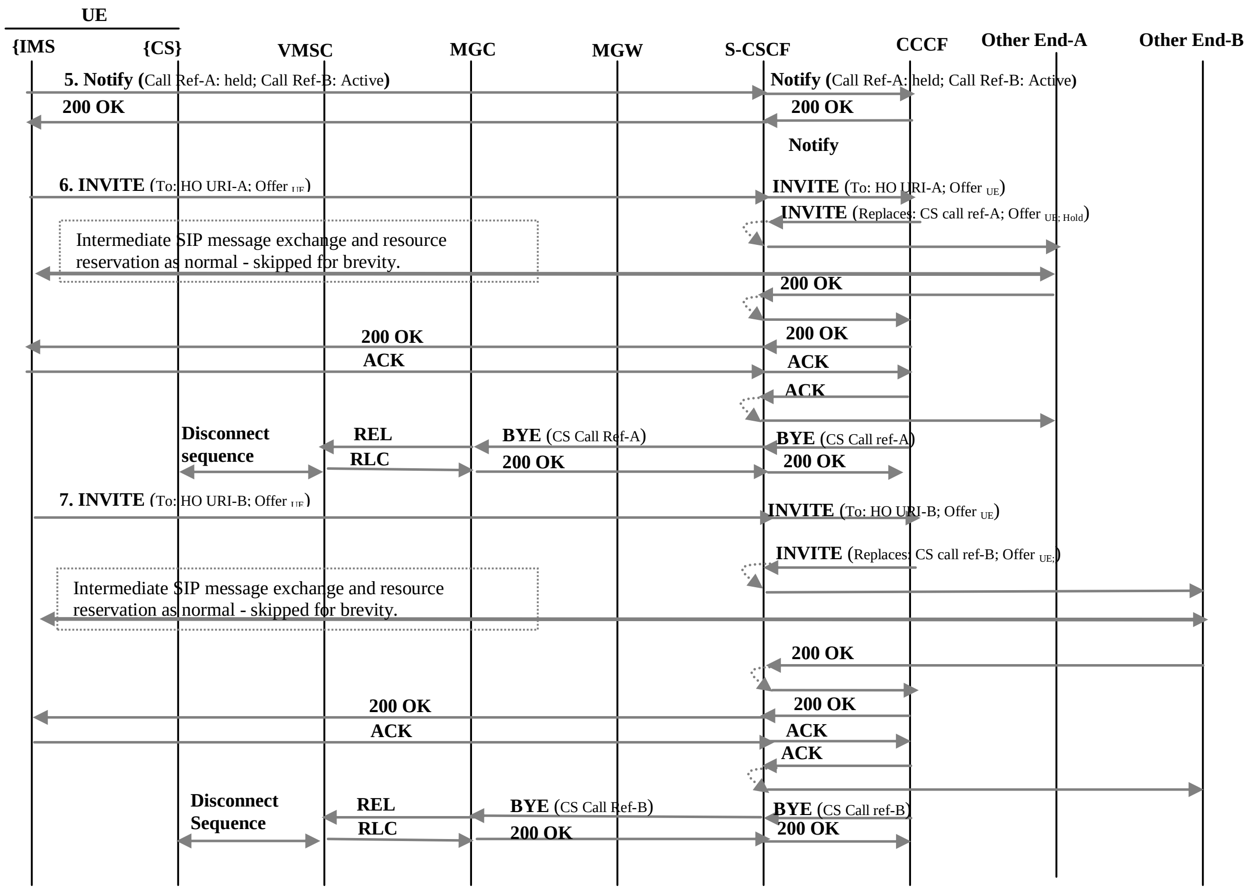


Figure 6.3.7.1-5: CS to IMS VCC of Held and Active sessions; IMS active at CS anchoring

5. Upon detection of border conditions, the UE updates CCCF with the current session state information to be used during the VCC procedure. Session Hold, Active states are passed to CCCF in a Notify with Mobility Event package. The UE uses the session call reference identifiers exchanged during anchoring of CS sessions to identify individual CS sessions to CCCF.

6. The UE performs VCC of the held CS leg to IMS using CS to IMS VCC procedures described in earlier sections. CCCF ensures that the held status is maintained at the other end A when transferring the CS IMS user from IMS to CS.

7. The UE subsequently performs VCC of the Active session to the other party B to IMS using CS to IMS VCC procedures described in earlier sections.

The bearer path interruption caused by transfer of Held/Active sessions is the same as the bearer path interruption of transfer of a single session as the media path is affected only when transferring the Active session.

The order of Held and Active session needs to be maintained when transitioning between domains to ensure replication of the original service state machine in the target ("handing-in") domain.

6.3.7.1.5 Anchoring of CS Held and Active sessions at CCCF without IMS Registration

Figure 6.3.7.1-6 below provides a walkthrough of a scenario in which the UE originates an IMS session to the other end A, holds the session toward other end A, and originates a session toward the other end B. It's assumed that the CS-IMS user is not registered in IMS at the time it establishes the CS calls.

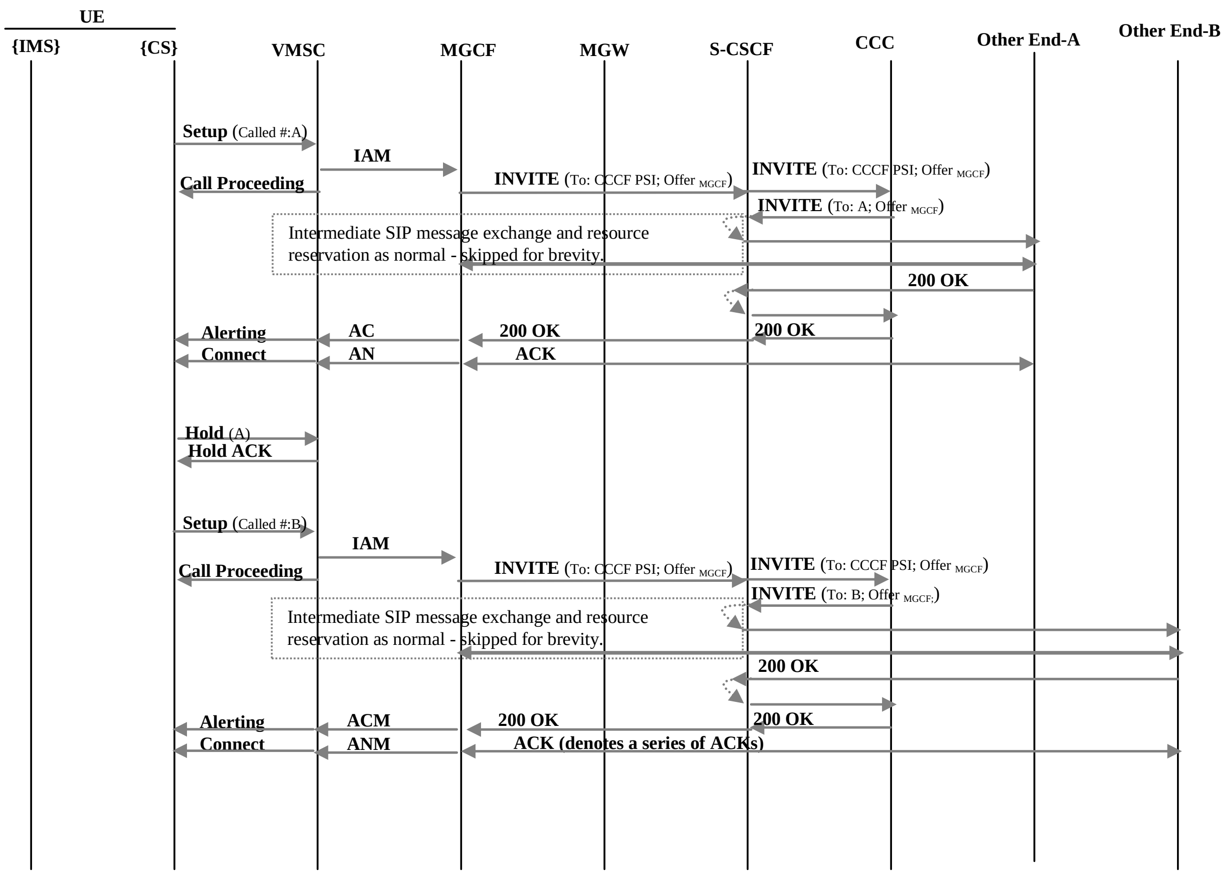


Figure 6.3.7.1-6: Anchoring of CS Held and Active sessions at CCCF

Since the user is not registered in IMS, exchange of session identifier is not possible with the UE. However, CCCF assigns and maintains these session identifiers for communication to the UE upon subsequent IMS Registration.

The rest of the procedure is similar to the procedure described for CS session anchoring with IMS Registration.

6.3.7.1.6 CS to IMS VCC of CS Held and Active sessions; IMS not active at the time of CS Anchoring

Figure 6.3.7.1-7 below provides a walkthrough of VCC of CS Held and Active sessions established as described in previous section.

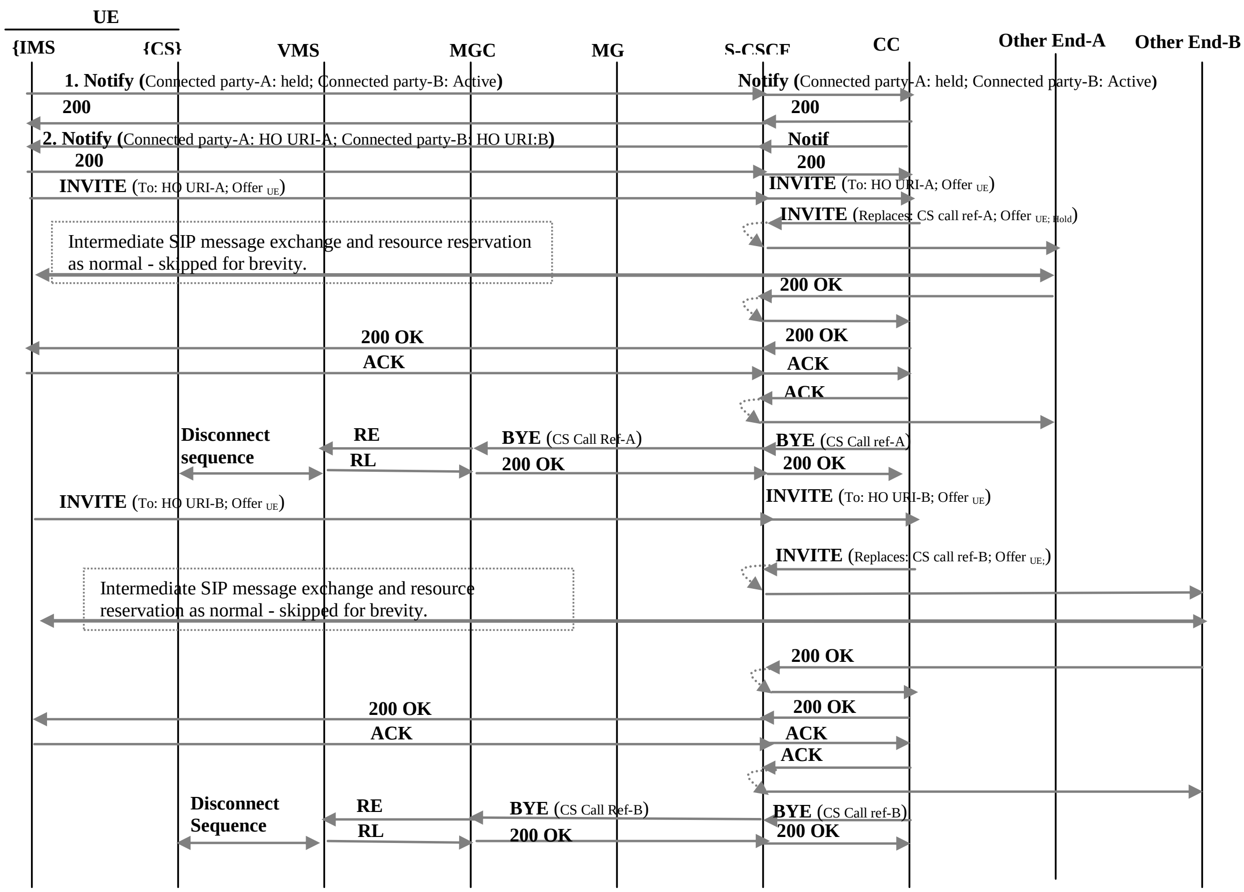


Figure 6.3.7.1-7: CS to IMS VCC of Held and Active sessions; IMS not active at CS anchoring

1. Upon detection of border conditions, the UE performs IMS Registration and updates CCCF with the current session state information to be used during the VCC procedure. Session Hold, Active states are passed to CCCF in a Notify with Mobility Event package. Since the session anchor reference could not be communicated to the UE upon CS anchoring as the IMS Registration was not active at the time of anchoring of CS sessions, the UE uses the connected party addresses to identify individual CS sessions to CCCF.

2. CCCF communicates SIP URIs to be used for VCC to IMS for individual sessions using connected address to identify individual CS sessions to the UE.

The rest of the procedure is same as described in CS to IMS VCC of Held/Active sessions with IMS Registration.

6.3.7.2 Voice Call Continuity for MPTY service

A MPTY service can be transferred across domains to maintain Voice Call Continuity using the same principles as applied to Held/Active sessions in previous sections. However, the VCC procedure for the MPTY session is different from the basic VCC procedure used to transfer Held/Active sessions. The MPTY VCC procedure is performed in 2 steps. In the first step the UE (with assistance of the CCCF) establishes all the sessions in the target ("handing-in") domain, followed by establishment of the conference bridge in the target domain. Then in step 2 the UE requests the CCCF to compete transfer of all connections subsequently followed by release of the conference bridge and associated legs in the source ("handing-out") domain.

The 2-step VCC procedure provides for simultaneous transfer of all connections of a conference session, making bearer path interruption comparable with the bearer path interruption caused by transferring a 2-party call.

It is possible to lose coverage in the source domain in case of VCC from WLAN to CS. However, loss of coverage in the source domain before completion of the VCC procedure is not a concern as the VCC procedure in the target domain continues without assistance from the source domain once the first set of Notify's has been exchanged between the UE and CCCF for exchange of information required to successfully perform VCC.

6.3.7.2.1 Transition of a Multi-party CS Call Session to an IMS-based Service Session

The following diagrams present an example call flows for the 2-step VCC procedure transferring an active multi-party call session in the CS domain to an AS/MRF based Service Session in the IMS domain.

The method used to invoke the multi-party conferencing of the transitioned call legs is the same method for initiating a multi-party call in the IMS domain, such that, after the transfer of the multi-party call session to the IMS domain, the UE continues to have full control of the newly established service session in the AS/MRF.

Pre-VCC Conditions: UE controls conference session in CS domain, UE moves into IMS domain, registers and receives NOTIFY with VCC information from CCCF

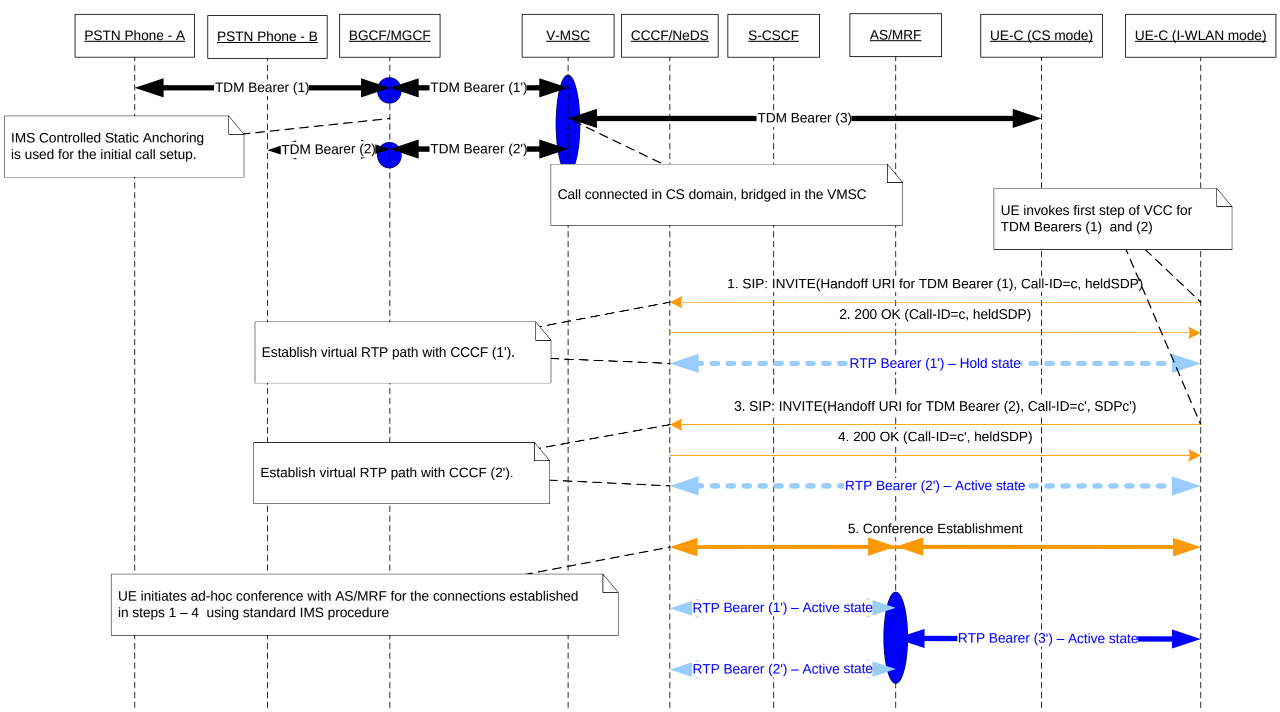


Figure 6.3.7.2.1-1: VCC – Step 1

Walkthrough:

1. UE initiates VCC for the connection with PSTN Phone – A by sending a SIP INVITE to CCCF/NeDS.
2. CCCF/NeDS sends a SIP 200 OK to the UE to place the RTP Bearer (1') on hold.
3. UE initiates VCC for the connection with PSTN Phone – B by sending a SIP INVITE to CCCF/NeDS.

4. CCCF/NeDS sends a SIP 200 OK to the UE to place the RTP Bearer (2') in the Active state.

5. UE and AS/MRF establish ad-hoc conference for the call legs established in steps 1-4 using standard IMS conferencing procedures [15].

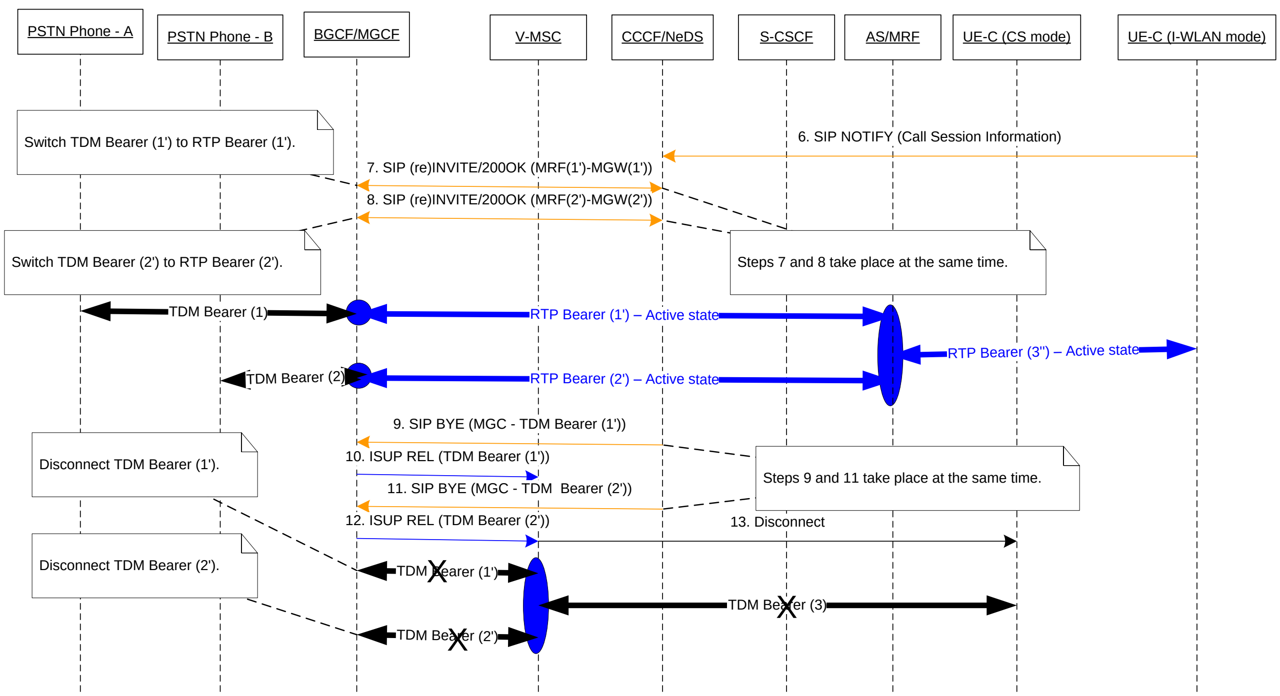


Figure 6.3.7.2.1-2: VCC- Second Step

Walkthrough:

1. Upon conference establishment UE instructs CCCF/NeDS to proceed to the Second Step of VCC

2. CCCF/NeDS (re)INVITEs MGCF and MRF to switch the connection associated with the TDM Bearer (1') to RTP Bearer (1')

3. CCCF/NeDS (re)INVITEs MGCF and MRF to switch the connection associated with the TDM Bearer (2') to RTP Bearer (2')

4. CCCF/NeDS sends a SIP BYE to release the connection associated with TDM Bearer (1')

5. MGCF maps the SIP BYE message to an ISUP REL message.

6. CCCF/NeDS sends a SIP BYE to release the connection associated with TDM Bearer (2')

7. MGCF maps the SIP BYE message to an ISUP REL message, and with other connection with CS side of the UE

8. VMSC sends Disconnect to the UE.

6.3.7.2.2 Transition of a Multi-party IMS-based Service Session to a CS Call Session

There are two potential methods to provide VCC support to a Multi-party AS/MRF-based service session during the transition from IMS to the CS domain. The first method is by maintaining the AS/MRF-based service session in IMS and transitioning the connection to the UE from IMS to CS using the defined VCC procedure. The second method is by creating a new multi-party call session in the CS domain and then transitioning all the connections in the AS/MRF service session to the new call session in CS.

The feasibility of these two methods requires further study.

6.3.7.3 Supplementary Service Implementation Options in the IMS Domain

In the IMS domain, the UE may control the invocation of supplementary services directly. Alternatively, services may be managed by a Services Application Server (AS). The discussion of the VCC procedures discussed in previous sections show the expected type of interactions needed when the UE controls services directly. When services are managed by a Services AS, the interactions are the same in principle in that the UE participates in the VCC procedure. In addition, the UE may need to inform the Services AS of the procedure.

Conferencing is an example service that may be managed by a Services AS that controls multi-port conference bridges. In the case of a session between the UE and the conference bridge controlled by the Services AS, the VCC procedure will control the transition between the CS and IMS domains. The Services AS may not need to be informed of the VCC.

6.3.7.4 IMS service control of VCC subscriber calls

When a service for VCC subscribers is to be realized in IMS, service control for VCC subscribers may be achieved by employing the following scheme that builds on fundamental principles shared with some of the techniques currently being evaluated as part of a R6 study on Combining CS Bearers with IMS (TR 23.899 [23]) in that the control for services is done via SIP/IMS using logic residing in an IMS Application Server, whereas the CS speech bearer control is done via CS signalling (as in TS 24.008 [13]):

1. IMS is used as the native network for VCC subscriber calls. All calls are anchored in IMS, with service control of all calls in IMS.

2. For calls made via the CS Domain network:

   i. A Call Control signalling channel is implicitly established between the UE and the CCCF for management of services in IMS. Two possible signaling protocols are proposed: USSD and SIP, which can be used only when simultaneous CS and IMS access is available, e.g., DTM capability (in case of GERAN access) and MultiRAB capability (in case of UTRAN access), . Considering limited payload size available with USSD (~190 octets), when using the USSD transport, a message protocol will need to be defined between the UE and CCCF which will be converted to SIP messages by the CCCF. One possible mechanism for USSD is to use SIP templates that are exchanged between the UE and the network at registration, to be used later for communication of the call control content required by the network for generation of SIP messages.

   ii. Static anchoring techniques described in section 6.2a are used to establish physical bearer access to the UE via the CS domain network. The circuit domain sees only one session for a VCC user, that is, the session which is used to establish this physical access.

   iii. The CCCF acts as a user agent on behalf of the UE and employs a 3pcc function for establishment and control of VCC user sessions with one or more remote parties.

   iv. All services for CS calls are controlled in IMS through iFC execution at the S-CSCF resulting in appropriate Application Server invocation with communication to the UE provided via the implicit call control signalling channel between the UE and the CCCF.

6.3.7.4.1 USSD enabled Service Control in IMS-Signalling and Bearer Architecture

Depicted in Figure 6.3.7.4-1 below are the signalling and bearer paths for a VCC subscriber engaged in a CS call when USSD is used for enablement of IMS service control of CS calls.

A physical bearer access is established for the UE, by using static anchoring techniques described in Section 6.2a over CS call control signalling channel. Additionally, an implicit call control signalling channel is established between the UE and the CCCF via USSD signalling exchange through the gsmSCF component of CCCF/NeDS. A USSD Gateway may be used in absence of CAMEL capability in the home CS network. A USSD/SIP translations function (USG: USSD-SIP-Gateway) is employed by the network for generation of SIP templates and mapping between USSD and SIP for communication between the CCCF and the UE. The CCCF manages and controls the CS session for the UE by employing a 3pcc function.

Note that the current specifications only allow execution of U-CSI and UG-CSI in the subscriber's home CS network, however, the USSD signalling path may be optimized by enhancements to allow U-CSI and UG-CSI execution in the visited CS network to facilitate a direct signalling path between the VMSC and the gsmSCF component of CCCF/NeDS.

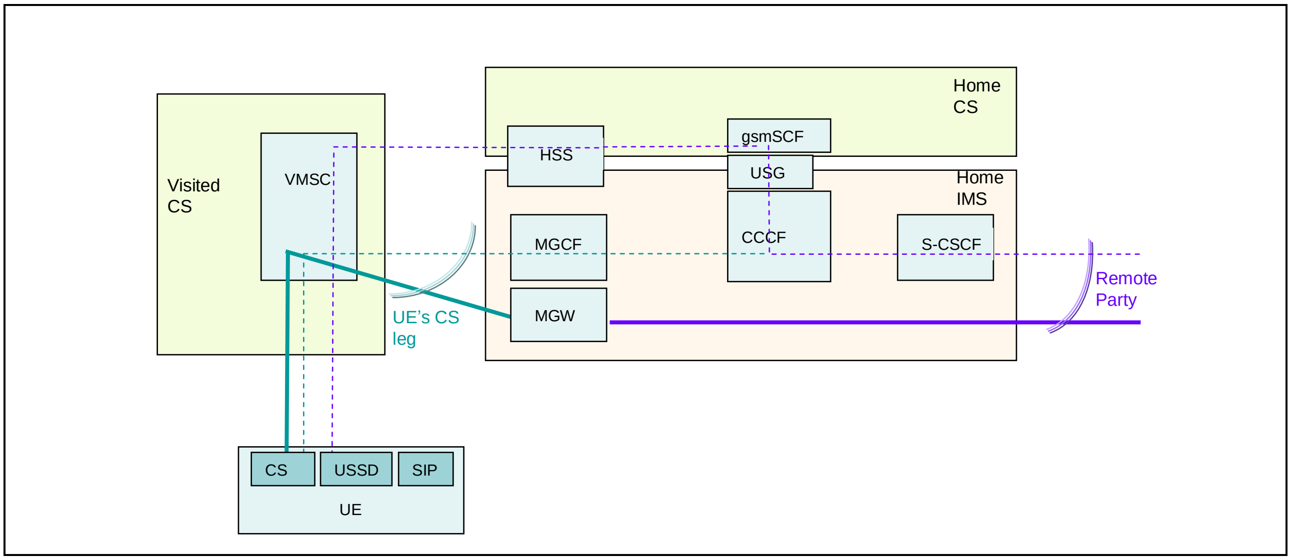


Figure 6.3.7.4-1 CS call signaling and bearer architecture with USSD enabled service control

6.3.7.4.2 SIP enabled Service Control of CS calls in IMS-Signalling and Bearer Architecture

Figure 6.3.7.4-2 below illustrates signalling and bearer architecture for a VCC subscriber engaged in a CS call when SIP is used for enablement of IMS service control of CS calls.

A physical bearer access is established for the UE via the CS domain network by using static anchoring techniques described in Section 6.2a over CS call control signalling channel. Additionally, an implicit call control signalling channel is established between the UE and the CCCF via SIP signalling exchange. The CCCF manages and controls the CS session for the UE by employing a 3pcc function.

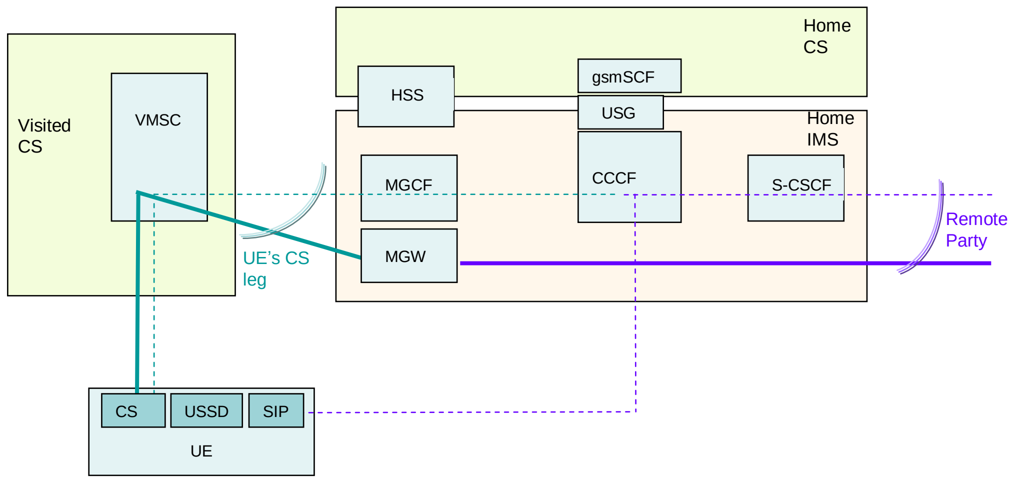


Figure 6.3.7.4-2: CS call signaling and bearer architecture with SIP enabled service control

6.3.7.5 Supplementary Service Impact summary

Table 6.3.7.5-1: Supplementary Service Impact of CCCF provides preliminary impact statements for commonly used supplementary services.

Table 6.3.7.5-1: Supplementary Service Impact of CCCF

Supplementary Service	Impact statement
Home Network Call Forwarding (CFU, CFNRc-HLR detached)	CCCF prevents unnecessary anchoring of the call by examining user's Call forwarding profile provided by the HSS
Visited Network Call Forwarding (CFB, CFNRy, CFNRc-VLR detached)	Call forwarding leg may be unnecessarily anchored as CCCF is unaware of user's availability
Incoming Call Barring	No impact.
Outgoing Call Barring	Outgoing Call Barring services should be provided in IMS for VCC users as some flavors of CS Outgoing Call Barring may bar the CCCF PSI used for execution of IMS to CS VCC. Outgoing call barring impact for use of VCC subscriber's SIM with a non-VCC capable UE is FFS.
Calling Line ID Presentation (CLIP)	CCCF ensures delivery of the originating party's CLIP information to the CS-IMS user when CS incoming calls are anchored via CCCF. CLIP presentation for target leg to be blocked by CCCF with appropriate use of screening indicators. Interactions with CLIP Override are for further study.
Connected Line Identity Presentation (COLP)	CCCF ensures delivery of the actual connected party's COLP information to the CS-IMS user when CS originating calls are anchored via CCCF. COLP presentation to the user is required to be blocked for the target leg, preferably at UE.
Closed User Group	CCCF PSI is required to be included in subscriber's Closed User Group profile.
Call Hold/Retrieve	No impact.
Call Wait	No impact.
Multi-party	With distributed service control model, transfer of Multi-party service upon IMS to CS VCC is possible only for Multi-party flavours supported by the CS domain. For example, it will not be possible to transfer an ad hoc IMS conference to CS with more than 6 parties.
Explicit Call Transfer	No impact.
Advice of charging	The CCCF preserves prepaid status
Freephone service	No impact.
Reverse charging	The CCCF preserves the reverse charging indication.
User to user signalling	IMS-ISUP UUS interworking to be defined for use of IMS as transit network.

NOTE: The impact in this table refers to the impact of VCC to the service behaviour; the standardization impact specific to the VCC application is documented in other subsections of 6.3.7.

The Supplementary Service subscriber profile synchronization between the CS and IMS subscriptions to be analyzed outside of the VCC study is critical in preserving seamless user experience during VCC across CS domain and IMS.

6.3.7.6 TISPAN's recommendation for Mandatory, Recommended, and Optional Supplementary Services in the IMS Domain

TISPAN has recommended a set of basic PSTN/ISDN simulation services for Release 1 being defined in DTS TISPAN-01002-NGN [21]. TISPAN has grouped the recommended services into three categories.

- Mandatory (regulatory requirements involved);
- Strictly recommended (should be provided);
- Optional.

TISPAN has defined the following set of features as mandatory:

- Orig ID Presentation;
- Orig ID Restriction;
- Term ID Presentation;
- Term ID Restriction;
- Malicious Call ID;
- Anonymous Call Rejection.

VCC has no impact upon the operation of the mandatory features in IMS. Originating ID Restriction applies between users and does not apply between the UE and the CCCF AS.

TISPAN has defined the following set of features as recommended:

- Communication Diversion;
- Communication Waiting;
- Communication Hold;
- Communication Barring;
- Completion of Communications to Busy Subscriber;
- Follow Me;
- Message Waiting Indicator.

Of the recommended features, VCC impacts Communications Diversion, Communication Waiting, Communication Hold, and Communications Barring as indicated in the previous table as Call Forwarding, Call Waiting, Call Hold, and Outgoing Call Barring, respectively.

The following features are considered as optional by TISPAN:

- Conference;
- AoC;
- CUG;

- Fixed Destination Communication;
- Inhibition of Incoming Forwarded Communications;
- DDI;
- ECT;
- Trunk Hunting.

Of the optional features, VCC impacts Conference, Closed User Group, and Explicit Call Transfer as indicated in the previous table. Fixed Destination Communication will be impacted in the same way as Closed User Group is impacted in that the CCCF PSI must be an allowed Fixed Destination.

6.3.8 Call Continuity within 3GPP radio

Table 6.3.8-1 below provides possible scenarios for call continuity within 3GPP radio and available solutions that provide service continuity in these scenarios.

Table 6.3.8-1: Call Continuity within 3GPP radio

Source >	**3G PS**	**3G CS**	**2G CS**
Target >	**Call Continuity Solutions**		
3G PS and CS available.	Classical PS Handover	Classical CS Handover	Classical CS Handover
3G PS registered; CS not available.	Invalid scenario	CS to PS VCC	Invalid scenario
3G PS registered; CS available.	Invalid scenario	Classical CS Handover	Invalid scenario
3G PS available but not registered; CS not available.	Classical PS Handover	FFS (may be possible to use VCC)	Solutions presented in this paper
3G CS registered; PS not available	PS to CS VCC	Invalid scenario	Invalid scenario
3G CS available but not registered; PS not available.	FFS (may be possible to use VCC)	Classical CS Handover	Classical CS Handover
2G CS; Voice not available on 2G PS	Solutions presented in this document	Classical CS Handover	Classical Handover

NOTE: It is assumed that Voice service is available in 3G PS for scenarios listed in this table.

Classical Handover procedures are used to transition user's active calls when roaming within the same domain.

The Voice Call Continuity between CS domain and IMS provides a solution for transition of user's services during inter-domain roaming between CS domain and IM Subsystem. The basic IMS Controlled Model based VCC solution requires simultaneous registration in the CS domain and the IMS at the time of initialization of the call continuity procedure. The basic IMS Controlled Model based VCC is recommended for scenarios wherever this is possible.

When direct application of IMS Controlled Model based VCC is not possible, Classical Handover may be used in conjunction with inter-domain transition via VCC to complete the overall voice call continuity procedure as described below:

3G PS voice to 2G CS voice:

1. UE detects a need to perform a 3G to 2G transition.
2. Before this takes place (i.e. while still in 3G coverage), it initiates VCC to 3G CS.
3. Normal 3G CS to 2G CS Handover occurs to transition to 2G CS.

2G CS voice to 3G PS voice:

1. UE waits for the network to perform the classical 2G CS to 3G CS Handover.

2. Upon completion of 2G CS to 3G CS Handover, UE initiates VCC to transition to 3G PS.

6.3.8a Security Impact of Handovers between the CS Domain and the IMS for IMS Controlled Method

Before a UE operating in the CS Domain can obtain service from IMS Domain, the UE must register with the IMS domain. For the IMS control method, the handover begins with a new session establishment originated by the dual mode UE to the CCCF/NeDS. Since the keys are established during IMS registration, the new session will be established securely.

Before a UE operating in the IMS Domain can obtain service from CS Domain, the UE must have performed a location update with the CS domain. For the IMS control method, the handover begins with a new call originated by the dual mode UE to the CCCF/NeDS. The new call will be established securely using the standard authentication procedure.

6.3.8b Charging Impact of Handovers Between the CS Domain and the IMS Domain for IMS Controlled Method

VCC requires business agreements that are expected to impact the online charging systems for the operators. For static anchoring IMS, the impacts are expected to be higher than for the ODC.

6.3.8b.1 General Description

In IMS controlled static anchoring option, all of the CS origination and termination call of the VCC user will be routed to his home IMS domain to get the CCCF involved to control the bearer path since the initial call setup. And then during the call continuity procedure, the CCCF shall reconnect with the VCC user in the handing-in domain, initiate the re-negotiation with the other end to change the direction of media exchange, and release the original connection with the VCC user in the handing-out domain.

The principle of charging strategy applied in the static anchoring option of IMS controlled alternative is that:

1. Provide cohesive charging records with a complete call continuity history for the duration of a voice session by CCCF.
2. For cases of CS origination and CS termination, correlate the charging records generated in CS and IMS for the single CS origination/termination, to avoid double billing to the subscriber. The information used for correlating these charging records is CCCF/NeDS PSI.
3. Treat the charging records generated in the handling-in domain for the call/session established during the inter domain call continuity as call continuity legs, and therefore do not impact the direction of the initial call/session for the purpose of charging.
4. Keep the start of charging in the handing-in domain align with the stop of charging in the handing-out domain, to avoid double billing to the subscriber in these period of time.

The detailed description for CS origination, CS termination, IMS origination, IMS termination, CS leg and IMS leg established for IMS to CS call continuity are provided separately in the following sections.

For VCC online charging, the following requirements apply:

1. Completeness and correctness of charging;
2. Avoid possible double billing in IMS and CS domains;

Applicable charging of VCC invocations, the information related to the initial call establishment as well as the information related to the call continuity procedure should be reported to OCS for correct credit control purpose.

6.3.8b.2 Charging Strategy Analysis

6.3.8b.2.1 CS Origination

Since all of the CS origination call shall be routed to the IMS domain to get the control of CCCF locates in IMS in this solution, for a single CS origination call, the routing path is from CS domain to IMS domain, and both CS and IMS domains shall generate the charging record for it as usual.

To avoid double billing the subscriber, the additional requirement is that, these generated charging records shall be correlated and be identified that they belong to the same call. The most interrelated charging records include:

1. The origination charging record for the initial CS origination captured by the originating MSC as usual, including the CCCF/NeDS PSI or the original called number prefixed with steering digits as the information of origination routing control, along with original called number.

2. Charging records captured by the CCCF during initial CS call set up as usual, including:

   a) A termination charging record for the termination session to the CCCF/NeDS PSI.

   b) An origination charging record for the origination session from the CCCF/NeDS PSI to the original called party.

The information used for correlating these charging records could be CCCF/NeDS PSI.

6.3.8b.2.2 CS Termination

Since all of the CS termination call shall be routed to the IMS domain to get the control of CCCF in IMS in this option, for a single CS termination call, the routing path is from the GMSC in CS domain to IMS domain, and finally return to CS domain to connect the user, both CS and IMS domains shall generate the charging record for it as usual.

To avoid double billing to the subscriber, the additional requirement is that, these generated charging records shall be correlated and be identified that they are belong to the same call. The most interrelated charging records includes:

1. The termination charging record captured by the gateway MSC for the initial CS termination as usual, including the IMRN which include the CCCF/NeDS PSI information, as the information of termination routing control, along with original called number, i.e. the number of the VCC user.

2. The termination charging record captured by the terminating MSC for the initial CS termination as usual, from the CCCF/NeDS PSI to the original called party, i.e. the VCC user.

3. Charging records captured by the CCCF during initial CS call set up as usual:

   a) A termination charging record for the termination session to the CCCF/NeDS PSI.

   b) An origination charging record for the origination session from the CCCF/NeDS PSI to the original called party, i.e. the VCC user.

The information used for correlating these charging records could be CCCF/NeDS PSI.

6.3.8b.2.3 IMS Origination

For an IMS origination, there is no inter-domains routing, so only the IMS domain shall generate the origination charging records as usual.

The most interrelated charging record includes:

1. A termination charging record generated at the CCCF for the termination session to the CCCF for insertion of B2BUA at CCCF.

2. An origination charging record generated at the CCCF for the origination session from the CCCF to the original called party.

6.3.8b.2.4 IMS Termination

For an IMS termination, there is no inter-domains routing, so only the IMS domain shall generate the termination charging records as usual; the charging strategy is almost the same as IMS origination, excepting the IMS domain shall generate the termination charging records as usual this time.

6.3.8b.2.5 CS leg established for IMS to CS call continuity

CCCF provides cohesive billing with a complete Call continuity history for the duration of the voice session.

During call continuity procedure in this option, the newly established session will always be established with a direction from the VCC user to the CCCF, no matter which side (e.g. origination or termination) in the initial call/session the VCC user is in.

Since different rate may be applied for these different sides, the call/session established during the inter domain call continuity should be captured as call continuity legs and therefore do not impact the direction of the initial call/session for the purpose of charging. The interrelated charging record includes:

1. Charging records generated in the CS domain, which needs to be charged as a call continuity leg instead of as a CS origination. With the fact that the call is made directly to a CCCF/NeDS PSI indicating that.

2. Charging records generated at CCCF for call continuity leg, which used as parts of a complete call continuity history for the duration of the voice session.

Furthermore, since the new connection between the VCC user and CCCF in handing-in domain may usually be established before the release of the old connection between them in handing-out domain, there is a period of time that these two connections exist synchronously.

To avoid double billing to the subscriber in these period of time, the start of charging in the handing-in domain shall align with the stop of charging in the handing-out domain.

6.3.8b.2.6 IMS leg established for CS to IMS call continuity

The charging strategy is almost the same as "CS leg established for IMS to CS call continuity", excepting that IMS domain generates the charging records for the call continuity leg instead of the CS domain, the fact that the IMS session is made directly to a CCCF/NeDS PSI indicates that it's call continuity leg but not an IMS origination.

6.3.8b.3 Accounting Strategy Analysis

Accounting is "the process of apportioning charges between the home environment, serving network and the user". The discussion in the former section is about the charging of the user; while this section provides the principle of settlement between different operators.

According to the Architectural Requirements,

1. The CS domain and IMS domain may belong to separate services providers.

2. The IP CAN for IMS access and the IMS Core Network may belong to separate services providers.

So, to perform the settlement between operators, the following criteria shall be applied:

1. Provide cohesive charging records with a complete call continuity history for the duration of a voice session by CCCF.

2. Use the charging records for call continuity leg generated in CS/IMS domain and the charging records generated in MGCF performing CS-IMS interworking, taking the complete call continuity history described above as reference, to perform the settlement between the providers of CS domain and IMS domain.

Use the access network information in IMS charging records, taking the complete call continuity history described above as reference, to perform the settlement between the providers of IPCAN and IMS Core Network.

6.3.9 Analysis of the IMS Controlled model

This clause presents an analysis of the IMS Controlled model as an architectural alternative for the Voice call Continuity service.

6.3.9.1 Evaluation of the IMS Controlled model

The VCC solution based on the IMS Controlled model employs an architecture that may be applied to any set of requirements criteria such as transition within 3GPP radio or between 3GPP radio and I-WLAN; single registration vs. dual registration; VCC across similar or dissimilar call control protocols.

The architecture model uses the serving domain to provide Basic Call Control with Supplementary Services Control optionally available in the serving domain or the IMS based on the operator policy. The VCC control is centralized in IMS for management of bearer and signalling components and consolidation of charging/accounting upon transfer of sessions in either direction.

The VCC procedure executes in the call control layer utilizing standard Radio layer and MM layer procedures to establish the call context in the new domain and release the call context in the old domain associated with VCC.

The benefits of this proposal include:

1. The IMS Controlled model provides an access agnostic approach with cohesive techniques applied for call continuity in both directions, CS to IMS and IMS to CS; providing solutions for VCC between CS and IMS accessed over I-WLAN and 3G PS.
2. The solution provides centralized control for all executions of VCC in both directions:
    a. CS to IMS
    b. IMS to CS
    c. Subsequent
3. Subsequent call continuity procedures are handled in an efficient manner. Such call continuity procedures do not result in daisy chain effect that results from use of disjoint techniques applied in CS to IMS and IMS to CS directions.
4. The solution facilitates simple configuration and deployment.
5. All VCC procedures are executed in the application layer without any dependency on the radio layer handover procedures.
6. Static anchoring of CS calls in IMS:
    a. Enables delivery of richer service set to the user when roaming in CS domain.
    b. Provides a migration path to early MMTel.
    c. Enables comprehensive billing with complete call continuity history.
7. The solution is based on a service in user's home IMS network; therefore, VCC service delivery to the user is not impacted when roaming in non supporting networks.
8. The solution applies to all roaming situations and has no restrictions on the location of the I-WLAN or CS MSC.
9. The VCC procedure is seamless to the user as the user can continue to control the supplementary services after the VCC procedures are complete.

10. Since the information required to complete the VCC procedure is exchanged between the UE and the CCCF via the pre-Notification procedure at the beginning of the execution of the VCC procedure, loss of radio/I-WLAN coverage in the transferring-out domain any time after the exchange of initial information exchange does not affect completion of the VCC procedure.

11. Converged services that are supported by both the CS domain and the IMS domain are supported during the VCC procedures. Converged services are those that are available for GSM/UMTS and are recommended by TISPAN for operation in the IMS domain for Release 1.

12. The solution does not impact performance or any other aspects of legacy terminals.

The drawbacks of this proposal include:

1. All CS calls made by VCC subscribers are routed via the user's home IMS network or a local IMS network, which may result in an extra hop for CS calls that otherwise might have remained in the local CS network, resulting in minor inefficiency in call setup delays and network resource usage when setting up such calls for VCC subscribers. However, it should be noted that this inefficiency is immediately offset by the resource usage inefficiency resulting from a VCC to IMS requiring that a trunk between the serving system and the IMS be set up if the services remain anchored in the CS domain even when the user is being served in IMS.

2. When using distributed service control in CS and IMS, VCC procedure setup time is proportionate to the number of sessions being transferred as the transfer happens serially. However, it should be noted that speech interruption is caused only during transfer of the active session(s); and the VCC procedure continues successfully even if coverage is lost in transferring-out domain during VCC procedure execution.

Please refer to section 6.3.7.5 for impact to Supplementary Services.

6.3a Extended VCC IMS controlled architecture

The extended VCC IMS controlled architecture is the same as the IMS controlled architecture, except that when the user is connected via a CS domain access, the service execution remains in the IMS domain.

The architecture contains a AGCF "Access Gateway Control Function". The AGCF confirms to the following:

1. Is a SIP user agent

2. Behaves as a UE towards the IMS network for calls connected via the CS domain.

Editors Note: The impact on the MSC is yet to be determined.

6.3a.1 IMS registration

The IMS registration for the extended IMS controlled architecture is the same as for the common IMS registration that is found in section 6.2a.1.2 "IMS Domain Registration".

6.3a.2 IMS origination

The IMS origination for the extended IMS controlled architecture is the same as for the common IMS origination that is found in section 6.2a.2.1 "UE Origination from IMS Domain".

6.3a.3 IMS termination

The IMS termination for the extended IMS controlled architecture is the same as for the common IMS termination that is found in section 6.2a.3.1 "IMS termination".

6.3a.4 Attaching/registering via the CS network

FFS.

6.3a.5 CS origination

The CS origination for the extended IMS controlled architecture is the same as for the IMS controlled origination that is found in section 6.3.4.2 with the exception that the AGCF is used in place of the MGCF.

6.3a.6 CS termination

The CS termination for the extended IMS controlled architecture is the same as for the IMS controlled termination that is found in section 6.3.5.2 with the exception that the AGCF is used in place of the MGCF.

6.3a.7 Call Continuity Scenarios (dual radio)

The call continuity scenarios for the extended IMS controlled architecture are shown in clauses 6.3a.7.1 and 6.3a.7.2. In both of these scenarios show the cases where the terminal has registered both in the IMS domain and via the CS domain.

6.3a.7.1 IMS to CS domain call continuity

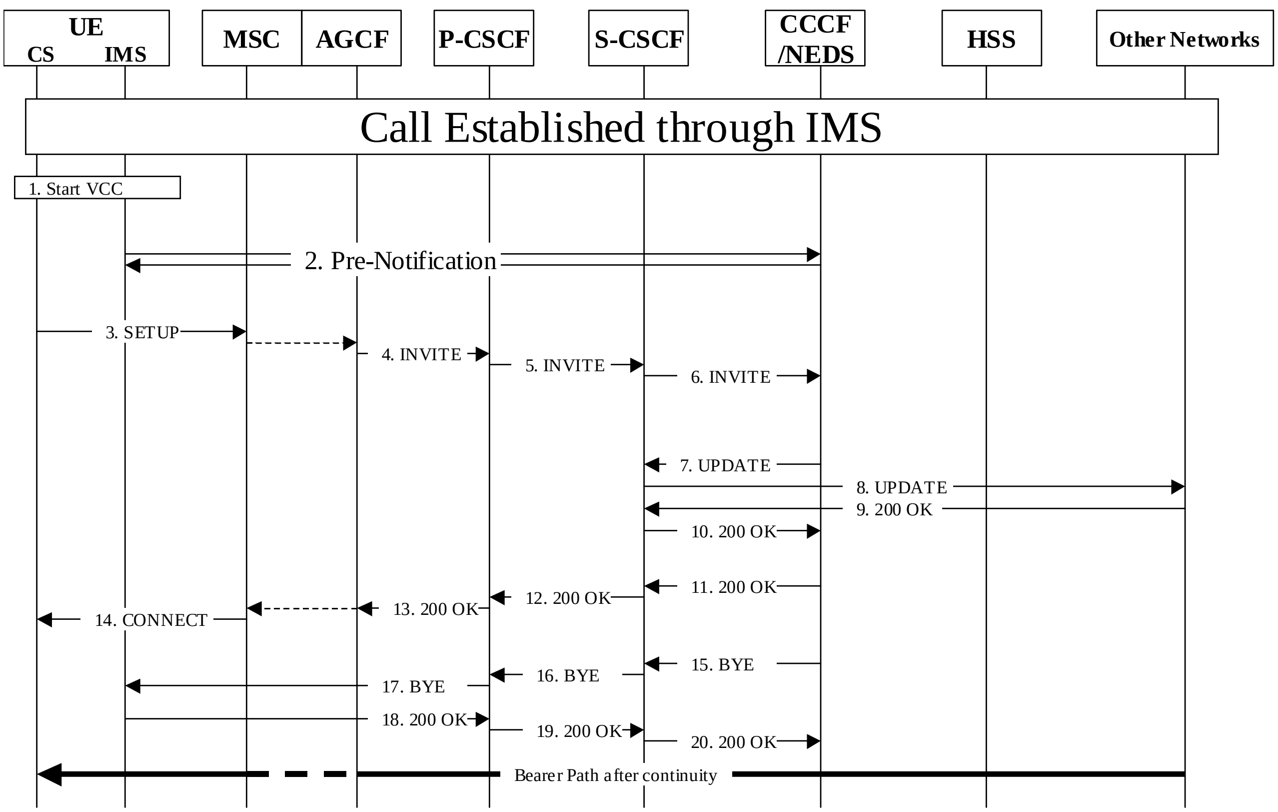


Figure 6.3a-1: IMS to CS domain call continuity

For call continuity from the IMS to the CS domain, the following steps are followed:

1. When the terminal detects that the call continuity procedure is to occur, the terminal shall initiate the call continuity procedure

2. The UE contacts the CCCF. The CCCF shall provide the PSI to be used for the call continuity procedure. In this case, the PSI shall be able to be expressed in a format usable on the CS domain.

3. The UE shall send a SETUP message to the MSC. The SETUP message shall contain the CS domain representation of the PSI retrieved in step 2 as the called party number in the SETUP message. The MSC forwards this to the AGCF.

4.-6. The AGCF forwards the INVITE, which is routed to the CCF as per normal IMS routing procedures.

7-8. The CCCF sends an Update to the other network with the media transport address of the AGCF that was received in step 6.

9-10. The other network responds, confirming whether the UPDATE was successful or not

10-13. The CCCF acknowledges the INVITE (steps 4-6) and includes the transport address of the peer network

14. The MSC completes the bearer establishment in the CS domain by sending the UE a CONNECT message.

15-20. The CCCF release the original IMS session between the UE and the CCCF.

6.3a.7.2 CS domain to IMS call continuity

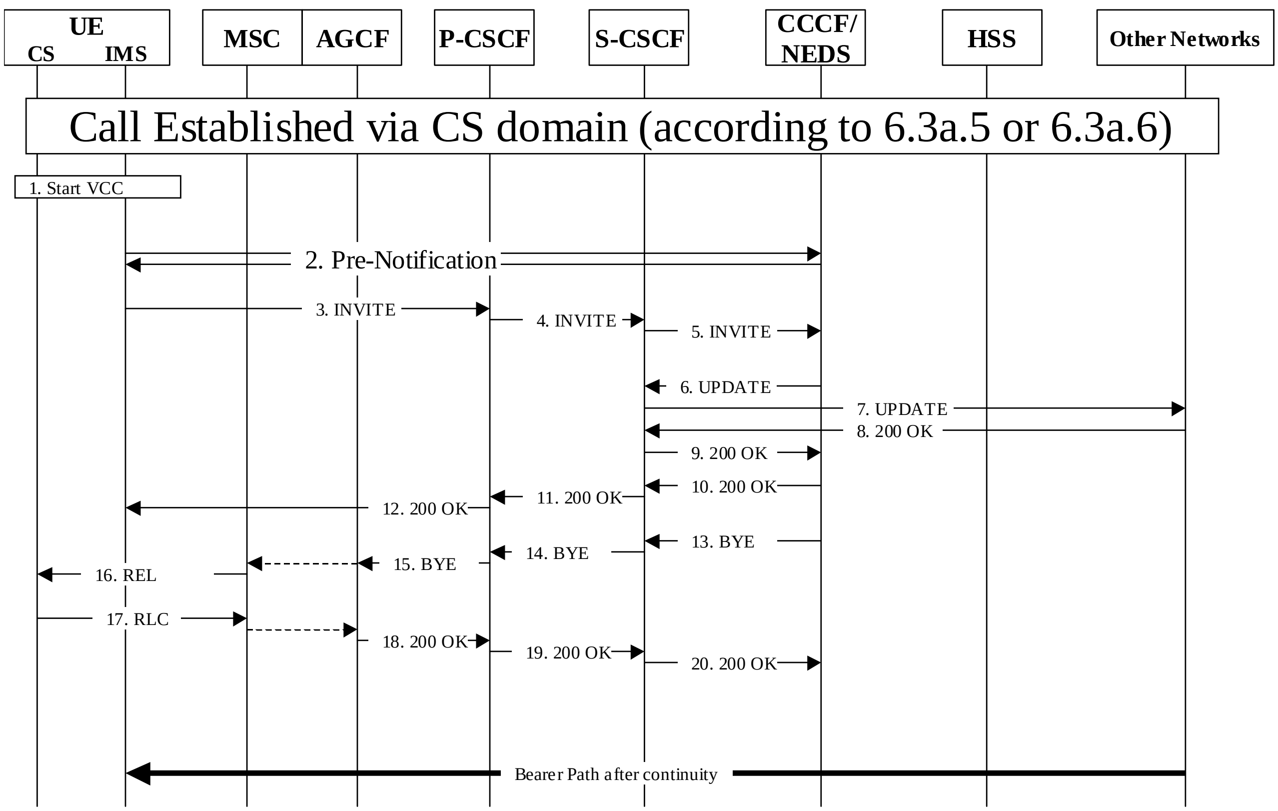


Figure 6.3a-2: CS to IMS domain call continuity

For call continuity from the CS domain to the IMS, the following steps are followed:

1. When the terminal detects that the call continuity procedure is to occur, the terminal shall initiate the call continuity procedure.

2. The UE contacts the CCCF. The CCCF shall provide the PSI to be used for the call continuity procedure.

3-5. The UE shall send an INVITE message P-CSCF which is routed to the CCCF as per normal IMS routing procedures.

6-7. The CCCF sends an Update to the other network with the media transport address of the AGCF that was received in step 6.

8-9. The other network responds, confirming whether the UPDATE was successful or not.

10-12. The CCCF acknowledges the INVITE (steps 3-5) and includes the transport address of the peer network.

13-20. The CCCF release the original IMS session between the UE and the CCCF.

6.3a.8 Call continuity procedures (single radio)

The flows for call continuity procedures (single radio) is the same as the IMS controlled that is found in 6.3.8.1.2.1.1 and 6.3.8.1.2.1.2 with the exception that the AGCF is used instead of the MGCF.

6.3a.9 Impact on supplementary services

FFS.

6.3a.10 Impact on accounting

For an MSC in the home network, there is no impact on accounting.

When the user is roaming, consideration is required to be given to the CS call that is setup in this manner. The CS call should not necessarily be considered as the start of voice call, but could also be considered as a continuation of an existing voice call. There is no impact in the case that the accounting agreements based upon bearer usage.

6.4 Service Continuity Model: Original Domain Control Model

6.4.1 General Description

6.4.1.1 Reference Architecture Model

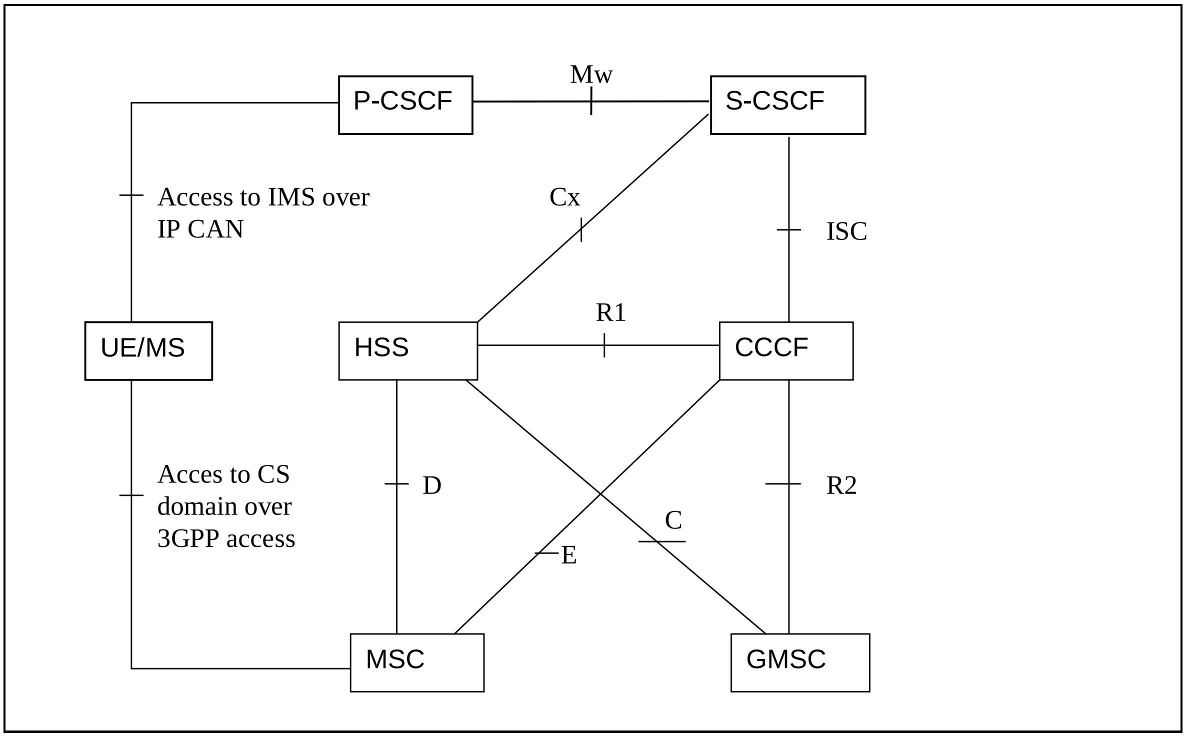


Figure 6.4.1.1-1: Reference Architecture Diagram

The anchored call control model follows the existing GSM inter-MSC handover model as defined in TS23.009 [4] Handover Procedures. To allow the HLR and MSC interworking, a Call Continuity Control Function (CCCF) logical functional element is introduced into the architecture.

6.4.1.2 Call Continuity Control Function

The Call Continuity Control Function (CCCF) Function manages the overall handoff process. It interfaces to the S-CSCF and has a role similar to an IMS Application Server. It interfaces to the CS Core Network

The CCCF consists the following functionalities:

- Subscribe to a UE for event notifications or publish information to a UE after registration.
- Perform Network Domain Selection function.
- Perform voice call continuity function (e.g. handover legs with MAP procedures).
- Assign MSRN on routing to IMS.
- Communicate with MSC, HSS, and CSCF.
- May generate charging records.

6.4.1.3 Call Continuity Control Function Reference Points

To support the VCC service, the CCCF uses the following reference points:

Reference Point ISC (S-CSCF - CCCF):

- Use for the VCC procedure

Reference Point R1 (CCCF - HSS):

- Use to determine routing of the call via the MAP C interface

Interface D (MSC - HSS):

- Use for CS registration

Interface E (MSC - CCCF):

- Use for call continuity between IMS to CS and CS to IMS

Reference Point R2 (GMSC - CCCF):

- Use for invoking NeDS function

6.4.6 Call continuity scenarios

6.4.6.1 CS to IMS call continuity

This Use Case illustrates the architecture used for handing off an active voice call from a GSM/UMTS system to a VoIP call on a WLAN/IMS system.

Figure 6.4.6.1-1 shows the bearer path for a UE in a CS domain call with a PSTN user; both before and after VCC procedures move the call to the IMS domain. The original call is from the UE to the MSC to the PSTN. After the UE moves to the IMS domain, the call leg from the MSC to the PSTN user remains, but the call leg from the MSC to the UE is now from the MSC to an IMS-MGW to the UE. Figure 6.4.6.1-2 shows the case where the UE is in a CS domain call to a UE in the IMS domain. After the UE moves to the IMS domain, the only change is to change the call leg from the MSC to the UE. In both cases, the MSC anchors the call.

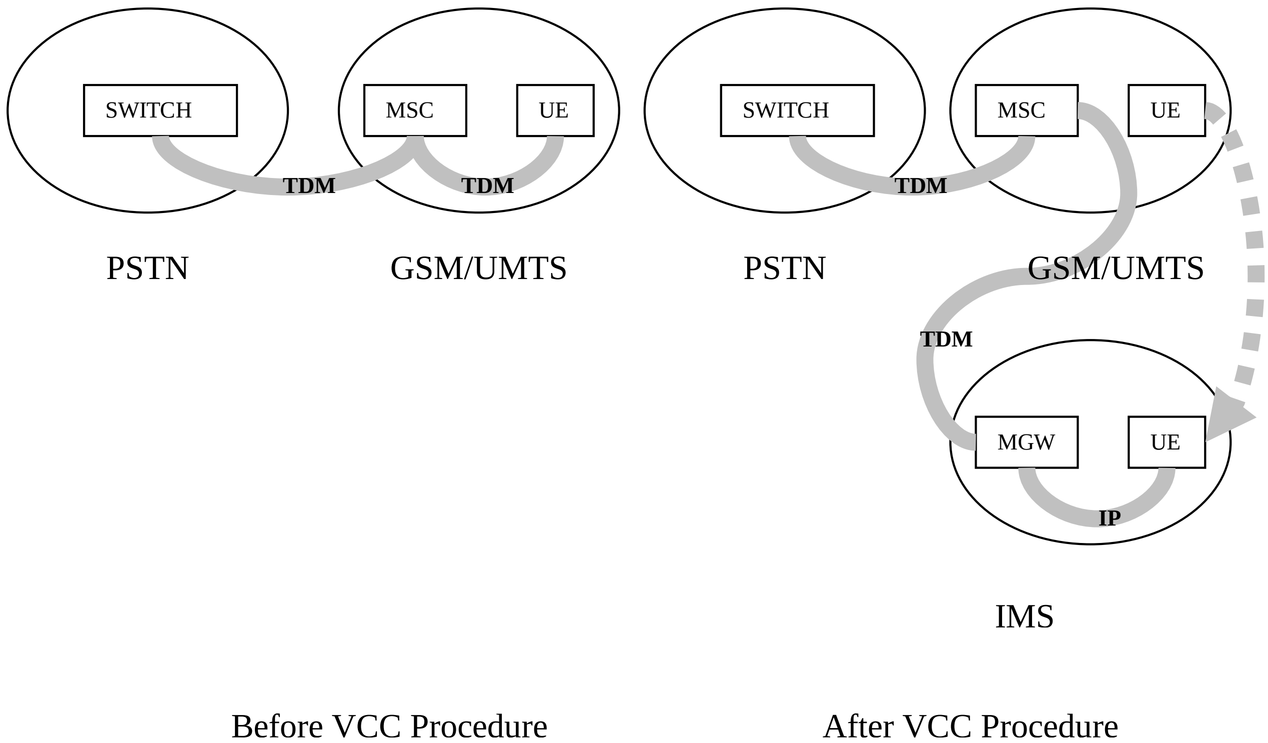


Figure 6.4.6.1-1: Bearer Path: CS domain UE call to a PSTN user

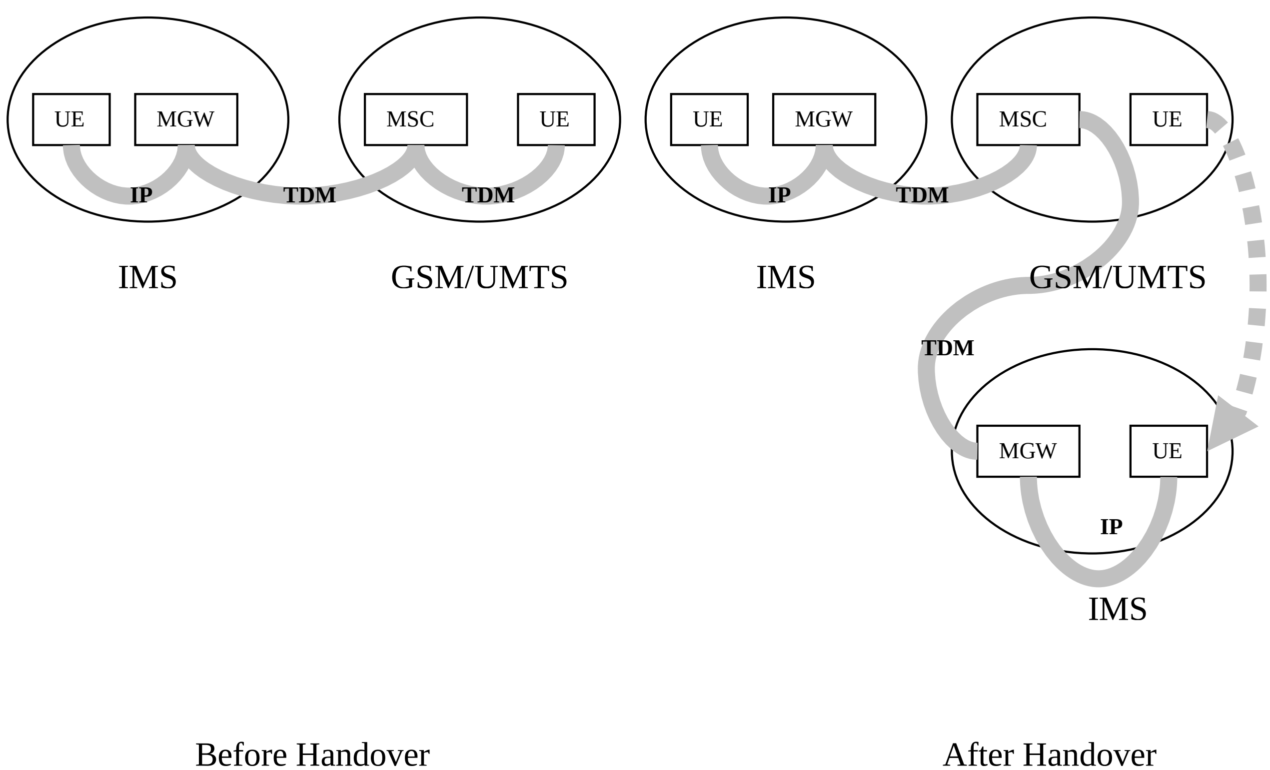
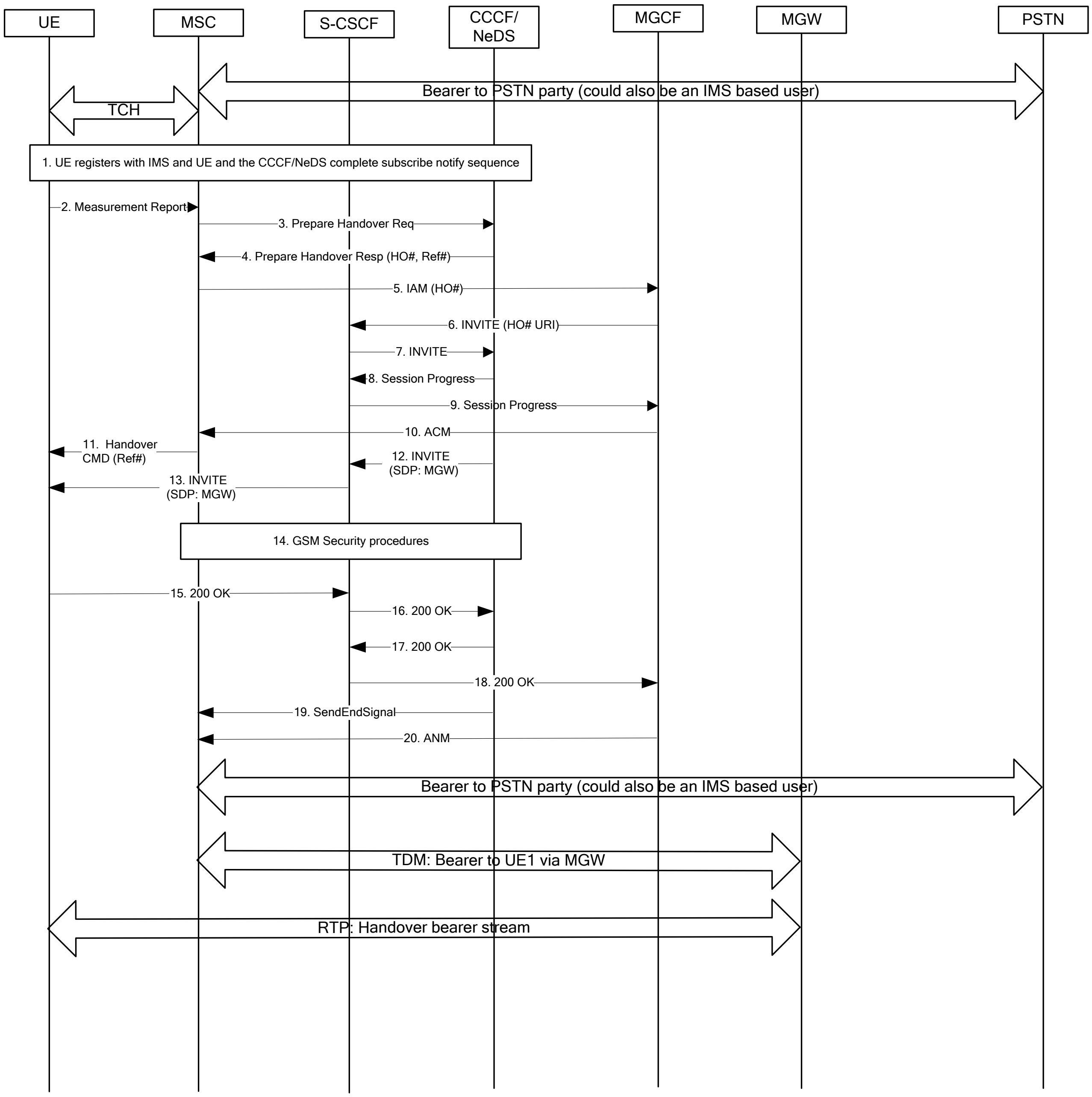


Figure 6.4.6.1-2: Bearer Path: CS domain call to an IMS UE



NOTE: The P-CSCF is not shown in the diagram or the flows.

Figure 6.4.6.1-3: CS Domain UE to PSTN user call continuity to IMS domain

The procedure for VCC from CS to IMS, as shown in Figure 6.4.6.1-3, is as follows:

The UE is a dual mode handset that is active in a CS domain voice call to a PSTN user, who could be a landline phone or CS domain user. (A similar flow applies when the PSTN is replaced by a CS domain user served by the same MSC as the UE.)

1. The UE detects an IMS network that is capable of supporting voice calls and registers with IMS. In the process, the UE and the CCCF/NeDS complete a subscribe notify sequence for mobility events.

2. When the UE determines that it is time to request call continuity, it sends a Measurement Report that contains channel information (e.g., an ARFCN and BSIC) configured to trigger call continuity to the CCCF/NeDS. The BSS translates this to a GCI and the MSC translates this to the CCCF/NeDS MSC ID.

3. The MSC sends a Prepare Handover Request to the CCCF/NeDS.

4. The CCCF/NeDS creates handover number and includes it along with a reference number in the Prepare Handover Response.

5. The MSC sends an IAM to the handover number contained in the Prepare Handover Response. The handover number routes to the MGCF. The network elements in the signalling path between the MSC and MGCF are not shown for simplicity of the flow.

6-7. The MGCF translates the handover number to a SIP URI and communicates with the MGW to create the SDP information for the connection to the circuit specified in the IAM. The SDP and SIP URI are put into an INVITE which routes to the CCCF/NeDS. (This may be direct or it may go through an S-CSCF.)

8-9. The CCCF/NeDS responds with a Session Progress message.

10. The MGCF sends an ACM message back to the MSC. This may occur when it receives one of the intermediate steps that was omitted in the flow.

11. The MSC sends a Handover Command towards the UE containing the reference number from the Prepare Handover Response.

12-13. At the same time as the CCCF/NeDS sent the Session Progress message, it forwards the INVITE towards the UE via the S-CSCF. The S-CSCF will need to use filter criteria or some other means to distinguish this INVITE from a normal terminating INVITE request. For this INVITE, normal filtering criteria rules are not applied and the INVITE is forwarded to the UE. This is to turn off normal service handling such as call forwarding unconditional that would prevent the INVITE from being sent to the UE.

NOTE 1: It is expected that the handover number will be sufficient for the CCCF/NeDS to identify the UE. If not, then the UE could send a NOTIFY to the CCCF/NeDS after step 9 which contains the reference number in the Handover Command.

NOTE 2: Filter criteria may need to be defined in the detail specification.

14. At this point, the CCCF/NeDS and the MSC can do any needed GSM security procedures if required.

15-16. The UE accepts the call and sends a 200 OK response. There are generally additional steps between the INVITE and the 200 OK, but they are not shown for simplicity of the flow.

17-18. The CCCF/NeDS sends the 200 OK to the MGCF. The MGCF uses this information to complete the connection to the MGW by providing the SDP contained in the 200 OK.

19. The CCCF/NeDS also sends a SendEndSignal message to the MSC.

20. The MSC responds with an ANM message and the path is now completed from the MSC to the MGW to UE and the VCC procedure to IMS for UE is completed.

The same procedures are used if the UE is in a complex call involving multiple parties. As in normal GSM handover, these procedures also apply to UEs with supplementary services, such as call waiting, active at the time of call continuity.

There is no difference if the terminating PSTN Telephone in the previous example is replaced by an IMS UE. The VCC procedures are the same. Call control remains in the CS domain and only the call leg handing off to IMS is affected.

Note that a pseudo (ARFCN, BSIC) tuple is proposed to be assigned to identify an MSCID representing a specific CCCF within an IMS network. The ARFCN-BSIC tuple assigned must not be one that is used by any GSM/UMTS operator(s) in the area(s).. When a CCCF within an IMS network with co-operation arrangements is available as a neighbour to a GSM/UMTS cell, the BSC/RNC controlling this GSM/UMTS cell should be configured to transmit the ARFCN (representing a specific CCCF) as one of the neighbour lists. This needs to be configured in the BSC/RNC and the CCCF.

NOTE: The neighbour list length is limited by existing GERAN/UTRAN specifications. The neighbour list length is an upper bound for the total number of CCCFs (i.e. both in this operator's network as well as in the IMS networks with cooperation arrangements) that can be mapped by the BSC/RNC.

Alternatively, if the neighbour list length proves to be an issue, a solution is to introduce a "pseudo MSC" function in the visited network whose purpose is to allow a network's radio protocols to use a single ARFCN to support handovers to an arbitrary number of IMS CCCFs. The goal is for the pseudo MSC to use IMSI to select the CCCF appropriate for the subscriber, if available. TS 29.010 [24] describes the conditions under which IMSI shall be available.

As in the "non-pseudo MSC" solution (above), during CCCF registration the CCCF informs the UE which pseudo (ARFCN, BSIC) tuple to use to represent the IMS network; by means of the VCC information exchange. The BSC/RNC controlling this GSM/UMTS cell should be configured to transmit the ARFCN. The pseudo MSC examines the IMSI in the MAP HORequest to determine the actual target network, GMSC, or home CCCF. The pseudo MSC is then responsible for relaying the MAP HORequest to the target system. The pseudo MSC remains in the signalling path between the MSC and home CCCF serving as a relay for handover signalling messages (not ISUP signalling).

Using the "pseudo MSC" function, the value of the pseudo (ARFCN, BSIC) tuple shared by the CCCF is statically determined (e.g. through cooperation agreements). Only one pseudo (ARFCN, BSIC) tuple is required for any number of CCCFs.

Editor Note: How would this be supported in 3G only environments

Figure 6.4.6.1-4 shows the case where the UE is in a CS domain multi-party call to a CS domain UE2 and a party in the PSTN. This could be a case of call waiting or the parties could be in a 3-way call managed by the MSC. After the UE moves to the IMS domain, the only change is to change the call leg from the MSC to the UE. The MSC continues to anchor the calls.

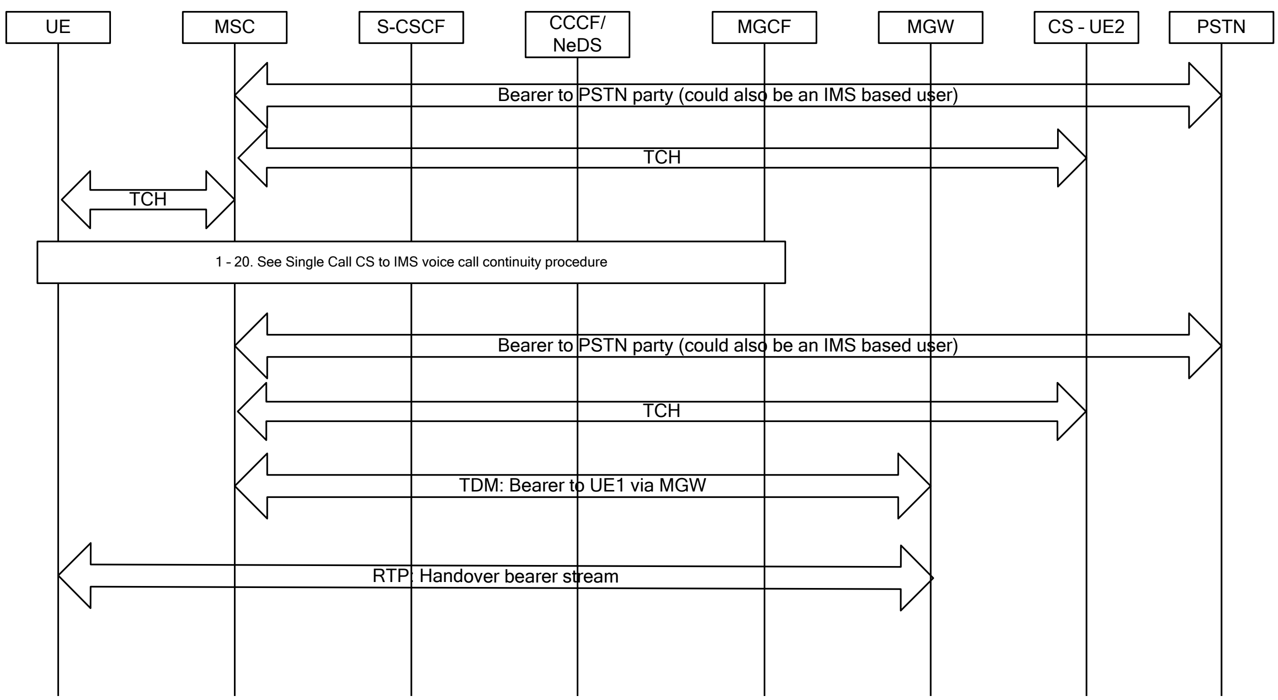


Figure 6.4.6.1-4: CS to IMS voice call continuity procedure for multi-party CS domain call

The procedure for CS to IMS voice call continuity for a multi-party CS domain call, as shown in Figure x-1, is as follows:

The UE is a dual mode handset that is active in a multi-party CS domain voice call to a CS domain UE2 and a PSTN user.

1-20. The voice call continuity procedure is the same as present above and shown in Figure 6.4.6.1-3, CS Domain UE to PSTN user call continuity to IMS domain.

6.4.6.2 Basic IMS to CS call continuity procedure

This section describes the basic call continuity scenario for 2-party calls, one in which the B party is in IMS, and another in which the B party is a fixed subscriber.

Bearer path, if B party is in IMS:

Before call continuity procedure: After call continuity procedure:

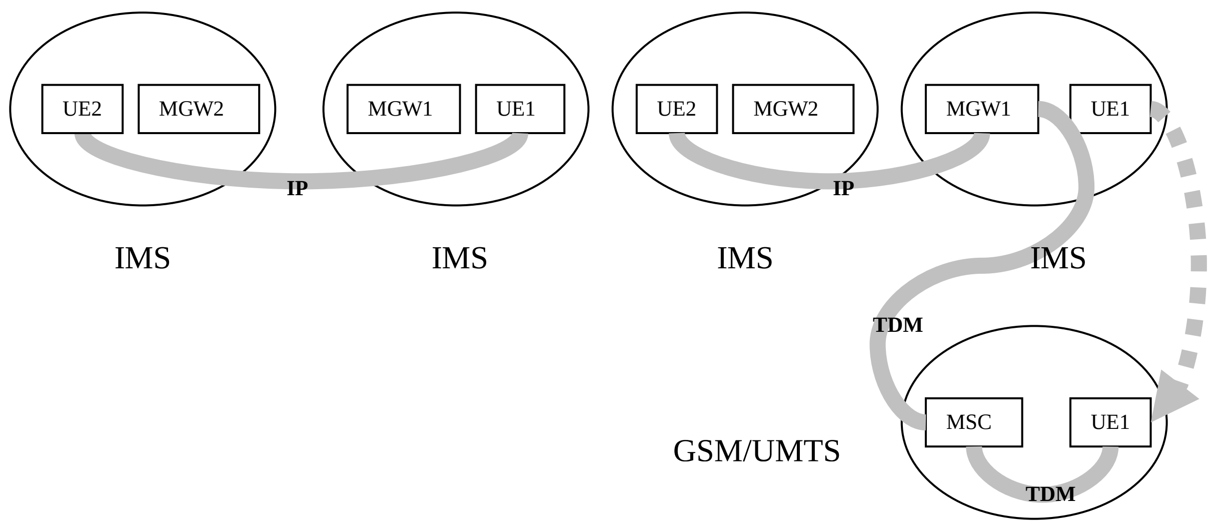


Figure 6.4.6.2-1: call continuity procedure bearer path - IMS VoIP to CS Voice, B party in IMS

Bearer path, if B party is a fixed subscriber:

Before call continuity procedure: After call continuity procedure:

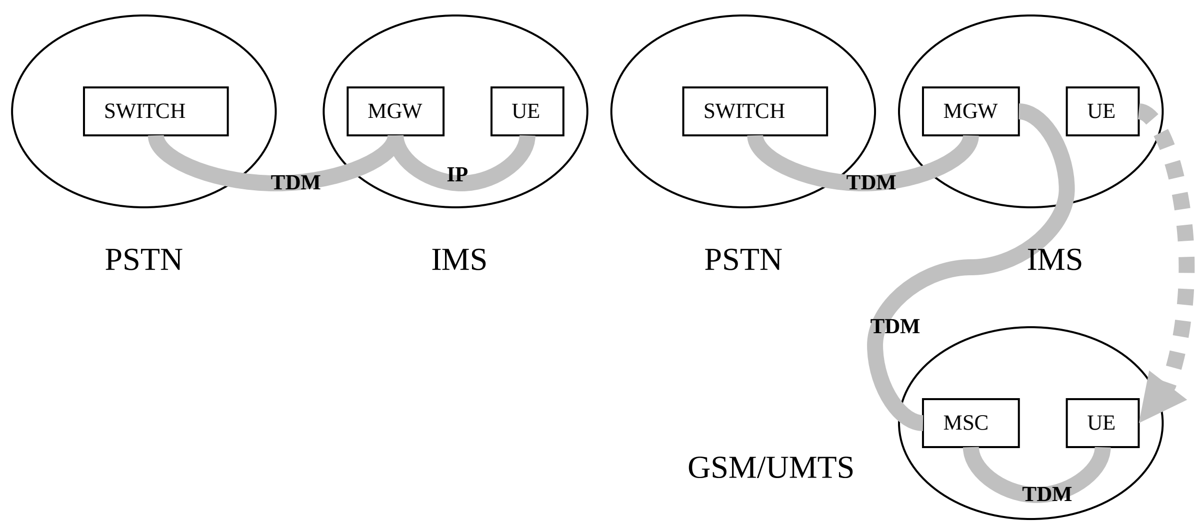


Figure 6.4.6.2-2: call continuity procedure bearer path - IMS VoIP to CS Voice, B party in PSTN

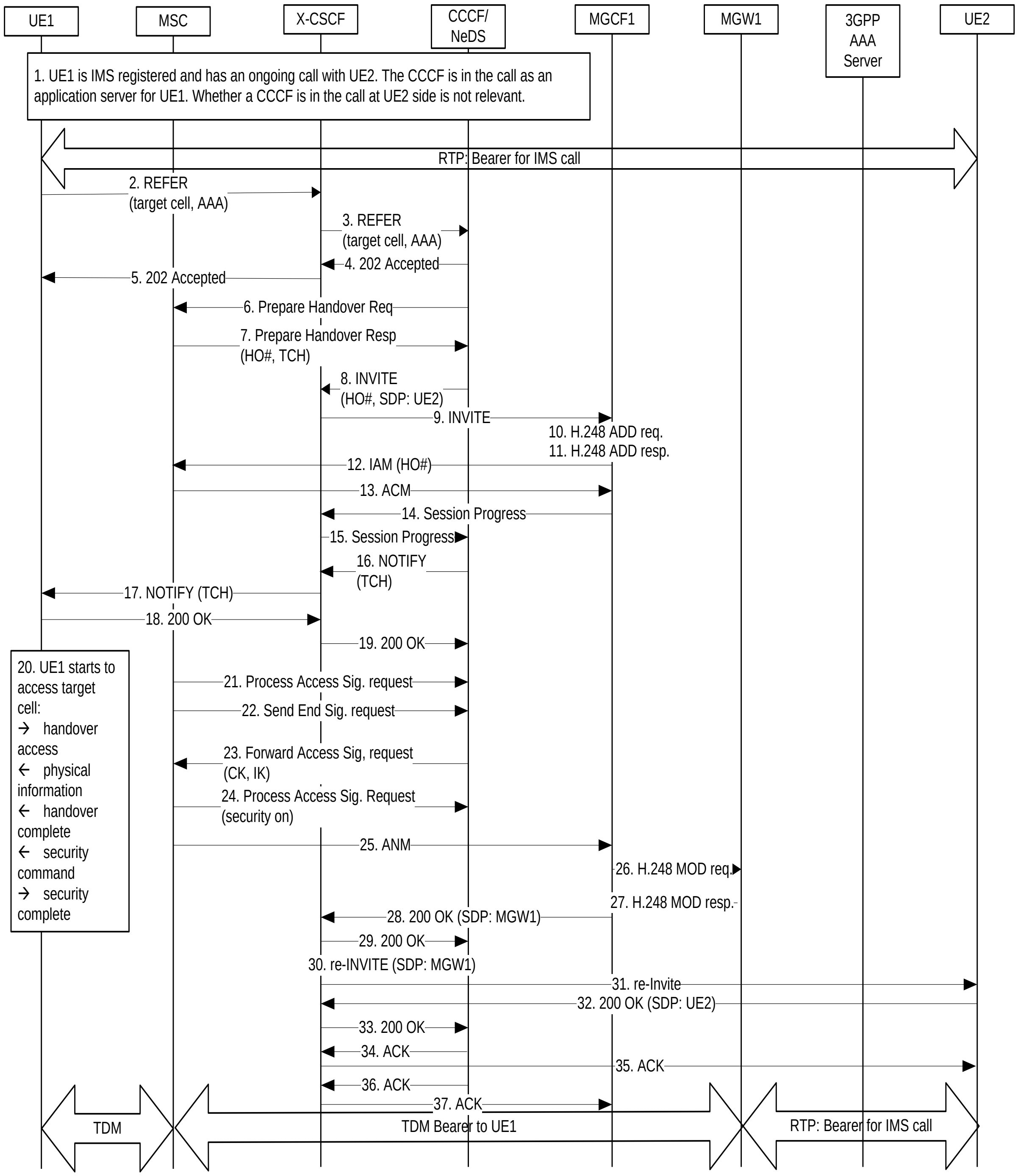


Figure 6.4.6.2-3

1. UE1 has an ongoing 2-party call with UE2. The CCCF/NeDS is informed by the S-CSCF about the used CK/IK (cipher and integrity key) in the 3rd party register. Note, for handovers to 2G RAN, the AKA keys shall be converted to SIM compatible keys according to the procedures specified in TS 33.102 [18].

2. The re-selection to GSM/UMTS criteria is fulfilled. UE1 requests call continuity to GSM/UMTS through a REFER method. The REFER request contains the target cell id within the GSM/UMTS system.

3. The REFER request is forwarded to the CCCF/NeDS based on initial filter criteria.

4. The CCCF/NeDS responds with a 202 Accepted and creates a subscription.

5. The S-CSCF forwards the 202 Accepted to UE1.

6. Based on the target cell id the CCCF/NeDS sends a MAP: PREPARE_HANDOVER request to the proper target MSC. The MAP: PREPARE_HANDOVER shall carry all information needed by the MSC for allocating a radio channel within the target GSM or UMTS cell.

7. The target MSC returns the MAP: PREPARE_HANDOVER response after retrieving a Handover Number from its associated VLR. The Handover Number (HO#) shall be used for routing the connection of the call from MGCF1/MGW1 to the target MSC. If a traffic channel is available in the target cell, a MAP: PREPARE_HANDOVER response containing the radio resource description of the target GSM or UMTS cell is sent back to the CCCF/NeDS. Otherwise, the MAP: PREPARE_HANDOVER contains a handover failure indication and the CCCF/NeDS stops the handover procedure.

8. The CCCF/NeDS requests the MGCF1 to establish an outgoing call to the target MSC (HO#) by sending an INVITE request. The SDP within the INVITE request contains the address of UE2, which uses the MGW1 for verification purposes.

9. The request goes via the S-CSCF.

10. Request IMS and remote resource configurations.

11. Acknowledges resource configurations.

12. Initiated by the INVITE request, the MGCF1 sends an Initial Address Message (IAM) to the target MSC addressed by the Handover Number (HO#) to establish a circuit connection between MGW1 and the target MSC.

13. The target MSC responds with the Address Complete Message towards the MGCF1 and waits for the capturing of the UE1 on the radio path in the target cell.

14. The MGCF1 receives the ACM and knows that a circuit link has been established to the target MSC. The Session Progress triggers sending the Handover Command to the UE1.

15. The Session Progress goes via the S-CSCF.

16. The CCCF/NeDS sends the radio channel description of the target cell via a NOTIFY to the CCCF/NeDS. The NOTIFY is part of the Mobility Event Package subscription performed by the UE right after IMS registration. The radio channel description of the target cell was provided by the Prepare Handover response.

17. The NOTIFY request is forwarded towards UE1. From now on the UE is able to do the call continuity even if the WLAN connection breaks.

18. UE1 answers with a 200 OK.

19. The UE starts accessing the target cell establish a radio link.

20. The 200 OK is forwarded to the CCCF/NeDS.

21. The MSC gets an HO detect from the access network and sends a MAP: Process Access Signalling request to the CCCF/NeDS.

22. As soon as the MSC receives the HO complete from the access network, it indicates the successful completion of call continuity to the CCCF/NeDS by the MAP: Send End Signal request.

23. The CCCF/NeDS transfers the keys by sending an MAP: Forward Access Signalling request to the target MSC. The UE already has the keys from using EAP-AKA in I-WLAN.

24. The MSC starts security procedures with UE1 and acknowledges security on to the CCCF/NeDS by a MAP: Process Access Signalling request.

25. The MSC accepts the call establishment request by sending an ANSWER message.

26. MGCF activates voice processing.

27. The circuit switched connection between UE1 and MGW1 is now established.

28. The MGCF1 acknowledges (200 OK) the INVITE from step 8 and indicates the address of MGW1 for the re-INVITE (step 33).

29. The 200 OK is forwarded to the CCCF/NeDS.

30. The CCCF/NeDS sends a re-INVITE to link UE2 towards IP address of MGW1 (needed if B party is an IMS subscriber).

31. The re-INVITE goes via the S-CSCF (needed if B party in an IMS subscriber).

32. UE2 acknowledges the re-INVITE with a 200 OK (needed if B party is an IMS subscriber).

33. The 200 OK goes via the S-CSCF (needed if B party is an IMS subscriber).

34. S-CSCF sends an ACK (needed if B party is an IMS subscriber).

35. The VoIP link between UE2 and MGW1 is now re-established (needed if B party is IMS subscriber).

36. The CCCF/NeDS sends a final ACK, which indicates to the MGCF that the voice connection between UE1 over the CS domain and UE2 can now be connected through.

37. The final ACK goes via the S-SCSF.

Editor Note: Inconsistency of the CS to IMS vs IMS to CS call model

Editor Note: Confirm that call continuity to serving MSC function properly for a CS user, who already registered in the targeted MSC

Editor Note: Set matching cipher key in the UE and BSS/RNS (pending LS to SA2)

Editor Note: Confirm MSC to MSC adhere to the existing security

Editor Note: How does the UE and CCCF/NeDS exchange supplementary services call control message via DTAP?

Editor Note: Target element addressing:
How does the visited network element know the CCCF/NeDS point code and vice versa?
Issue due to use of target cell ID, in particular for the case of 3G target cells

Figure 6.4.6.2-4 shows the case where UE1 is in a IMS domain multi-party call with two IMS domain parties, UE2 and UE3. This could be a case of call waiting or the parties could be in a 3-way call managed by UE1. After UE1 moves to the CS domain, the CCCF/NeDS has inserted an MRF in the media path (assuming that it was not already in the path) and a single media stream is sent to the CS domain. The CCCF/NeDS remains in the session path and manages the IMS sessions for the UE.

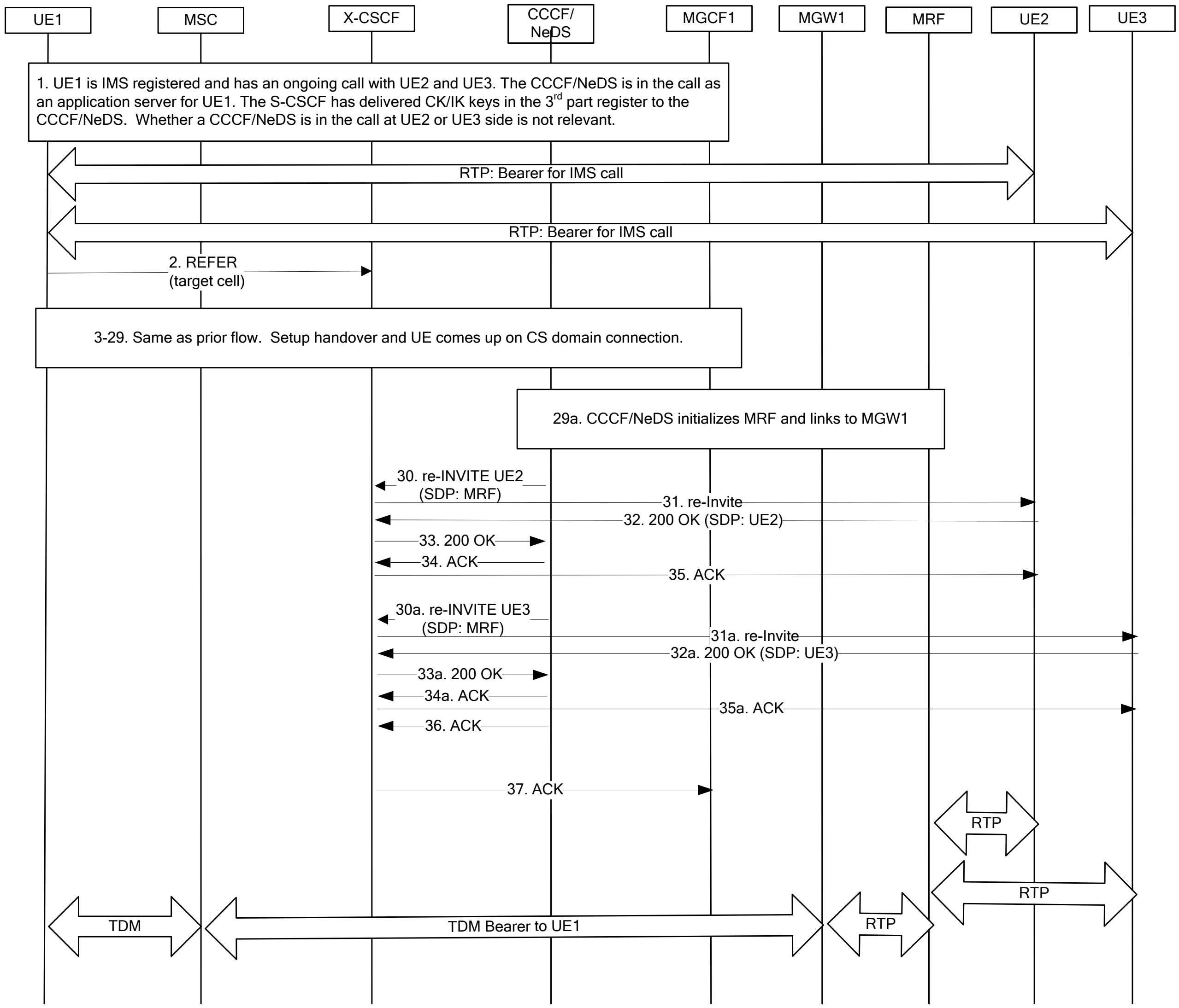


Figure 6.4.6.2-4: IMS to CS voice call continuity procedure for multi-party IMS domain call

1. UE1 has an ongoing 2-party call with UE2 and UE3. The CCCF/NeDS is informed by the S-CSCF about the used IK/CK/IK (cipher and integrity key) in the 3rd party register.

2. The re-selection to GSM/UMTS criteria is fulfilled. UE1 requests call continuity to GSM/UMTS through a REFER method. The REFER request contains the target cell id within the GSM/UMTS system. The REFER request may contain additional information about the sessions to handover.

3-29. These are the same as the steps in the preceding figure, except that an MRF is added into the bearer path and MGW1 is configured to point to it and not UE2.

31a. At some time during steps 3-31, the CCCF/NeDS will configure an MRF and provide bearer path links to MGW1.

NOTE: If an MRF was already inserted into the media stream for call control purposes, this step could be eliminated.

30-35. These are the same as the steps in the preceding figure, except that UE2 is pointed to a port on the MRF and not MGW1.

30a-35a. These are similar steps for UE3.

36-37. Finally, the ACK is sent back to MGCF 1 completing the voice call continuity procedure.

Clause 6.4.7.2 will describe how supplementary service messages are processed after the voice call continuity procedures have completed.

6.4.7 Impact on Supplementary Services

Both the CS and IMS domains may have voice call supplementary services (e.g. call waiting or 3-way calling) active when the voice call continuity service is activated, as discussed in clause 5.3.4, "Supplementary services are active when call continuity procedures occur." In the Original Domain Control Model, these services remain under the control of the original domain until the active voice calls are terminated.

During and after GSM handover procedures, the MSC that initially set up a call retains call control for the entire duration of that call; it is the so called anchor MSC. It this role, it also retains control of supplementary services. After a handover, the anchor MSC can exchange supplementary service related DTAP messages with the UE using inter MSC messaging. Such DTAP messages are passed transparently to the UE by MSC-B. The Original Domain Control Model uses this same approach.

In the case of a CS call that has been handed over to IMS, the CS domain MSC remains as the call's anchor MSC, and continues to manage call state. Call state transition messages are exchanged between the UE and the anchor MSC via the CCCF/NeDS, which emulates an MSC-B. On the anchor MSC facing interface, the CCCF/NeDS behaves as an MSC, supporting inter MSC messaging. On the dual mode handset facing interface (the IMS facing interface), the CCCF/NeDS relays DTAP messages in the body of SIP messages. This DTAP communication allows all supplementary services to be supported.

In the case of a call originally established through the IMS, that has been handed over to the CS domain, the CCCF/NeDS emulates the anchor MSC. In this role, the CCCF/NeDS retains call control and remains in the UE DTAP signaling path. In this case, DTAP messages are exchanged between the CCCF/NeDS and the UE across the CS domain. This exchange uses standard CS domain inter MSC messaging between the CCCF/NeDS and the CS domain MSC-B.

6.4.7.1 Supplementary services after CS to IMS call continuity procedures

In the CS to IMS voice call continuity case, the anchor MSC behaves as it normally does for inter-MSC handover and expects the target MSC (in this case, the CCCF/NeDS) to relay messages between the anchor MSC and the UE. The details are shown in Figure 6.4.7.1-1 below.

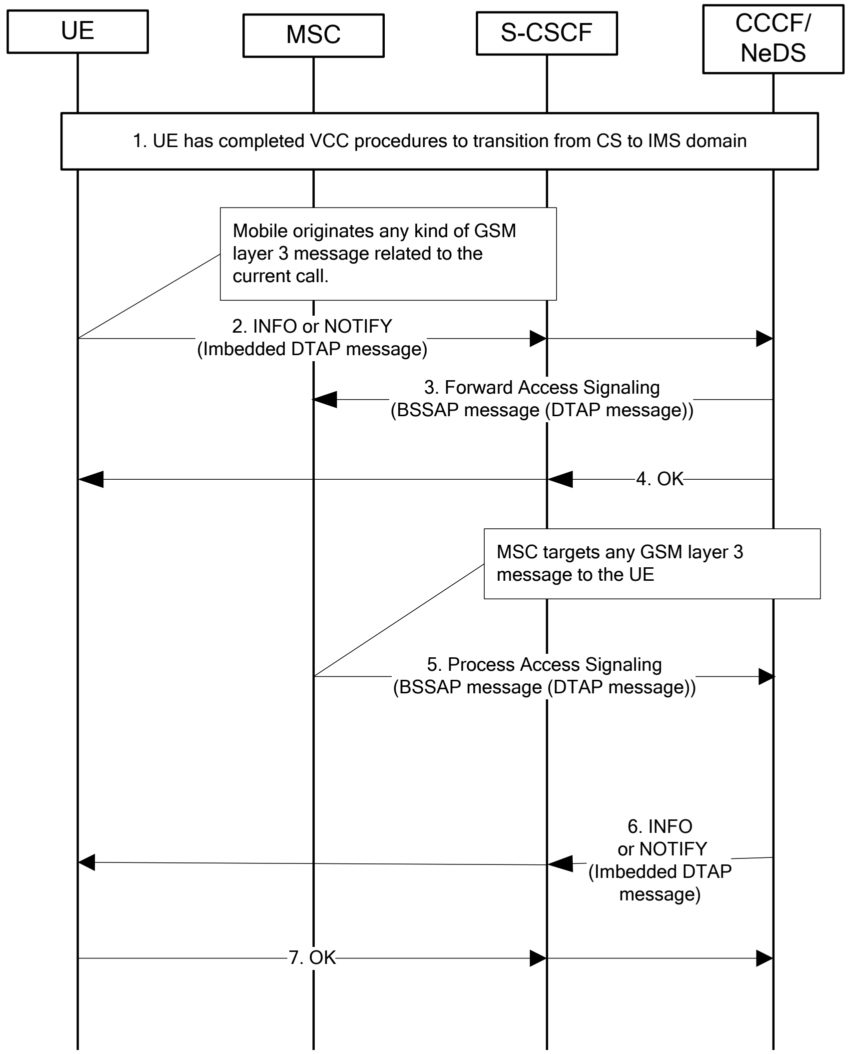


Figure 6.4.7.1-1: Supplementary Service Activation After CS to IMS Voice Call Continuity Procedure

The procedure for Supplementary Service Activation After CS to IMS Voice Call Continuity Procedure, as shown in Figure 6.4.7.1-1, is as follows:

1. The UE is a dual mode handset that has completed the voice call continuity procedure to move from the CS to the IMS domain and is in an active call. The UE and the anchor MSC continue to communicate via GSM layer 3 signaling messages for the current call.
2. The UE encapsulates the desired DTAP message and sends it as the body of a SIP message, which gets forwarded to the CCCF/NeDS. While INFO or NOTIFY are shown above, these are just examples of the possible SIP message that can be used. These messages are sent within the existing SIP dialogue that established the call.
3. The CCCF/NeDS forwards the embedded DTAP message to the anchor MSC using existing post handover inter-MSC messaging.
4. The CCCF/NeDS responds to the UE with an OK.
5. When the anchor MSC needs to send a message to the UE, it creates a DTAP message and sends it to the CCCF/NeDS (acting as a target MSC) using existing post handover inter-MSC messaging.
6. The CCCF/NeDS puts the DTAP message in the body of a SIP message and sends it to the UE.
7. The UE responds with an OK.

These message flows can be used to carry any standard CS domain messages between the anchor MSC and the UE. The UE and MSC both continue to use the call state machines that were in effect at the time of voice call continuity. The only change is that the messages are now carried in the body of SIP messages between the UE and the CCCF/NeDS.

6.4.7.2 Supplementary services after IMS to CS call continuity procedures

In the IMS to CS voice call continuity case, the CCCF/NeDS takes on the role of voice call control after the voice call continuity procedures have been completed. The UE uses DTAP messages in the CS domain to communicate with the CCCF/NeDS. Two methods are possible:

1) If IMS adopts the ISDN call model, the existing GSM DTAP messages can be used, or

2) The DTAP messages can contain embedded compressed SIP messages, segmented, as needed, to meet the size limits for DTAP messaging.

NOTE: The choice of specific DTAP messages and the details of how SIP compression is established between the UE and the CCCF is for further study

At the target MSC in the CS domain, the UE to MSC communication is unchanged. The target MSC just forwards messages between the UE and the anchor MSC, in this case the CCCF/NeDS, without analyzing the contents. The details are shown in Figure 6.4.7.2-1, 6.4.7.2-2 and 6.4.7.2-3 below.

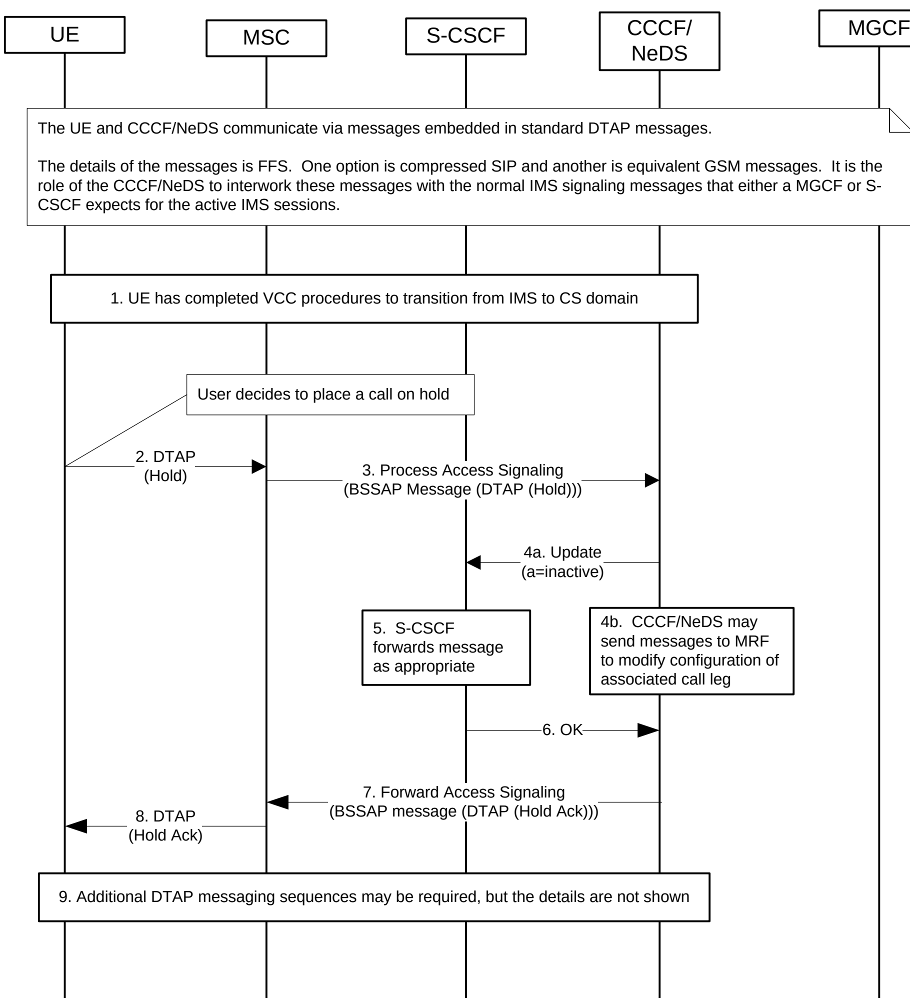


Figure 6.4.7.2-1: Placing Call On Hold After IMS to CS Voice Call Continuity Procedure Completed

The procedure for placing a call on hold after the IMS to CS voice call continuity procedure has completed, as shown in Figure 6.4.7.2-1, is as follows:

1. The UE is a dual mode handset that has completed the voice call continuity procedure to move from the IMS to the CS domain and is in an active call. The UE and the CCCF/NeDS communicate with each other to control the state of the various active IMS voice sessions.

2. The User decides to place the current call on hold and the UE constructs a DTAP message which signifies placing the session on hold. This could be a SIP UPDATE for the active session or an equivalent GSM layer 3 message. If more than 1 DTAP message is required, steps 2 and 3 will be repeated.

3. The target MSC forwards the embedded DTAP message to CCCF/NeDS, which is acting as an anchor MSC, using existing post handover inter-MSC messaging.

4a. The CCCF/NeDS sends an Update message to the S-CSCF to forward to the other end for this session.

4b. If an MRF has been placed in the media path, the CCCF/NeDS may send messages to the MRF to modify the configuration of the associated call leg.

5. The S-CSCF forwards the message as appropriate to the end point.

6. When the S-CSCF receives an OK response, it forwards it to the CCCF/NeDS.

7. The CCCF/NeDS forwards a Hold acknowledgement back to the target MSC using existing post handover inter-MSC messaging.

8. The target MSC sends the DTAP message on to the UE. If more than 1 DTAP message is required, steps 7 and 8 will be repeated.

9. If additional messages are required for call hold, they would be send between the UE and CCCF/NeDS using the same procedure shown above.

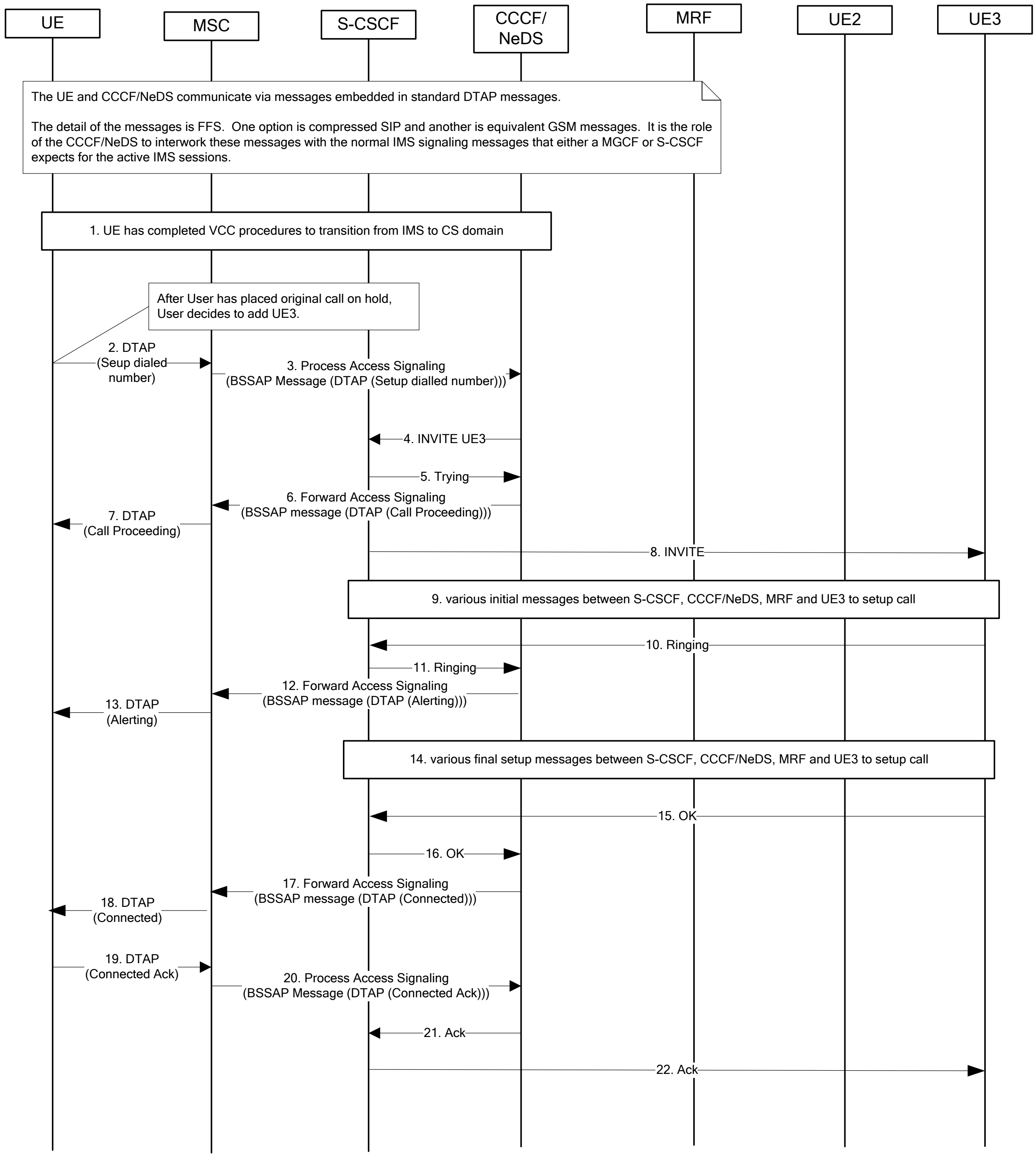


Figure 6.4.7.2-2: Adding on a Third Party After IMS to CS Voice Call Continuity Procedure Completed

The procedure for adding on a third party after the IMS to CS voice call continuity procedure has completed, as shown in Figure 6.4.7.2-2, is as follows:

1. The UE is a dual mode handset that has completed the voice call continuity procedure to move from the IMS to the CS domain and is in an active call. The UE and the CCCF/NeDS communicate with each other to control the state of the various active IMS voice sessions.

2. After the user has placed the original call(s) on hold, the user decides to add UE3. The UE constructs a DTAP message which signifies setting up a new session. This could be a SIP INVITE for the new session or an equivalent GSM layer 3 message. If more than 1 DTAP message is required, steps 2 and 3 will be repeated.

3. The target MSC forwards the embedded DTAP message to CCCF/NeDS, which is acting as an anchor MSC, using existing post handover inter-MSC messaging.

4. The CCCF/NeDS sends an INVITE to the S-CSCF to INVITE UE3.

5. The S-CSCF responds with a Trying message

6. The CCCF/NeDS forwards a call proceeding message back to the target MSC using existing post handover inter-MSC messaging.

7. The target MSC sends the DTAP message on to the UE.

8. At the same time as the S-CSCF sends the trying, it forwards the INVITE to UE3 (this may be direct or via another S-CSCF or MGCF).

9. Various call setup messages occur at this time. If needed, an MRF is added to the call.

10. Ringing is sent from the UE to the S-CSCF.

11. The S-CSCF forwards the Ringing to the CCCF/NeDS.

12. The CCCF/NeDS creates a DTAP Alerting message and forwards it to the target MSC using existing post handover inter-MSC messaging.

13. The target MSC sends the DTAP message on to the UE.

14. Other final setup messages occur between the S-CSCF, CCCF/NeDS, MRF and UE3 to setup the call.

15. The final OK is sent from UE3 to the S-CSCF.

16. The S-CSCF forwards the final OK to the CCCF/NeDS.

17. The CCCF/NeDS creates a DTAP Connected message and forwards it to the target MSC using existing post handover inter-MSC messaging.

18. The target MSC sends the DTAP message on to the UE.

19. The UE responds with a DTAP Connected Ack message.

20. The target MSC forwards the embedded DTAP message to CCCF/NeDS.

21. The CCCF/NeDS sends an Ack to the S-CSCF.

22. The S-CSCF forwards the Ack to UE3 and the call setup is complete.

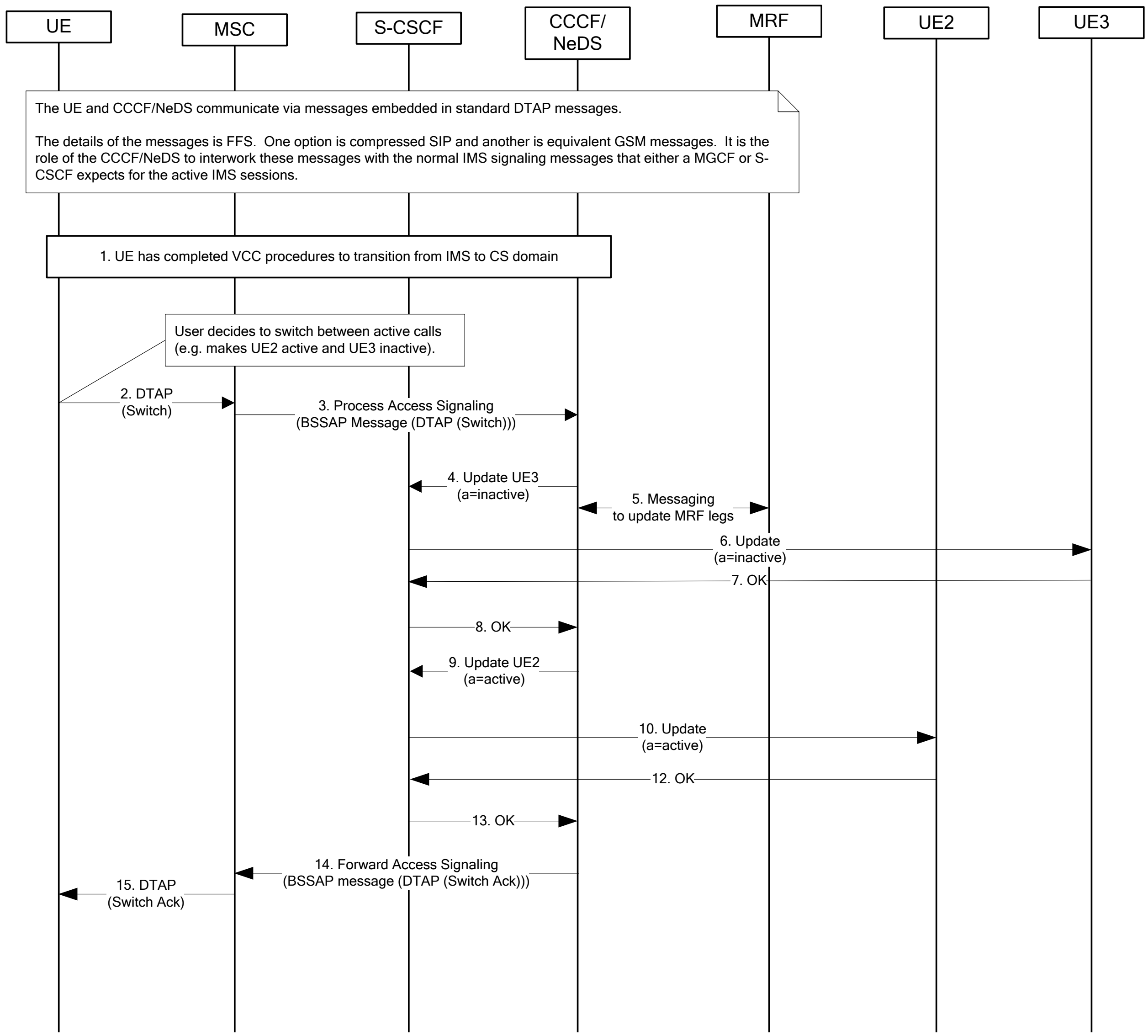


Figure 6.4.7.2-3: Switching Between Two Active Call Sessions After IMS to CS Voice Call Continuity Procedure Completed

The procedure for switching between two active call sessions after the IMS to CS voice call continuity procedure has completed, as shown in Figure 6.4.7.2-3, is as follows:

1. The UE is a dual mode handset that has completed the voice call continuity procedure to move from the IMS to the CS domain and is in an active call. The UE and the CCCF/NeDS communicate with each other to control the state of the various active IMS voice sessions.

2. The user decides to switch between active calls, making the session to UE2 active and the session to UE3 inactive. The UE constructs a DTAP message which signifies switching between sessions. This could be multiple SIP UPDATEs for each session or an equivalent GSM layer 3 message. If more than 1 DTAP message is required, steps 2 and 3 will be repeated.

3. The target MSC forwards the embedded DTAP message to CCCF/NeDS, which is acting as an anchor MSC, using existing post handover inter-MSC messaging.

4. The CCCF/NeDS sends an Update to UE3 via the S-CSCF which will make the session to UE3 inactive.

5. If needed, the CCCF/NeDS sends appropriate messages to the MRF to update the MRF call legs.

6. The S-CSCF forwards the Update to UE3.

7. UE3 responds with an OK.

8. The S-CSCF forwards this to the CCCF/NeDS.

9-13. A similar sequence of messages is sent to UE2 to make it active.

14. The CCCF/NeDS creates a DTAP Switch Ack message and forwards it to the target MSC using existing post handover inter-MSC messaging.

15. The target MSC sends the DTAP message on to the UE.

6.4.8 Charging Impact of Handovers Between the CS Domain and the IMS for Original Domain Control

Charging records are generated as specified today. Correlation may be needed for CS to IMS transition.

VCC requires business agreements that are expected to impact the online charging systems for the operators. For static anchoring IMS, the impacts are expected to be higher than for the ODC.

6.4.9 Security Impact of Handovers Between the CS Domain and the IMS for Original Domain Control

No security impact has been identified.

6.5 HandOver Application Server for voice continuity between the IMS and CS domain

6.5.1 General Description

In this alternative the CS-IMS continuity is solved by employing Call Continuity Control Function (CCCF) in the user's home IMS network.

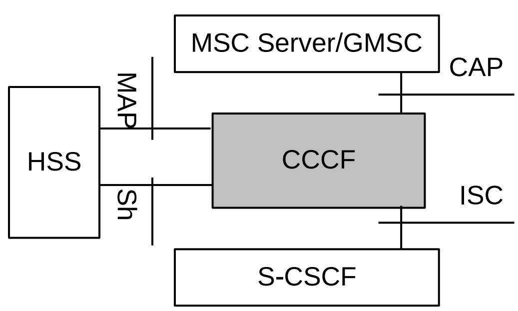


Figure 6.5.1-1: CCCF functionality

Some CCCF fundamentals are listed below.

1. The CCCF provides the voice call continuity service between the CS domain and IMS.

2. The CCCF is a functionality which can act as the gsmSCF of Camel Service in CS domain and also act as the Application Server in IMS domain. The architecture of the CCCF is FFS.

3. In IMS, the CCCF is served as the PSI based application server. The CCCF enables a Routing B2BUA 3pcc function to control the bearer path for the session . It can originate a new session request or terminate a session request.

4. In CS domain, the CCCF is served as the gsmSCF of Camel Service. If the CS domain supports camel 4 service, then the the CCCF can manipulate call legs which includes creating new parties in a call, placing individual call parties on hold, reconnecting them to the group of call parties and disconnecting individual call parties.

5. The CCCF can store the call/session information for both domains. The CCCF can correlate this information by using the MSISDN of the user but it is not exclusive. Some examples of this information are: MSISDN of the user, session identifier and SDP in IMS domain, CAP dialogue information and MSC Server number in CS domain, etc.

6. The CCCF is routed in IMS domain by using a Public Service Identity. The CCCF also has two E.164 addresses(one is a gsmSCF address and the other is a tel number which can be used as a called party number) in CS domain. The gsmSCF address can be sent to the VLR in Location update procedure in CS domain. The PSI and the tel number can be automatically transferred to the UE during the registration in IMS domain or statically configured before the voice call request.

7. The CCCF also has two reference points to the HSS in CS domain and in IMS domain. This two reference points are used to query the registration state in both domains which can be considered in the routing selection.

8. The CCCF can correlate the charging information in both domains.

6.5.4 Origination

6.5.4.1 IMS origination

This section describes CS-IMS subscriber originate a voice call from IMS domain firstly. The procedure of the voice call establishment is the same as the one described in TS 23.228 [9].

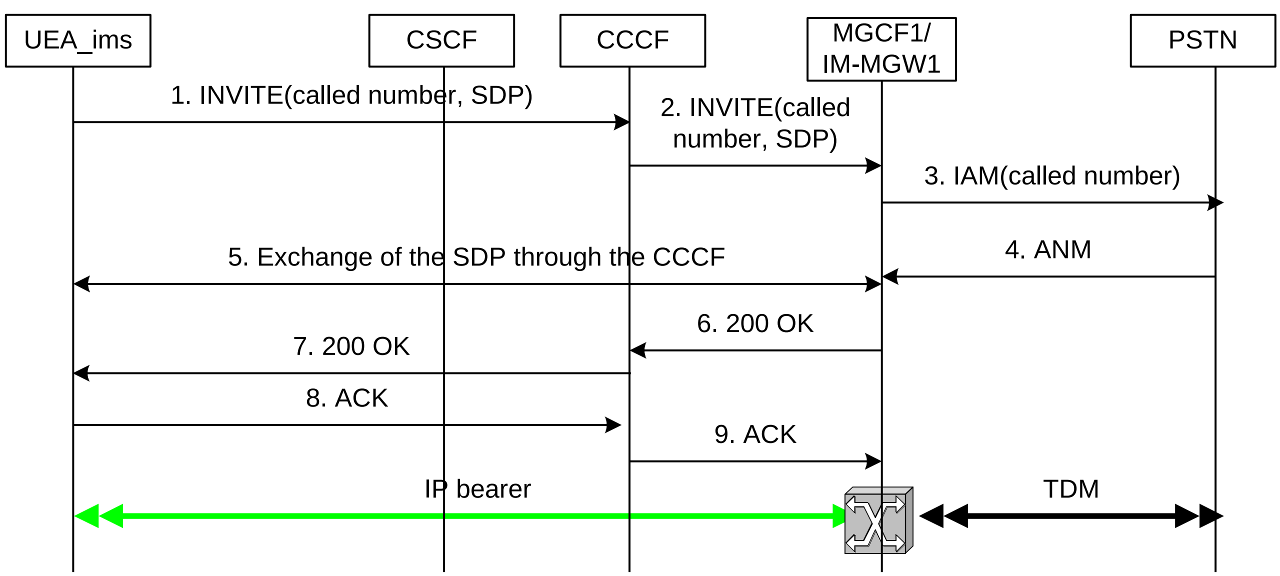


Figure 6.5.4.1-1: Call flow of IMS origination

1. The user A send a INVITE request to the IMS network(PSTN number of user B, the MSISDN of user A, initial SDP). This request is routed to the home IMS network of user A. According the Initial filter criteria configured in the home IMS PLMN, the INVITE request is routed to the CCCF. The iFCs are stored in the HSS as part of the CS-IMS user's service subscription profile and downloaded to the currently assigned S-CSCF at the time of subscriber's registration with IMS network.

2. After receiving the INVITE request ,the CCCF store the MSISDN of user A and the initial SDP. Then the CCCF shall originate a session request to user B(PSTN number of user B, initial SDP). This session request arrives a proper MGCF1 .

3. The MGCF1 send an IAM message to the PSTN user B.

4. When the user B send an ANM back to MGCF1 to answer the call.

5. Then the CCCF exchanges the SDP between the MGCF1 and the user A.

6,7,8,9. After the exchange of SDP the MGCF1 send a 200 OK to the CCCF and the CCCF send a 200 OK to user A. The user A send an ACK message to the CCCF and the CCCF send a ACK to MGCF1.

Then a voice call between the user A and user B is established. The whole bearer path include two parts: one is based on IP from the user A to IM-MGW1 and the other is based on TDM from the IM-MGW1 to PSTN user, as described in Figure 6.5.6.1.1-1.

6.9.4.2 GSM/UMTS CS origination

This section describes CS-IMS subscriber originate a voice call from CS domain firstly. The procedure of the voice call establishment is the same as the one described in TS 23.018 [5].

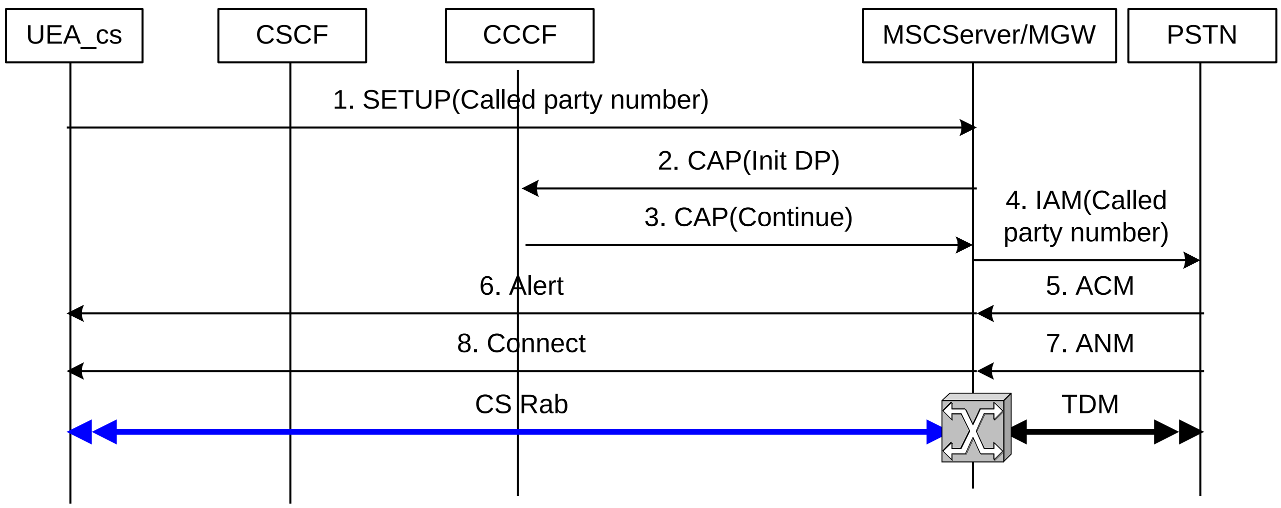


Figure 6.9.4.2-1: Call flow of CS origination

1. The user A originates a CS call to the user B through the MSC Server in visiting CS PLMN. User B is a PSTN user.

2. The MSC Server trigger a CAP dialogue to CCCF according to the camel criteria in the MSC Server.

3. The CCCF sends a CAP(Continue) to the MSC Server to continue the CS call to the PSTN user.

4. The MSC Server send an IAM message to the PSTN user.

5. When the PSTN user rings the PSTN shall send an ACM message to the MSC Server.

6. The MSC Server sends a Alert to the user A.

7. When the PSTN user answers the CS call the PSTN shall send an ANM message to the MSC Server.

8. The MSC Server sends a Connect to the user A.

So the CS call between the user A and PSTN user is established. There are two bearer paths: one is based on CS Rab between the user A and MSC Server/MGW , the other is based on TDM between the MSC Server/MGW and PSTN, as described in Figure 6.5.6.1.2-1. During the CS call procedure the MSC Server establishes a CAP dialogue to CCCF.

6.5.5 Termination

6.5.5.1 IMS termination

6.5.5.2 GSM/UMTS CS termination

6.5.6 Call continuity scenarios

6.5.6.1 Two party UE to PSTN calls

6.5.6.1.1 IMS to CS call continuity

This section describes CS-IMS subscriber call continuity from the IMS to the CS domain. In this scenario a CS-IMS subscriber has an active call in the IMS to a PSTN subscriber firstly and the CS-IMS UE decides to requestcall continuity from the IMS to the CS domain.

The following figure shows the detailed call continuity procedure.

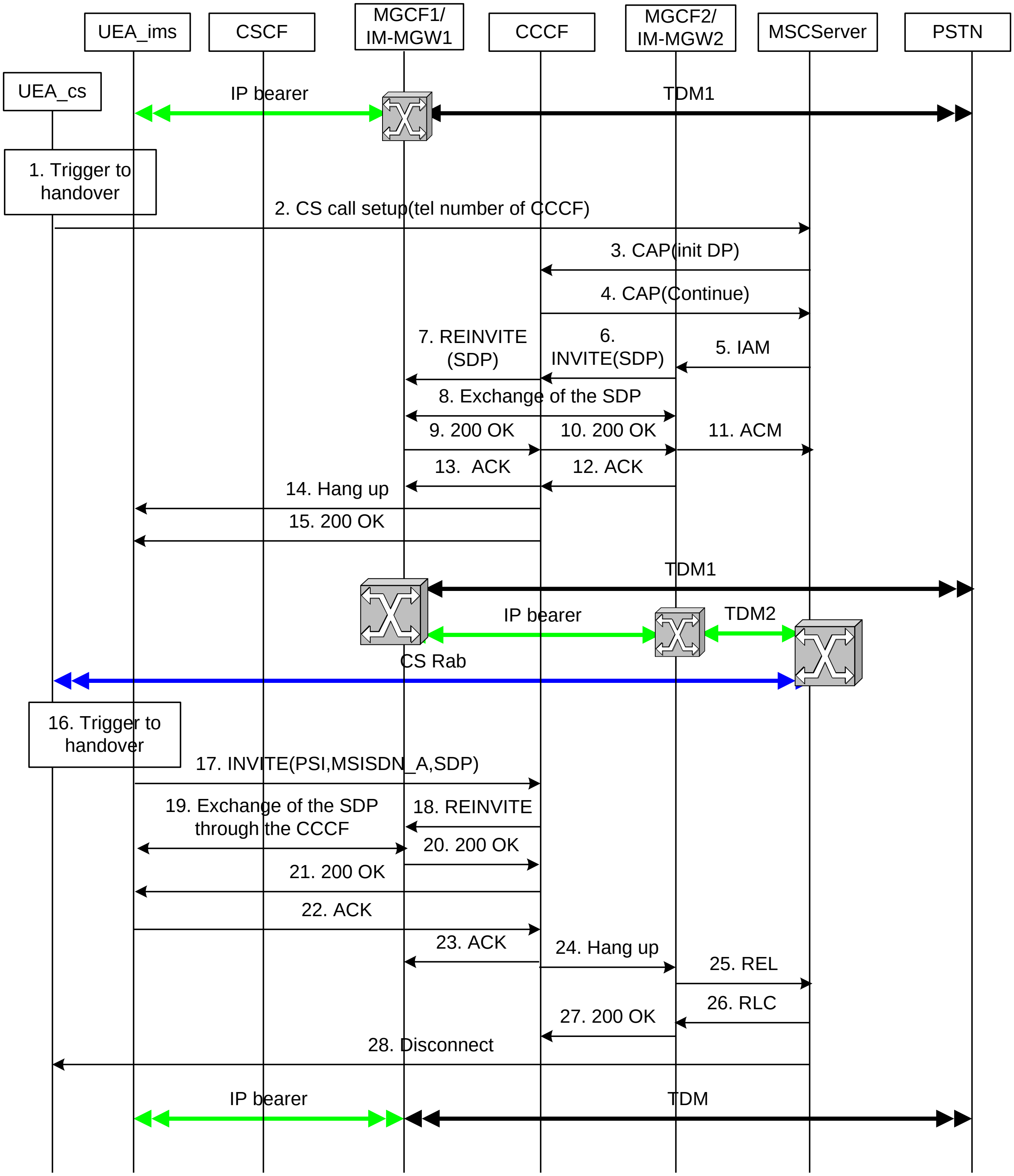


Figure 6.5.6.1.1-1: Call flow for call continuity from IMS to CS and back

1. According the radio condition and/or user preference the UEA decides to request call continuity from IMS domain to CS domain

2. User A originates a CS call request to CS domain. The called party address is the tel number of CCCF. The tel number of the CCCF can be automatically transferred in the registration or statically configured in the user A.

3. During the CS call procedure the MSC Server establishes a CAP dialogue to the CCCF according to the camel criteria in the MSC Server.

4. The CCCF sends a CAP(Continue) message to indicate the MSC Server to continue the voice call to CCCF.

5. After analysis on the called party address the MSC Server selects a proper MGCF2 and sends an IAM message to this MGCF2, containing the tel number of the CCCF and the MSISDN of user A.

6. The MGCF shall translate the tel number of the CCCF to a SIP routable SIP URI using an ENUM DNS translation mechanism. Then the MGCF2 sends an INVITE request to the CCCF(SIP URI, MSISDN of user A, SDP offer). The MGCF1 and MGCF2 may be one MGCF.

7. After receive the INVITE request from the MGCF2, the CCCF shall find the session between the CCCF and the MGCF1 by using the MSISDN of user A and send a REINVITE request to the MGCF1 to modify the SDP of this session.

8. The CCCF shall exchange the SDP between the MGCF1 and MGCF2.

9,10. The MGCF1 send a 200 OK to the MGCF2 through the CCCF.

11. The MGCF2 send an ACM message to the MSC Server. So the IP bearer between the MSC Server/MGW and the MGCF2/MGW2 is established.

12,13. The MGCF2 send an ACK message to the MGCF1 through the CCCF. So the IP bearer between the IM-MGW1 and the IM-MGW2 is established.

14,15.After the whole bearer paths are established the CCCF shall release the IMS session from user A to the CCCF. When the user A receive the release message it shall change its voice channel to CS domain.

So the bearer path from user A in CS domain to user B is established. See figure 6.9.6.1.1. When the MGCF1 and MGCF2 is the same MGCF then there is no need to exchange the SDP between this two MGCFs.

16. According the radio condition and/or user preference the UEA decides to handback to IMS domain

17. The user A shall register in the IMS domain firstly. After the registration the user A shall send a INVITE request to the CCCF, containing the PSI of CCCF ,the MSISDN of user A and the SDP. The INVITE request is routed to the CCCF.

NOTE: The registration in IMS domain may increase the duration of completion of call continuity. It is FFS how to handle this issue.

18. After receive the INVITE request, the CCCF shall find the session between the CCCF and the MGCF1 by using the MSISDN of user A and send a REINVITE request to the MGCF1 to modify the SDP of this session.

19. The CCCF shall exchange SDP between the user A and MGCF1 and establish the IP bearer path between the user A and IM-MGW1 controlled by MGCF1.

20,21,22,23. The MGCF1 sends a 200 OK to the user A through the CCCF and the user A send an ACK message to the MGCF1 through the CCCF. So the bearer path between the user A and the IM-MGW1 is reestablished.

24,25,26,27. The CCCF shall release the IMS session between the CCCF and the MGCF2. The MGCF2 shall release the path between the MGCF2 and MSC Server.

28. The MSC Server shall release the CS call between the MSC Server and user A. When the user A receive the release it shall change its voice channel to IMS domain.

Therefore, the voice call is transfered back to IMS domain.

6.5.6.1.2 CS to IMS call continuity

This section describes CS-IMS subscriber call continuity from the CS domain to the IMS. In this section a CS-IMS subscriber has an active call in the CS domain to a PSTN subscriber firstly and the CS-IMS UE decides to request call continuity from the CS domain to the IMS.. In this scenario we use the Camel 4 service to handle the call legs in CS domain. So we assume the MSC Server support Camel 4 service and the CCCF acts as a gsmSCF of Camel 4 service. The CS-IMS suberscirber has camel 4 subscription. in CS domain.

The following figure shows the detailed call continuity procedure.

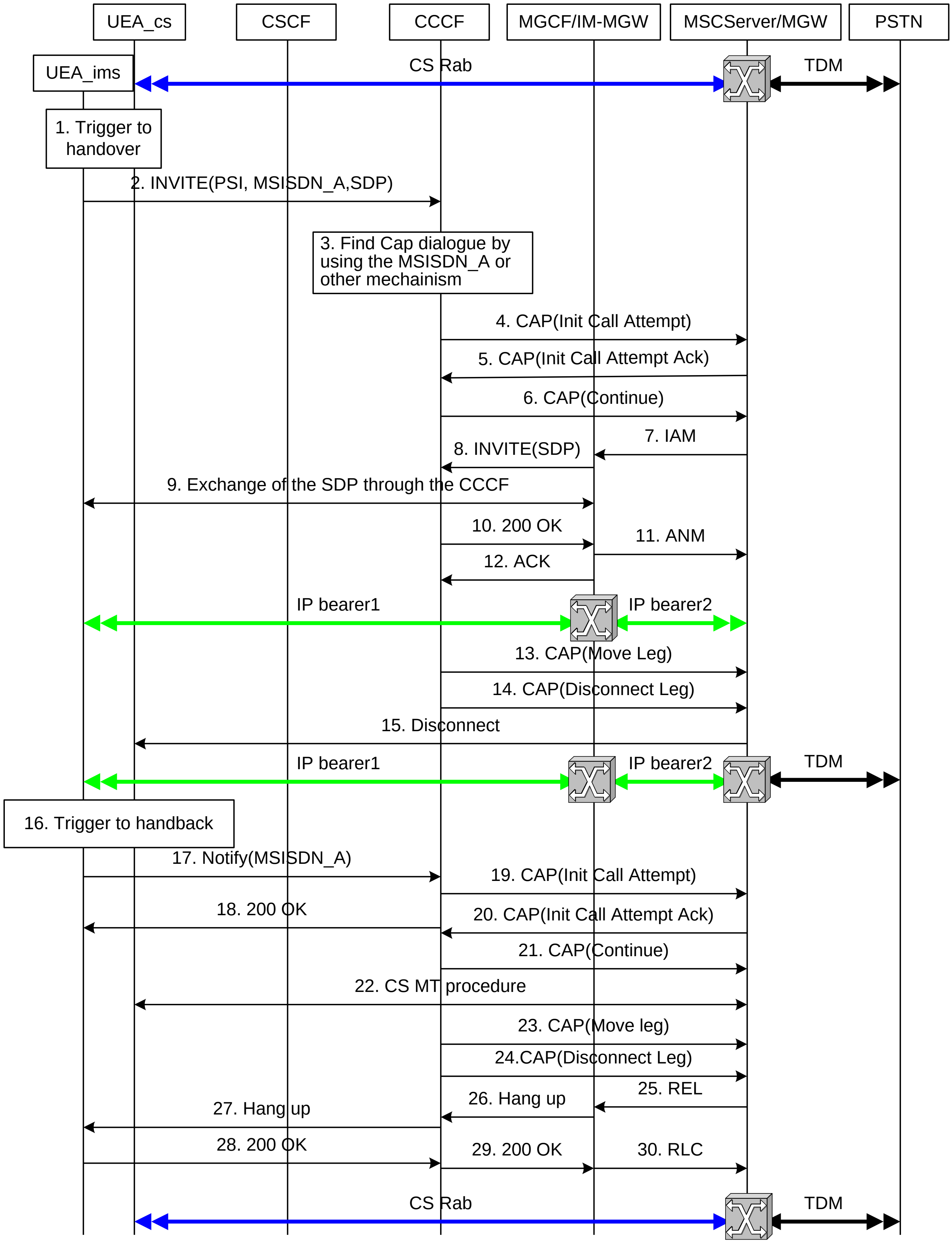


Figure 6.5.6.1.2-1: Call flow for call continuity from CS to IMS and back

1. According to the radio condition and/or user preference, the UEA decides to request call continuity from CS domain to IMS. User A register in IMS firstly.

NOTE 1: The registration in IMS domain may increase the duration of completion of call continuity. It is FFS how to handle this issue.

2. After the registration, user A send a INVITE request to the IMS core network(PSI of the CCCF, MSISDN of the user A, initial SDP). The INVITE request is routed through the P-CSCF in visiting IMS PLMN, the I/S-CSCF in the home IMS PLMN, then arrives the CCCF. The PSI of the CCCF can be automatically transferred in the registration or statically configured in the user A.

3. The CCCF stores the initial SDP included in the INVITE request. By using the MSISDN of user A or other mechanisms, the CCCF finds the CAP dialogue of the user established in the CS domain.

4,5,6. The CCCF shall send a CAP(initial call attempt) through this CAP dialogue to the MSC Server to create a CS call leg to the CCCF。This message includes the Tel number of CCCF. The MSC Server responeses a CAP(initial call attemp Ack) and the CCCF send a CAP(Continue) to indicate the MSC Server continue the CS call.

7. Once receive the indication, the MSC select a proper MGCF and send an IAM to this MGCF, the called number is the Tel number of CCCF..

8. After receive the IAM the MGCF shall translate the tel number of the CCCF to a SIP routable SIP URI using an ENUM DNS translation mechanismCCCFCCCF. Then the CCCF shall initiate a INVITE request(CCCF SIP URI, MSISDN of user A, SDP). The INVITE request is then routed to the CCCF.

9. When receive the INVITE request, the CCCF shall find the SDP information of the user A by using the MSISDN of user A. The CCCF shall exchange the SDP between the user A and the MGCF.

10,11. After the SDP exchange the CCCF sends a 200 OK to the MGCF and the MGCF send an ACK to the CCCF. So the bearer path between the user A and the IM-MGW controlled by the MGCF is established.

12. After establishment of the bearer path(IP bearer1) between the user A and the IM-MGW, the MGCF send a ANM to the MSC Server. So the bearer path(IP bearer2) between the MSC Server/MGW and IM-MGW is established.

13. The CCCF shall send a CAP(move leg) message through the CAP dialogue to the MSC Server to connect the call leg between the MSC Server/MGW and User B with the call leg between the MSC Server/MGW and the IM-MGW.

14,15. The CCCF shall send a CAP(disconnect leg) messagethrough the CAP dialogue to the MSC Server to release the CS connection between the MSC Server/MGW and user A in CS domain. When the user A receives the disconnect message it shall change it's voice channel to IMS domain.

So the bearer path from user A in IMS to user B is established. The CCCF in CS domain can still control the voice call, that means the Camel service of this voice call remain unchanged . The CCCF store the session information in IMS and the CAP dialogue information in CS domain. The CCCF associates this information by using the MSISDN of user A.

16. According the radio condition and user preference, UEA decide to handback to CS domain.

17. User A will send a Notify to CCCF using the existing session in IMS, containing the MSISDN of user A.

18. When the CCCF receive the Notify message, the CCCF shall return a ACK to the UE.

19,20,21. The CCCF shall look for whether the CAP dialogue of the user A have already existed. If it find the CAP dialogue then the CCCF shall send a CAP(initial call attempt) message through this dialogue to the MSC Server to originate a CS call leg to user A in CS domain..

22. When the MSC Server receives the message, the MSC shall page the user A and establish the signalling path and bearer path between the user A and the MSC Server as the normal CS termination call procedure.

NOTE 2: This step which including the paging procedure may increase the duration of completion of call continuity. So how to reestablish the CS call of user A is FFS.

23. After the CS call from MSC to user A has been established, the CCCF shall send a CAP(move leg) message to the MSC Server to connect the call leg between user A and MSC Server/MGW with the call leg between MSC Server/MGW and user B.

24. The CCCF shall send a CAP(disconnect leg) message to the MSC Server to release the call leg from MSC Server to CCCF.

25,26,27,28,29,30. The MSC Server release the ISUP/BICC connection between the MSC Server and the MGCF. The MGCF shall release the IMS session between MGCF and the CCCF. The CCCF shall release the IMS session between the CCCF and user A.. When the user A receive the release message it shall change it's voice channel to CS domain.

Therefore, the voice call of user A handback to CS domain.

6.5.8 Evaluation of the Model

The benefits of this solution are:

1. CS domain and IMS domain can belong to different operators.
2. This solution has no impact on the existing UMTS/GSM specifications and IMS specifications.
3. This solution is an access-agnostic solution in IMS domain.
4. The time delay of call continuity in this solution is very limit because the user A change it's voice path after the second voice path is established.
5. This solution has no modification in CS domain. So the supplymental services in CS domain can provide to the user unchanged before the call continuity. After the completion of call continuity from CS domain to IMS domain, the IM service in IMS domain can also provide to the user. The Camel dialogue exist during the whole call so it is possible that the camel logic in CCCF can also control this voice call even after the call continuity.

The drawbacks of this solution are:

1. For the IMS to CS call continuity procedure the MSC Server do not need support CAMEL 4 service. For the CS domain to IMS call continuity procedure the MSC Server must support CAMEL 4 service.
2. The subscriber who wants to transfer from the CS domain to IMS must have CAMEL 4 subscription.

7 Security

7.1 Access security for CS Domain Handover

TS 33.102 [18] describes the Security Architecture for GSM and UMTS subscribers, including the impact of handover on security. VCC places no additional requirements upon the CS domain above those described in TS 33.102 [18].

7.2 Access Security for IP-based Services

TS 33.203 [19] describes Access Security for IP-based services. VCC places no additional requirements upon the IMS domain above those described in TS 33.203 [19].

8 Charging

9 Comparison of the Architecture Model

10 Conclusion

During the course of this study, a number of alternative architectural solutions that enable Voice Call Continuity between CS and IMS domains have been proposed and documented in this report. As a result of analysis, some options have been removed or placed in annexes of this document (thereby ensuring that the work is not completely lost). Because of the fact that the almost all of the current deployed CS networks do not support CAMEL 4, the CAMEL 4 based-solution has not been further investigated. The outcome is that two alternatives remained: Original Domain Controlled (clause 6.4) and IMS Controlled Static Anchoring (clause 6.3).

Work has been undertaken to identify and separately document any common aspects of the two alternatives. In working on the study, it was determined that the VCC solution impacts Registration, Origination, Termination, Network Domain Selection as well as mid-call supplementary services. When these areas were deemed common or only had minor differences, they were placed in common sections. The common aspects have been captured in clauses 6.2 and 6.2a. These clauses retain some solution-specific details where necessary.

Finally, the study has shown that both the Original Domain Controlled and IMS Controlled Static Anchoring solutions have their advantages and disadvantages. However, to facilitate interoperability it is believed that the interests of the industry would be best served by pursuing a single solution. It is further believed that on balance the comparative advantages of the IMS Controlled Static Anchoring model make it the preferred way forward. This report therefore concludes that further effort on the VCC Work Item should focus on a solution based on the IMS Controlled Static Anchoring approach, as the basis for standardization.

Annex A: IMS based HO control Model

A.1 IMS based HO control Model

In the following section, a reference logical model of IMS based HO control is provided, and 4 general HO control modes and two directions of new session establishment mechanism during call continuity procedure was described and discussed based on it.

A.1.1 Logical Model Introduction

For the flexibilities from IMS network, call continuity from IMS to CS may be provided in many ways, so in present paper an abstract system model is provided. Based on this abstract system model, different control modes can be analysed and so help to determine a preferred one.

A system model to support IMS controlled bidirectional CS2IMS call continuity is proposed shown in the figure A.1.

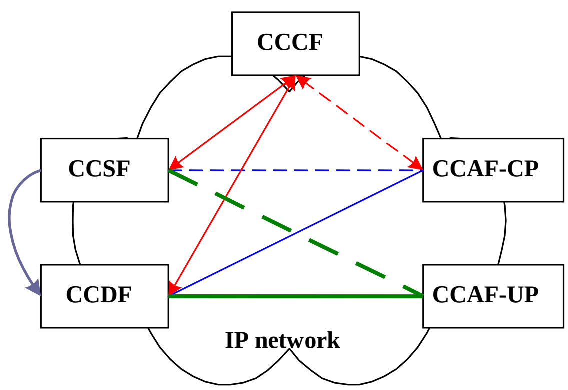


Figure A.1: system model

In the figure, CCCF, CCSF, CCDF and CCAF are logical functional units and shall be mapped onto actual IMS entities and/or CS entities. Some brief introduction below:

CCSF (Call Continuity Source Function): CCSF is the functional unit, by which the user establishes session to the remote UE side before call continuity, including control connection at control plane to CCAF-CP and media connection at user plane to CCAF-UP.

CCDF (Call Continuity Destination Function): CCDF is the functional unit, by which the call continuity user establishes session to the remote UE side after call continuity, including control connection at control plane to CCAF-CP and media connection at user plane to CCAF-UP. The new session between CCDF and CCAF will replace the old session between CCSF and CCAF after call continuity.

CCAF (Call Continuity Anchor Function): The CCAF acts as an anchor in call continuity procedure. It establishes the connection with CCSF before call continuity and with CCDF after call continuity. CCAF consists of two parts: CCAF-CP (for control plane) and CCAF-UP (for user plane), which may reside in the same entity or in the different entities.

- **CCAF-CP**: CCAF-CP acts as an anchor in control plane in call continuity procedure, establishes control connection at control plane with CCSF and CCDF separately before/after call continuity. During the course of call continuity procedures, CCAF-CP is able to establish new session control connection with CCDF while maintaining the old control connection with CCSF, and when new control session with CCDF is established successfully, the old control connection can be replaced with new-established control connection.

- **CCAF-UP**: CCAF-CP acts as an anchor in user plane in call continuity procedure, establishes media connection at user plane with CCSF and CCDF separately before/after call continuity. During the course of call continuity, under the control of CCAF-CP, the old media connection can be replaced with new-established media connection.

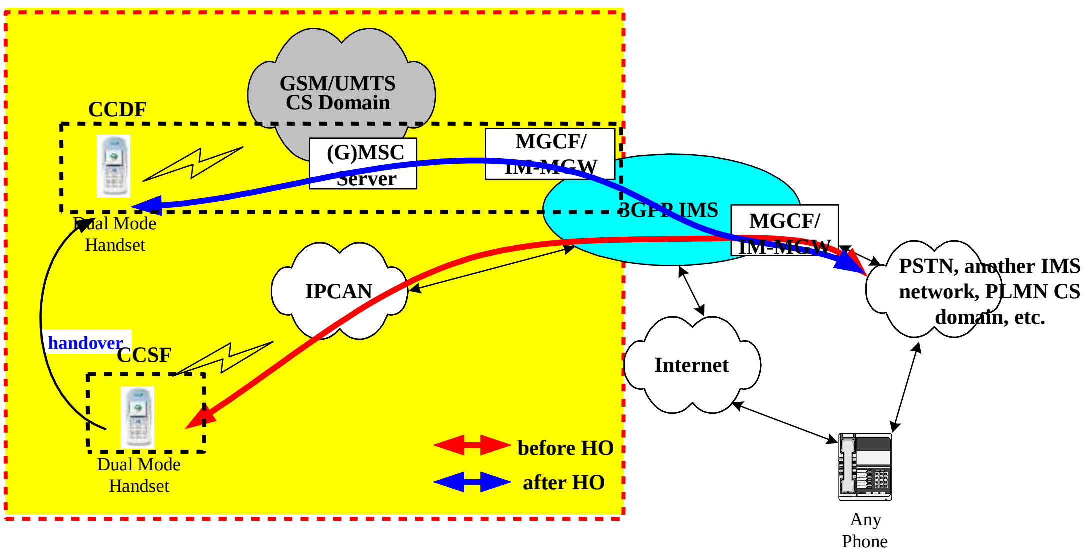
CCCF (Call Continuity Control Function): CCCF is an optional unit and is to enable:

- Authenticating call continuity request
- Allocating and Controlling CCAF-CP

As we discuss on IMS based control modes, it was benefiting to think the call continuity as interaction between three SIP UAs. As shown in figure A.1, the remote SIP UA (i.e. CCAF) connects with local SIP UA (i.e. CCSF) by dashed before call continuity. The same remote SIP UA (i.e. CCAF) connects with another local SIP UA (i.e. CCDF) by real line after call continuity.

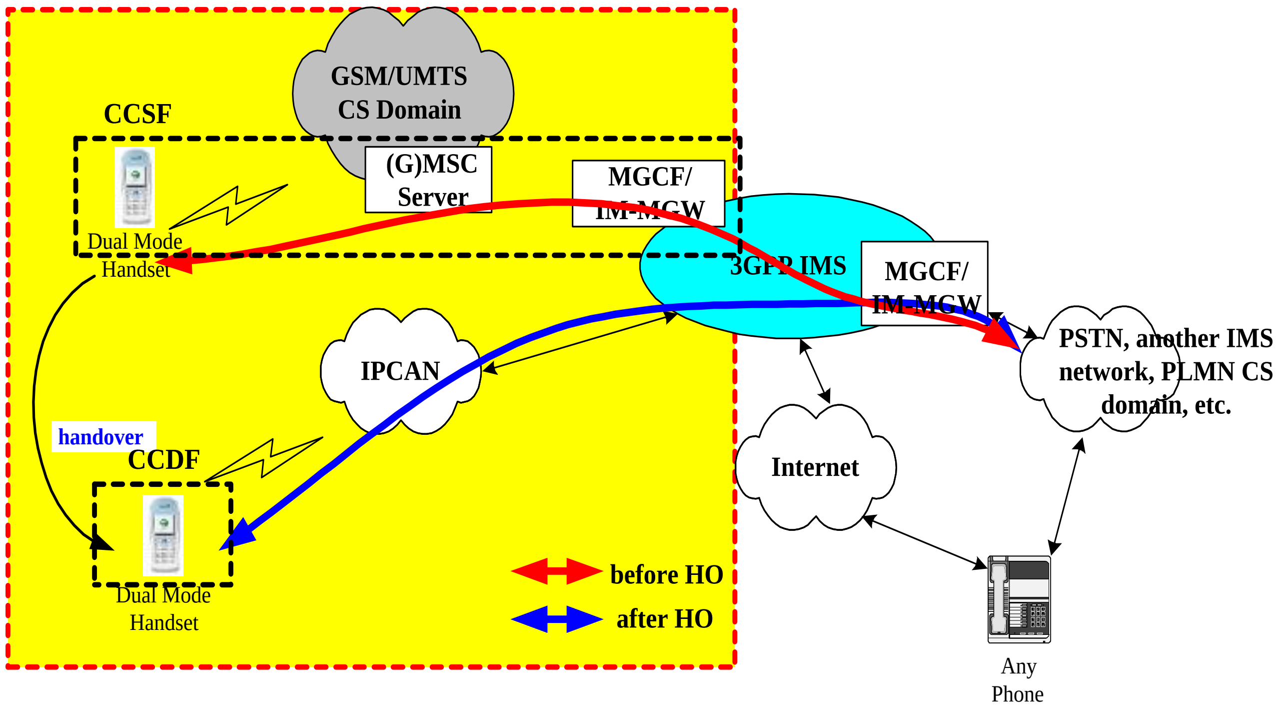
Figures A.2 and A.3 below show the session connection scenarios when user equipment switches between IMS-controlled VoIP call and CS call.

To unify different terminals in SIP UA mode, we need using the MGCF/MGW to convert the CS mode terminal to SIP mode terminal. That is to say: in case of the UE is in CS domain, the combination of CS terminal and MGCF/IM-MGW is to perform the function of CCSF/CCDF as shown in figures A.2 and A.3.



NOTE: If the remote user is an IMS user, the MGCF/IM-MGW is not needed in the figure.

Figure A.2: Call continuity from IMS-controlled VoIP call and CS call



NOTE: If the remote user is an IMS user, the MGCF/IM-MGW is not needed in the figure.

NOTE: For simplicity, old or original session indicates the session before HO, and new session indicates the session after HO. Meanwhile HO UE means the UE performing call continuity.

Figure A.3: Call continuity from CS call to IMS-controlled VoIP call

A.1.2 Control Mode Analysis

Based on the above system model, four different control modes may be adopted. In this section these four control modes will be introduced in detail:

- End-to-end mode, including terminal-controlled mode and network-controlled mode;
- Segmented mode, including CP-segmented (control-plane-segmented) mode and CPandUP-segmented (control-plane and user-plane segmented) mode.

1. End-to-end/Terminal-controlled Mode:

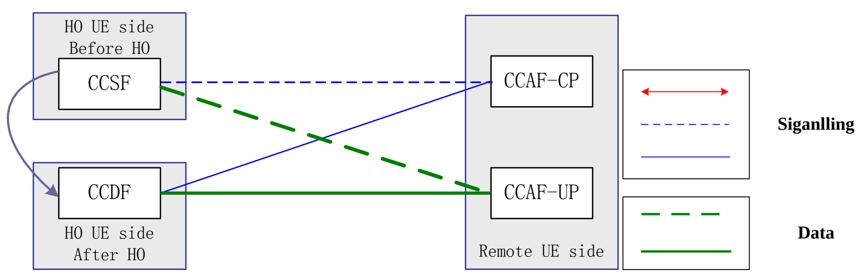


Figure A.4: Terminal-controlled Mode

In this mode CCAF resides in the remote UE side (inside the remote SIP UE in case of it is in IMS, or in MGCF/IM-MGW which performing IMS-CS interworking for the remote UE in case of the remote UE is in CS/PSTN). During HO procedure, the HO UE indicates the CCAF in remote UE side to replace the old session with the new session. Detailed behaviour includes:

- Original session establishment (before HO): CCSF establishes session with CCAF. CCSF may be either the calling part or the called part;
- Call continuity procedure, including call continuity detection, initiation and execution:
    - Call continuity detection is executed by CCSF;
    - Contnuity initiation: CCSF indicates to CCDF or CCAF to perform call continuity;
    - Call continuity execution: either CCDF or CCAF may be the master controller of call continuity, means to initiates the establishment of the new session. The old session between CCSF and CCAF is replaced with the new session between CCDF and CCAF in this step.

Pros:

i. This scheme is the simplest one;

ii. Network resource utilised to implemented this scheme is the smallest in all modes (only need to support corresponding session establishment).

Cons:

i. Increase more requirements for the remote SIP UE or MGCF/IM-MGW (in case of the remote UE is in CS/PSTN), e.g. support to establish new session with CCDF while maintaining the old session with CCSF, and perform session replacement when finish the establishment of new session;

ii. A network can not control any of the call continuity procedure;

iii. In the view of the network, session before HO is completely different from the one after HO, so the service control upon old session can not be maintained on the new session;

iv. From charging aspect, the old session and new one are regarded as different sessions. This is not reasonable, especially for the remote side.

2. End-to-end/Network-controlled Mode:

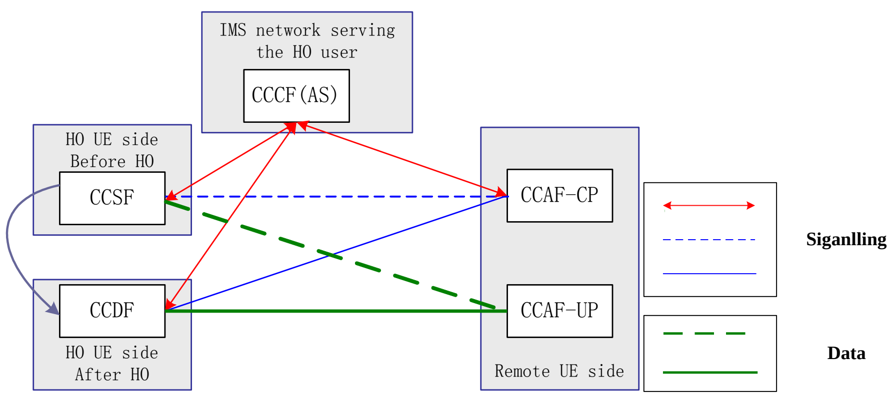


Figure A.5: Network-controlled Mode

In this mode CCAF resides in the remote UE side (inside the remote SIP UE in case of it is in IMS, or in MGCF/IM-MGW which performing IMS-CS interworking for the remote UE in case of the remote UE is in CS/PSTN). The HO UE establishes sessions with CCAF through CCSF before HO and CCDF after HO, and indicates the remote UE side to replace the old session with the new session. Different to the previous mode, CCCF is included which resides generally in home IMS networks serving the HO UE. Detailed behaviour includes:

- Original session establishment (before HO): CCSF establishes session with CCAF. During the session establishment CCCF is triggered;
- Call continuity procedure, including call continuity detection, call continuity authentication, call continuity initiation and call continuity execution:
  - Call continuity detection is executed by CCSF;
  - Call continuity authentication: CCSF sends call continuity request to CCCF and CCCF authenticates the request;
  - Call continuity initiation: After completion of call continuity authentication, CCCF may either indicate directly to CCAF/CCDF to perform call continuity, or return authentication acknowledgement to CCSF and then CCSF sends call continuity indication;
  - Call continuity execution: the old session between CCAF and CCSF is replaced by the new session between CCAF and CCDF, which may be initiated by CCAF or CCDF.

Pros:

i. This scheme is similar as End-to-End/Terminal-Controlled Mode and utilised resource is also small;

ii. A network can control the procedure at a certain extent and the handling of CCCF is simple.

Cons:

i. Similar as the End-to-End/Terminal-Controlled Mode, it increase more requirements for the remote SIP UE or MGCF (in case of the remote UE is in CS/PSTN);

ii. A network can only control part of the call continuity procedure, i.e. Authentication;

iii. In the view of the network, session before HO is a completely different one to the session after HO, then service control upon old session can not be maintained on the new session;

iv. From charging aspect, the old session and new one are regarded as different sessions, which is not reasonable.

3. CP-segmented Mode:

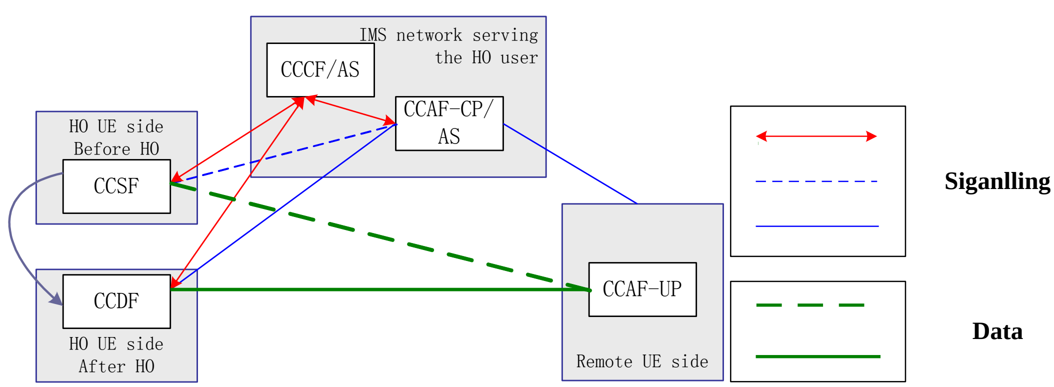


Figure A.6: CP-segmented Mode

In this mode CCCF is included and CCAF-CP and CCAF-UP are located in different entities. CCAF-CP and CCCF are in the home IMS network serving the HO UE. The CCAF-UP reside in the remote UE side (inside the remote SIP UE in case of it is in IMS, or in IM-MGW which performing IMS-CS interworking for the remote UE in case of the remote UE is in CS/PSTN). Under control of CCCF, CCAF-CP terminates the session with CCSF, and re-establishes the session with the remote UE side at original session establishment. At control plane CCAF-CP splits the control session between the HO UE and the remote UE side into two segments and controls these two sessions in 3PCC mode. Since CCAF-UP still resides in the remote UE side, media exchange between the HO UE and the remote UE side still works in end-to-end mode. Detailed behaviour:

- Original session establishment (before HO):
  - CCSF establishes session with CCAF;
  - CCCF is triggered during the initial session establishment procedure;
  - CCCF allocates the control instance to perform the function of CCAF-CP for present session. Under the control of CCCF, CCAF-CP splits the control session between UE performing HO and the remote UE side into two segments and controls these two sessions in 3PCC mode. Media flow between CCSF and CCAF-UP communicates directly.
- Call continuity procedure, including call continuity detection, call continuity authentication, call continuity initiation and call continuity execution:
  - Call continuity detection, call continuity authentication, call continuity initiation and call continuity execution procedure is the same as that in End-to-End/Network-controlled mode;
  - During the call continuity procedure, connection at control plane between CCAF-CP and the remote UE side remains unchanged excepting re-negotiation to change the media exchange direction and the exchanged media attributes if the media capabilities of CCSF and CCDF are different, IMS service control relationship upon this session is not affected by call continuity.

Pros:

i. During call continuity procedure, the session between CCAF and the remote UE side is only need to perform re-negotiation. Service control and charging handling upon this session segment is not affected;

ii. Requirements for the remote UE side is low: the remote UE side is only needed to support re-negotiation;

iii. Utilised network resource is also small comparing with CPandUP-Segmented Mode.

Cons:

i. More complicated and 3PCC capability need to be supported by AS comparing with End-to-end Mode.

4. CPandUP-segmented Mode:

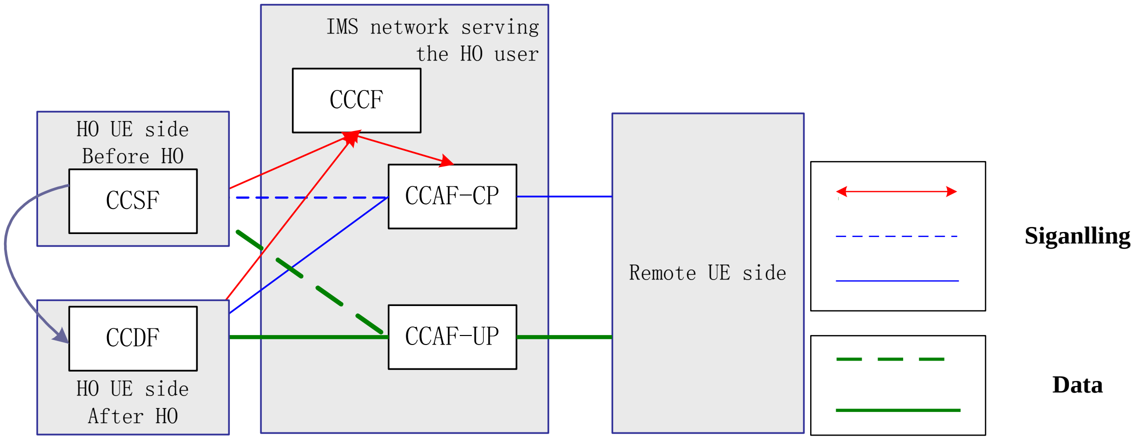


Figure A.7: CPandUP-segmented Mode

In this mode CCCF is included and CCAF-CP and CCAF-UP/CP are all located in the home IMS network serving the HO UE. Under control of CCCF, CCAF-CP terminates the session between CCSF and CCAF-CP, and re-establishes the session with the remote UE side at original session establishment. At control plane CCAF-CP splits the session between HO UE and the remote UE side into two segments and controls these two segments of session in 3PCC mode. Meanwhile, under the control of CCAF-CP, IMS network allocates media resource performing the function of CCAF-UP for present session, and then media exchange at user plane between HO UE and the remote UE side is also split into two segmented, too. Detailed behaviour:

- Original session establishment (before HO):
  - CCSF sets up session with CCAF;
  - CCCF is triggered during the initial session establishment procedure;
  - CCCF allocates the control instance to perform the function of CCAF-CP for present session. Under the control of CCCF, CCAF-CP splits the control session between HO UE and the remote UE side into two segments and controls these two sessions in 3PCC mode;
  - CCCF/CCAF-CP applies media resource in network as CCAF-UP, establishes two media flows: one between CCSF and CCAF-UP, another between CCAF-UP and the remote UE side, and through connect these two media flows in CCAF-UP.
- Call continuity procedure, including call continuity detection, call continuity authentication, call continuity initiation and call continuity execution:
  - Call continuity detection, call continuity authentication, call continuity initiation and call continuity execution procedure is the same as that in End-to-End/Network-controlled mode. When finish the establishment of new session with the CCDF (and the re-negotiation procedure with the remote UE if needed), the CCAF-UP will change the connection of the segmented media flows;
  - During the call continuity procedure, connection at control plane and user plane between CCAF-CP/UP and the remote UE side remains changeless excepting re-negotiation to change the exchanged media attributes in case of that the media capability of CCSF and CCDF is different, and IMS service control relation upon this session is not be affected by call continuity.

To provide a better service continuity in view of media exchange, the CCAF-UP can provide some optional function, such as media duplication and filtering, that means, during the call continuity procedure, the CCAF-UP duplicates the media flow from the remote side and sends it to the CCSF and CCDF simultaneously, and filters the media flow from CCSF and CCDF to send to the remote side (in case of it is implemented in the network) or present to the remote user (in case of it is located inside the remote UE).

Pros:

i. During call continuity procedure, the session between CCAF and the remote UE side is needed to perform re-negotiation only if the media capabilities of CCSF and CCDF are different. Service control and charging handling upon this session segment are not be affected;

ii. Requirements for the remote UE side is low: The remote UE side is only needed to support re-negotiation;

iii. Users' feeling is best in view of media exchange continuity when MRF provides media duplication/filtering function.

Cons:

i. More complicated and 3PCC capability is needed to be supported by AS and AS needs to control MRF;

ii. Need more network resource, especially media resource to perform CCAF-UP function.

5. Brief summary:

From the discussion above, in segmented mode, session between CCAF-CP and the remote UE side is only needed to support re-negotiation, and other aspects will not be affected, e.g. IMS service control and charging for the remote user. In segmented mode the remote UE side is not required to maintain the old session while establishing new session and support session replacement;

While in End-to-End mode, the CCAF function is located in the remote UE side and the remote UE side is asked to support session replacement. It is difficult to ask all of current SIP UEs to support it, and in case of the remote UE is in CS/PSTN, it will have some special requirements for MGCF to support session replacement;

As for the network-controlled models (including Segmented modes and End-to-end/Network-controlled Mode), the HO control point (CCCF, may be include CCAF-CP) is implemented in an AS (HO-AS) located in IMS network serving the HO user, which will be triggered when original session established.

A.1.3 Direction of Session Establishment during HO

During HO, new session between CCDF and CCAF-UP is established to replace the old one for service continuity. There are two directions to establish the new session: CCDF-initiated session establishment and CCAF-initiated session establishment. Two directions stated here can be supported by the all control models showing above.

A.1.3.1 CCDF-initiated new session establishment

CCDF sends new session establishment Req. to CCAF-CP through CS and CS/IMS IW gateway in case of call continuity from IMS to CS, or directly from IMS in case of call continuity from CS to IMS. In request message HO indication and old session-related information are delivered. HO indication is used to indicate CCAF to perform session replacement. Old session-related information is used to indicate the session should be replaced. According to the information receiving in the request message, CCAF sets up new session to replace the old one.

According to the information receiving in the request message, entities involved in HO between CCAF and CCDF can perform some specific operations, e.g. avoid of duplicated service trigger, specific charging handling. CCAF can also perform some specific operations, e.g. avoid of ringing or responding 180.

Pros:

i. It's easy to avoid affecting ongoing services (*1);

ii. It's easy to realize the CS routing optimization (*2);

iii. In segmented mode, CS CDR may be picked out to be handled distinguishingly based on the special called party number, i.e. the E.164 number assigned to the CCAF (corresponding to HO PSI);

iv. In segmented mode, CCAF-CP is implemented in network, so it is easy to avoid ringing/responding 180 when new session is establishing, which can provide better service feeling;

v. It has low dependency to the old session since it is unnecessary for CCSF to send REFER to indicate the CCAF-CP to initiate new session establishment through the old session.

Cons:

In case of IMS to CS call continuity:

i. Specific E.164 number (e.g. E.164 number corresponding to HO PSI) or prefix need to be allocated to control the routing of new session to CCAF-CP via IMS;

ii. Routing information need to be configured to this specific E.164 number.

A.1.3.2 CCAF-initiated new session establishment

CCAF sends new session establishment Req. to CCDF through CS/IMS IW gateway and CS domain to set up an IMS-CS IW call in case of call continuity from IMS to CS, or directly through IMS domain in case of handover from CS to IMS. In the new session establishment request message, HO indication and/or CCSF-related information may be optionally included. HO indication is used to indicate the requested session is a handover-related session. CCSF-related information is used to check the validation of the request by CCDF. On receiving the session establishment request, CCDF establishes new session with CCAF.

If the request message for establishing new session includes the optional HO indication and/or CCSF-related information, CCDF and entities involved in HO between CCAF and CCDF can also perform some specific operations based on it, e.g. avoid of ringing for CCDF, avoid of duplicated service trigger, and/or perform specific charging handling for corresponding network entities.

Pros:

In case of IMS to CS call continuity:

i. A user's MSISDN can be used to address the user during new session establishment, and no special routing info. is needed to be configured in CS domain;

ii. An operator can determine whether provides E.164 number or not to CCAF-CP. If the CCAF is assigned the E.164 number, CS CDR may be picked out to be handled distinguishingly based on the special calling party number for CCAF.

Cons:

i. It's not easy to avoid affecting ongoing services (*1);

ii. It's not easy to realize the CS routing optimization (*2);

iii. It is not easy to avoid ringing/ responding 180 when new session is establishing;

iv. In this case, it is CCSF needs to send HO indication to CCAF and provides CCDF-related info through the old session. In environment where attenuation of signalling is rapid, e.g. in a WLAN, the steady connection is not easy to be guaranteed.

NOTES:

(*1) Avoid the effects from other services.

To insure establishment of new session, it should be avoided the effects from some services which may result in call transfer or call reject. In IMS domain, benefiting from the flexibilities of SIP and powerful service control capabilities of IMS network, effects from these services can be avoided with configuration of iFC or minor modifications to network entities. But in CS domain, the modification on network entities should be as little as possible, and the extensibility of CS signalling is limited. In CS domain, to meet such requirements, the mechanism of elusion of originating side of services based on the called number is mature (just as been adopted in the CS domain to avoid the impact of originating side supplementary service to the emergency service and special number service (for example, 800)). On the contrary, for elusion of terminating side of services, no appropriate mechanism is available. So when call continuity happens, it is easy for CCDF-initiation new session establishment to provide elusion of effects from other services.

(*2) Routing optimisation in CS domain (avoiding utilisation of long distance trunk)

In segmented mode, if a user call continuity from IMS to CS, CCSF resides in the user's home network, so:

- If the new session is initiated by CCAF and the call usually will be routed from MGCF in home network to CS domain no matter whether the user is roaming, then resource of long distance trunk is utilised unnecessarily.
- If the new session is initiated by CCDF, according to the routing information configuration of the specific E.164 number, CS domain is able to select an IW gateway locally to route the call to IMS, unnecessary utilisation of long distance trunk/toll can be avoided.

So based on the discussion above, CCDF-initiated session establishment procedure can avoid utilization of long distance trunk.

Annex B: Routing Selection Decision Logic

B.1 NeDS function is invoked from the CS Domain

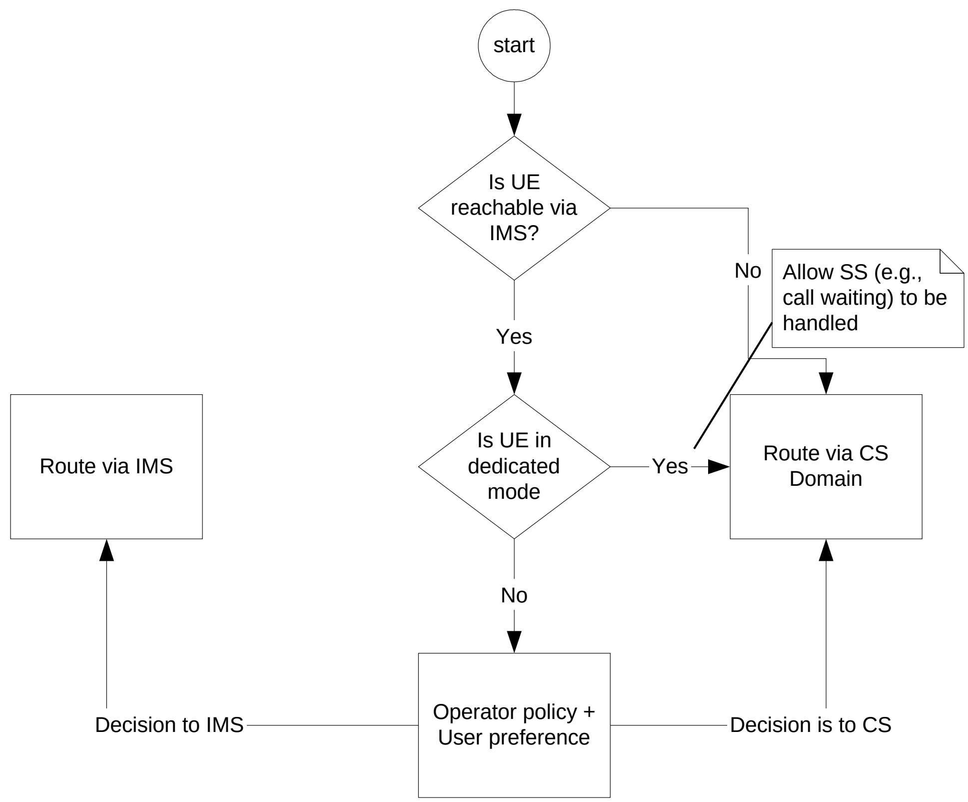


Figure B.1: NeDS logic (CS Domain triggered)

If UE is already engaged in CS call then additional call should be continuing directed to CS domain. This is important for supplementary services like CW, conferencing, CFB, HOLD.

When CS is idle, then user preference may be indicated during IMS Registration phase that voice call should be done via CS or IMS. However, operator's rule is deciding which way to route the call.

B.2 NeDS function is invoked from the IMS Domain

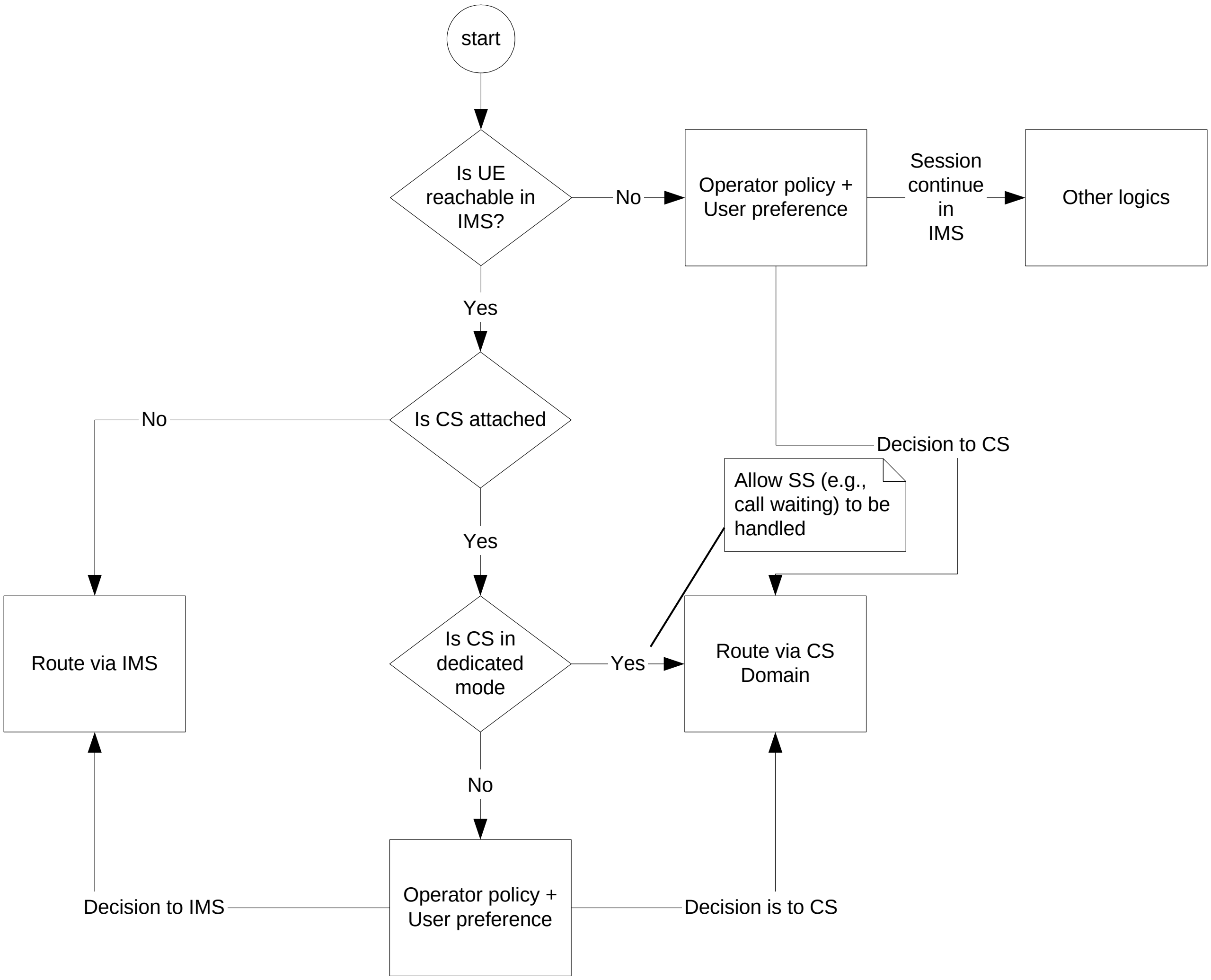


Figure B.2: NeDS logic (IMS Domain triggered)

If UE is not registered (or reachable) in the IMS domain then it should be possible to invoke other IMS service besides VCC if desired.

If UE is already engaged in CS call then additional call should be continuing directed to CS domain. This is important for supplementary services like CW, conferencing, CFB, HOLD.

User preference may be indicated during IMS Registration phase that voice call should be done via CS or IMS. However, operator 's rule is deciding which way to route the call.

Annex C: Bearer optimisation for static anchoring of CS calls in IMS

Bearer backhaul can be optimised by making use of functions in the "local IMS" that are local to the VCC user. This requires certain local IMS functions to be made available to support roamed-in users.

Bearer path optimisations for static anchoring of CS calls with IMS can be achieved by placing the CS/IMS interworking function in a local IMS network as opposed to a CS/IMS interworking function in the home IMS. It is proposed to use an MGCF in the local network to access the CCCF in home network. New inter operator reference points are introduced requiring the need for a detailed study on trust relationships and accounting agreements for the inter operator interfaces.

The routing of calls from CS domain network into IMS takes place by the use of a DN associated with the CCCF PSI. A user is assigned a CCCF DN upon IMS registration. The selection of a CCCF Application Server for a particular user is determined by network scalability and the user's current location. The CCCF DN provided to the user may be assigned in a manner to facilitate routing of CS calls into IMS such that the CS/IMS interworking function occurs at the local IMS network.

When routing calls from IMS to CS, the BGCF in the home IMS network may be configured such that it does not select a CS/IMS interworking function in home IMS for the case when the user is roaming outside of home network. The BGCF in home IMS network forwards the session to a BGCF in a network closer to the CS domain network which is where the CS/IMS interworking function is invoked.

Figure C.1 below provides an example of the bearer optimisation achieved by using this strategy for a CS origination terminating to a remote party located in user's home IMS network.

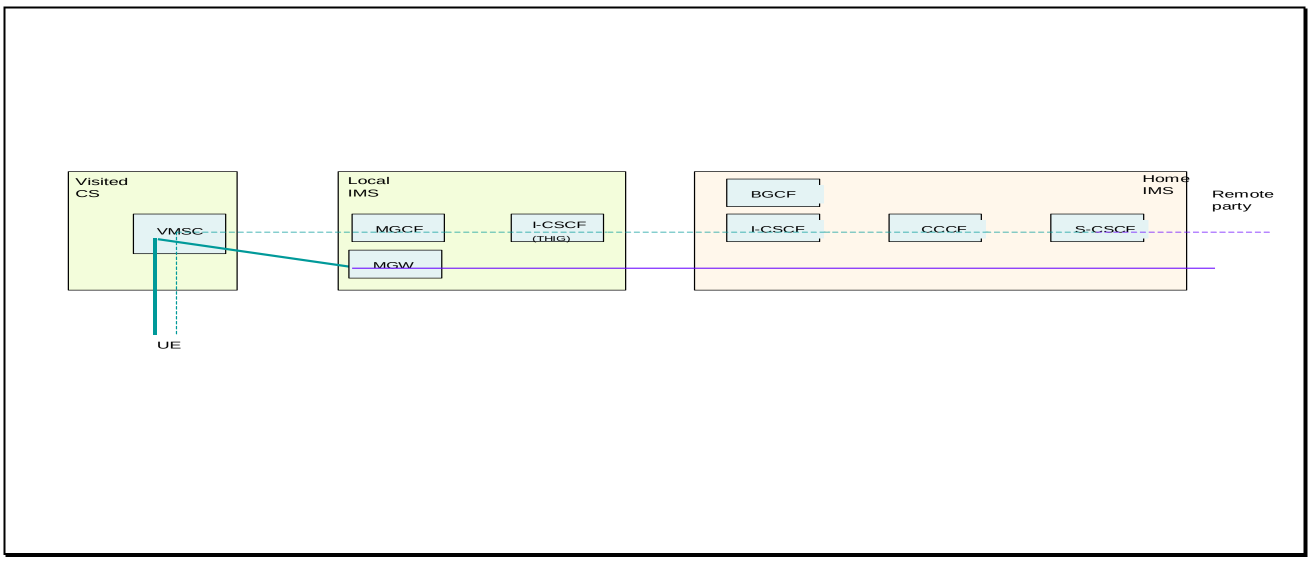


Figure C.1: CS origination to an IMS subscriber

Allocation of the DN used for routing CS calls to the CCCF is performed such that the CS call terminating to the CCCF is routed via local IMS network with the CS/IMS interworking function for this call executed in the local IMS network. The MGCF in the local IMS network may either forward the session directly to an I-CSCF in the user's home IMS network or via an I-CSCF in the local IMS network in case where network configuration hiding (THIG) is in place at the local IMS network.

The TDM bearer backhaul stops at the MGW in the local IMS network. The RTP flows all the way to the remote party in the user's home IMS network.

Figure C.2 below provides an example of the bearer optimisation achieved by using this strategy for a CS origination terminating to a remote party served by the VMSC used for the originating leg of the call. Note that the originating and terminating users are assumed to belong to the same network and the GMSC function for the terminating CS user is not shown for the sake of brevity.

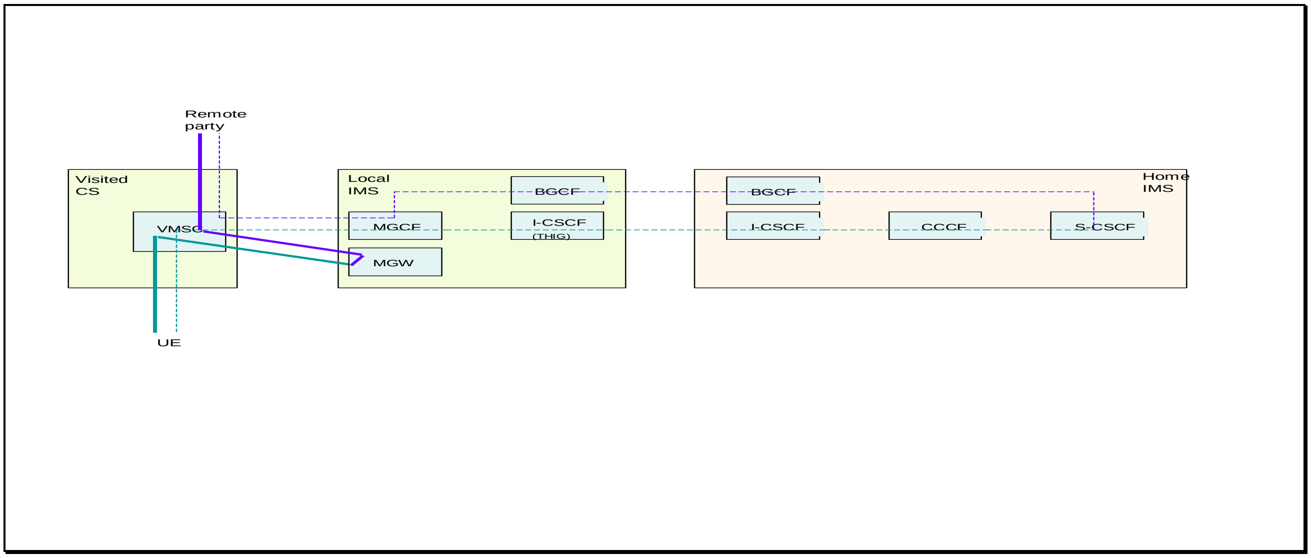


Figure C.2: CS origination to CS subscriber

The procedure for routing the CS call for static anchoring in the IMS network is the same as the scenario described in Figure C.1. For routing of call back to the CS domain, the BGCF in the home IMS network of the originating user forwards the session to a BGCF in an IMS network closest to the CS termination for placement of CS/IMS interworking function in that IMS network. Note that the originating and terminating paths may not use the same IMS network or the same MGCF or MGW in the IMS network if using the same IMS network for placement of CS/IMS interworking function as shown in this example.

As depicted by the bearer flow, the bearer path remains within the visited network without extending to the home IMS network.

Annex D: Evaluation Criteria

Criteria for evaluating call continuity architecture solution.

Table D.1

Criteria	Definition	Relative Weighting Factor
Impact on Circuit Switch (CS)	Identify and define changes and new requirements to the Circuit Switch (CS) System and CS part of UE.	High. Changes should be minimized
Impact on IMS	Identify and define changes and new requirements to the IMS System and IMS part of UE.	Medium. Fewer Changes are preferred.
Support of Supplementary Service	Identify the types of CS and IMS supplementary services that can be supported with the architecture.	High. Identify those supplementary services that can be supported.
Support of Roaming	Identify whether the architecture allows subscriber to use the service when roaming.	Medium.
Scalability	Identify how the network scales as the service usage grows.	High.
User Experience	Identify and define changes and new requirements to the subscriber behavior.	High.
Resource Usage	Identify which network resources are impacted to implement the architecture.	Medium. e.g., identify any routing impact to resource usage
Service Interactions	Identify impact on other services to support call continuity.	High. Identify impacts to existing services, e.g., SMS, PS domain services.
Voice Break	Identify the length of voice break introduced by the architecture solution during transition from one network to another network.	High. The length of the voice break should be reasonable to not impact user experience.
Procedure Latency	Identify the time it takes from start to completion of the call continuity procedures. Identify critical intervals that need to be completed in order for the procedure to complete successfully.	High.
Service Scenarios	Identify the degree to which the architecture solution supports envisioned interworking scenarios.	High.
Requirements on the Visited Network	Identify the requirements placed on the visited network in order for the voice continuity procedure to work.	Medium.
Applicability to 3GPP PS IPCAN	Identify whether the architecture solution can support VCC between CS voice and VoIP/IMS call over I-WLAN, 3GPP GPRS, LTE, etc.	High.
Bearer Path Detour	Identify whether the architecture solution introduce bearer path detour after transition from one network to another network.	Medium.
Flexibility for operator control	Identify what kind of operator control can be supported by the architecture solution.	High.
Robustness	Identify whether the voice service would continue when the serving domain becomes unstable, but the other domain is available.	Medium.
Accounting Data Collection	Identify the architecture's impact to the operator's ability for consistent charging across domains and the accounting settlement between home and roaming operator.	High.
Emergency Call Support	Identify whether the architecture solution supports call continuity for emergency call.	High.
Impact to Security	Identify whether the call continuity procedure impacts existing security in CS and IMS domain.	High.
Idle Mode Call Termination	Identify how the architecture solution supports Call Termination in Idle mode.	High.

Annex E: "Call Reestablishment on Domain Transfer" for IMS-controlled static anchoring

Radio coverage may not always be available to perform Classical Handover in conjunction with VCC, e.g. the case where the 3G cell is configured as PS-only. A new service Call Reestablishment on Domain Transfer (CReDT) is introduced to address such cases.

E.1 Call Reestablishment on Domain Transfer (CReDT)

CReDT provides reestablishment of voice service upon transition between CS and PS domain when enablement of the basic IMS Controlled Model based VCC is not possible. Voice service is reestablished in the new domain after releasing user's interface in the old domain. CCCF anchors the bearer path of the remote party during the execution of CReDT. The user is notified via appropriate MMI of the ongoing CreDT procedure; and comfort tone/announcement is provided to the remote party during CReDT.

E.1.1 Triggers for CReDT

2G to 3G:

- The trigger for "CS to PS transition notification" is under study. It may be based, e.g., on a downlink radio link failure criteria, or any similar process by which the terminal would make the decision to re-establish the 2G-CS service on a 3G-PS network.
- The time for service re-establishment is shortened if the source BSS provides the terminal with 3G cells in the neighboring cell list, so that the terminal can measure and monitor potential 3G target cells in advance, before losing 2G coverage.
- When in 2G-CS, the terminal needs to know in advance how voice service is supported in the target cell. This information needs to be provided along with the 3G neighboring cell list, so that:
    - If the target 3G cell supports voice over CS, the 2G to 3G voice mobility would be handled through existing handover (over A interface) and relocation (over Iu interface) procedures, under the control of the network.
    - If the target 3G cell only supports voice over PS, the voice service need to be re-established as described in the document, based on a trigger implemented in the terminal.

Editor's note: It is FFS on how to notify the terminal in advance on the voice service that is supported in the target cell.

3G to 2G:

- The trigger for "Notification (PS to CS transition indication)" is under study. It may be equivalent to the criteria being used for the 2G to 3G transition.
- As for the 3G to 2G case, the time for service re-establishment is shortened if the source RNS provides the terminal with 2G cells in the neighboring cell list, so that the terminal can measure and monitor potential 2G target cells in advance, before loosing 3G coverage.
- As for the 3G to 2G case, the terminal needs to know in advance how voice service is supported in the target 2G cell, as GERAN TSGs are currently studying the possibility to support Voice over PS in GPRS/EDGE cells.

E.1.2 CReDT solution options

CEDT releases the source radio prior to establishment of the target radio potentially resulting in significant procedure execution time, and potentially noticeable service interruption; two different options are presented with their benefits and liabilities to help select a solution that provides optimal procedure execution time without sacrificing user's quality of experience.

E.1.2.1 CReDT (option 1) – Uncontrolled Release of Source Radio by UE

This section presents a solution for CReDT that optimizes the procedure execution time, thereby minimizing the overall service interruption time. The UE releases the source radio as soon as it notifies CCCF of a need for CReDT. It then follows with establishing of the target radio. The source radio is released prior to application of announcement to the remote party, potentially resulting in a noticeable speech gap toward the remote party.

Note that USSD enabled CAMEL service is used as a mechanism for initiation of CS to PS CReDT in the example calls walk-throughs presented in this document. However, it should be possible to use other mechanisms for enablement of CS to PS CReDT at CCCF.

E.1.2.1.1 2G CS to 3G PS CReDT

Figure E.1-1 below provides a walk-through of a 2G CS call that has been anchored in user's IMS network using static or dynamic anchoring techniques and undergoes CS to PS CReDT with source release initiated by the UE.

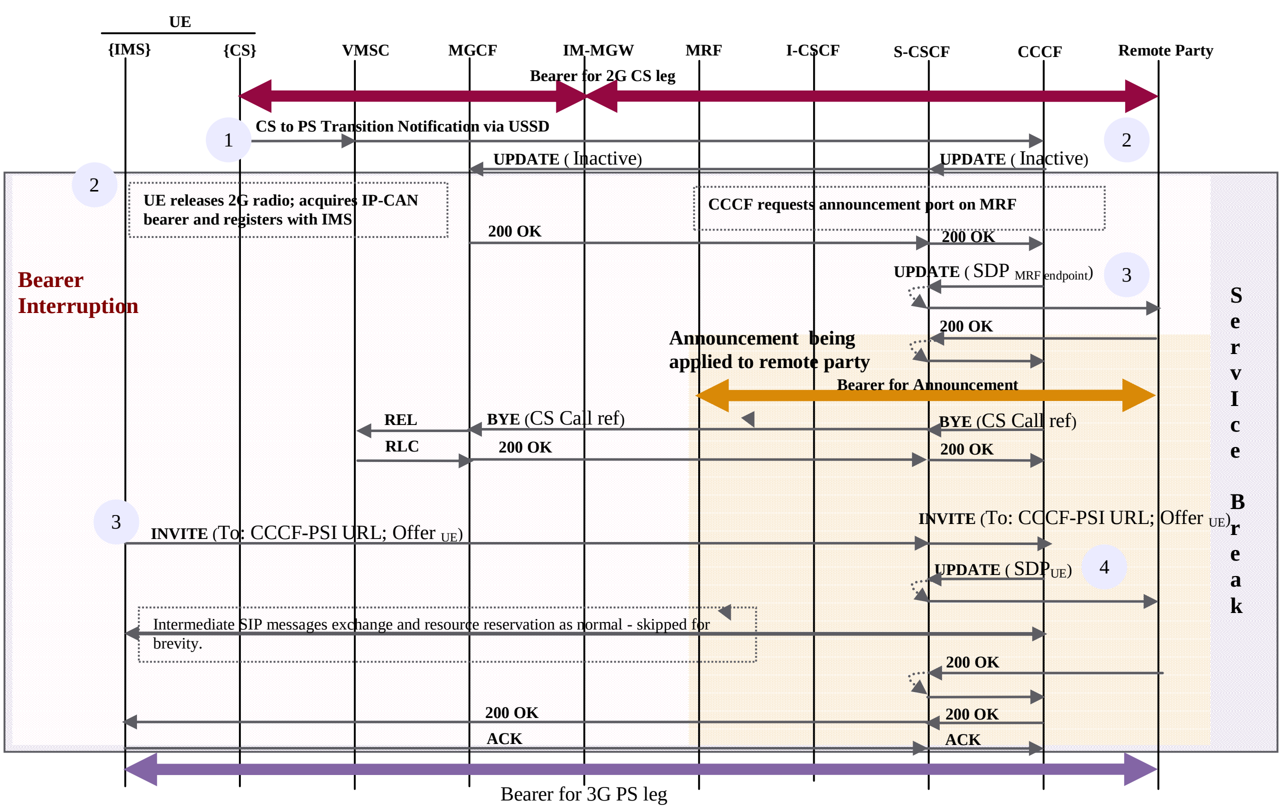


Figure E.1-1: 2G CS to 3G PS CReDT-UE initiated source release

1. The UE detects a need for CS to PS CReDT based on trigger criteria discussed in Section X.1.1 Triggers for CReDT, and notifies CCCF via a USSD initiated CAMEL service in user's CS home network. It also notifies the user of the CS to PS transition via proper MMI. A USSD Gateway may be used to communicate the CS to PS CReDT notification to CCCF in absence of CAMEL.

2. UE subsequently initiates release of source radio and acquires IP-CAN RAB in the target radio.

2. Upon receipt of CS to PS CReDT Notification, CCCF starts the procedure to acquire an announcement port on MRF for application of announcement toward the remote party. CCCF also sends a SIP message (for example, an UPDATE) to the MGCF hosting the user's CS bearer to report inactivity on the user's CS connection to prevent inactivity timer expiry resulting in premature release of the call at the IM-MGW.

3. CCCF updates the remote party with the SDP of the announcement port at the MRF to establish announcement toward the remote party.

3. The UE initiates CS to IMS transition request toward CCCF after acquiring IP-CAN bearer.

4. CCCF updates the remote party with the UE's IMS mode information, completing the transition from 2G CS to 3G IMS.

E.1.2.1.2 3G PS to 2G CS CReDT

Figure E.1-2 below provides a walk-through of an IMS session that undergoes PS to CS CReDT via CCCF with source radio release initiated by the UE.

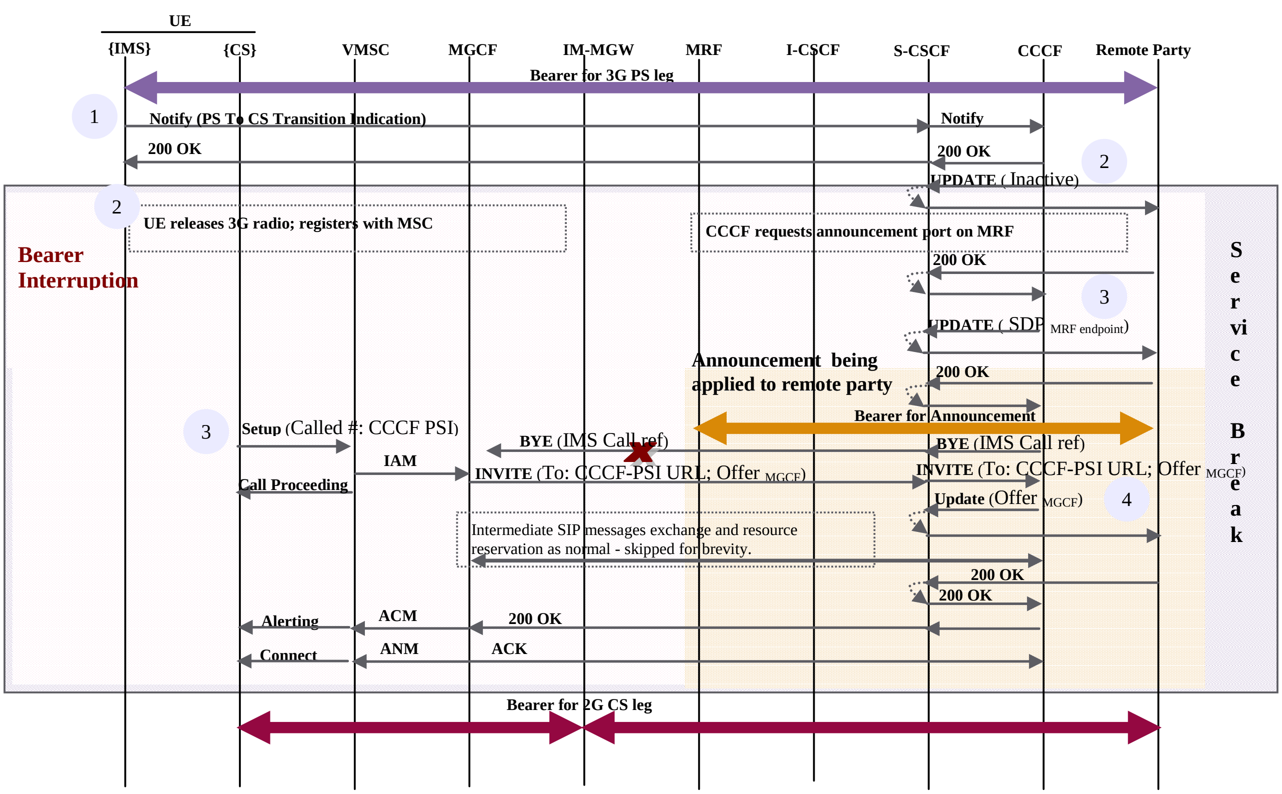


Figure E.1-2: 3G PS to 2G CS CReDT-UE initiated source radio release

1. The UE detects a need for PS to CS CReDT based on trigger criteria discussed in Section X.1.1 Triggers for CReDT, and notifies CCCF via a Notify with Mobility Event Package. It also notifies the user of the PS to CS transition via proper MMI.

2. UE subsequently initiates release of source radio and registers with the 2G MSC.

2. Upon receipt of a Notify indicating PS to CS CReDT Notification, CCCF starts the procedure to acquire an announcement port on MRF for application of announcement toward the remote party. CCCF also sends a SIP message (for example, an UPDATE) to the remote party to report inactivity on the user's connection to prevent inactivity timer expiry resulting in premature release of the call by the other end.

3. CCCF updates the remote party with the SDP of the announcement port at the MRF to establish announcement toward the remote party.

3. The UE initiates IMS to CS transition request toward CCCF after registering with the 2G MSC.

4. CCCF updates the remote party with the user's CS connection information, completing the transition from 3G IMS to 2G CS.

Benefits:

1. Shorter Reestablishment time as Registration in the target domain happens in parallel to the establishment of announcement toward the remote party.

Drawbacks:

2. Potential speech gap before the application of announcement to the remote party.

E.1.2.2 CReDT (option 2) – CCCF controlled Release of Source Radio by UE

This section presents a solution for CReDT that improves the remote party's quality of experience such that the remote party is not subject to a potential speech gap during the CReDT of the other end. The UE in this option does not initiate the release of the source radio until it receives an indication from CCCF that the remote party has been connected to a comfort tone/announcement. Since the establishment of the target radio is not started until later in the procedure, this option incurs significantly longer overall service interruption.

E.1.2.2.1 2G CS to 3G PS CReDT

Figure E.1-3 below provides a walk-through of a 2G CS call that has been anchored in user's IMS network using static or dynamic anchoring techniques and undergoes CS to PS CReDT via CCCF when source radio release is controlled by CCCF.

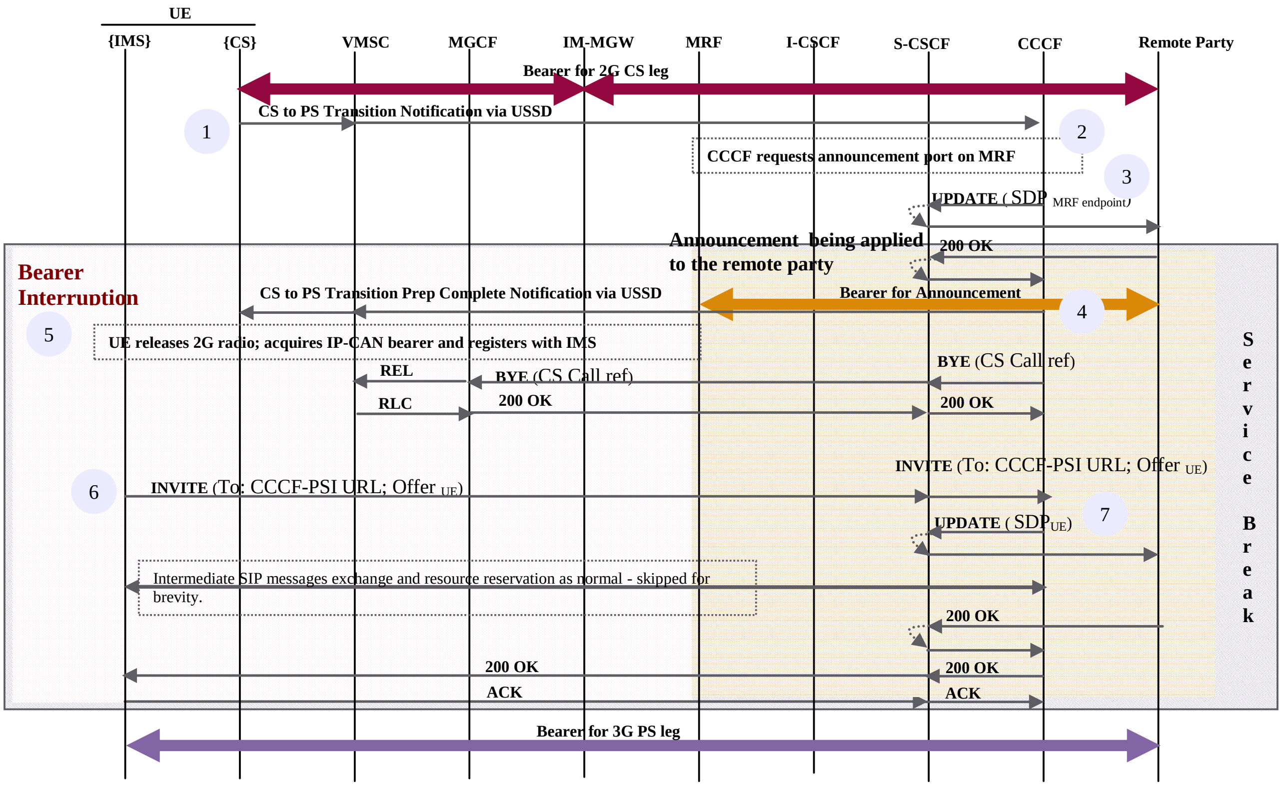


Figure E.1-3: 2G CS to 3G PS CReDT-CCCF Controlled source radio release

1. The UE detects a need for CS to PS CReDT based on trigger criteria discussed in Section X.1.1 Triggers for CReDT, and notifies CCCF via a USSD initiated CAMEL service in user's CS home network. It also notifies the user of the CS to PS transition via proper MMI.

2. Upon receipt of CS to PS CReDT Notification, CCCF acquires an announcement port on MRF for application of announcement toward the remote party.

3. CCCF updates the remote party with the SDP of the announcement port at the MRF to establish announcement toward the remote party.

4. CCCF sends a CS to PS Transition Prep Complete Notification to the UE via a CAMEL service initiated USSD message.

5. UE initiates release of source radio upon receipt of CS to PS Transition Prep Complete Notification from CCCF and acquires IP-CAN RAB in the target radio.

6. The UE initiates CS to IMS transition request toward CCCF after acquiring IP-CAN bearer.

7. CCCF updates the remote party with the UE's IMS mode information, completing the transition from 2G CS to 3G IMS.

E.1.2.2.2 3G PS to 2G CS CReDT

Figure E.1-4 below provides a walk-through of an IMS session that undergoes PS to CS CReDT via CCCF with source radio release controlled by CCCF.

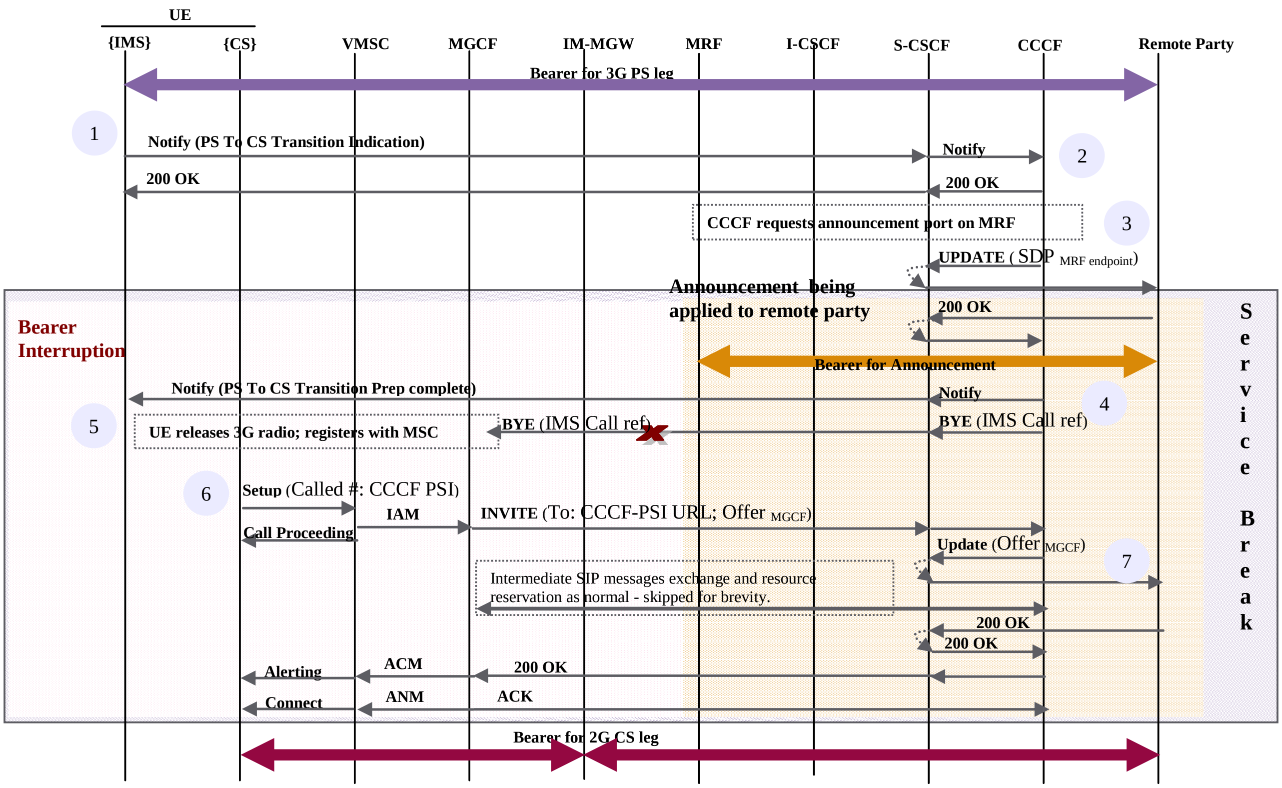


Figure E.1-4: 3G PS to 2G CS CReDT-CCCF Controlled source radio release

1. The UE detects a need for PS to CS CReDT based on trigger criteria discussed in SectionX.1.1 Triggers for CReDT, and notifies CCCF via a Notify with Mobility Event Package. It also notifies the user of the PS to CS transition via proper MMI.

2. Upon receipt of a Notify indicating PS to CS CReDT Notification, CCCF starts the procedure to acquire an announcement port on MRF for application of announcement toward the remote party. CCCF also sends a SIP message (potentially an UPDATE) to the remote party to report inactivity on the user's connection to prevent inactivity timer expiry resulting in premature release of the call by the other end.

3. CCCF updates the remote party with the SDP of the announcement port at the MRF to establish announcement toward the remote party.

4. CCCF sends a Notify to the UE to indicate PS to CS Transition Prep completion.

5. UE initiates release of source radio upon receipt of Notify with PS to CS Transition Prep completion and registers with the 2G MSC.

6. The UE initiates IMS to CS transition request toward CCCF after registering with the 2G MSC.

7. CCCF updates the remote party with the user's CS connection information, completing the transition from 3G IMS to 2G CS.

Benefits:

1. Better quality of experience for the remote party.

Drawbacks:

2. Longer Reestablishment time, hence longer overall service interruption time as Registration in the target domain happens after the establishment of announcement toward the remote party.

NOTE: Re-establishment time may be equal to the redial of a new call set up time.

Editor Note: Clarify the impact of supplementary services and regulatory services (e.g. emergency call)

Annex F: Service Continuity Model: IMS Controlled Alternative-Dynamic Anchoring Extensions

Use of static anchoring as described in sections 6.3.4 Origination and 6.3.5 Termination is recommended for enablement of VCC for the following reasons:

- It provides a uniform solution for anchoring CS and IMS calls/sessions at CCCF with support of supplementary service continuity along with basic CS-IMS voice call continuity.
- CCCF is in control of the bearer path since the initial call setup, therefore, Voice Call Continuity is guaranteed any time after the call/session is established. It is not susceptible to race conditions or loss of radio/WLAN coverage that may affect Voice Call Continuity when dynamic anchoring techniques are applied.
- It provides optimal execution time for completion of Voice Call Continuity procedure as additional signalling procedures are not required to establish the anchor prior to execution of VCC procedures.

It may not always be possible to employ static anchoring techniques due to network conditions such as non-supporting CS roaming partners. Dynamic anchoring techniques as discussed below may be employed in such situations.

NOTE: The solutions presented in this section are independent of each other. Also note that these solution options have not been evaluated so as to assess their feasibility as an implementation option for VCC.

F.1 ECT enabled Dynamic Anchoring

Although ECT has significant drawbacks when used as the only solution for voice call continuity from CS domain to IM Subsystem, it may be used as a means for dynamically anchoring CS calls at CCCF when techniques for static anchoring are not possible. The use of ECT is limited to setting up an anchor reference for a CS call in IMS upon first CS to IMS VCC transition. CCCF is used for all subsequent VCC transitions from CS to IMS and from IMS to CS.

F.1.1 Supplementary Service control for ECT enablement

Certain supplementary services like Call Forward Unconditional and Incoming Call Barring can prevent ECT enabled VCC procedures due to interactions of these services with ECT. Call Independent Supplementary Services operations can be used by the UE to disable such services prior to initiating the VCC procedure and re-enable these services just before the CS radio is released as part of the overall VCC procedure.

F.1.2 CS to IMS Voice Call Continuity using ECT

ECT can be used to enable first CS to IMS VCC transition and establish the anchor at CCCF so that subsequent CS to IMS and IMS to CS VCC transitions are executed via CCCF. Figure F.1-1 below provides a walkthrough of first CS to IMS VCC transition using ECT.

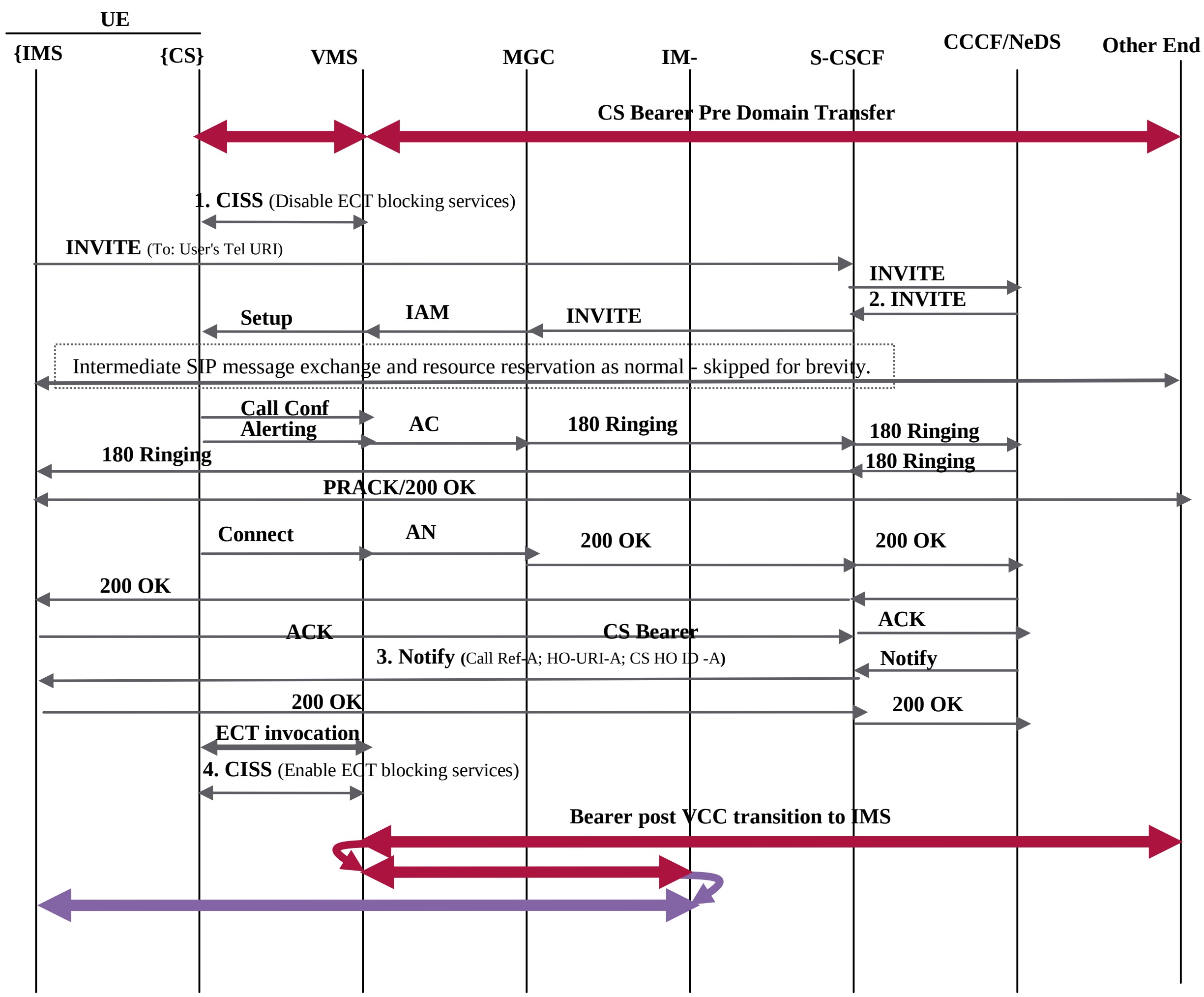


Figure F.1-1: CCCF anchoring via ECT

1. UE registers with IMS if required upon detection of VCC triggers. If a CCCF call anchor reference is available, the UE executes CS to IMS VCC transition via CCCF as described in Section 6.3.6.2.1, Procedures for CS to IMS Voice Call Continuity. If a CCCF call anchor reference is not available at the UE, then it enables ECT as a mechanism to execute VCC transition to IMS.The UE disables any supplementary services like Call Forward Unconditional that could potentially block ECT and initiates a call to its CS mode via IMS.

2. CCCF inserts a routing B2BUA function upon completion of the call toward the CS Domain as a result of filter criteria execution at S-CSCF.

3. Upon successful execution of a Routing B2BUA function for IMS session to the CS domain, CCCF assigns a unique call reference identifier to the session for identification between the UE and CCCF in subsequent dialogue. It also assigns a unique identifier which can be used for VCC transition of this session to CS Domain. These new identifiers are communicated to the UE via a Notify with Mobility Event package.

4. The UE re-enables the supplementary services disabled previously upon successfully receiving the incoming CS call.

F.1.3 Subsequent Executions of VCC Procedure

Use of ECT for subsequent VCC transitions results in a daisy chain effect s shown in Figure F.1-2 below, due to the fact that the call anchor moves between CS and IMS upon each VCC transition.

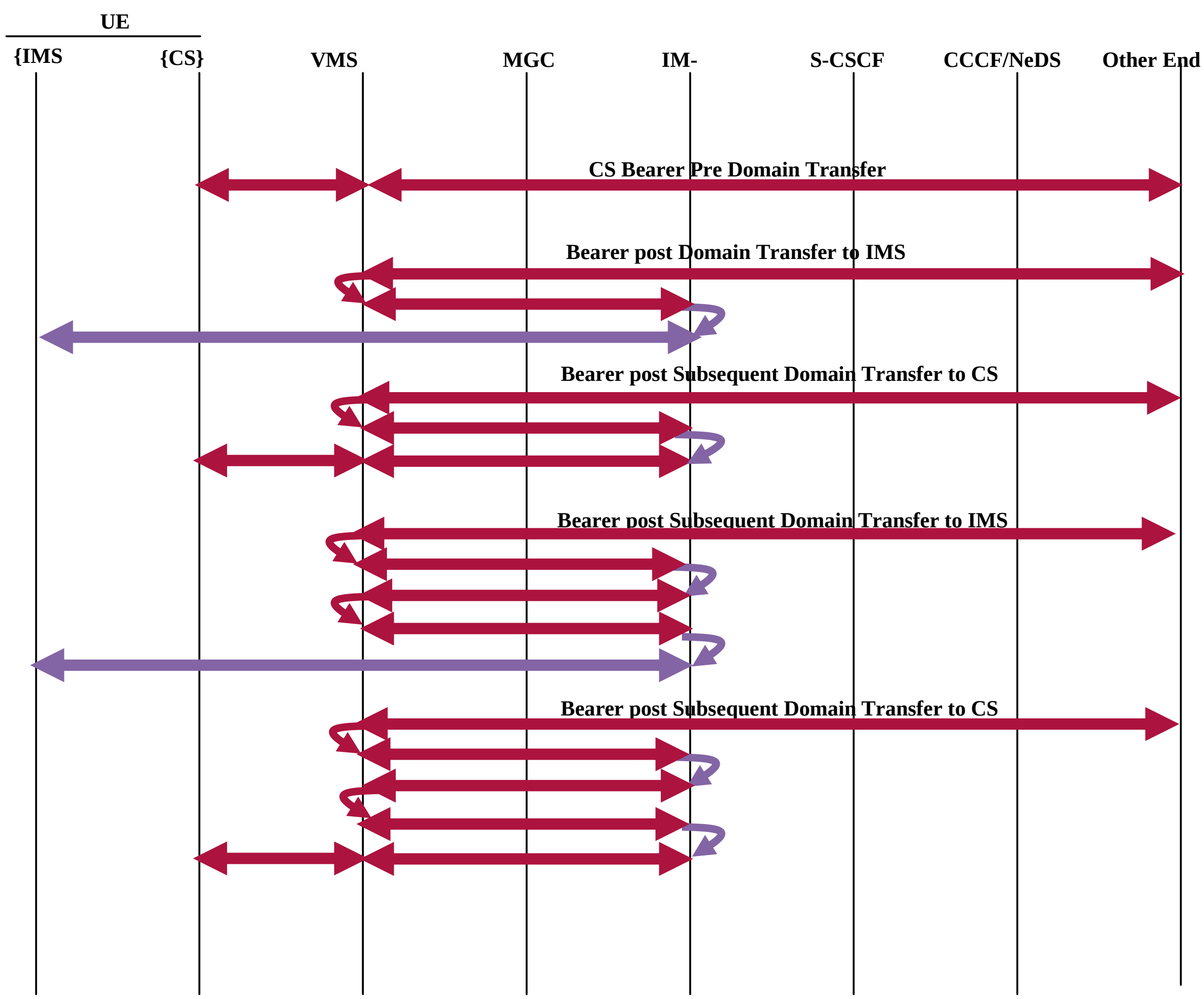


Figure F.1-2: Undesirable Resource Daisy Chain with use of ECT for subsequent executions of VCC procedure

Once a session anchor has been established at CCCF and the session identifiers have been communicated to the UE, the UE executes all VCC transitions via CCCF. This helps eliminate the daisy chain effect as shown in Figure F.1-3 below, because the call remains anchored in the IMS MGW after the first CS to IMS VCC transition.

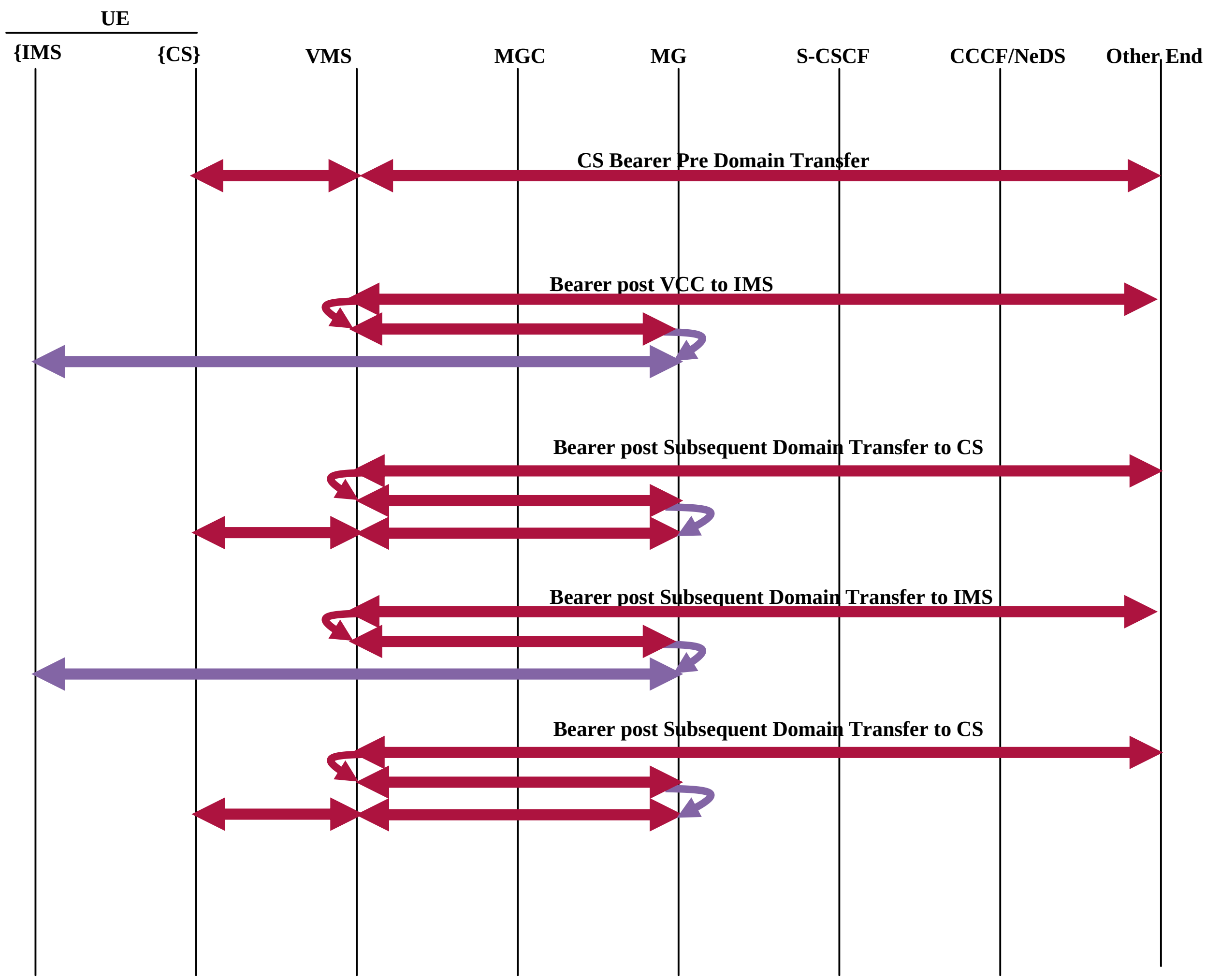


Figure F.1-3: Resource optimization with use of CCCF for subsequent executions of VCC procedure

ECT enabled dynamic anchoring technique cannot be used when the user is engaged in more than one CS sessions or is engaged in an Emergency Call.

F.1.4 Evaluation of Dynamic CS Anchoring for VCC using ECT

The benefits of this proposal include:

- This approach enables dynamic anchoring with CCCF that alleviates concerns around resource usage with static anchoring.

The drawbacks of this proposal include:

- ECT cannot be used to enable call continuity from CS to IMS if multi-session services like CW are active at the time of call continuity.

- ECT cannot be used to dynamically anchor CS Emergency calls with CCCF as ECT Supplementary Service is not applicable for Emergency calls.

F.2 DACCI enabled Dynamic Anchoring

Dynamic Anchoring of CS Calls in IMS, DACCI is a new Call Control service which can be requested by the UE via USSD, Facility Invocation or any other enabler as determined by a further study on DACCI enablers. DACCI transfers the Call Control Protocol State Machine of a user's CS-IMS call from CS domain to the IM Subsystem upon UE's request.

Enablement of first CS to IMS VCC transition for a CS call using DACCI provides dynamic means of anchoring CS calls in IMS. DACCI anchors CS bearer in an IM-MGW upon first CS to IMS VCC transition in order to enable subsequent VCC transitions between CS and IMS via CCCF.

DACCI uses the Mobility Event package to communicate with CCCF, session information required to perform VCC transitions from CS to IMS.

DACCI is applicable to all voice teleservices including the Emergency Call Service.

F.2.1 First CS to IMS Execution of VCC Procedure using DACCI

Figure F.2-1 below provides a walk through of a first CS to IMS VCC transition enabled by DACCI.

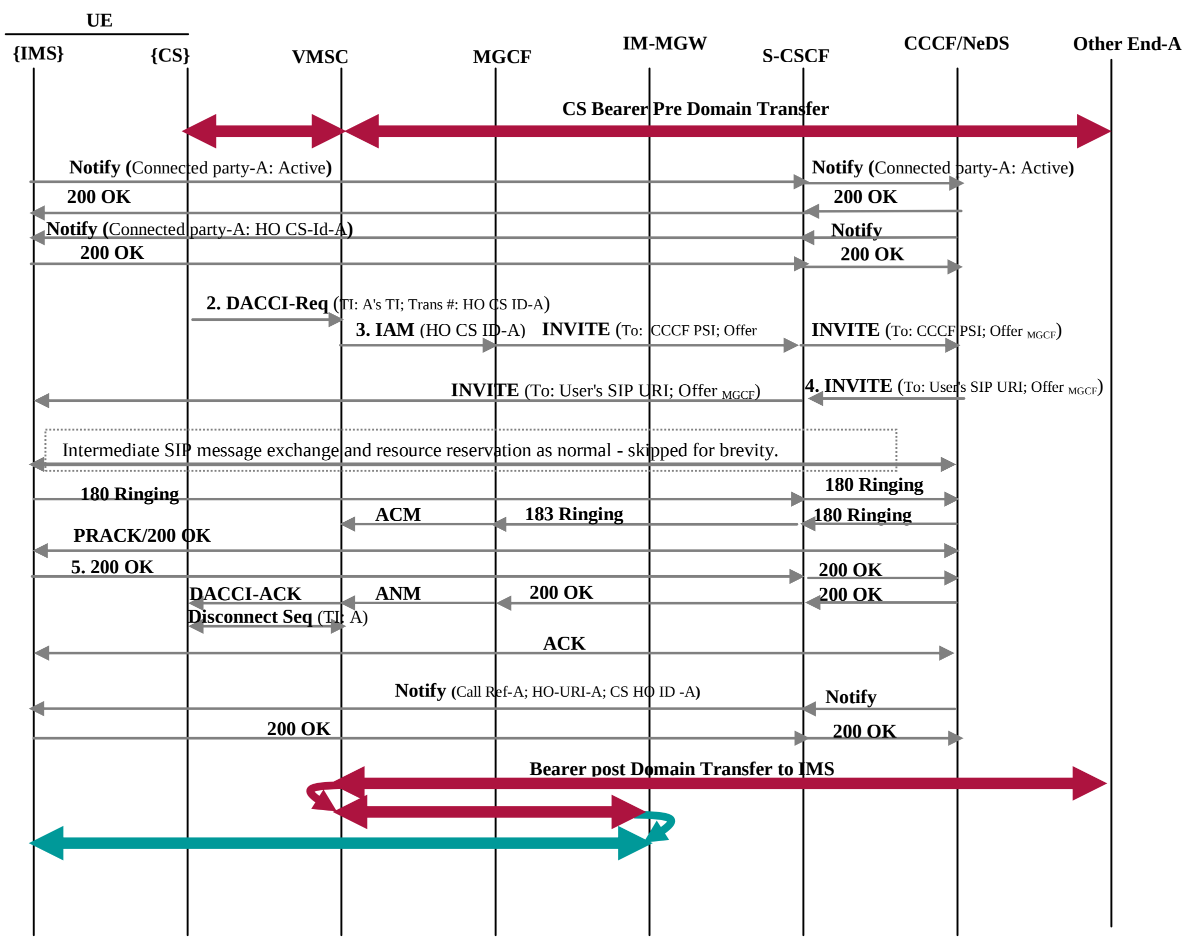


Figure F.2-1: First CS to IMS VCC transition using DACCI

1. The UE registers with IMS as it detects border conditions requiring VCC transition to IMS. If a CCCF call anchor reference is available, the UE executes CS to IMS VCC transition via CCCF as described in a Section 6.3.6.2.1

Procedures for CS to IMS Voice Call Continuity If a CCCF call anchor reference is not available at the UE, then it uses DACCI to execute VCC transition to IMS.

The UE communicates connected party information with intent to execute DACCI enabled CS to IMS VCC transition to CCCF in a Notify. The connected party information is communicated to CCCF for use in subsequent VCC transition from CS to IMS for the case when HO URI cannot get communicated to the UE due to lack of WLAN coverage at the time of establishing of CS leg upon subsequent VCC transition to CS. CCCF assigns a unique CS Identifier to be used for the first CS to IMS VCC transition using DACCI and communicates it to the UE in a Notify.

2. UE invokes DACCI at the CS Core Network, providing a routing number derived from the CS HO Identifier to set up the VCC leg towards IMS. It also provides the CC Transaction ID of the session to be handed over.

3. The MSC uses the routing number to establish an ISUP circuit connection toward CCCF in IM Subsystem using procedures similar to the establishment of circuit connection toward target MSC for GSM/UMTS CS Domain Inter MSC Handovers. One way bearer path is established from the other end point for the extended bearer leg, resulting in a downlink bi-cast of the other endpoint's bearer towards CS-IMS user's CS and IMS legs.

4. CCCF establishes a Routing B2BUA function before extending the IMS session toward UE for the new IMS session and communicates the session information to the UE to be used for subsequent VCC transition to CS, upon successful setup of the IMS session.

5. Upon sending 200 OK in response to the INVITE extended from CCCF, the UE switches its bearer plane to IMS. An ISUP Answer message subsequently reports successful establishment of circuit connection to the MSC, thereby resulting in switch of the uplink bearer path for the user and release of the CS radio link at the MSC.

   The MSC remains in the bearer path, but the call control is moved to CCCF and the bearer is anchored at IM-MGW.

F.2.2 Subsequent VCC Transitions with DACCI enabled anchoring

Once a session anchor has been established at CCCF and the session identifiers have been communicated to the UE, the UE executes all VCC transitions via CCCF using VCC procedures described in Section 6.3.6.2.1 Procedures for CS to IMS Voice Call Continuity; 6.3.6.2.2, Subsequent VCC Transitions to CS and 6.3.6.4, Procedures for IMS to CS Voice Call Continuity). Subsequent VCC transitions result in establishment of new leg toward the UE followed by release of the old leg. The bearer for the other end remains anchored at the IM-MGW used to establish bearer for the first CS to IMS VCC transition as shown in Figure F.2-2 below.

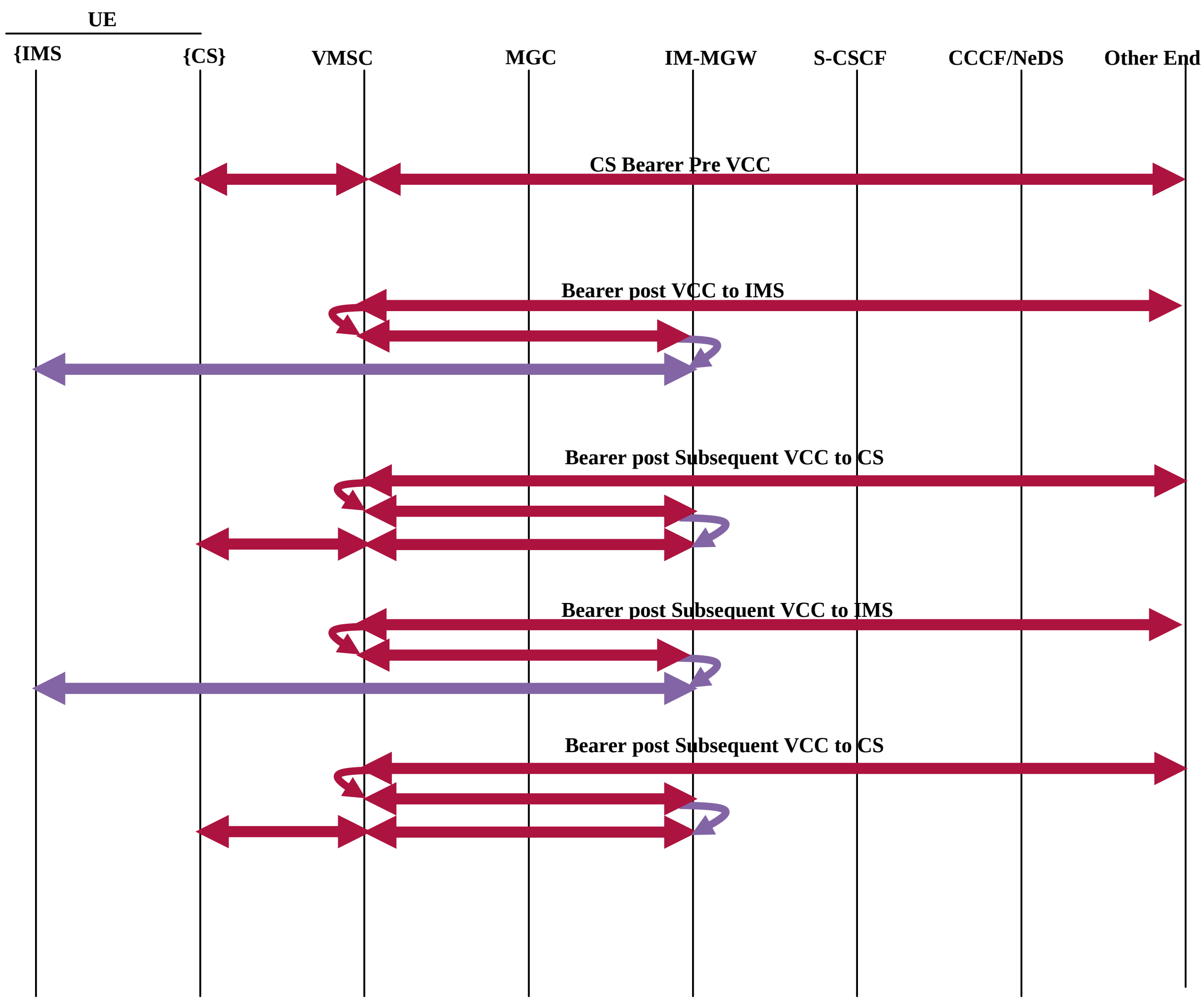


Figure F.2-2: Use of CCCF for subsequent VCC transitions

F.2.3 Multi-Session services support with DACCI

DACCI uses techniques similar to the techniques described in the previous section to transfer a CS call with multiple sessions from CS domain to IMS.

Figure F.2.-3 below provides a walkthrough of a CS to IMS transfer of a CS call to party A in the Held state and CS call to party B in the Active state.

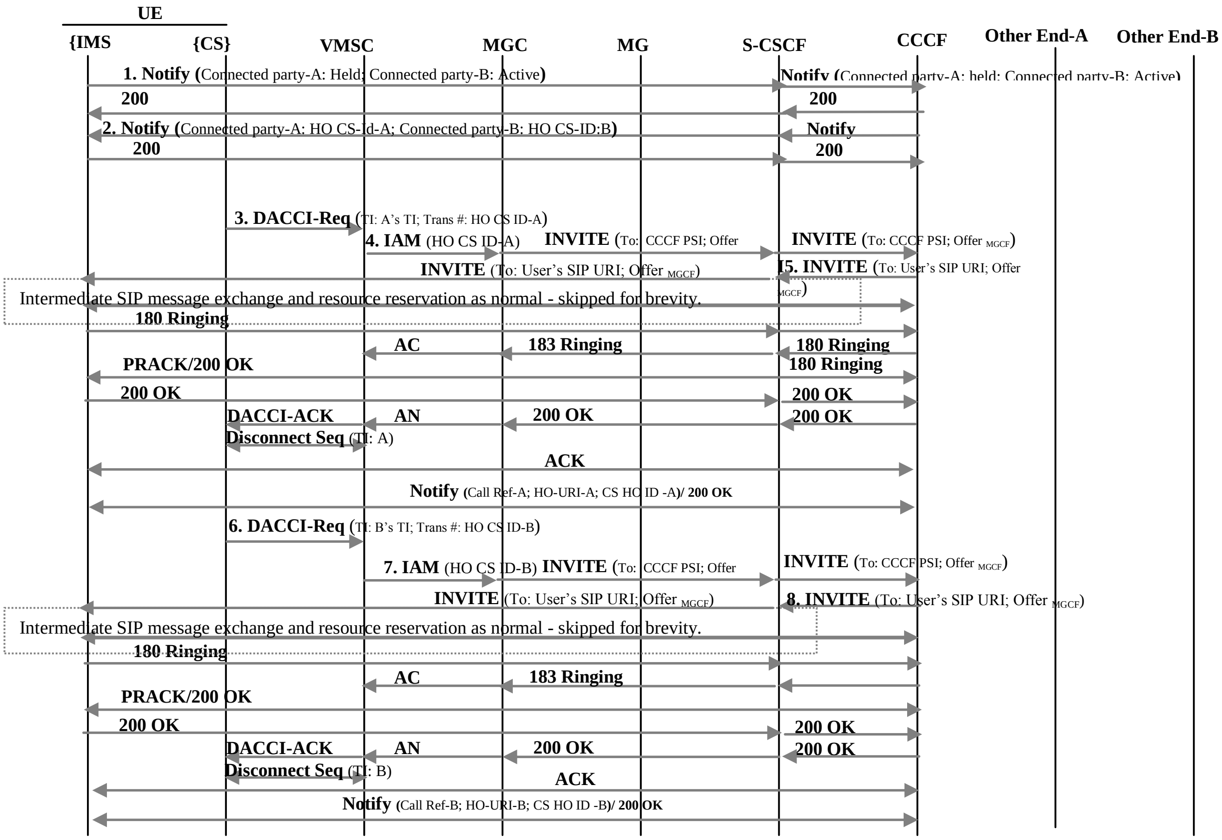


Figure F.2-3: VCC of Held/Active CS Sessions via DACCI

1. The UE detects a need for DACCI enabled VCC, registers with IMS, and provides session state information to CCCF.
2. CCCF allocates CS HO Identifiers for A's and B's session and communicates to the UE.
3. UE executes VCC of A's session to IMS using DACCI. It uses the CC Transaction Identifier to identify the session that needs to be handed over to the MSC.
4. The MSC allocates a new circuit connection for A's session, extending it's bearer toward CCCF as described in previous section.
5. CCCF invokes a routing B2BUA function for A's session communicating session specific information to the UE upon successful set up of the IMS leg for this session. CCCF establishes A's session in the Held state as suggested by session state information received from the UE.
6. Upon completion of VCC of A's session, UE executes VCC of B's session.
7. The MSC allocates a new circuit connection for B's session, extending it's bearer toward CCCF as described in previous section.
8. CCCF invokes a routing B2BUA function for B's session, communicating session specific information to the UE upon successful set up of the IMS leg for this session.

F.2.4 Evaluation of Dynamic CS Anchoring for VCC using DACCI

The benefits of this proposal include:

- This approach enables dynamic anchoring with CCCF that alleviates concerns around resource usage with static anchoring.
- The approach provides dynamic anchoring without any interactions with CS Supplementary Services that ECT based dynamic anchoring is subject to.

The drawbacks of this proposal include:

- Impacts CS domain nodes (requires an MSC software upgrade).
- Connected party address availability is required at the UE.

F.3 Dynamic Anchoring using CAMEL 4

In this alternative:

- The Call Continuity Control Function (CCCF) acts as a SIP Application Server in the IMS domain interfacing to the S-CSCF via ISC interface.
- The CCCF also acts as a gsmSCF interfacing to HPLMN through MAP/CAP interface.
- VCC subscriber calls in CS mode are CAMEL triggered to the CCCF for routing decision. The CCCF provides routing direction for both incoming and outgoing calls to and from the CS mode of the UE/MS.
- The MS/UE signals the CCCF when it detects IMS coverage. The MS/UE monitors the received IMS signal strength and necessary QoS parameters. If it has not already done so, once the MS/UE determines that there is sufficient QoS to provide voice service the MS/UE signals the CCCF, e.g., using a NOTIFY.
- Anchoring:
  - When the MS/UE is operating in CS mode and the other connected party is also in CS domain:
    - The call is anchored in the VMSC.
    - On the first instance of roaming from CS domain to IMS, an additional anchor is introduced at the MGW.
    - Once the call is anchored in the MGW the call remains anchored in the MGW for the remainder of the call.

When the MS/UE is operating in IMS mode and the other party is in the CS domain the call remain anchored in the MGW when MS/UE roams across CS and IMS domains.

F.3.1 Origination

CS mode originated:

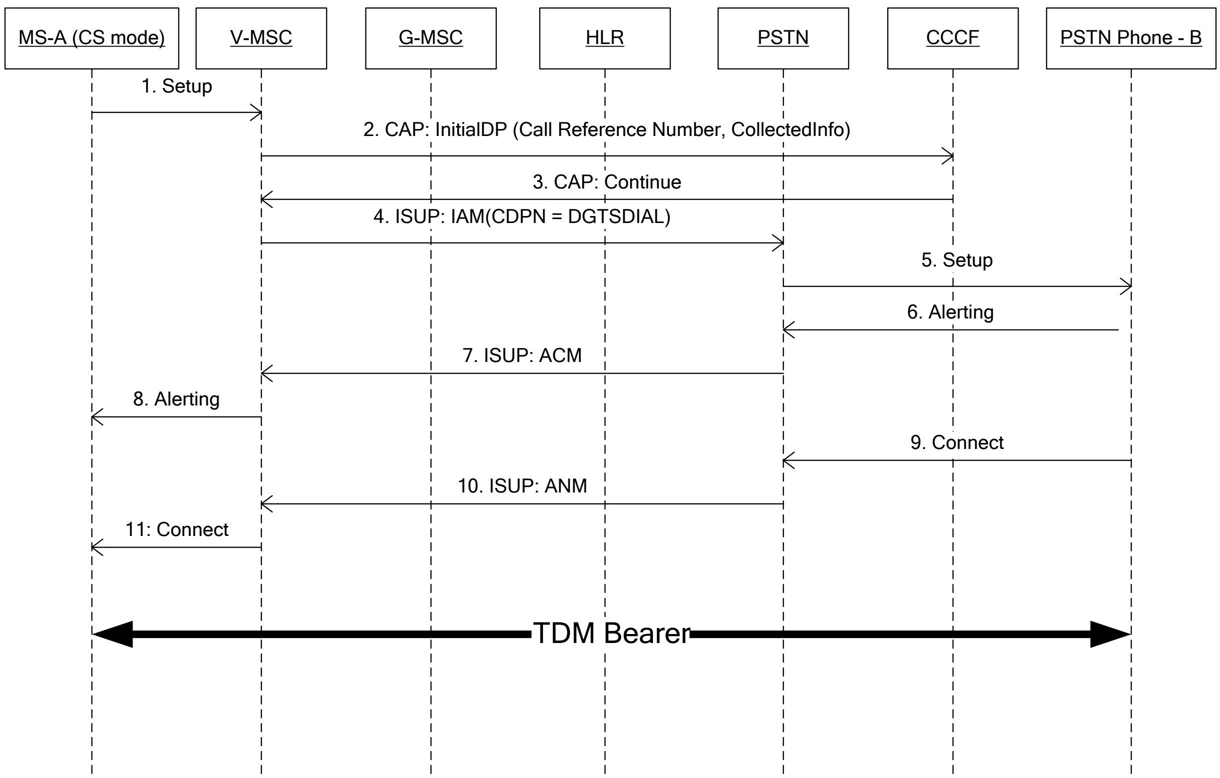


Figure F.3-1: MS/UE in CS mode -to- PSTN termination

Walkthrough:

1. Dual mode MS/UE sends a call setup request to the VMSC
2. VMSC detects the CollectedInfo trigger which has been armed for the MS/UE with the CCCF point code; VMSC queries CCCF for further routing decision
3. CCCF sends a CAMEL Continue response to the VMSC; CCCF saves CallReferenceNumber for call continuity processing
4. VMSC routes to the PSTN network
5. PSTN connects the call to the called party
6. PSTN receives an Alerting indication from the called party
7. PSTN sends an ISUP ACM message back to the VMSC
8. VMSC sends an Alerting indication to the dual mode MS/UE
9. The called party answers the call and sends a Connect message to the PSTN
10. PSTN sends an ISUP ANM message to the VMSC

11. VMSC sends an Connect indication to the dual mode MS/UE

F.3.2 Termination

CS mode termination:

Precondition: MS/UE registered in GSM HLR and the terminating CAMEL event armed in the VMSC.

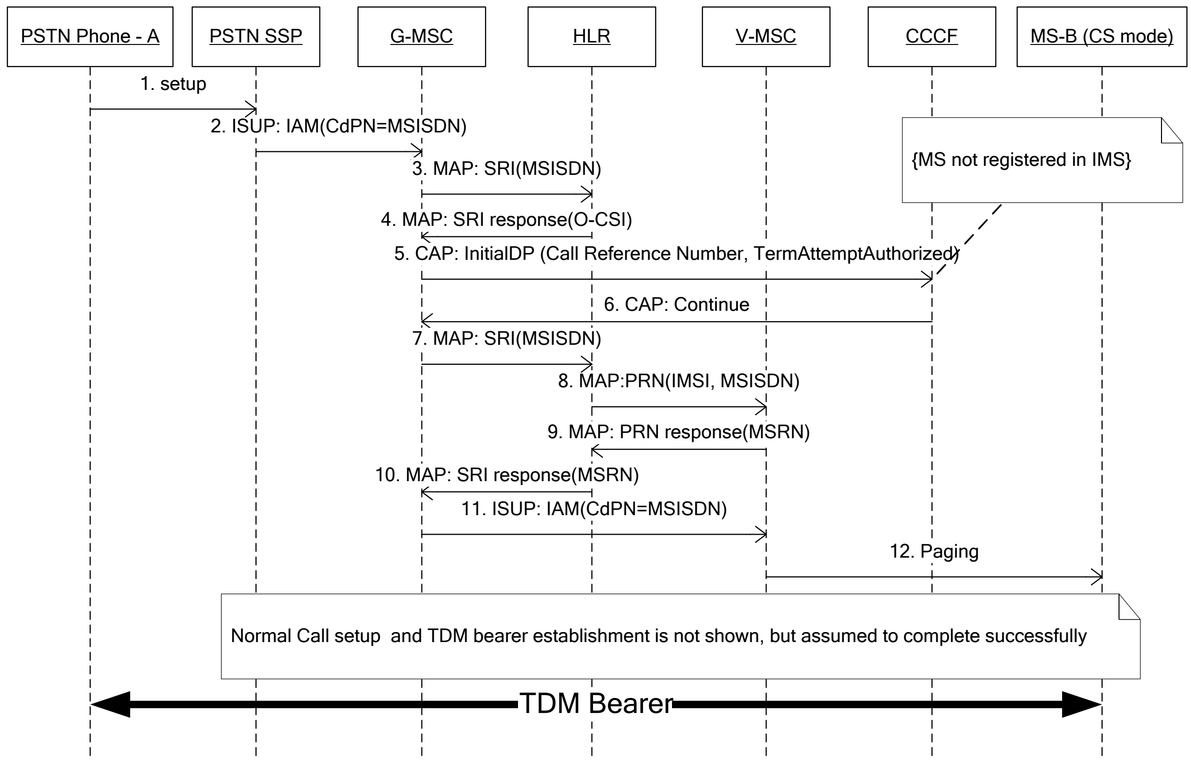


Figure F.3-2: PSTN -to- MS/UE in CS mode termination

Walkthrough:

1. PSTN (or PLMN) caller initiates a request to the MSISDN of the dual mode MS/UE
2. IAM sent to the gateway MSC in the HPLMN of the called dual mode MS/UE
3. Gateway MSC queries HLR for routing information
4. HLR response indicates further interrogation of CCCF
5. VMSC detects the TerminatingAttemptAuthorized trigger which has been armed for the MS/UE with the CCCF point code; VMSC queries CCCF for further routing decision
6. CCCF verifies whether the MS/UE is registered in IMS, either having a list of access network location for the subscribers list constructed from third party registration or Sh-Pull from the HSS. If CCCF determines that the MS/UE is not able to receive a call over IMS then it instructs the VMSC to continue termination to the CS mode.
7. Gateway MSC queries HLR for routing information (i.e. MSRN)

8. HLR queries the VLR (generally coexists with the VMSC) where the MS/UE is attached

9. VLR allocates a MSRN and responds to the HLR

10. HLR returns the MSRN to the gateway MSC

11. GMSC sends IAM towards the VMSC where the MS/UE is attached

12. Through the radio access network VMSC completes the call in the CS mode.

IMS mode termination:

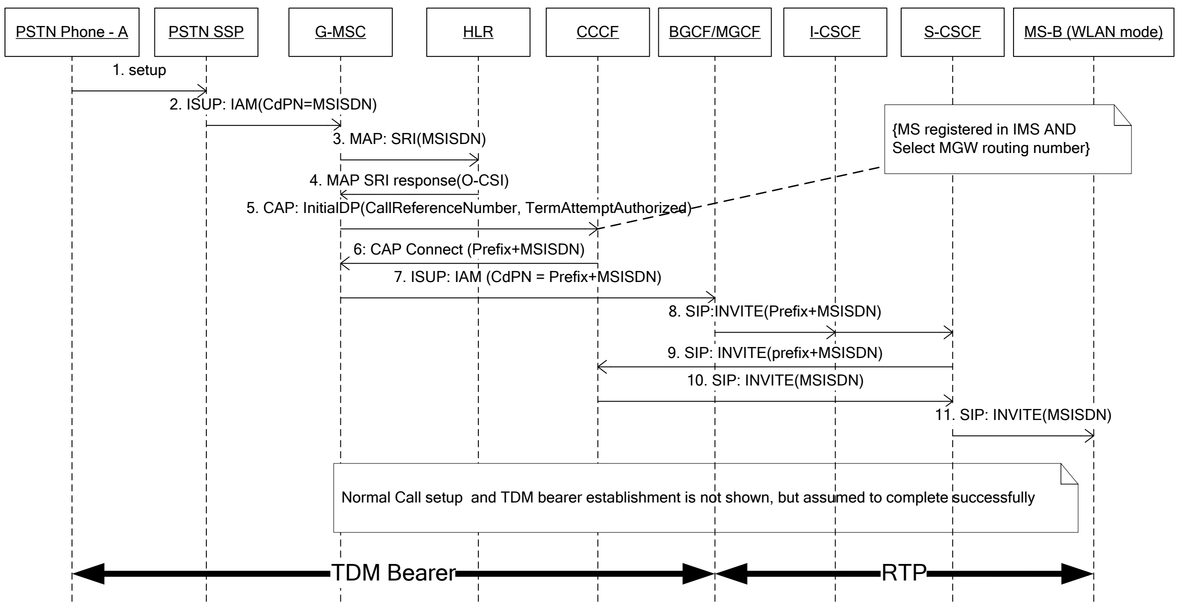


Figure F.3-3: PSTN -to- MS/UE in IMS mode termination

Walkthrough:

1. PSTN (or PLMN) caller initiates a request to the MSISDN of the dual mode MS/UE

2. IAM sent to the gateway MSC in the HPLMN of the called dual mode MS/UE

3. Gateway MSC queries HLR for routing information

4. HLR response indicates further interrogation of CCCF

5. VMSC detects the TerminatingAttemptAuthorized trigger which has been armed for the MS/UE with the CCCF point code; VMSC queries CCCF for further routing decision

6. CCCF verifies whether the MS/UE is registered in IMS, either having a list of access network location for the subscribers constructed from third party registration or Sh-Pull from the HSS. The CCCF determines that the MS/UE is registered in IMS and able to receive an IMS voice call and instructs the VMSC to route the call to IMS MGW by prefixing a routing code. This method requires no additional capability in the MSC

7. Gateway MSC routes the call to a MGW that is part of the home IMS

8. The MGCF in the home MSC sends INVITE to the I-CSCF; iFC is based on MSISDN, extracted after the prefix length. I-CSCF selects registered S-CSCF for the MSISDN and forwards the INVITE.

NOTE 1: IMS elements other than CCCF do not need to know the prefix, only the length of the prefix is required for CSCF to route based on MSISDN.

9. S-CSCF process the request as mobile terminating request, detects iFC based on MSISDN and forwards the INVITE to CCCF

10. CCCF correlates the INVITE with service context created previously during GSM interaction based on <prefix, TO and FROM headers>, strips the prefix and route the INVITE towards the S-CSCF

11. S-CSCF forwards the INVITE to the MS/UE in IMS mode using normal IMS procedures.

F.3.3 Domain Transfer Scenarios

MS/UE call continuity from GSM CS domain to IMS:

Precondition: Dual mode terminating subscriber is connected in GSM CS mode to the PSTN caller. CCCF is aware of the Call Reference of the CS terminated call.

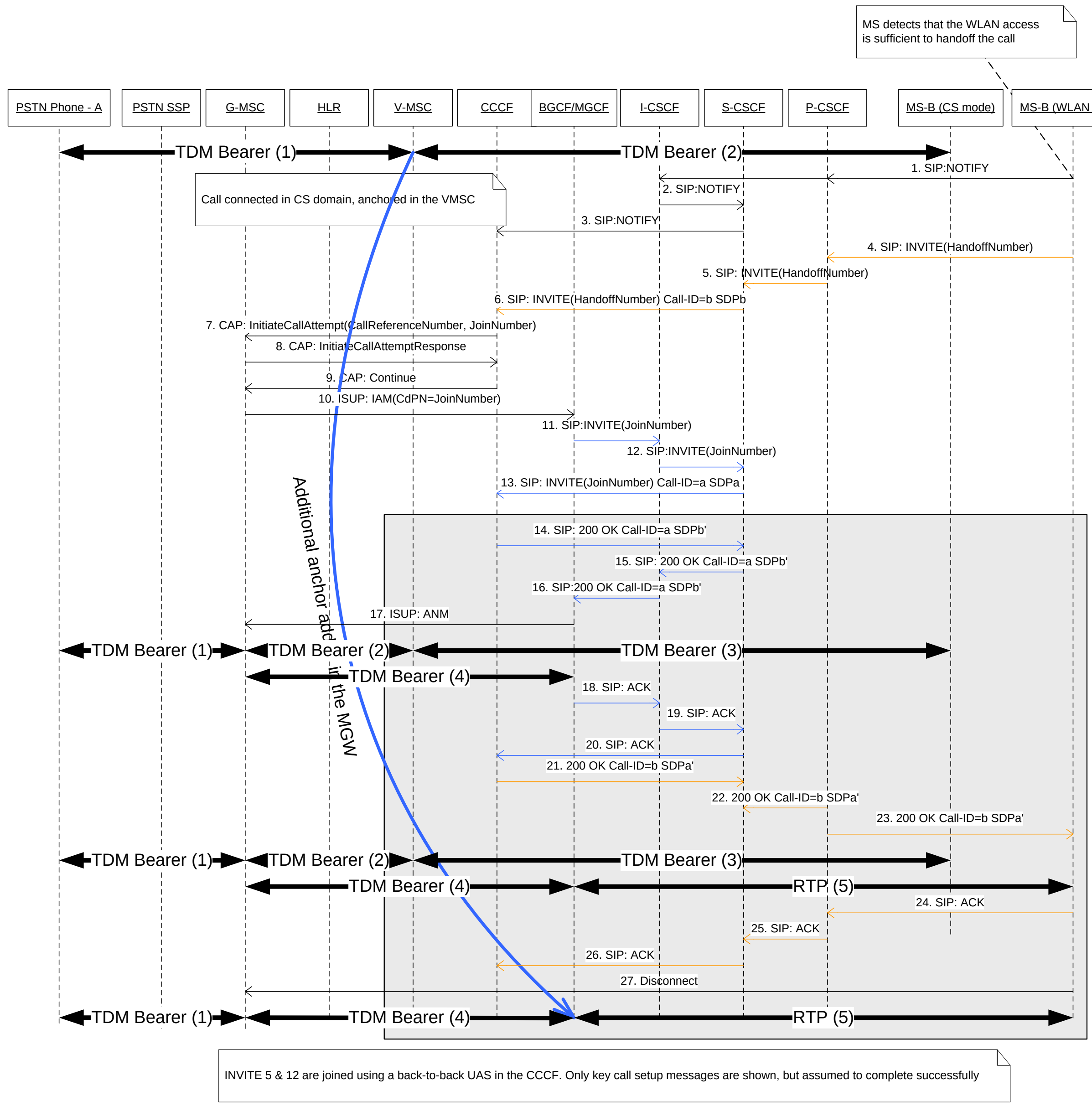


Figure F.3-4: Call Continuity from CS mode to IMS mode

Walkthrough:

1. The MS/UE is connected in the CS mode. MS/UE detects IMS coverage and decides to domain transfer to IMS and continue the same conversation. If not already, the MS/UE registers in IMS mode. Dual mode MS/UE sends a NOTIFY to the CCCF
2. P-CSCF forwards the NOTIFY towards the S-CSCF selected previously during REGISTER processing

3. S-CSCF forwards the NOTIFY towards the CCCF. CCCF updates access network availability of the dual mode MS/UE

4. MS/UE in IMS mode initiates a call to the predefined routing number called "Domain transfer number" towards P-CSCF.

5. P-CSCF forwards to the request towards S-CSCF.

6. S-CSCF based on the iFC forwards the INVITE to the CCCF.

7. CCCF sends a CAMEL InitiateCallAttempt request, which contains the corresponding call reference number, to the Gateway MSC where the dual mode MS/UE is attached.

8. Gateway MSC acknowledges the Initiate Call Attempt request and wait for the next set of instructions from the CCCF.

9. CCCF sends a Continue request to the Gateway MSC

10. Gateway MSC establishes an additional call leg to the existing call session by sending an ISUP IAM message towards the MGC.

11. MGCF translates the IAM into a SIP INVITE and send it to the I-CSCF

12. I-CSCF selects a S-CSCF for this subscriber based on the original dialled number, and forwards the INVTE to the selected S-CSCF

13. S-CSCF using iFC determines that CCCF is providing the voice call continuity service. S-CSCF forwards the INVITE to the CCCF for termination call handling

14. CCCF using its internal call leg correlation methods determines the INVITE in step 13 and step 6 need to be joined. The join process can be achieved in different ways, e.g. answering the first leg based on the offer SDP from the second request, and based on the answer from the first leg generate the answer SDP for the second leg. CCCF interrogates the SDP offer-answer protocol to synchronize the SDP and avoid infinite iterations. (steps 14-26)

27. When the MS/UE detects that the IMS call is connected, it releases the CS mode call.

MS/UE call continuity to CS from IMS:

Precondition: The terminating dual mode subscriber is connected to the CS caller.

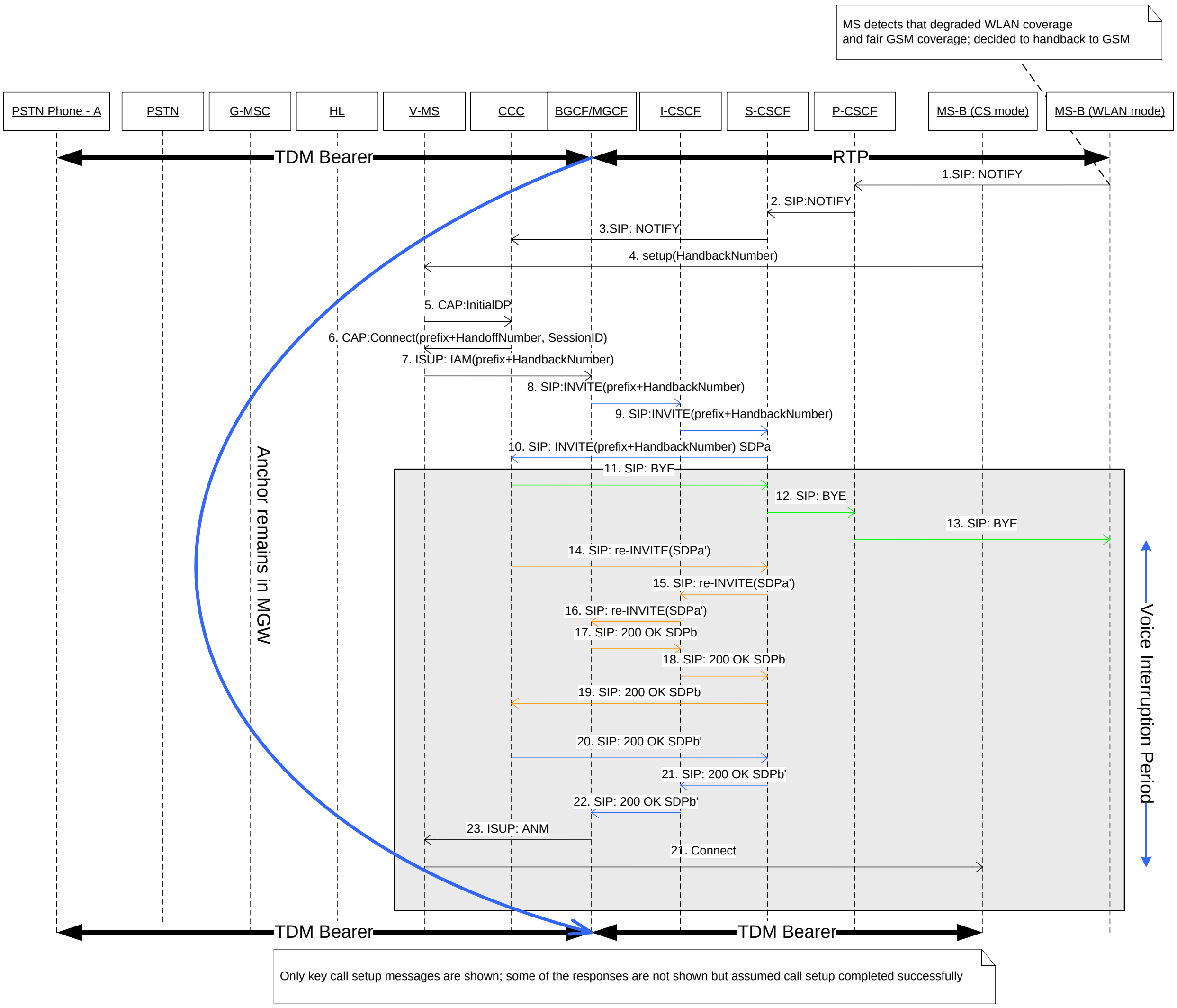


Figure F.3-5: Call continuity from IMS to CS mode

Walkthrough:

1. Dual mode MS/UE detects that IMS signal degradation, and will send a NOTIFY to the CCCF
2. P-CSCF forwards the NOTIFY towards the S-CSCF selected previously during REGISTER processing
3. S-CSCF forwards the NOTIFY towards the CCCF. CCCF updates access network availability of the dual mode MS/UE
4. MS/UE initiates a call in CS mode to a preconfigured domain transfer number
5. The VMSC detects a terminating CAMEL trigger and sends event notification to the CCCF
6. CCCF correlates the event notification to the existing call session to the MS/UE, provides a routing number that is prefixed the same way as in IMS termination
7. VMSC routes the call towards the IMS, IAM received at the MGCF

8. MGCF translates the IAM to INVITE and forwards to the I-CSCF

9. I-CSCF selects the S-CSCF based on mobile originated iFC and forwards the INVITE to the S-CSCF. The selected S-CSCF may be different from the one handling the IMS terminated call to the MS/UE.

10. S-CSCF process the INVITE as per mobile originated case and forwards the INVITE to the CCCF based on iFC

11. CCCF correlates the INVITE with an existing call session that is IMS terminated through the CCCF back-to-back UA. Releases the IMS leg of the call by sending BYE towards the S-CSCF

12. S-CSCF forwards the BYE towards the P-CSCF

13. P-CSCF forwards the BYE to the MS/UE. 200 OK propagation is not shown to conserve space. It is assumed processing of BYE as per RFC 3261.

14. CCCF refresh the caller leg by sending a re-INVITE towards the MGCF with the SDP received in step1. This step and rest of the call completion messages causes the TDM bearer from caller and MS/UE in CS mode hair pinned through the MGW.

F.3.4 Handoff URI

The UE uses a Handoff URI in the request URI of the SIP INVITE message to signify the request of a domain transfer of an established call connection from CS to IMS. The Handoff URI, for the CS-to-IMS domain transfer request, is independent of the UE and the MSISDN of the service subscriber. The Handoff URI could also be a sip: or tel: URI, and the format and allocation is operator dependent. To support the VCC procedure, the selected Handoff R-URI must be specified in the CCCF/NeDS as CS-to-IMS Handoff Request identifiers.

There are two potential methods for communicating the Handoff URI to the UE. These methods are static provisioning during initial service activation and dynamic provisioning during the registration procedure with CCCF and/or via the NOTIFY method provided by the Mobility Event Package upon first detection of the IMS connectivity. For the dynamic provisioning method, the Handoff URI provisioning procedure should take place regardless of the number of CS call connections present on the UE.

If the NOTIFY method from the Mobility Event Package is used, the UE would notify the CCCF on every CS call establishment, and correspondingly, the CCCF would notify the UE with the corresponding Handoff URI for that CS call.

To maintain the support of the 2-party CS call connection and the CS-based 3-party supplementary services, such as Call Waiting and 3-Way conference, during the VCC procedure, the UE needs to be provisioned with two Handoff Numbers. These two (2) Handoff Numbers are associated with the two (2) potential CS call connections (i.e. one Handoff number for each potential CS call connection) that could be established with the UE from the invocation of the CS-based 3-party supplementary services.

F.3.5 Use of Join Number for Establishing Additional CS Call Leg

For the VCC procedure supported by the CAMEL-based dynamic anchoring method, a Join Number is included in the CAMEL ICA request, from the CCCF/NeDS, which extends an additional call leg of the CS session from the GMSC or VMSC to the MGW under IMS control. This Join Number would have to be a network routable number.

The Join Number is dynamically assigned from a pool of numbers managed by the CCCF/NeDS, and the assigned Join Number should be unique to each active call session that requested a CS-to-IMS domain transfer. The primary purpose of the Join Number is for enabling the CCCF/NeDS to correlate the additional call leg to the call session that has initiated the CS-to-IMS domain transfer request. Once the correlation is completed, the Join Number is available for reassignment.

To maintain the support of the CS-based 3-party supplementary services, such as Call Waiting and 3-Way Conference, the CCCF would need to assign two (2) Join Numbers (i.e. one Join Number for each CS call connection) for the domain transfer of the two (2) CS call connections from CS to IMS.

F.3.6 Notification of CS Call Connection State to CCCF/NeDS

If required, the UE could use the NOTIFY method provided by the Mobility Event Package to send the state information of the CS call connection to the CCCF/NeDS, prior to the initiation of the VCC procedure. As indicated in Section 6.3.6.3.3.3.1, the UE and the CCCF/NeDS could invoke the NOTIFY method on every CS call establishment.

F.3.7 Emergency Call Services

CAMEL trigger is generally not invoked for emergency calls. Therefore, an established emergency call will not be able to be transferred from the CS domain to IMS. Despite the unavailability of the voice call continuity option, the UE will still be able to make emergency calls in the CS or IMS domain.

F.3.8 Reliable CS-to-IMS Transfer with Fallback to the CS Mode

The dynamic anchoring method using CAMEL 4 trigger provides reliable domain transfer from the CS mode to IMS. This is achieved by not releasing the CS connection to the UE until the IMS connection to the UE has been successfully established.

If the call continuity procedure fails at any point, before the IMS connection to the UE is established, the CCCF shall terminate the Call Continuity procedure by cancelling the ICA request from the corresponding CS call session and releasing the associated call legs in both CS and IMS domains.

F.3.9 Multi-Session services support with CAMEL 4

F.3.9.1 General

The method of using the CAMEL 4 ICA trigger to transfer the CS connection of a basic 2-party CS-to-CS call from the CS domain to IMS also supports the transfer of a CS call with multiple sessions. This enables the CCCF to support of supplementary services such as Call Hold, Three Way Conference and Call Waiting during the Call Continuity procedure. The corresponding message flows are described in the following subsections.

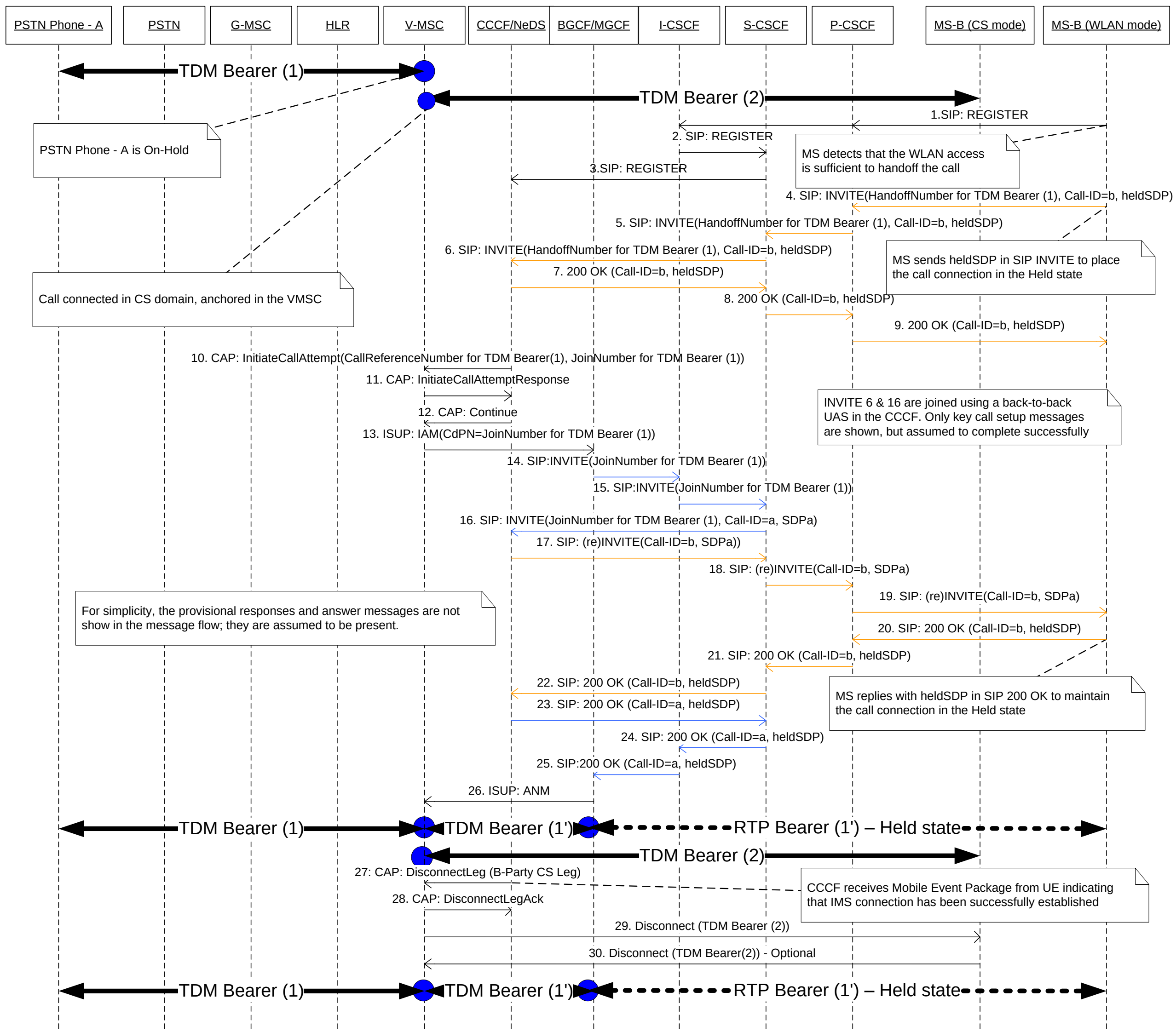
The 3-Way conference function for a 3-party call session in the IMS domain can be provided in the UE or the MRF. For the support of the 3-Way Conference service in the IMS domain, there would be two (2) RTP bearers connected to the UE if the UE provides the bridging function. On the other hand, if the bridging function is provided in the MRF, there would be only one RTP bearer between the UE and the MRF.

The message flow provided in the Section 6.3.7.5.4 assumes that the 3-Way conference function is provided in the UE.

Editor Note: The TR should cover the method of using the MRF to provide the call bridging function for multi-party call session in the IMS domain.

F.3.9.2 Call continuity for Call Hold service (Single Session)

Figures F.3-6 below provides a walkthrough of a CS to IMS transfer of a CS call to Party A in the Held state.



Figures F.3-6: Call Continuity for a 2-Party Call (Call Hold) – an UE with one Held Party

The call flow starts when the UE (Party B) is connected in the CS mode and has a call (Party A) in the Held state.

1. UE detects WLAN coverage and registers in IMS mode.
2. S-CSCF authenticates the UE subscriber.
3. S-CSCF registers to the CCCF on behalf of the UE subscriber.
4. UE initiates call handoff to WLAN by sending a SIP INVITE containing the "Handoff Number" to the P-CSCF, while maintaining the connected in an Hold state.
5. P-CSCF forwards the SIP INVITE request towards S-CSCF.
6. S-CSCF based on the iFC forwards the SIP INVITE to the CCCF.
7. CCCF establishes the call session by sending a SIP 200 OK response back to the S-CSCF. The SIP 200 OK contains Null SDP such that no RTP stream would be established.
8. S-CSCF forwards the SIP 200 OK response to the P-CSCF.
9. P-CSCF forwards the SIP 200 OK response to the UE. At this point, a session has been established between the CCCF and the UE.
10. CCCF sends a CAMEL InitateCallAttempt (ICA) request, which contains the Call Reference number of the established call connection to the VMSC (or GMSC, which is not shown in the above message flow) where the dual mode UE is attached.
11. VMSC (or GMSC) acknowledges the request by sending an ICA response back to the CCCF and then wait for the next set of instructions from the CCCF.
12. CCCF sends a Continue request to the VMSC (or GMSC).
13. VMSC (or GMSC) establishes an additional call leg to the existing call session (in the CS domain) by sending an ISUP IAM message towards the MGCF.
14. MGCF translates the ISUP IAM message into a SIP INVITE request and sends it to the I-CSCF.
15. I-CSCF selects the corresponding S-CSCF for the subscriber based on the "Handoff Number" and forwards the SIP INVITE request to the selected S-CSCF.
16. S-CSCF forwards the SIP INVITE request to the CCCF.
17. CCCF correlates the incoming SIP INVITE request to the call session associated with the SIP INVITE request received in Step 6, and sends an updated SIP INVITE request (i.e. (RE)INVITE) to the S-CSCF.
18. S-CSCF forwards the updated SIP INVITE request to the P-CSCF.
19. P-CSCF forwards the updated SIP INVITE request to the UE.
20. UE acknowledges the updated SIP INVITE request by sending a SIP 200 OK response back towards the P-CSCF. The SIP 200 OK contains a Null SDP to emulate a call connection in an Hold state.
21. P-CSCF forwards the SIP 200 OK response to the S-CSCF.
22. S-CSCF forwards the SIP 200 OK response to the CCCF.
23. The corresponding B2BUA in the CCCF sends a SIP 200 OK response the S-CSCF.
24. S-CSCF forwards the SIP 200 OK response to the I-CSCF.
25. I-CSCF forwards the SIP 200 OK response to the MGCF.

26. MGCF translates the SIP 200 OK response to an ISUP ANM message and sends it to the VMSC (or GMSC). At this point, a new call leg is added to the existing CS call session and this new call has a TDM bearer between the VMSC (or GMSC) and the MGW in the MGCF.

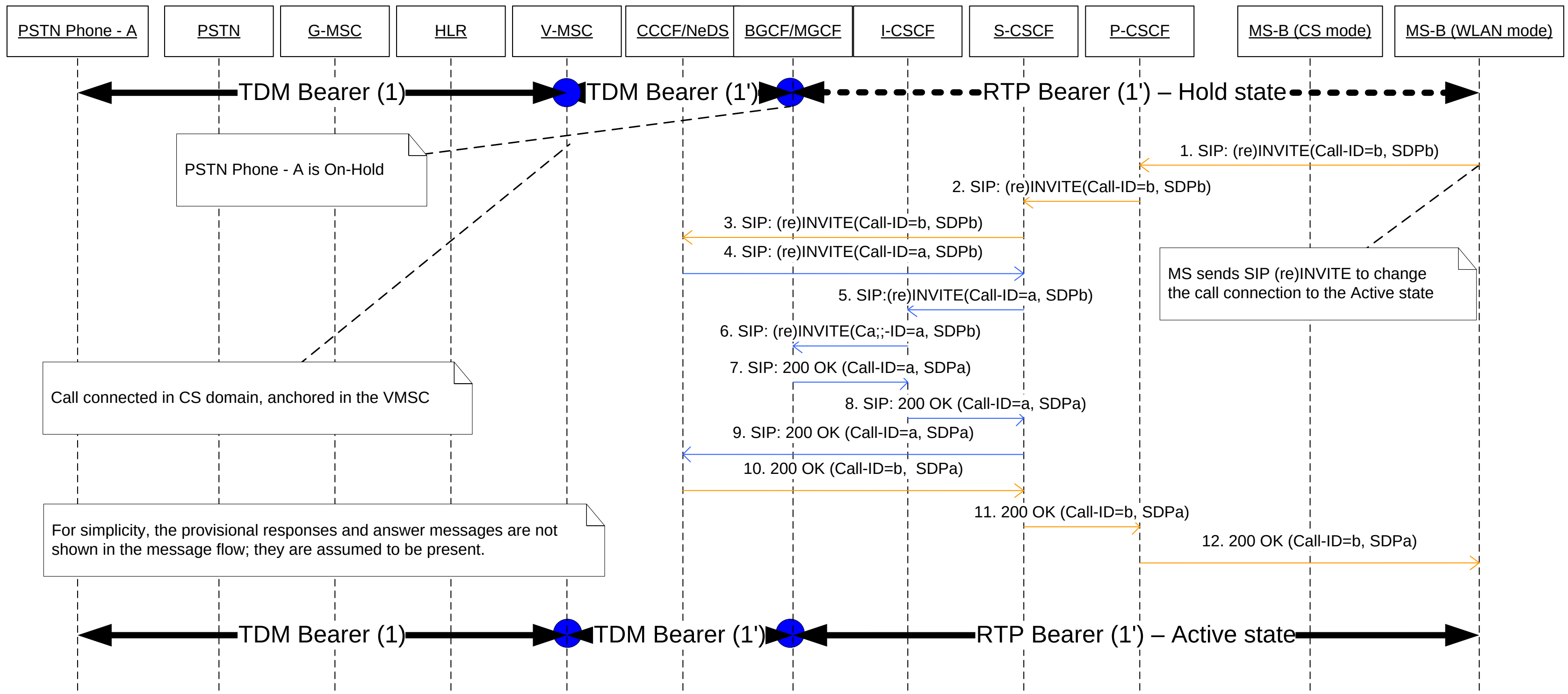
27. CCCF sends a CAMEL DisconnectLeg request to the VMSC when it receives a Mobility Event Package indicating that an IMS connection to the UE has been successfully established.

28. VMSC acknowledges the CAMEL DisconnectLeg request

29. VMSC releases the CS connection towards the UE.

30. UE releases its CS connection with the VMSC when it detects that an IMS call session has been established. This step is optional if the UE receives the Disconnect request from the VMSC first.

Figures F.3-7 below provides a walkthrough of changing the call to Party A from the Held state to the Active State after the call has been transfer to the IMS mode.



Figures F.3-7: Call Transition from the Held State to the Active State after Transfer to IMS Domain

The call flow starts when the UE is connected in the I-WLAN mode and has a call in the Held state.

1. UE sends a SIP INVITE request to establish the RTP connection to the UE).
2. P-CSCF forwards the SIP INVITE request to the S-CSCF.
3. S-CSCF forwards the SIP INVITE request to the CCCF.
4. The corresponding B2BUA in the CCCF sends a SIP INVITE request to the S-CSCF.
5. S-CSCF forwards the SIP INVITE request to the I-CSCF.
6. I-CSCF forwards the SIP INVITE request to the MGCF.

7. MGCF acknowledges the SIP INVITE request by sending a SIP 200 OK to the I-CSCF.

8. I-CSCF forwards the SIP 200 OK to the S-CSCF.

9. S-CSCF forwards the SIP 200 OK to the CCCF.

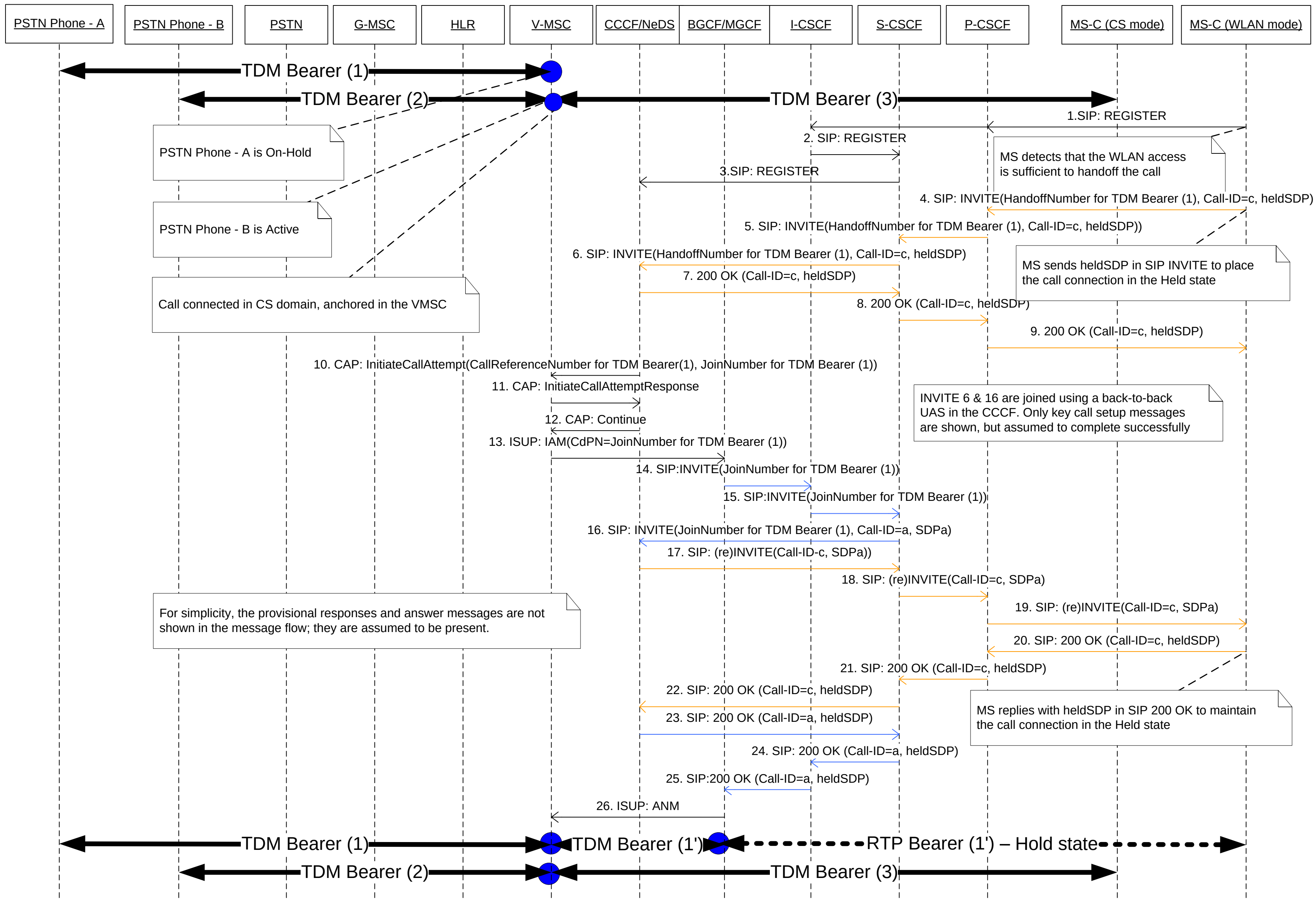
10. The corresponding B2BUA in the CCCF sends a SIP 200 OK to the S-CSCF.

11. S-CSCF forwards the SIP 200 OK to the P-CSCF.

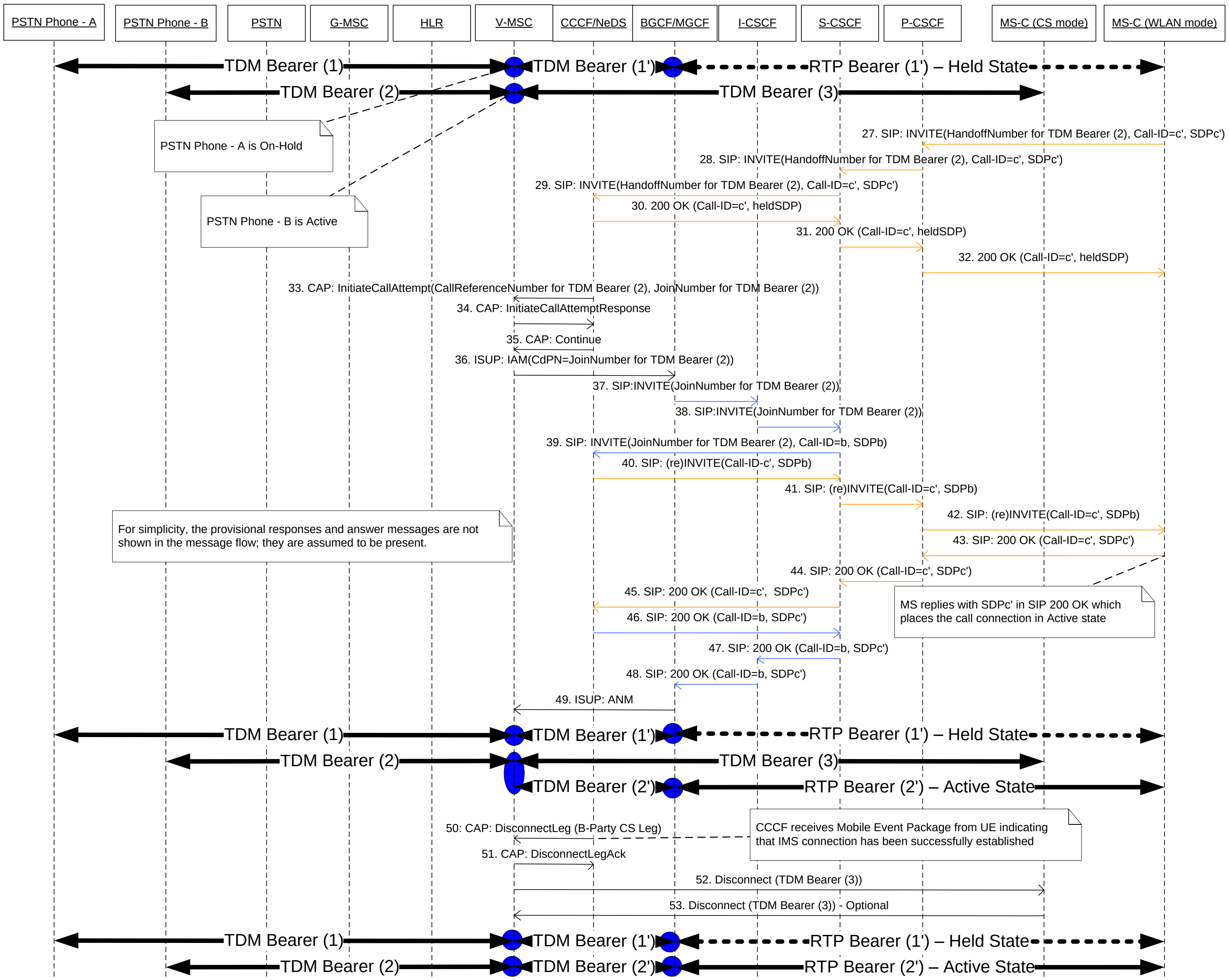
12. P-CSCF forwards the SIP 200 OK to the UE. The RTP connection to the UE is established after the Acknowledgement messages are sent from the UE to the CCCF and from CCCF to the MGCF (which are not shown in the message flow).

F.3.9.3 Call continuity for Call Waiting service (Multi-Session)

Figure F.3-8 below provides a walkthrough of a CS to IMS transfer of a CS call to Party A in the Held state and CS call to party B in the Active state.



Figures F.3-8: Call Continuity for a 3-Party Call (Call Waiting) – an UE with one Held Party and one Active Party (1 of 2)



Figures F.3-9: Call Continuity of a 3-Party Call (Call Waiting) – an UE with one Held Party and one Active Party (2 of 2)

The call flow starts when the UE is connected in the CS mode and it is in a Call Waiting mode (i.e. Toggle), which the UE has one CS call (Party A) in the Held state and another CS call (Party B) in the Active state.

Steps 1 to 26 are same as the first 26 steps in Figures 6.3.7.5-1. After Step 26, the UE still have two CS call sessions, the first one (Party A) in the Held state and the second one (Party B) in the Active state, and one new call session in IMS mode in the Held state.

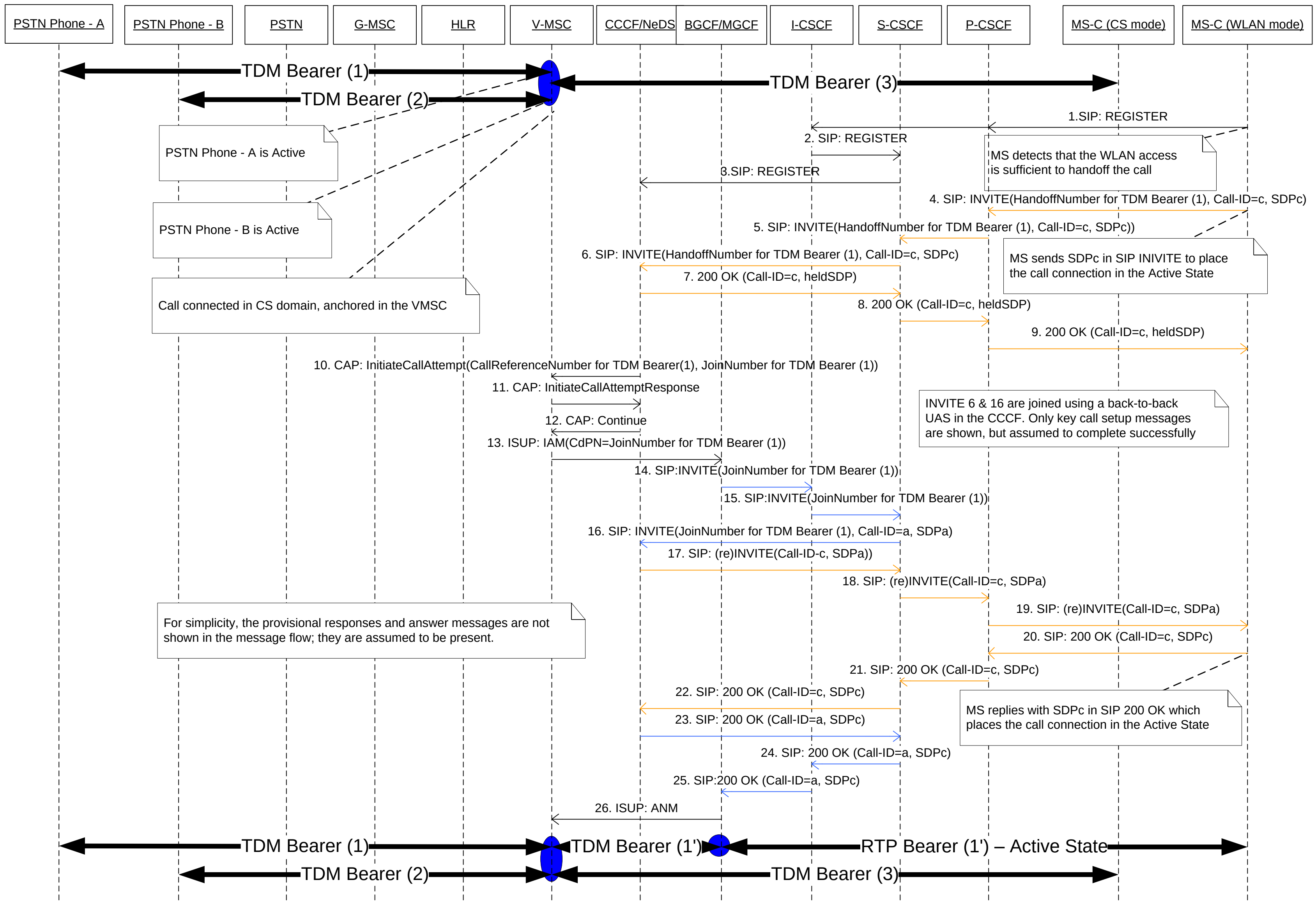
Steps 27 to 49 are same as Steps 4 to 26 in Figures 6.3.7.5-1, except that in Steps 44 to 49, a valid SDP is provided in the SIP 200 OK response, instead of Null SDP. The valid SDP enables the call connection to be in the Active state.

After Step 49, the UE still have two CS call sessions, one session (Party A) in the Held state and the second session (Party B) in the Active state, and two call sessions in IMS mode, one in the Held state and the second one in the Active state. In the UE, the IMS call session in the Held state corresponds to the CS call session that is the Held state. Likewise, in the UE, the IMS call session in the Active state corresponds to the active call session in CS mode.

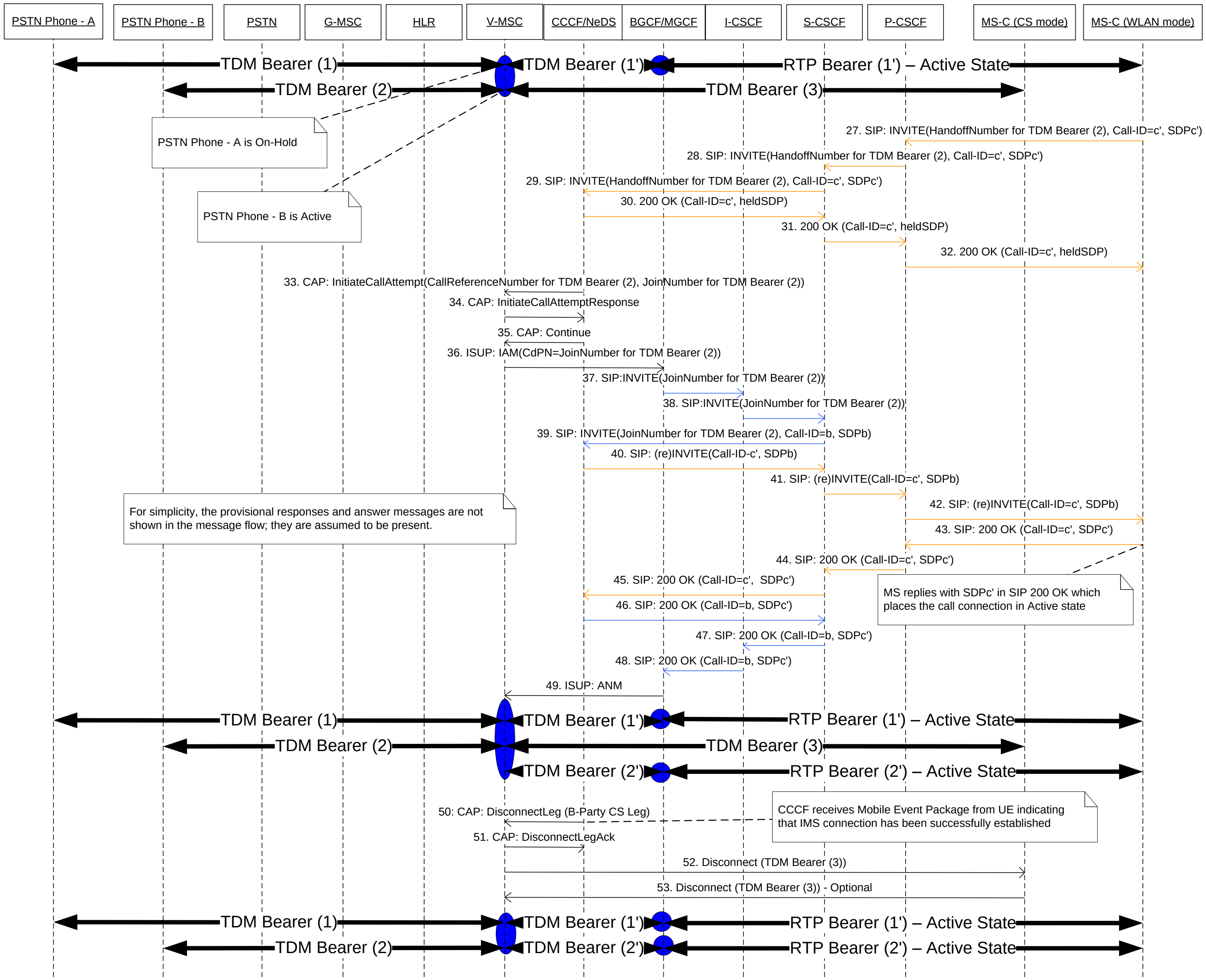
Steps 50 to 53 are same steps 27 to 30 in Figure 6.3.7.5-1, which shows the release of the CS connection to the UE after the new call sessions have been established in the IMS mode.

F.3.9.4 Call continuity for 3-Way Conference service (Multi-Session)

Figure F.3-10 below provides a walkthrough of a CS to IMS transfer of a CS call to Party A in the Active state and a second CS call to party B, which is also in the Active state.



Figures F.3-10: Call Continuity of the a 3-Party Call (3 Way Conference) – an UE with Two Active Parties (1 of 2)



Figures F.3-11: Call Continuity of a 3-Party Call (3-Way Conference) – an UE with Two Active Parties (2 of 2)

F.3.10 Evaluation of Dynamic Anchoring using CAMEL4 model

This alternative has the following advantages:

1. Dynamic anchoring of the call in the CS domain provides resource efficiency in terms of MSC ports and MGW ports.
2. Makes use existing standard based call setup procedure within IMS that requires no new method development for domain transfer.
3. Existing call control capabilities of MS/UE on both CS and IP modes are used in this alternative makes no special hardware/firmware development. An application software making use of existing telephony functions on both GSM and IMS are made use of to create the domain transfer logic.
4. Voice continuity is preserved during call continuity without major interruption of voice path during domain transfer.
5. Minimizing the call continuity procedure set up time for a multi-session call by transferring the held/active sessions in a concurrent fashion, instead of a serial manner (i.e. one session at a time).

The drawbacks:

'Break-before-make' – this can be improved by standardizing MGW capability to create a three way port for a brief period during domain transfer. i.e. PCM->RTP<-PCM and RTP->PCM<-RTP. Once the subscriber is transferred to a new domain the three-way port can be reconfigured to become a cost effective two way port.

Annex G:
UE Functions – IMS Controlled Model with static anchoring

NOTE: This is based on the distributed supplementary service model in the IMS Controlled Model.

G.1 UE Registration

CS registration:

VCC phone application informs its CS status to CCCF/NEDS if it is also IMS registered.

NOTE: This requirement is from 6.2a.1.1, it stated that "If the UE is also registered to the IMS domain, the UE indicates to the CCCF/NeDS Function about its CS status (i.e., IMSI attach state)." It should be noted that Network based query via HSS can also be used to determine UE CS status. If network based query is chosen in the TS stage then this will not be a UE requirement.

IMS registration:

VCC phone application exchange information with the CCCF/NeDS such as:

i. Send User's preference;

ii. Receive/store Operator policy;

iii. Indicate VoIP capabilities on IP-CAN (e.g., UE detects that the IP-CAN is not stable enough for VoIP application). This info can be send anytime after registration as well;

iv. Indicate CS status. Note: depended whether network based query is used or not.

G.2 UE Origination

VCC Phone Application retrieves the operator policy for originating domain selection from the VCC DB cache. The operator policy may force the UE to use certain domain to place the call, and override the user preferences.

IMS Origination:

VCC Phone Application receives PSI (and call reference) from CCCF/NeDS. VCC Phone Application stores the PSI and the call reference for handover.

CS Origination:

VCC Phone Application needs to aware whether to use USSD, SIP NOTIFY, or Do-Nothing (i.e., CAMEL case) for signalling the destination address of the CS call to CCCF/NeDS during call setup.

When UE is also IMS registered, VCC Phone Application receives a PSI (and call reference) from CCCF/NeDS. VCC Phone Application stores the PSI and the call reference for handover.

G.3 UE Termination

IMS Termination:

VCC Phone Application receives a PSI (and call reference)from CCCF/NeDS. VCC Phone Application stores the PSI and the call reference for handover.

VCC Phone Application reports (dynamically) VoIP capabilities on the IP-CAN to CCCF/NeDS in order to avoid terminating call to a non-VoIP capable IP-CAN.

CS Termination:

When UE is also IMS registered, VCC Phone Application receives PSI (and call reference) from AS. VCC Phone Application stores the PSI and the call reference for handover.

G.4 VCC Triggers

VCC Phone Application is responsible to start the VCC process from one domain to another. The logic is expected to be vendor dependent and proprietary. It is expected that the VCC Phone Application detects that the IP-CAN, via Access capability monitoring, is good enough for VoIP session before the VCC to IMS procedure is started. Additionally, Ping-pong type of VCC procedure should be avoided.

Editor's Note: The interaction between Network controlled HO and VCC procedure, as per operator preference, is FFS.

G.5 VCC Procedures

G.5.1 From CS to IMS

From CS to IMS(WLAN):

If the VCC Phone Application has already stored the PSI (and call reference) during the CS call phase, it can begin the VCC procedure right away. Otherwise the UE must register to IMS first, exchange information with CCCF/NeDS including operator policy for originating domain selection, and PSI and call reference for ongoing CS call.

VCC Phone Application uses PSI (received from CCCF/NeDS) in INVITE to CCCF. VCC Phone Application must allow the existing CS voice call uninterrupted while the new SIP session is being established (i.e., existing CS voice call can not be put on HOLD). VCC Phone Application should swap the media to the IMS side as early as possible, i.e., meaning when the UE receives the 200 OK for INVITE. After media is swapped, VCC Phone Application disconnects the call from the CS side.

From 3G CS to IMS with UMTS PS RAB:

When UMTS radio layer supports both CS and PS RAB, the VCC Phone Application can initiate the PS RAB to establish the IP-CAN to IMS. With both the CS and PS RABs established, the same procedure as defined above for "From CS to IMS(WLAN)" applies.

UMTS radio layer supports only PS RAB is FFS!

From 2G CS to IMS with UMTS PS RAB:

This requires Intersystem HO from 2G CS to 3G CS before VCC can apply. Intersystem HO is network controlled HO and no VCC components are defined. Therefore, no specific UE VCC procedures are needed.

From 2G CS to IMS with GSM/EDGE GPRS:

DTM capabilities are required to established simultaneous CS and PS connections. With both the CS and PS established, the same procedure as defined above for "From CS to IMS(WLAN)" applies.

G.5.2 From IMS to CS

From IMS(WLAN) to CS:

VCC Phone Application UE uses PSI (received from CCCF/NeDS) in set up a new call on the CS side to CCCF/NeDS. VCC Phone Application must allow the existing IMS VoIP session uninterrupted while the new CS call is being established (i.e., existing SIP session cannot be put on HOLD). VCC Phone Application should swap the media to the CS side as early as possible meaning, i.e., when the UE receives the CONNECT from the MSC. After media is swapped, VCC Phone Application disconnects the SIP session from the IMS side.

From IMS with UMTS PS RAB to 3G CS:

When UMTS radio layer supports both CS and PS RAB, the VCC Phone Application can initiate the CS RAB to establish the CS call to CCCF/NeDs. With both the CS and PS RABs established, the same procedure as defined above for "From IMS(WLAN) to CS" applies.

UMTS radio layer supports only PS RAB is FFS!

From IMS with UMTS PS RAB to 2G CS:

If UMTS supports both CS and PS RABs then VCC Phone Application can invoke VCC to 3G CS side. After that, 3G CS to 2G CS transition is done with Intersystem HO which is a network controlled HO.

From IMS with GSM/EDGE GPRS to 2G CS:

DTM capabilities are required to established simultaneous CS and PS connections. With both the CS and PS established, the same procedure as defined above for "From IMS(WLAN) to CS" applies.

G.5.3 Emergency Service

This is FFS.

G.5.4 Supplementary Services

This is FFS.

Annex H: Operator Deployment Scenarios

A variety operator scenarios are possible that could impact Voice Call Continuity architectural decisions. Interworking scenarios that need to be considered include various permutations of CS and IMS Core Network and Access Network components, and related subscriber information provisioning. The following scenarios should be taken into consideration. Note that the CS domain should include considerations for both the PLMN and PSTN. Decomposition of existing functionality allocated between the home networks is out of scope of 3GPP. All entities are defined as per TS 23.002 [3].

1 Scenario 1 No Interworking:

In this scenario all network elements are provided by operator A.

Core Network entities:	Operator A
Access Network:	Operator A
IP Multimedia (IM) Core Network entities:	Operator A
3GPP/WLAN Interworking entities:	Operator A

2 Scenario 2 CS Interworking:

In this scenario Core Network entities provided by Operator Aand IP Multimedia (IM) Core Network entities provided by B. Operator B is also providing the subscriber's E.164 number. Operators B's subscribers receive IMS based services when they are in Operators B's network and can roam into operators A's network and only receive Core network services.

Core Network entities:	Operator A
Access Network:	Operator A
IP Multimedia (IM) Core Network entities:	Operator B
3GPP/WLAN Interworking entities:	Operator B

3 Scenario 3 IMS Access Interworking:

In this scenario Core Network entities and IP Multimedia (IM) Core Network entities are provided by operator A. Operator A is also providing the subscriber's E.164 number Operator B has only deployed 3GPP/WLAN Interworking entities.

Core Network entities:	Operator A
Access Network:	Operator A
IP Multimedia (IM) Core Network entities:	Operator A
3GPP/WLAN Interworking entities:	Operator B

4 Scenario 4 Service Interworking:

In this scenario Core Network entities are provided by Operator A and the IP Multimedia (IM) Core Network and 3GPP/WLAN interworking entities provided by B. The subscriber may be reachable by two separate identities, operator A provides one E.164 number and Operator B provides a second E.164 number. In addition, Operator B may provide an addutional IMS Public Identity

Core Network entities:	Operator A
Access Network:	Operator A
IP Multimedia (IM) Core Network entities:	Operator B
3GPP/WLAN Inter-working entities:	Operator B

Annex I: Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2005-05	SA#28	SP-050347	-	-	Updated editorially by MCC for presentation to TSG SA for information. Many figures re-numbered.	0.2.1	1.0.0
2005-11	SA#30	SP-050675	-	-	Updated editorially by MCC for presentation to TSG SA for approval.	1.7.0	2.0.0
2005-12	-	-	-	-	Updated to version 7.0.0 by MCC after TSG SA#30 approval	2.0.0	7.0.0