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Abstract of document:

TR 23.806 “Voice Call Continuity between CS and IMS Study” is a 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).

Basic assumptions, architectural requirements and session and traffic scenarios have been defined that document the functionality that candidate architectures will need to provide. A number of alternative candidate architectures have been documented. They broadly fall into two approaches known as IMS controlled and Anchored Call Control and there is scope for merging several of the alternatives. The TR is structured such that each solution documents CS-IMS and IMS-CS inter-system handover, terminating network selection, call origination and termination and impact on supplementary services. Documentation of the architecture alternatives will continue in subsequent meetings.

Changes since last presentation to TSG SA:

This is the first time that this technical report is being presented to TSG SA.

TSG SA2 believes that this version of TR 23.806 is more than 50% complete.

Outstanding Issues:

Requirements from SA1 have yet to be finalised and any necessary Basic Assumptions and Architectural Requirements resulting from additional SA1 requirements will need to be incorporated.

The supported operator scenarios need to be agreed upon.

The criteria for selection of a solution need to be agreed.

There are proposals for an Anchored Call Control solution that should be aligned in subsequent meetings to identify commonalities and document differences.

A proposal for Terminating Domain Selection currently documented within the IMS Controlled Handover solution section should be refined to allow operator policy control and investigated for more common applicability.

Impacts of the solutions on accounting needs to be elaborated.

A number of Editor's Notes which document identified open issues within the technical content have been created during the agreement of contributions in SA2#46 and these will need to be resolved and removed.

The impacted standards have not yet been identified, and may be different depending upon the solution chosen.

Contentious Issues:

None identified.

3GPP TR 23.806 V1.0.0 (2005-05)

Technical Report

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Voice Call Continuity between CS and IMS Study
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GLOBAL SYSTEM FOR
MOBILE COMMUNICATIONS

Keywords

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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 IPCANs, 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 IPCANs.

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 Tele phone (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 41.001: "GSM Release specifications".

[2] 3GPP TS 23.234: "3GPP system to Wireless Local Area Network (WLAN) interworking".

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the [following] terms and definitions [given in ... and the following] apply.

<*defined term*>: *<definition>*.

example: text used to clarify abstract rules by applying them literally.

3.2 Symbols

For the purposes of the present document, the following symbols apply:

<symbol> <Explanation>

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

I-WLAN	Interworking WLAN
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 IP Connectivity Access Network.
- 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, 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, TS 33.234) shall be reused.
- The UE will be capable of transmitting and receiving simultaneously in the CS domain on GERAN/UTRAN, and on the IP Connectivity Access Network.
- 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 inter domain constraints.
- When a user is attached to both the CS and IMS domains, the network has the responsibility for selecting the terminating service domain, depending on operator policy and possibly user preference.
- Support for Voice call and Emergency call handover shall be provided, if the target domain supports it.
- Use of available QoS mechanisms need to be considered, however, the Impact on the QoS mechanism is out of the scope.

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

Editor Note: The CS domain in roaming should be supported.

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 select the appropriate access Network (such as GSM/UMTS radio or IP Connectivity Access Network). The selection may be based, e.g., on operator policy for real-time voice service and user's preference.
- 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.
- While not in CS or IMS voice call, the UE shall be able to detect and automatically select the appropriate access Network (such as GSM/UMTS radio or IP Connectivity Access Network). The selection may be based, e.g., on operator policy for real-time voice service and user's preference.
- 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.
- Handoff should be provided such that from the end user's perspective minimal service disruption is perceived. Handoff 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

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 handover occurs

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 Voice 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.

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

- <u>Mostly IMS network traffic</u>
  This is an traffic scenario where the network supports predominately IMS based traffic.

- <u>Mixed CS – IMS network traffic</u>
  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: This section illustrates the reference model to support service continuity

6.2.1 Call Continuity Control Function (CCCF)

CCCF provides functions for service 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.

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 Terminating Domain Selection

The ability to select the correct domain in which to terminate the call is required. While, 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.

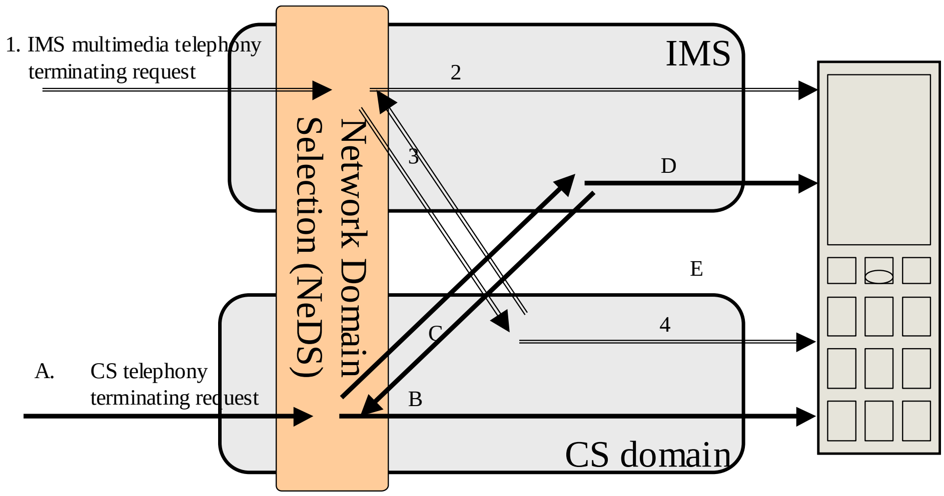


Figure 6.2.1: Logical view of Terminating Network Domain Selection Functionality

Figure 6.2.1 above depicts a logical representation of the network domain selection functionality. This could be one or several logical entities.

A telephony call to the CS domain (A) could be directly routed to the UE via the CS domain (B), or via the IMS domain (C, D) for terminating treatment.

An IMS terminating multimedia call (1) could be terminated on the IMS (2) or via the CS domain (3).

There are cases where the network domain selection may require to be re-invoked (indicated by the return arrows (3) and (C).

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.

There are a number of potential solutions for network domain selection. In order to generalise the discussion and understand the requirements before delving into protocol details, this contributions proposes a general approach to the problem.

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

- The Terminating NeDS function needs to be 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 Terminating NeDS function needs to be aware of whether the terminal is attached to the CS domain.

The Terminating NeDS functional can make a decision as to the appropriate terminating domain, taking into account the operator, user and service preferences.

6.3 Service Continuity Model: IMS Controlled Alternative

6.3.1 General Description

In this alternative the CS-IMS continuity is solved by employing CS IMS Voice Continuity Service (CIVCS) in the user's home IMS network.

CIVCS allows for Handing over, Subsequent handing over and Handing back of Call Control functions of a CS-IMS capable UE's voice call for active mode roaming across CS Domain and IM Subsystem with seamless user experience.

CIVCS is a home IMS network service that can be enabled either via static or dynamic anchoring techniques. Fundamental principles of CIVCS as applicable to static and dynamic anchoring are listed below:

- CIVCS 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 DTAP in SIP for Handovers from GSM/UMTS CS to IMS over I-WLAN, significant changes to GSM/UMTS Core Network and Access Network nodes are required for SIP encapsulation in BSSAP for Handover from IMS to GSM CS, and SIP encapsulation in RANAP for Handover from IMS to UMTS CS; it is therefore not feasible to maintain the same anchor control model with Handovers involving I-WLAN.

- CIVCS anchors the bearer for CS originations and terminations at a MGW controlled by an MGCF in user's home IMS network. Although the MGCF belongs to user's home IMS network, the MGW used to hairpin the bearer is located in close proximity with the CS network to save TDM backhaul cost.
- CIVCS is realized by using IETF Third Party Call Control (3pcc) function.

- CIVCS provides the Handover Control Functions for CS-IMS voice continuity. All CS-IMS Handovers are executed and controlled by CIVCS upon UE's request.

- Since Handovers are executed across CS Domain and IM Subsystem with different call control protocols, DTAP is used in CS Domain whereas SIP is used in IMS for call control procedures; the handover procedures execute 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. As a result, Handover execution can only be guaranteed in the active state of the CS call or IMS session.

- CIVCS provides cohesive billing with a complete Handover 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 inter domain transition are captured as Handover 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.

- CIVCS 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.

- 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 transition.

CIVCS does not influence supplementary service execution in the serving network node prior to or post handover. 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.

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

CIVCS 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 CIVCS to facilitate control of the bearer path upon Handover 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.

6.3.1.2 Dynamic CS Anchoring for CIVCS using DACCI

DACCI is a new CS domain service that provides dynamic anchoring of CS calls at CIVCS.

Enablement of first CS to IMS voice 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 transition in order to enable subsequent transitions between CS and IMS via CIVCS. This helps eliminate the requirement for static anchoring of CS calls whereby calls are anchored with CIVCS at initial call setup resulting in resource inefficiencies.

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.

DACCI uses the Mobility Event package to communicate with CIVCS, session information required to perform transition from CS to IMS.

DACCI is applicable to all voice teleservices including the Emergency Call Service.

6.3.1.3 Dynamic CS Anchoring for CIVCS using ECT

6.3.1.3.1 General

Although ECT has significant drawbacks when used as the only solution for Handovers from CS domain to IM Subsystem, it may be used as a means for dynamically anchoring CS calls at CIVCS 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 transition. CIVCS is used for all subsequent Handover and Handbacks from CS to IMS and from IMS to CS.

6.3.1.3.2 Supplementary Service control for ECT enablement

Certain supplementary services like Call Forward Unconditional and Incoming Call Barring can prevent ECT enabled Handovers 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 Handover procedure and re-enable these services just before the CS radio is released upon Handover.

6.3.2 Routing Selection Decision

6.3.2.1 Alternative A

6.3.2.1.1 Inter-Domain Routing Policy Definition

The Inter-domain routing policy is that set of service logic functions available to the MSC, GMSC, and IMS CSCF servers which examine the current state of registration within the domains, and, based on that knowledge and operator and subscriber preferences, route the call for completion in the appropriate domain.

6.3.2.1.2 Scenarios

The scenarios to be supported by this architecture are:

- Registered in the Home CS Network;
- Registered in a visited CS Network;
- Registered on a non-3GPP Access Network capable of supporting IMS VoIP.

The UE may or may not be simultaneously registered in the CS and IMS networks.

6.3.2.1.3 Logical Architecture

The following diagram depicts the architecture for implementing the routing policy.

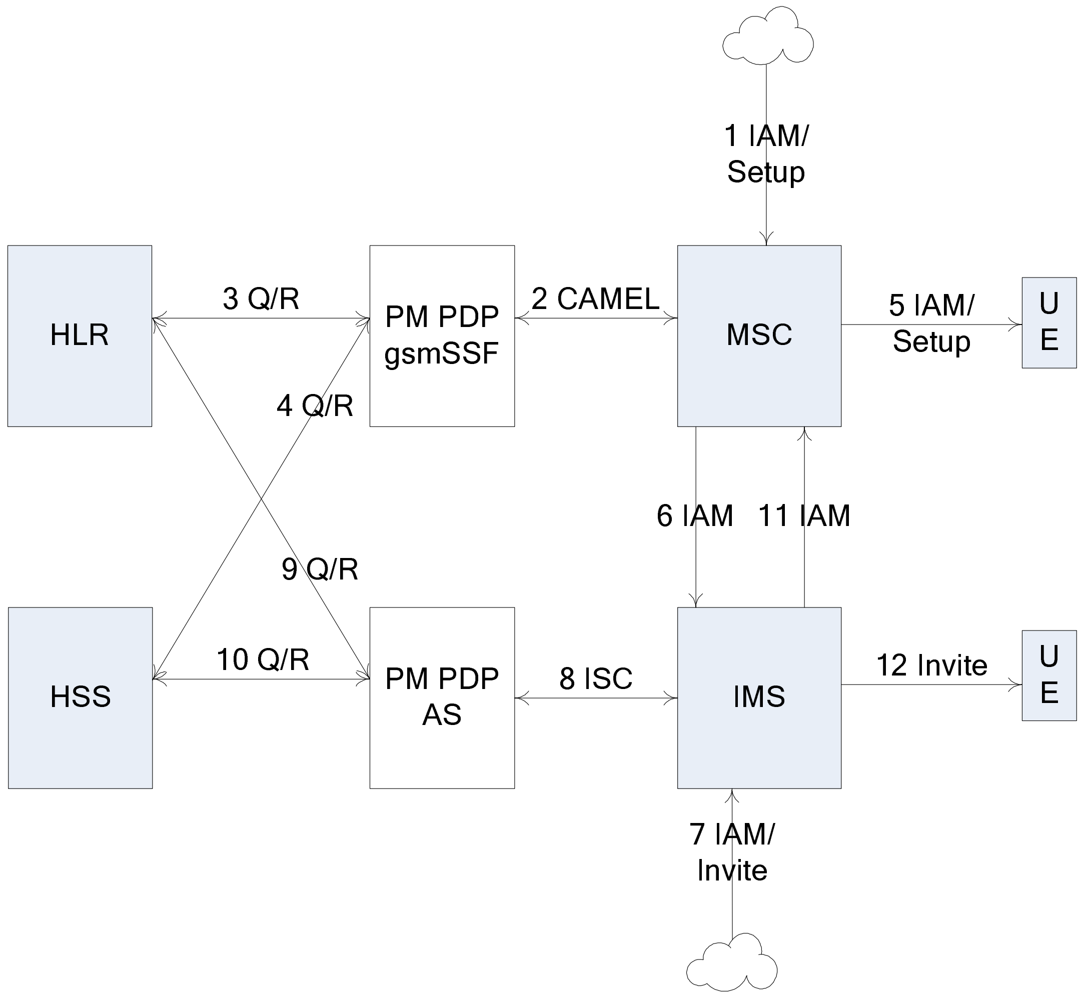


Figure 6.3.1

In the architecture above, the service logic is executed as follows:

1. An IAM or call setup arrives at an MSC ar a GMSC from an external network or via a originating subscriber in the GSM network.

2. Service Logic is invoked via a termination trigger

3 and 4. The service logic causes queries to both the HLR and the HSS to obtain the current registration states for both CS and IMS domains.

5 or 6. Depending upon the registration states and the operator policy, the MSC routes the call either towards the IMS system, or towards the UE in the CS domain.

Similarly, in steps 7 through 12, an IAM or an invite entering the IMS system receives similar treatment.

6.3.2.1.4 Routing Policy Execution in CS Domain

In the CS domain, an SCP serves as the Policy Decision Point. The Policy Enforcement Point is the GMSC function, which envokes CAMEL triggers to the Policy Decision Point using existing standardized CS triggers to query the Policy Decicion Point. A CS termination will cause the gsmSSF acting as the Policy Decision Point to execute the queries and determine the routing based on operator policy. The response to the CAMEL trigger returns the routing information to the GMSC, which as the Policy Enforcement Point routes the call.

The change to the existing CS network is the addition of a gsmSSF that can query both the HLR and HSS and determines policu based upon the query responses, operator preferences, and subscriber preferences.

6.3.2.1.5 Routing Policy Execution in IMS Domain

In the IMS domain, the Serving CSCF (S-CSCF) queries the AS PDP to determine routing. The Policy Decision Point executes the queries to the HSS and HLR and determines the routing based upon the query responses and operator policy. The interface between the S-CSCF and the Policy Decision Point is the ISC interface.

Based upon the response from the Policy Decision Point, the S-CSCF routes either to the CS network via the BGCF or through the IMS network to the subscriber's current Proxy CSCF.

6.3.2.1.6 Conclusion

Sufficient flexibility exists within already standardized capabilities such that it is not necessary to define an architecture that attempts to combine or synchronize possibly separate HLRs and HSSs with new protocol for the purposes of flexible routing control. Note that this architecture makes no special distinctions about either HLR/HSS "ownership". All that is required is that cooperating networks allow access. Likewise, the architecture would support the PDP for both networks residing on a single platform.

Editor's Note: The following are open issues:

- Error Scenarios "IMS registered but UE not reachable" , "CS attached but UE not reachable" , "Registered in IMS but not with 802xx voice(e.g., GPRS)" need to be considered.
- Correctly label the query/response interfaces.

6.3.2.2 Alternative B

6.3.2.2.1 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: The term "Reachable" is used throughout this document to designate that the user has registered in the domain, and no corresponding incoming call barring service been activated.

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

6.3.2.2.2 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 Routing Policy Decision Point (RPDP), to perform routing policy decision function based on the combination of information from different aspects. Accordingly, HLR/HSS should be enhanced with some necessary improvements to support decision information provisioning and interaction with a RPDP. A RPDP is shown in the figure below.

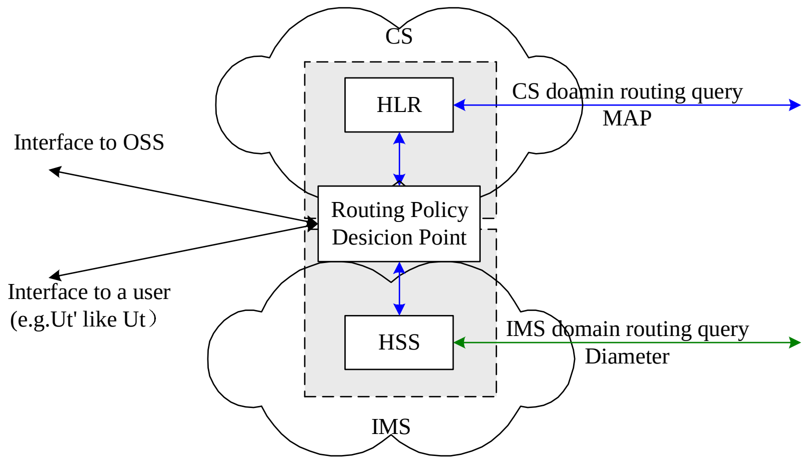


Figure 6.3.2

A Routing Policy Decision Point (RPDP) is a newly-added logical functional entity for the enhancement of inter-domain routing control, which can be a separate physical entity, or be a logical entity embedded in HLR or HSS. A RPDP stores routing policy of a user and provides routing policy decision to the HLR/HSS that initiates routing decision query. In addition, A RPDP 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 HLR/HSS can make routing decision based on the information owned by them, they will not quire the RPDP for the routing policy. A typical procedure shown as follows:

NOTE 1: The detailed enhancement on HLR/HSS, i.e. interface and procedure between HLR/HSS and RPDP, is FFS.

NOTE 2: From Release 5, there is only a HSS, which consists of the original HLR/AUC functionality required by the CS/PS Domain, and so HLR is no longer a separate network entity as described in TS 23.002. If operator only owns the HSS for IMS and CS, the RPDP was embedded on the HSS and the interaction shown below was all performed internal.

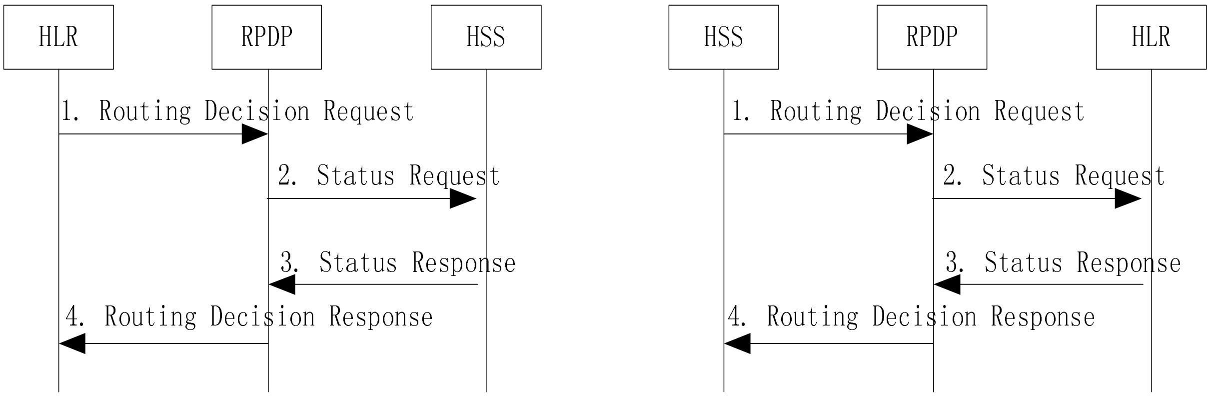


Figure 6.3.3

1. Once receiving the routing information query from GMSC/I-CSCF, the HLR/HSS sends a Routing Decision Request (User's reachable status, User Identifier) to the RPDP. User's reachable statuses consist of the Registration status i.e. CS attached for HLR or IMS registered for HSS, and status of incoming call barring service. As for User Identifier identified the user, if queried from HSS, it may be a Public User ID. As for HLR, it may be a MSISDN.

2. According to the current routing policy, the RPDP determine whether the user's status in the other domain is needed for routing decision. If user's status is necessary, RPDP sends a Status Request (User Identifier) to HSS/HLR. Otherwise, the RPDP perform step 4.

3. After received Status Request form RPDP, the HSS/HLR response a Status Response (User status) to RPDP. Including the User's Registration status i.e. CS attached for HLR or IMS registered for HSS.

4. Based on the combination of routing policy and the Status information, the RPDP determines the network through which the terminating call should be routed and sends a Routing Decision Response (Routing Decision) to HLR/HSS. Then the HLR/HSS can get the routing information based on this routing decision and responds the routing information to GMSC/I-CSCF.

Therefore, a RPDP is also to enable to:

- provide interfaces to HLR/HSS through which the HLR/HSS can retrieve the routing information and routing decision based on the current routing policy, and the RPDP can retrieve the user's subscription information in CS/IMS domain, status information about registration, location update and location information. This interface may be implemented by using MAP or other appropriate protocols and related extended protocols;

- provide interfaces to OSS. Operators can utilise this interface to configure routing policy flexibly based on the operation policy;

provide interfaces to a user to configure routing policy based on the user's preference. This interface may be implemented by means of, e.g. like Ut interface in IMS.

6.3.2.2.3 An Example of Routing Policy Decision

Here is an example of routing policy decision. In this case, when a terminating call from PTSN arrives at a GMSC, the GMSC interrogates a HLR which the called user is subscribed to get the routing information. At receiving the query request, the HLR communicates with a RPDP to get routing policy decision. The RPDP makes a decision based on the current configured routing policy and called user's status information (e.g. reachable flag) stored in HLR and HSS. In this case, the result of decision is that the call should be routed through IMS domain, and then the HLR return the routing information to the GMSC. Detailed behaviour is shown as follows:

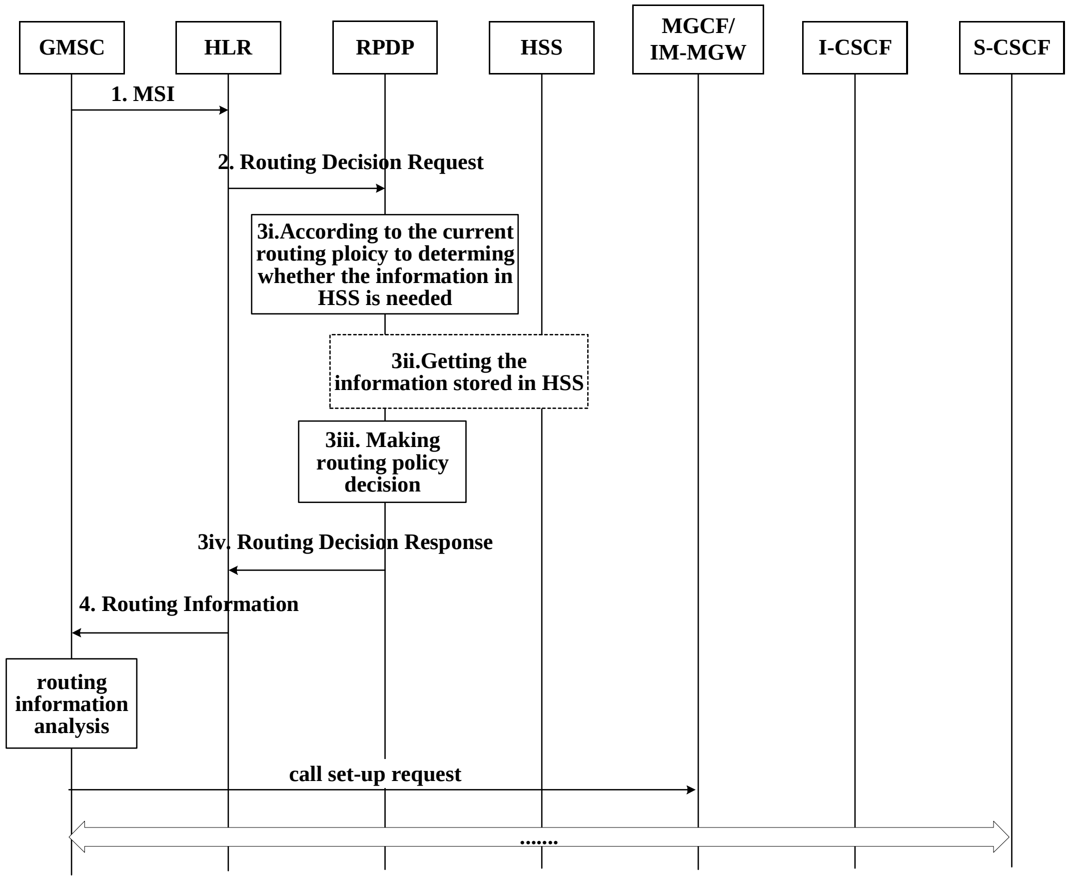


Figure 6.3.4

1. The GMSC receiving a terminating call from PSTN initiates a routing information query to the HLR using MSI message (MAP_SEND_ROUTING_INFORMATION).

2. The HLR sends a Routing Decision Request to the RPDP querying routing policy decision. In this request, the user's status information in CS domain should also be delivered to the RPDP.

3. After received the Routing Request, the RPDP makes routing policy decision and return the result of decision to the HLR, including some sub-steps:

   i. According to the current routing policy, the RPDP determine whether the user's status is needed for routing decision. If user's status is necessary, continue the following step. Otherwise, perform step iii.;

   ii. The RPDP interacts with HSS to get the user's status in IMS domain;

   iii. Based on the combination of routing policy, and the status information, the RPDP determines the network through which the terminating call should be routed (i.e. routing policy decision);

   iv. The RPDP returns the result of routing policy decision to the HLR and indicates the incoming call should be routed through the IMS domain. In the result, an identifier of MGCF is included for further call process.

4. The HLR returns the Routing Information (the MSRN getting from VLR based on the routing policy decision) to the GMSC and the GMSC routes the call to the appointed network and related entity, e.g. a MGCF.

The routing through IMS domain is similar to the above case and not to be listed here.

6.3.3 Registration

The UE and CIVCS subscribe to each other for the Mobility Event package upon UE Registration with IMS to enable exchange of session information that is necessary to perform Inter domain Voice Call Continuity for a CS IMS user engaged in multiple sessions.

Figure 6.3.5 below describes a sequence for UE Registration with IMS to enable Voice Call Continuity using CIVCS.

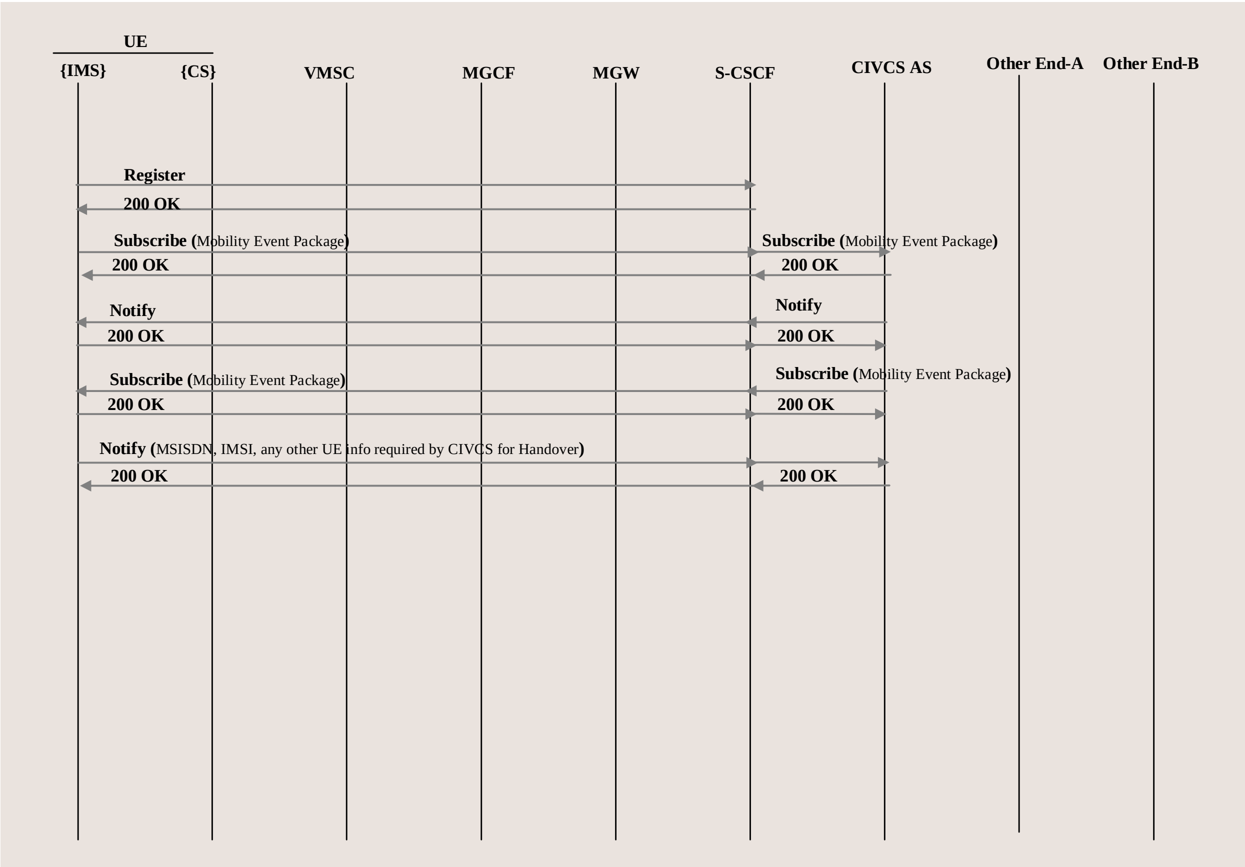


Figure 6.3.5: IMS Registration to enable CIVCS-UE information exchange

- The filter criteria set for the CS IMS user on Subscribe with Mobility Event directs the Subscribe for Mobility Event initiated by the UE post IMS Registration to the CIVCS AS assigned to the registered user.
- The CIVCS AS replies with a 200 OK and a Notify followed by a Subscribe for the Mobility Event toward the UE.
- The UE responds with a Notify consisting of all user identities available at the UE along with any other information that may be required by CIVCS for execution of session anchoring and enablement of Voice Call Continuity procedures.

6.3.4 Origination

6.3.4.1 IMS origination

6.3.4.1.1 Static Anchoring: CIVCS controlled IMS originating sessions

6.3.4.1.1.1 General

Filter criteria associated with CIVCS are stored in the HSS as part of the CS-IMS user's service subscription profile. The CIVCS filter criteria are downloaded to the currently assigned S-CSCF as part of Initial Filter Criteria at the time of subscriber's registration with IMS network. Initial Filter Criteria are executed at the S-CSCF upon IMS session initiation from the CS-IMS user that results in routing of the user's session to CIVCS. CIVCS enables a Routing B2BUA 3pcc function to control the bearer path for the session.

In order to maximise network efficiencies with the use of CIVCS, it is recommended that subscriber's current location be used at the S-CSCF or at CIVCS such that it results in invocation of 3pcc function at CIVCS only when the user is being served in the areas with overlapping coverage or with borders between the two domains.

6.3.4.1.1.2 IMS originations walkthrough

Figures 6.3.6a and 6.3.6b describe how signalling and bearer paths are established for IMS originations from CS-IMS users.

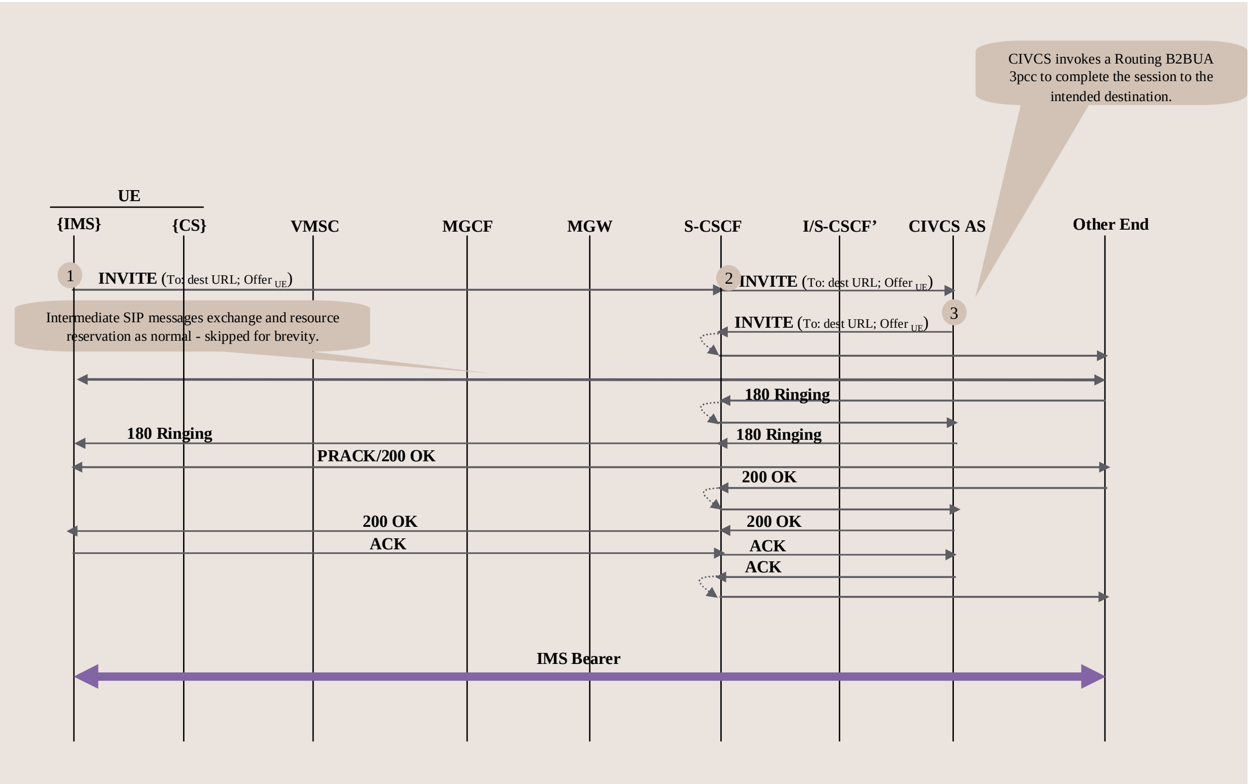


Figure 6.3.6a: IMS Origination controlled at CIVCS walk-through

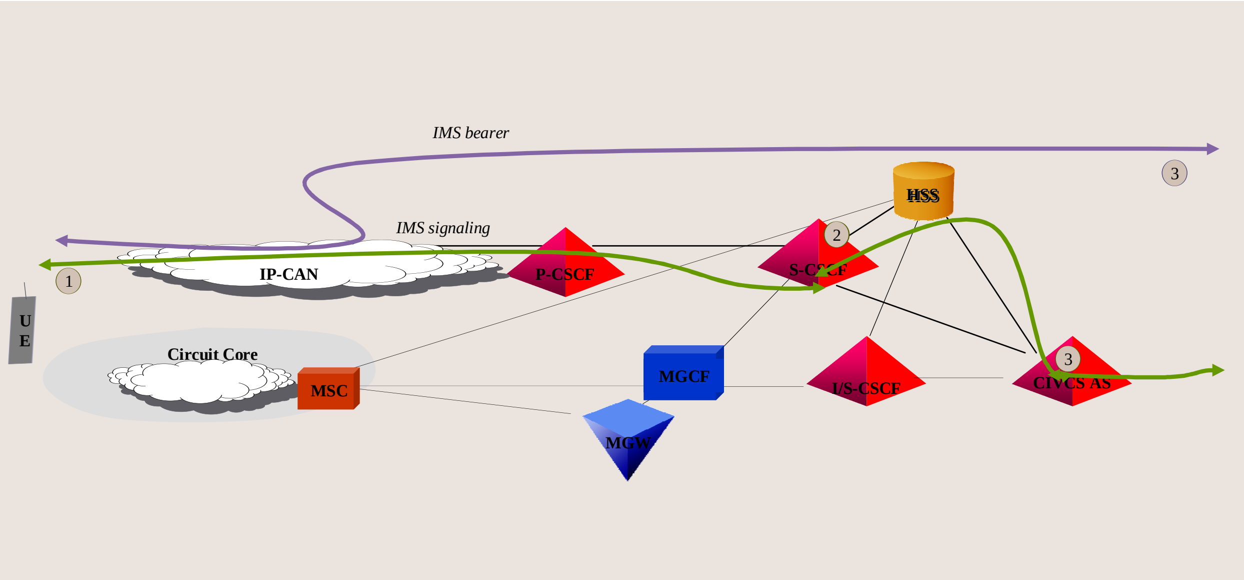


Figure 6.3.6b: IMS Origination controlled at CIVCS walk-through

1. The UE registers with IMS resulting in an assignment of S-CSCF. It subsequently initiates an originating session as normal.

2. Filter criteria at the S-CSCF result in forwarding of the INVITE to the CIVCS application server.

CIVCS completes the session establishment by setting up the terminating leg of the session. It enables 3pcc function to maintain session states for control of the originating and terminating leg of the CS-IMS user's session in order to control bearer upon Handover requests from the UE.

6.3.4.2 GSM/UMTS CS origination

6.3.4.2.1 Static Anchoring: Originating call anchoring at CIVCS for users roaming in CS Domain

6.3.4.2.1.1 General

Special routing techniques are established at the Visited MSC in areas with overlapping coverage and with borders between the two domains so that CS originations for CS-IMS users are routed via CIVCS in user's home IMS network. The Visited MSC routes the CS originating call to CIVCS which enables a Routing B2BUA 3pcc function to control the bearer path of the call. The original called number is passed to the CIVCS application so that it can perform final translations to route the call to the originally intended destination.

This routing function can be realized by using appropriate translation techniques or using a CAMEL service at the Visited MSC that steer the calls made by CS-IMS Voice Continuity service subscribers via the user's home IMS network. A few potential options to realize this function are specified below:

- Use CAMEL service or special translation techniques at the Visited MSC to prefix steering digits to the called party number in the outgoing IAM message. Note that special translation techniques may not be available at the networks owned by roaming partners, whereas a standardized R99 CAMEL trigger is the only requirement for enablement of the CAMEL service.

- Use CAMEL service to manipulate the outpulsed digits such that the CIVCS PSI is sent as the called party number and the original called number is communicated in another, to be determined, ISUP parameter to the IMS network.

6.3.4.2.1.2 CIVCS Anchored CS Origination walkthrough

Figures 6.3.7a and 6.3.7b describe 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.

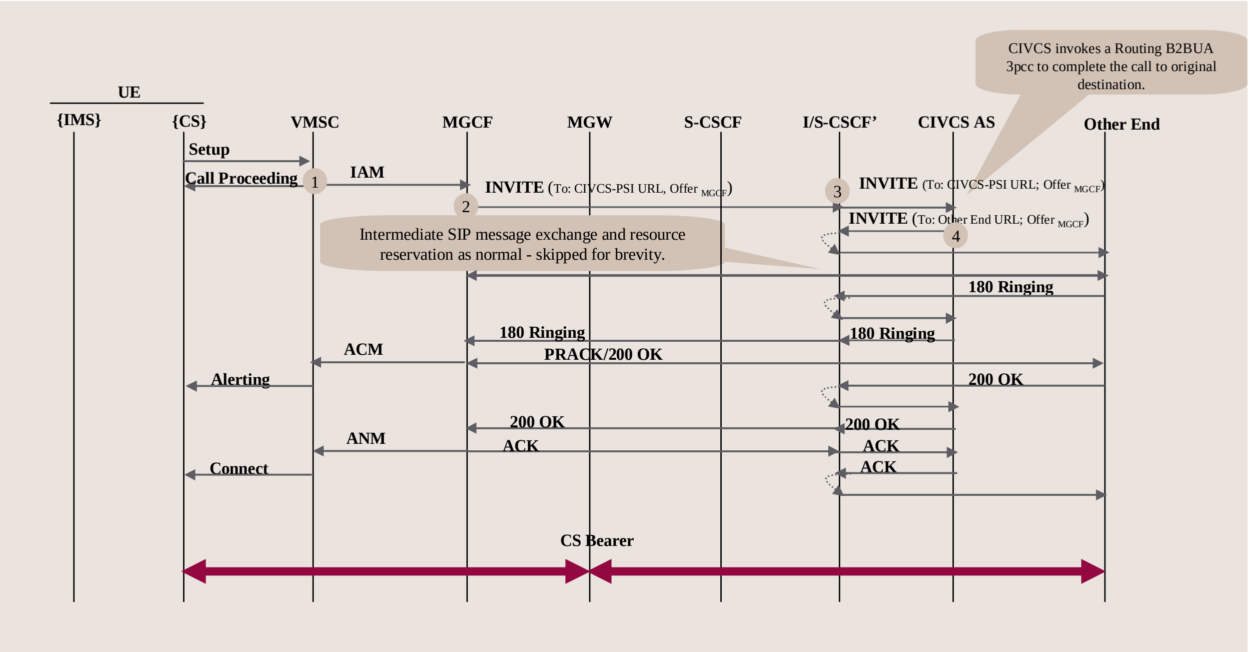


Figure 6.3.7a: CS Origination Anchored at CIVCS walk-through

NOTE: The CS bearer indicates the bearer path for the user when being served in CS Domain, whereas the IMS bearer shows bearer path for the user when being served in IMS.

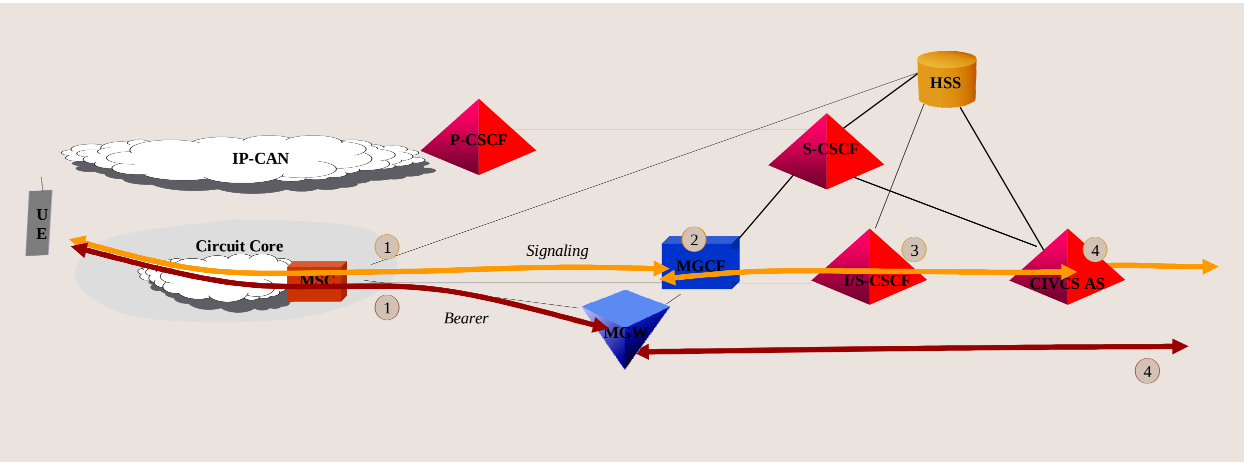


Figure 6.3.7b: CS Origination Anchored at CIVCS walk-through

1. The user originates a call after registering with the MSC. Techniques described previously in this section are used to route the call via the user's home IMS network.

2. The MGCF acts as a user agent on behalf of the CS user initiating an INVITE towards CIVCS. The original called number is passed to CIVCS for routing to the terminating party.

3. The MGCF or the I-CSCF discovers the CIVCS PSI using information received from the originating network. An INVITE addressed to CIVCS PSI is routed to an S-CSCF assigned.

CIVCS completes the call by routing it 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. CIVCS 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.

6.3.5 Termination

6.3.5.1 IMS termination

6.3.5.1.1 Static Anchoring: CIVCS controlled IMS terminating sessions

6.3.5.1.1.1 General

Filter criteria associated with CIVCS are stored in the HSS as part of the CS-IMS user's service subscription profile. The CIVCS filter criteria are downloaded to the currently assigned S-CSCF as part of Initial Filter Criteria at the time of subscriber's registration with IMS network. Initial Filter Criteria are executed at the S-CSCF upon incoming IMS session delivery toward the user that result in routing of the user's session to CIVCS. CIVCS enables a Routing B2BUA 3pcc function to control the bearer path for the session.

In order to maximise network efficiencies with the use of CIVCS, it is recommended that subscriber's current location be used at the S-CSCF or at CIVCS such that it results in invocation of 3pcc function at CIVCS only when the user is being served in the areas with overlapping coverage or with borders between the two domains. IMS Originations and Terminations

6.3.5.1.1.2 IMS terminations walkthrough

Figures 6.3.8a and 6.3.8b describe how signalling and bearer paths are established for IMS terminations from CS-IMS users.

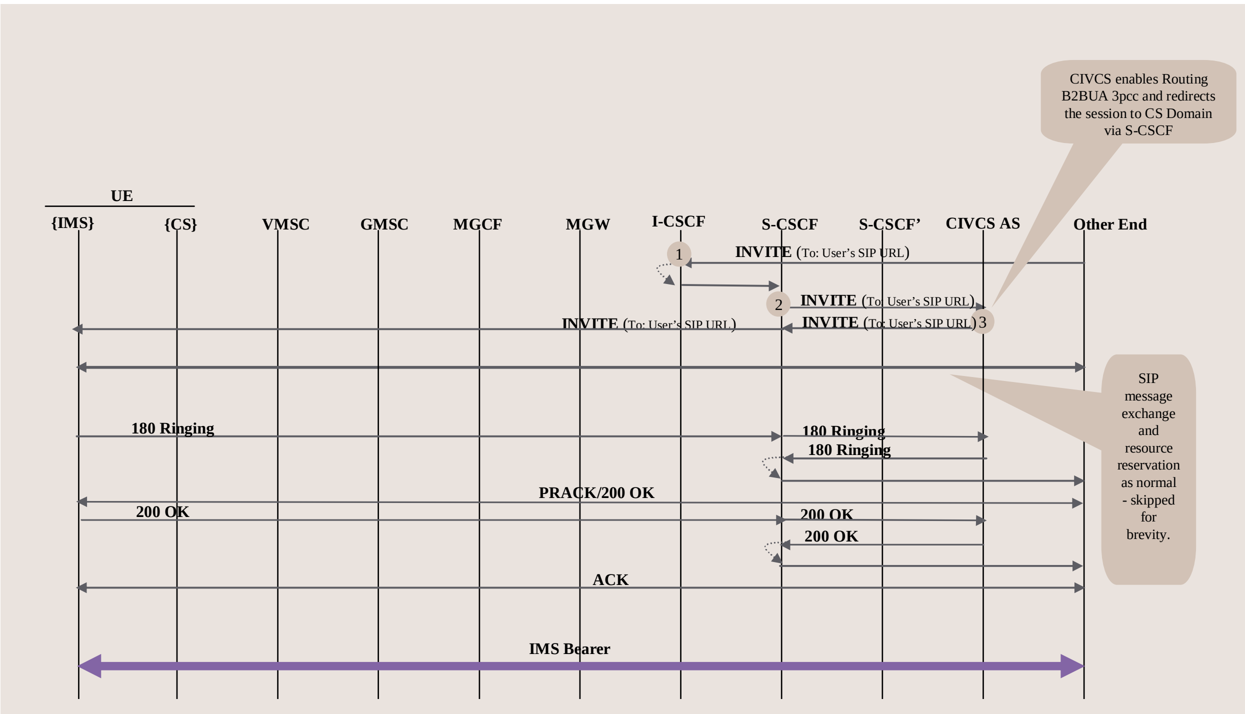


Figure 6.3.8a: IMS Termination controlled at CIVCS walk-through

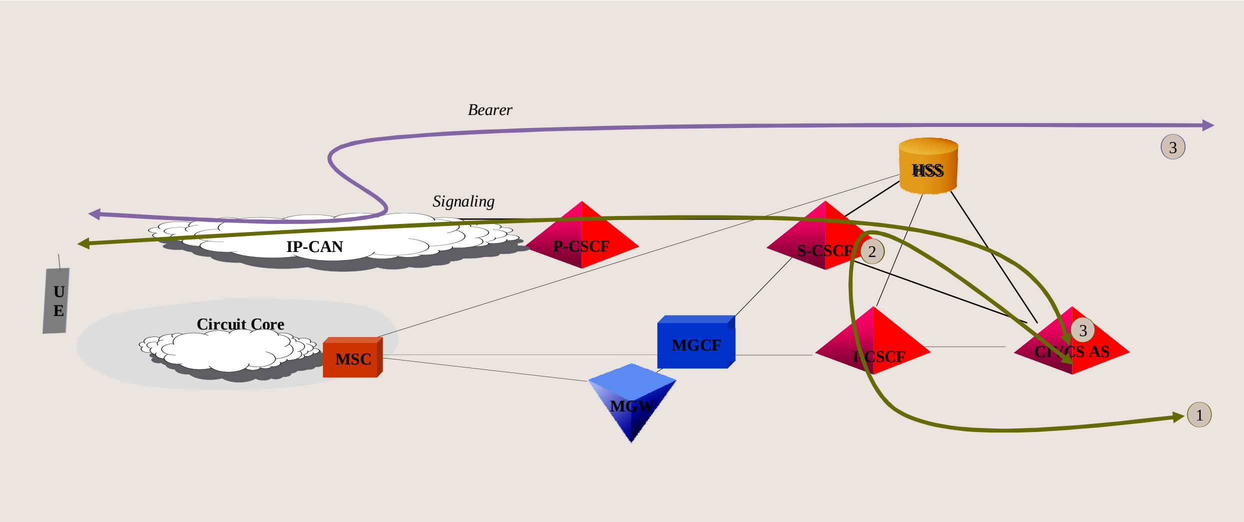


Figure 6.3.8b: IMS Termination controlled at CIVCS walk-through

1) A session originated in IMS is delivered at the user's home I-CSCF function. A call originated in CS Domain and/or PSTN is delivered to the MGCF function in the user's home IMS network. Location Query at I-CSCF results in forwarding of the INVITE to S-CSCF assigned to the user upon IMS Registration. Filter criteria at the S-CSCF

2) Filter criteria at the S-CSCF result in forwarding of the INVITE to CIVCS.

3) CIVCS completes the session establishment by setting up the terminating leg of the session. It enables 3pcc function to maintain session states for control of the originating and terminating leg of the CS-IMS user's session in order to control bearer upon Handover requests from the UE.

6.3.5.2 GSM/UMTS CS termination

6.3.5.2.1 Static Anchoring: Terminating call anchoring at CIVCS for users roaming in CS Domain

6.3.5.2.1.1 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 CIVCS. Incoming calls originated in the CS domain network, PSTN or other IMS networks, which are destined for CS-IMS users can be anchored at CIVCS by setting up routing functions at the originating nodes such that the incoming calls for CS IMS users are delivered to the user's home IMS network. CIVCS 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.

6.3.5.2.1.2 CIVCS Anchored CS Terminations walkthrough

Figures 6.3.9a and 6.3.9b describe how signalling and bearer paths are established for calls terminating to CS-IMS users when roaming in CS Domain. The walk-through assumes that the user is not registered in IMS at the time of incoming call delivery.

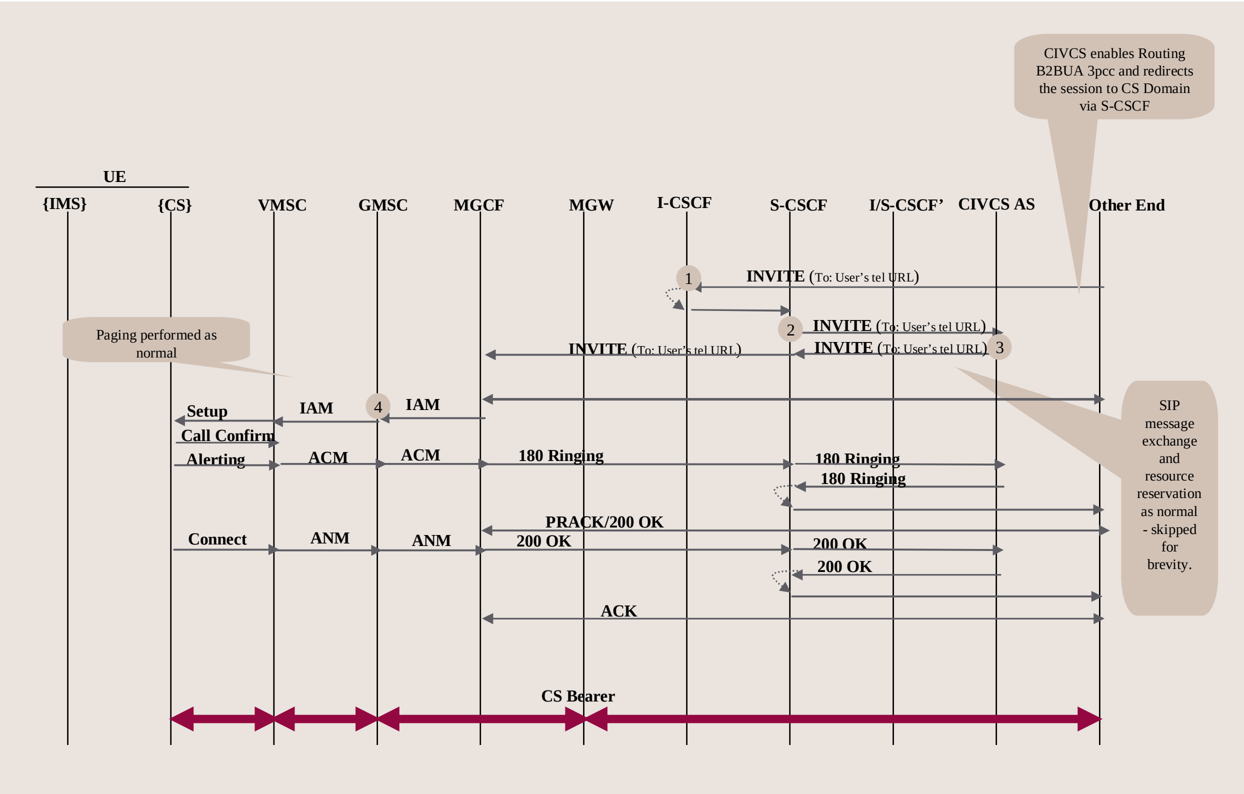


Figure 6.3.9a: CS Termination Anchored at CIVCS walk-through

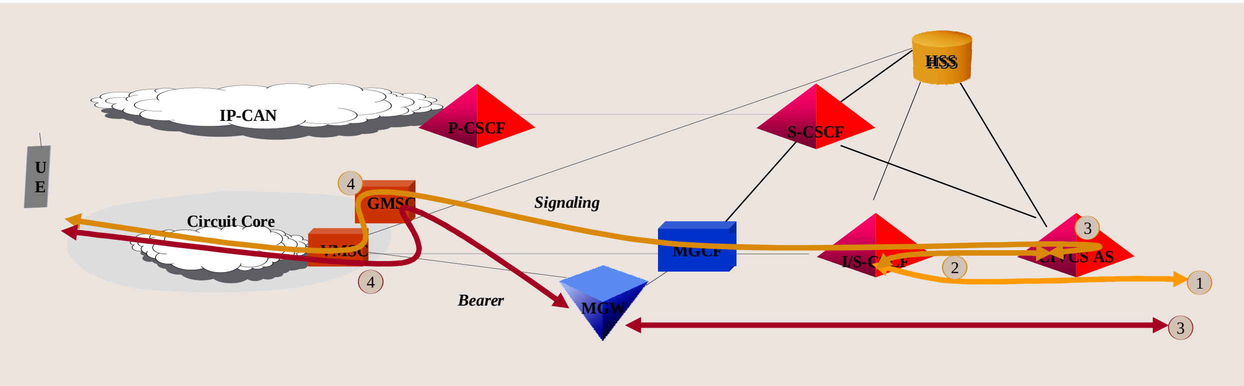


Figure 6.3.9b: CS Termination Anchored at CIVCS walk-through

1. A call originated in IMS is delivered at the user's home I-CSCF function. A call originated in CS Domain and/or PSTN is delivered to the MGCF function in the user's home IMS network. "Mobile Terminating call procedures to unregistered IMS Public User Identity that has services related to unregistered state" as specified by 3GPP TS 23.228, "IP Multimedia Systems, Stage 2" are applied to route the call to a temporary S-CSCF. Procedures for S-CSCF assignment when user is registered in IMS are applied when the user is registered in IMS at the time of incoming call delivery.

2. Filter Criteria at temporary S-CSCF result in forwarding of the INVITE to the CIVCS application server.

3. CIVCS enables a 3pcc function to maintain session states for the originating and terminating legs of the call in order to control bearer upon Handover requests from the UE. "Mobile termination, CS Domain Roaming procedures" as described in 3GPP TS 23.228, "Stage 2 IMS specification" and "Routing Sessions from the IMS to the CS domain"

as described in 3GPP TS 23.221, Architecture requirements, are subsequently applied to route the terminating leg to the GMSC in the CS Domain via an MGCF.

4. GMSC performs normal GMSC procedures to route the call to the user via the Visited MSC that the user is currently registered at.

NOTE: Direct application of 3GPP procedures referenced in this section requires a direct connection of GMSC with the home IMS network which may not always be possible. Use of these procedures will result in a circular routing loop between the PSTN network and the IMS network if a direct connection is not possible. Furthermore, it results in two HSS/HLR dips, one at the I-CSCF in the user's home IMS network and the other at the GMSC in user's home CS network. It is therefore required that "Mobile Terminating call procedures to unregistered IMS Public User Identity that has services related to unregistered state" and "Mobile termination, CS Domain Roaming procedures" as specified by 3GPP TS 23.228, "IP Multimedia Systems, Stage 2", be reviewed to evaluate possible corrections and optimizations of the incoming call delivery for users with subscriptions in both domain which are roaming in CS Domain.

6.3.6 Handover Scenarios

6.3.6.1 CS UE to CS UE call

6.3.6.2 CS UE to IMS UE call

6.3.6.2.1 CS to IMS Handovers

Figures 6.3.10a and 6.3.10b describes how signalling and bearer paths are established for execution of Handover of CS originations from CS Domain to IM Subsystem. 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.

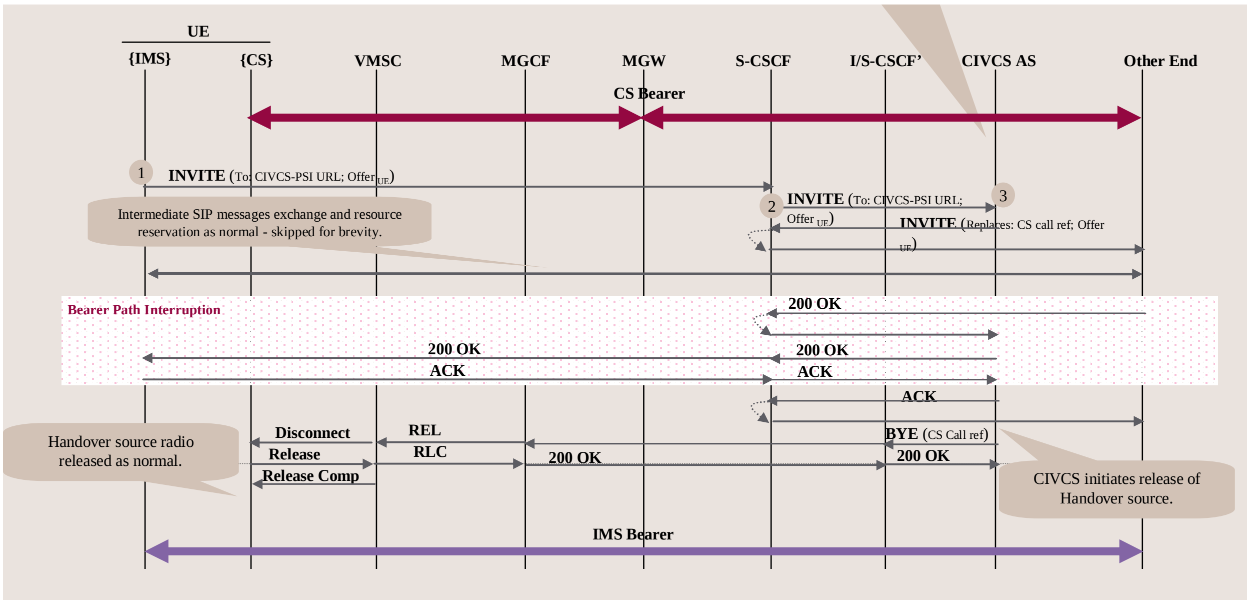


Figure 6.3.10a: CS to IMS Handover walk-through

NOTE: The CS bearer indicates the bearer path for the user when being served in CS Domain, whereas the IMS bearer shows bearer path for the user when being served in IMS.

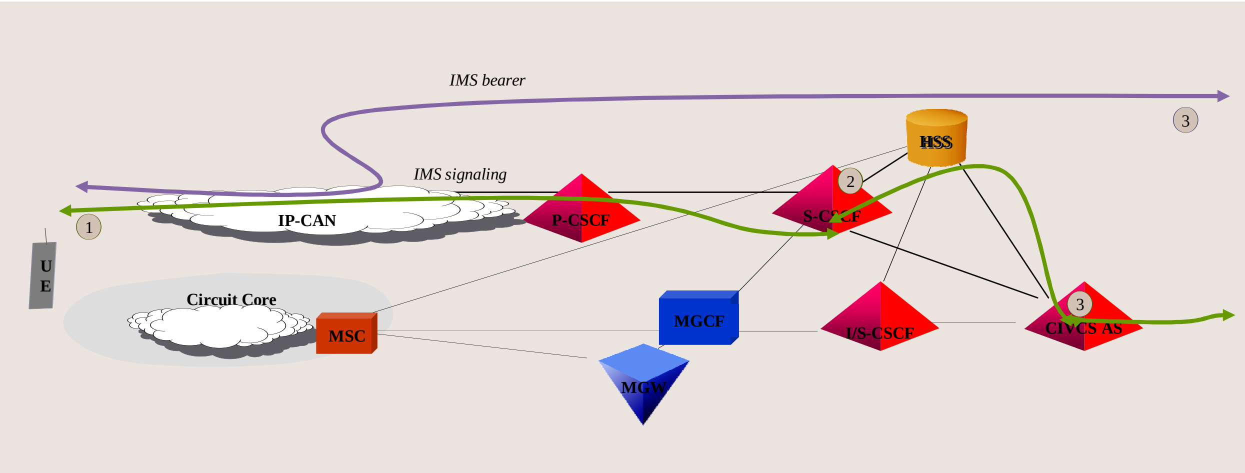


Figure 6.3.10b: CS to IMS Handover walk-through

1. If the user is not registered with IMS at the time when the UE determines a need for Handover to IMS, the UE initiates Registration with IMS. It subsequently sends an INVITE to CIVCS using CIVCS PSI requesting it to perform Handover of the active CS call to IM Subsystem.

2. User's S-CSCF routes the INVITE to CIVCS service instance assigned to the user upon execution of filter criteria.

3. CIVCS 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 INVITE with Replaces header consisting of the SDP of the IMS leg is illustrated here; however, other options such as SIP REFER and UPDATES methods 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. The CS bearer and signalling legs are released upon successful execution of SIP Transfer.

Editor's Note: IETF enhancements to enable SIP Session Transfer with "make before break" sequence which will result in quality of user experience similar to Handovers within GSM/UMTS CS are for further study.

6.3.6.2.2 Subsequent Handback to CS

Figure 6.3.11a and 6.3.11b describes how signalling and bearer paths are established for execution of subsequent Handback to CS Domain.

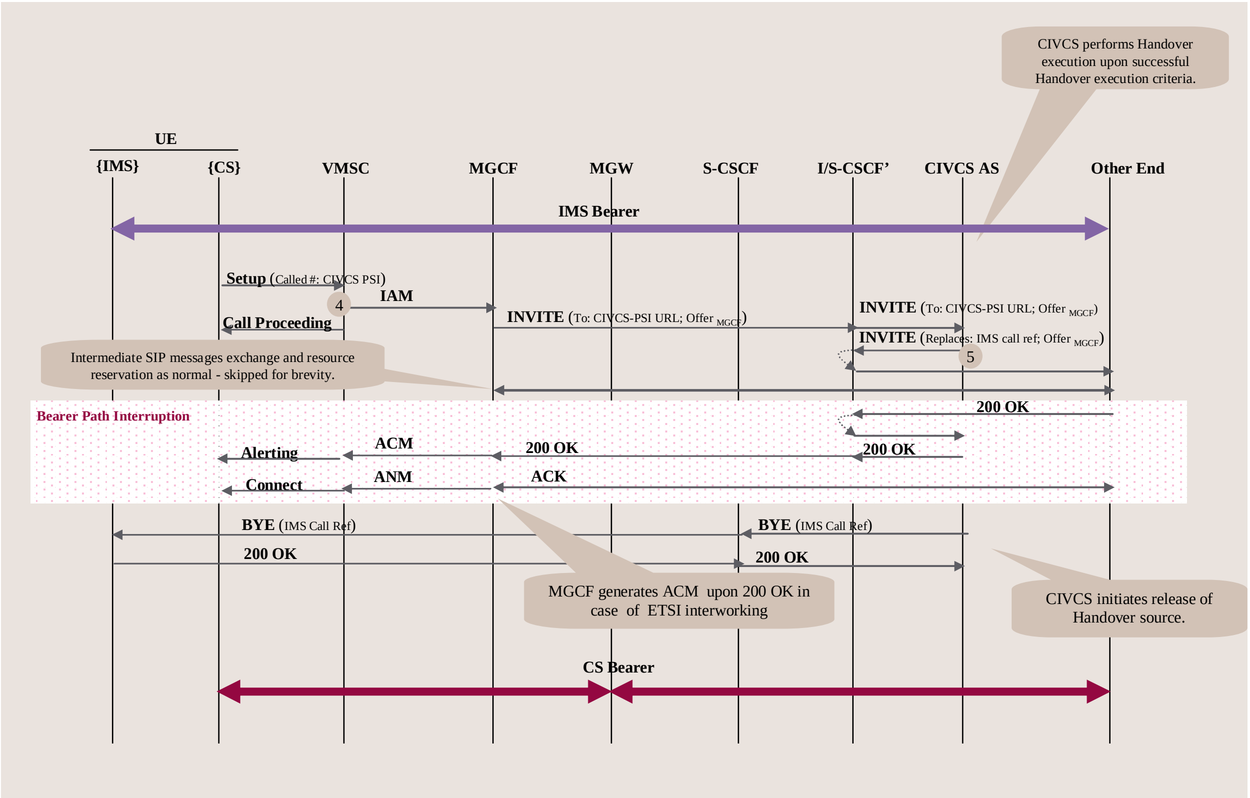


Figure 6.3.11a: Subsequent Handback to CS walk-through

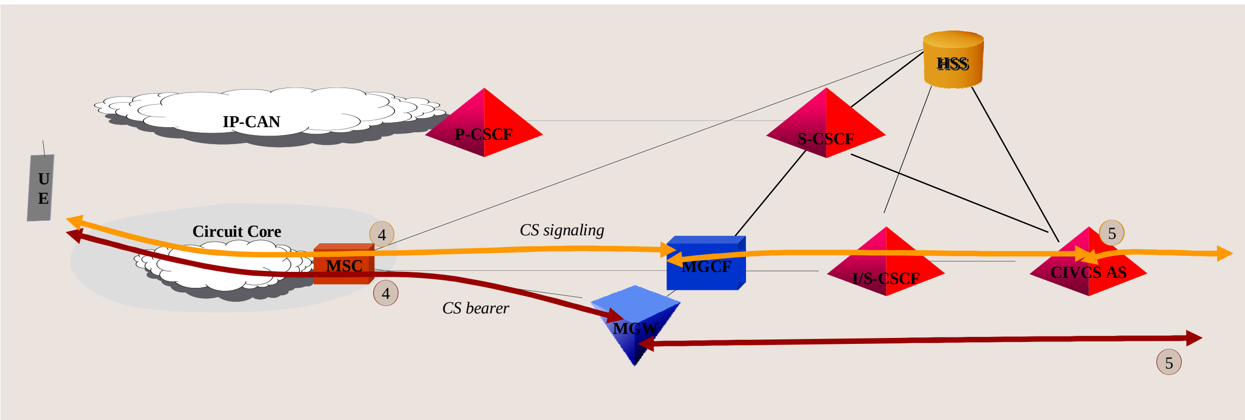


Figure 6.3.11b: Subsequent Handback to CS walk-through

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

5 CIVCS 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 Handover walk-through. The IMS bearer and signalling legs are released upon successful execution of SIP Transfer.

6.3.6.2.3 AS discovery in CS to IMS Handovers

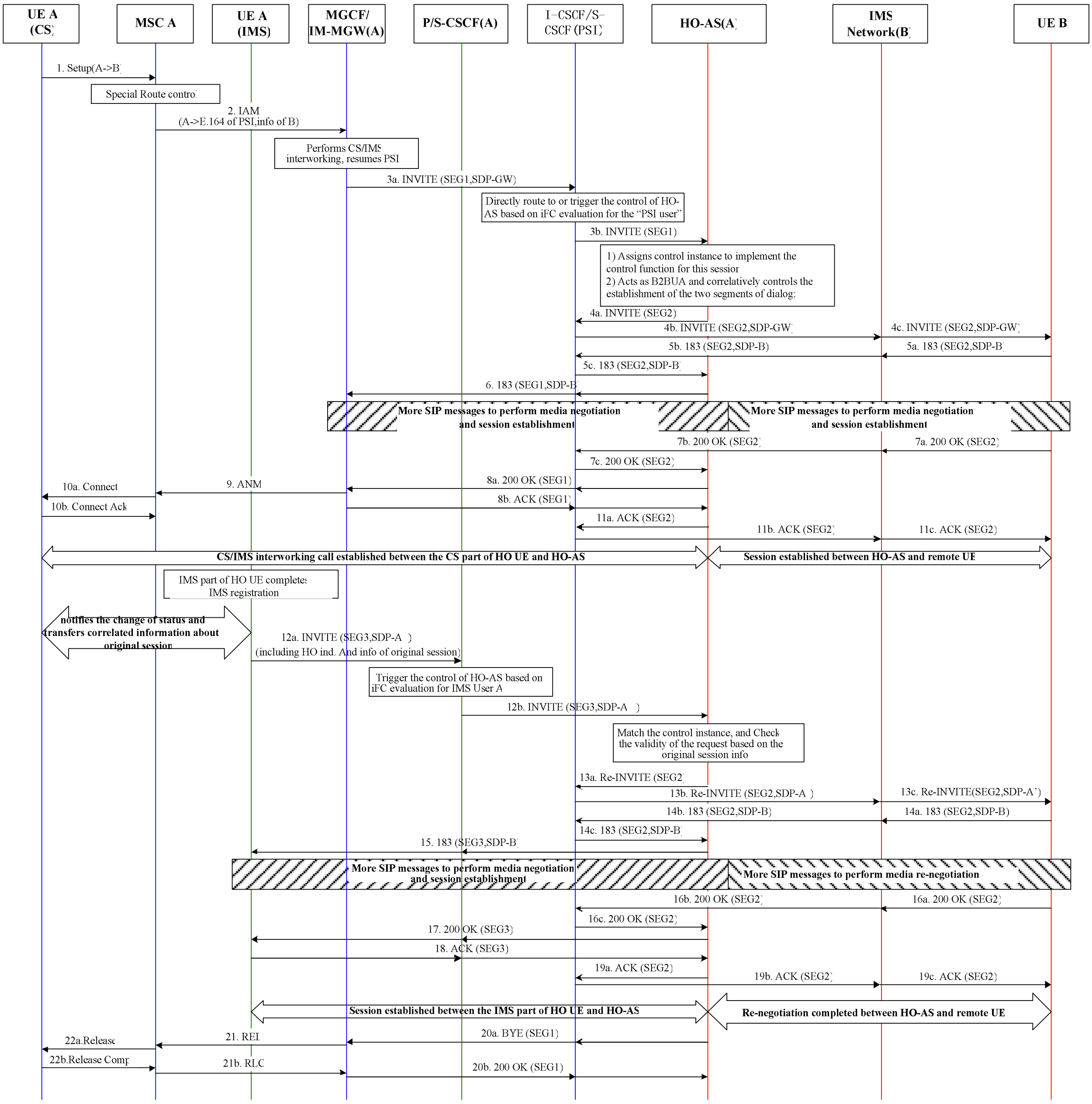


Figure 6.3.12: CS handover to IMS (Calling Party handover for example)

1. When user A uses his dual-mode UE to establish a communication with remote user B from CS domain, the corresponding MSC (MSC-A) shall perform special routing control based on the user's subscription data or based on a special prefix the user dialled. Then the called party number is changed to an E.164 number or with a service prefix corresponding to a HO-PSI, and the originating CS call is routed to a CS-IMS interworking gateway (MGCF/IM-MGW). The IAM message sent to MGCF should also include the original dialled number of user B, how to do it is FFS. (1~2)

2. The MGCF/IM-MGW performs CS-IMS interworking and resume the HO-PSI based on its local configuration or inquiry of ENUM server, then forwards the INVITE message to an AS hosting the HO-PSI according to the standard procedure of "PSIs on the terminating side" described in TS 23.228. (3)

NOTE 1: In case of that there are more than one HO AS in a domain to serve a huge amount of users, more PSIs may be needed to distinguish the different ASs and to ensure the sessions correlating with the same user can be triggered to the same AS.

NOTE 2: There provide two ways to route towards the AS hosting the HO-PSI in TS 23.228, one is directly from I-CSCF to AS based on a HSS query, the other one is from I-CSCF and a S-CSCF assigned for the "PSI user". The former one is recommended here since it is more efficient.

3. HO-AS assigns control instance to implement the control function for this session, acts as B2BUA and correlatively controls the establishment of the two segments of dialog, respectively with the CS part of UE A via the MGCF (dialog 1) and the remote UE (dialog 2); The original dialled number of user B is then used to route the session to the remote UE. These two segments of dialog finally establish the media exchange between IM-MGW and remote UE. (4~11).

4. When the dual-mode UE A decides to initiate a handover, the CS part and IMS part interact with each other to notify the change of status and/or exchange information about the original session. The information about the original session may be Call-ID or a serial number assigned by the HO-AS, and the CS part of the dual-mode UE A may get it during the establishment of original session. How to create it, how to transfer it in SIP/CS signalling is FFS.

5. The IMS part of the UE A then initiates an INVITE to the remote UE or HO PSI to establish a dialog 3, including HO indication and original session information. The so-called HO indication may be an implicit indication such as the appearance of HO-PSI or the original session information, which can be understanded by the HO-AS. The INVITE message is transferred to the S-CSCF assigned for user A during his IMS registration, and then triggered to the same HO-AS based on the user's iFC downloaded from HSS. (12)

NOTE 3: In case of that there are more than one HO AS in a domain and different PSIs are used to distinguish these different ASs, it should keep the configuration in user's iFC consistent with configuration in the user's UEs to ensure the sessions correlating with the same user can be triggered to the same AS.

6. The procedure of signalling and bearer paths establishment for execution of CS to IMS Handover, refer to handover procedure in [1]. Finally, the HO-AS initiates the release of its dialog 1, the CS-IMS interworking call with the CS part of UE A via MGCF/IM-MGW, and completes the procedure of handover from CS to IMS. (13-20)

NOTE 4: It is also possible for the UE A to initiate the release of the original connection after handover, but the process of HO-AS shall not relay on it since the UE may be unable to do so as a result of having lost of access in the handling-out domain.

6.3.6.2.4 CS to IMS Handover using ECT

ECT can be used to enable first CS to IMS Handover and establish the anchor at CIVCS so that subsequent CS to IMS and IMS to CS Handovers are executed via CIVCS. Figure 6.3.13 below provides a walkthrough of first CS to IMS Handover using ECT.

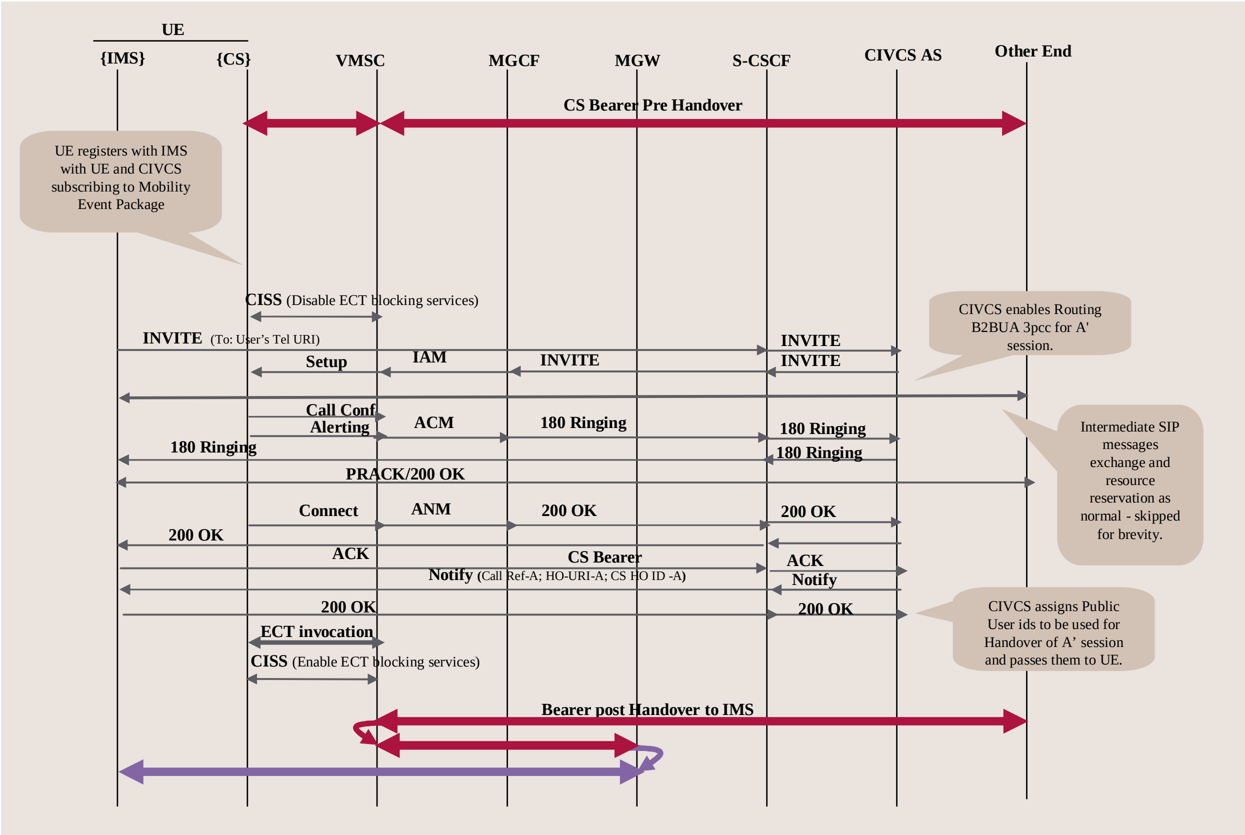


Figure 6.3.13: CIVCS anchoring via ECT

- UE registers with IMS as it detects border conditions requiring handover to IMS. If a CIVCS call anchor reference is available, the UE executes CS to IMS Handover via CIVCS as described in a companion paper (ref [1]). If a CIVCS call anchor reference is not available at the UE, then it enables ECT as a mechanism to execute Handover 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.

- CIVCS inserts a routing B2BUA function of completion of the call toward the CS Domain as a result of filter criteria execution at S-CSCF.

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

The UE re-enables the supplementary services disabled previously upon successfully receiving the incoming CS call.

6.3.6.2.5 First CS to IMS Handover using DACCI

Figure 6.3.14 below provides a walk through of a first CS to IMS transition enabled by DACCI.

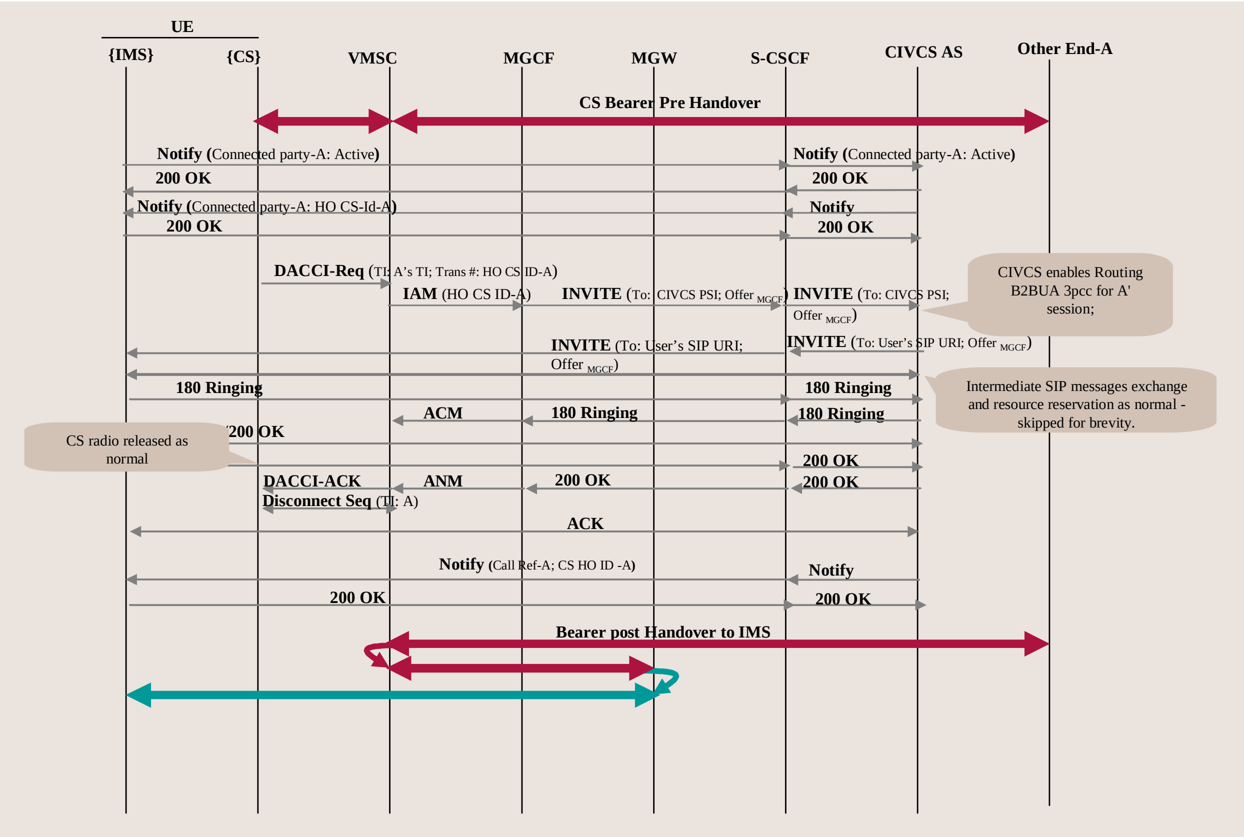


Figure 6.3.14: First CS to IMS Handover using DACCI

- The UE registers with IMS as it detects border conditions requiring handover to IMS. If a CIVCS call anchor reference is available, the UE executes CS to IMS Handover via CIVCS. If a CIVCS call anchor reference is not available at the UE, then it uses DACCI to execute Handover to IMS.
- The UE communicates connected party information with an intent to execute DACCI enabled CS to IMS Handover to CIVCS in a Notify. The connected party information is communicated to CIVCS for use in subsequent Handover 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 Handback to CS. CIVCS assigns a unique Handover CS Identifier to be used for the first CS to IMS transition using DACCI and communicates it to the UE in a Notify.
- UE invokes DACCI at the CS Core Network, providing a routing number derived from the CS HO Identifier to set up the Handover leg towards IMS. It also provides the CC Transaction ID of the session to be handed over.
- The MSC uses the routing number to establish an ISUP circuit connection toward CIVCS 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.
- CIVCS 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 Handback to CS, upon successful setup of the IMS session.
- Upon sending 200 OK in response to the INVITE extended from CIVCS, 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 CIVCS and the bearer is anchored at IM-MGW.

6.3.6.3 IMS UE to IMS UE call

6.3.6.4 IMS UE to CS UE call

6.3.6.4.1 IMS to CS handovers

Figure 6.3.15a and 6.3.15b describes how signalling and bearer paths are established for execution of IMS to CS Handover.

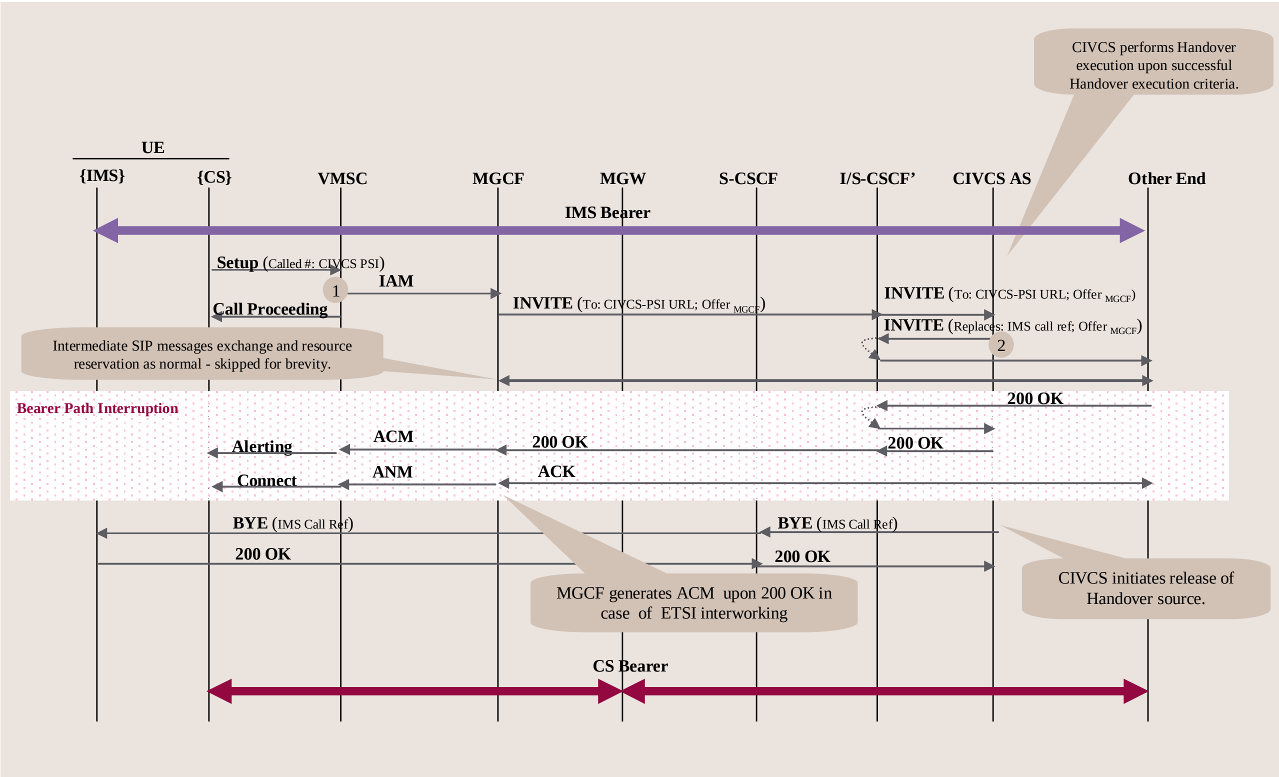


Figure 6.3.15a: IMS to CS Handover walk-through

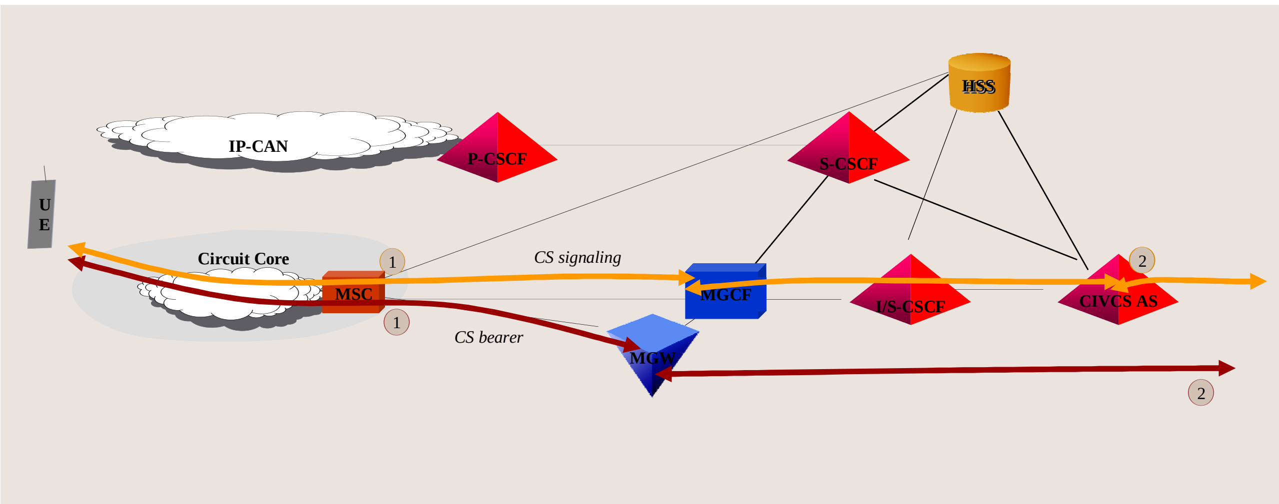


Figure 6.3.15b: IMS to CS Handover walk-through

Note that the Handover procedures executed in steps 1 and 2 are the same as Handover procedures executed in steps 4 and 5 in Figures 6.3.15a and 6.3.15b as the Handover procedure between CS Domain and IMS is agnostic of the previous Handover history.

6.3.6.4.2 AS discovery in IMS to CS Handovers

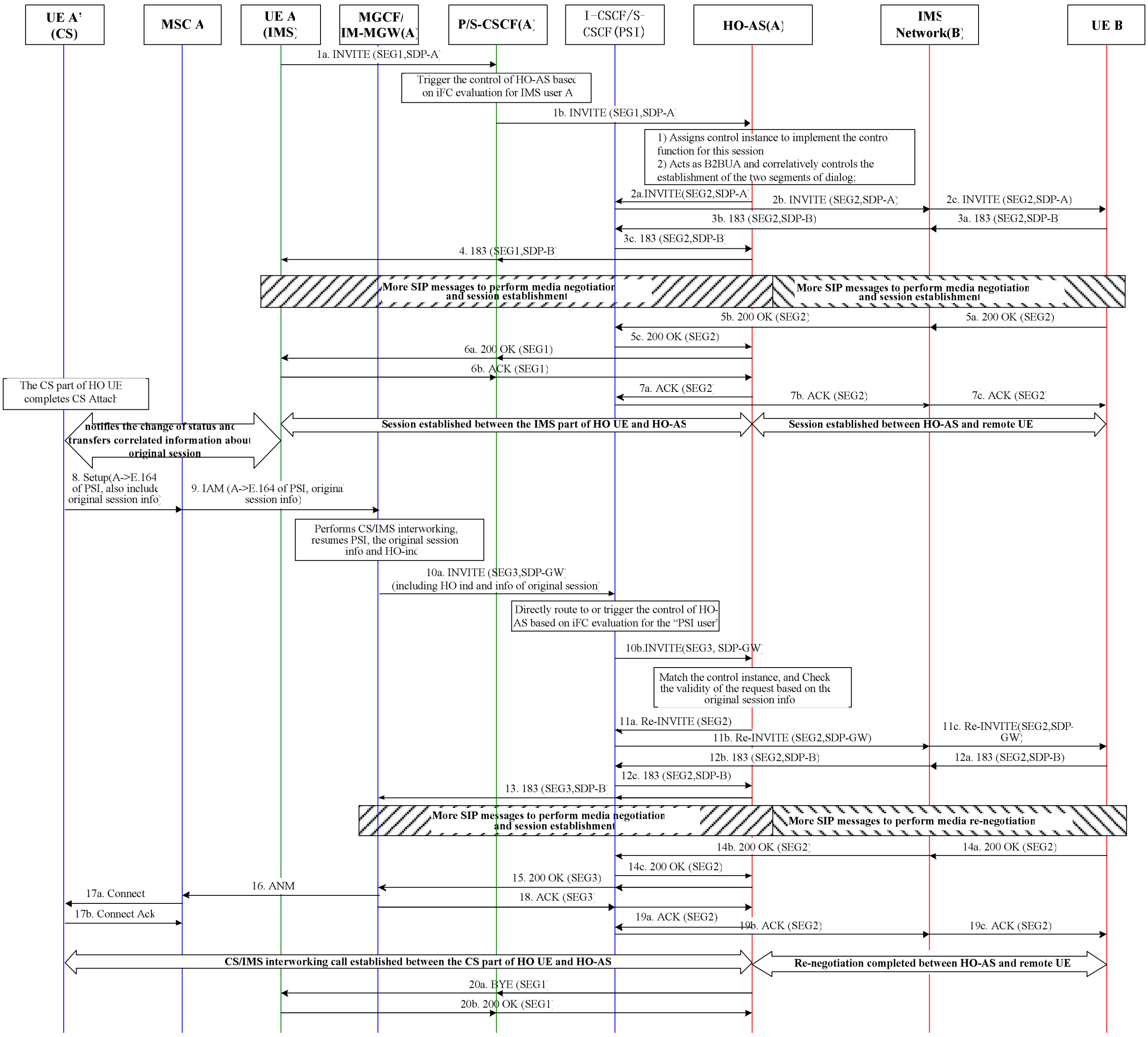


Figure 6.3.16: IMS handover to CS (Calling Party handover for example)

1. When user A uses his dual-mode UE to establish a communication with remote user B from IMS domain, the IMS part of UE A initiates an INVITE to the remote UE. The INVITE message is transferred to the S-CSCF assigned for user A during his IMS registration, and then triggered to a HO-AS based on the user's iFC downloaded from HSS (1) .

2. The HO-AS assigns control instance to implement the control function for this session, acts as B2BUA and correlatively controls the establishment of the two segments of dialog, respectively with the IMS part of UE A (dialog 1) and the remote UE (dialog 2); These two segments of dialog finally establish the media exchange between the IMS part of UE A and remote UE. (4~7)

3. When the dual-mode UE A decides to initiate a handover, the CS part and IMS part interact with each other to notify the change of status and/or exchange information about the original session.

4. The CS part of the UE A then initiates a setup request to the corresponding MSC (MSC-A), while the called party number is the E.164 number corresponding to the HO-PSI, and the corresponding MSC (MSC-A) then

routes the CS call to a CS-IMS interworking gateway (MGCF/IM-MGW) based on the analysis of the E.164 number corresponding to the HO-PSI. The Setup and IAM message should include the original session information, which will be used to find the original control instance in the HO-AS. The information about the original session may be Call-ID or a serial number assigned by the HO-AS, and the IMS part of the dual-mode UE A may get it during the establishment of original session. How to create it, how to transfer it in SIP/CS signalling is FFS. (8~9)

5. The MGCF/IM-MGW performs CS-IMS interworking, resumes the HO-PSI based on its local configuration and/or inquiry of ENUM server, the original session information and HO-indication, then forwards the INVITE message to the HO-AS hosting the HO-PSI to establish a dialog 3, the routing of AS hosting a PSI is performed according to the standard procedure of "PSIs on the terminating side" described in TS 23.228. The so-called HO indication may be an implicit indication such as the appearance of HO-PSI or the original session information, which can be understanded by the HO-AS. (10)

NOTE 1: In case of that there are more than one HO AS in a domain to serve a huge amount of users, more PSIs may be needed to distinguish the different ASs, and should keep the configuration in user's iFC consistent with configuration in the user's UEs to ensure the sessions correlating with the same user can be triggered to the same AS.

NOTE2: There provide two ways to route towards the AS hosting the HO-PSI in TS 23.228, one is directly from I-CSCF to AS based on a HSS query, the other one is from I-CSCF and a S-CSCF assigned for the "PSI user". The former one is recommended here since it is more efficient.

NOTE 3: Another choice is that, the CS part of UE A initiates a Setup request while the called party number is its own MSISDN with a special prefix, used to indicate that the call should be routed to the IMS domain. IMS domain entity then transforms this number to the UE A's IMPU and send the INVITE to the S-CSCF assigned for user A, then trigger to the HO-AS based on iFC evaluation for user A. This will ensure the INVITE of dialog 3 be sent to the same S-CSCF which control the original session.

6. The procedure of signalling and bearer paths establishment for execution of IMS to CS Handover, refer to handover procedure in [1]. Finally, the HO-AS initiates the release of its dialog 1 with the IMS part of UE A, and completes the procedure of handover from IMS to CS (11-20)

NOTE 4: It is also possible for the UE A to initiate the release of the original connection after handover, but the process of HO-AS shall not relay on it since the UE may be unable to do so as a result of having lost of access in the handling-out domain.

6.3.6.4.3 Subsequent Handovers and Handbacks with DACCI enabled anchoring

Once a session anchor has been established at CIVCS and the session identifiers have been communicated to the UE, the UE executes all Inter domain Handovers via CIVCS using Handover procedures described elsewhere. Subsequent Handovers 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 transition as shown in Figure 6.3.17 below.

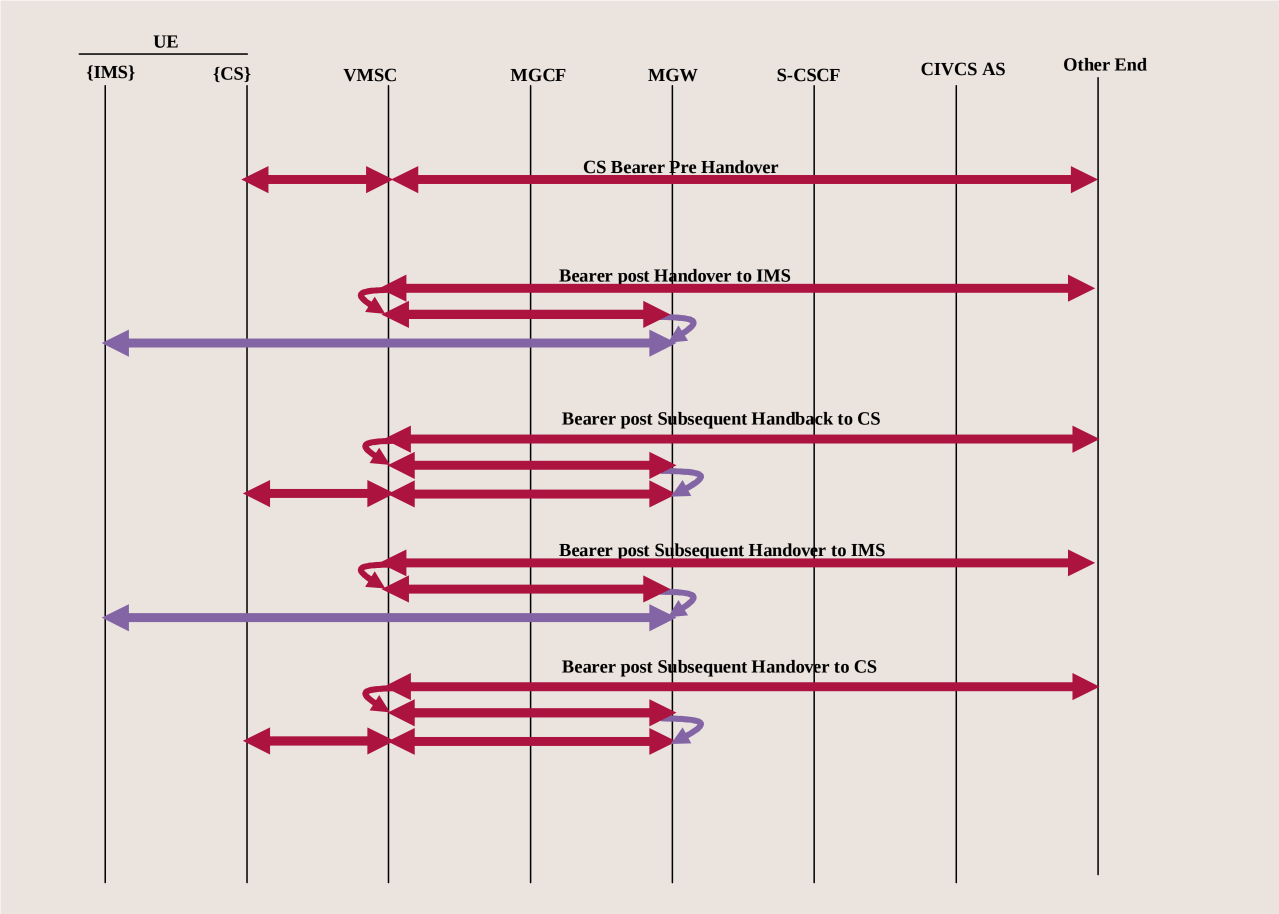


Figure 6.3.17: Use of CIVCS for subsequent Handovers

6.3.6.4.4 Subsequent Handovers and Handbacks

Use of ECT for subsequent Handovers and Handbacks results in a daisy chain effect as shown in Figure 6.3.18 below, due to the fact that the call anchor moves between CS and IMS upon each Handover.

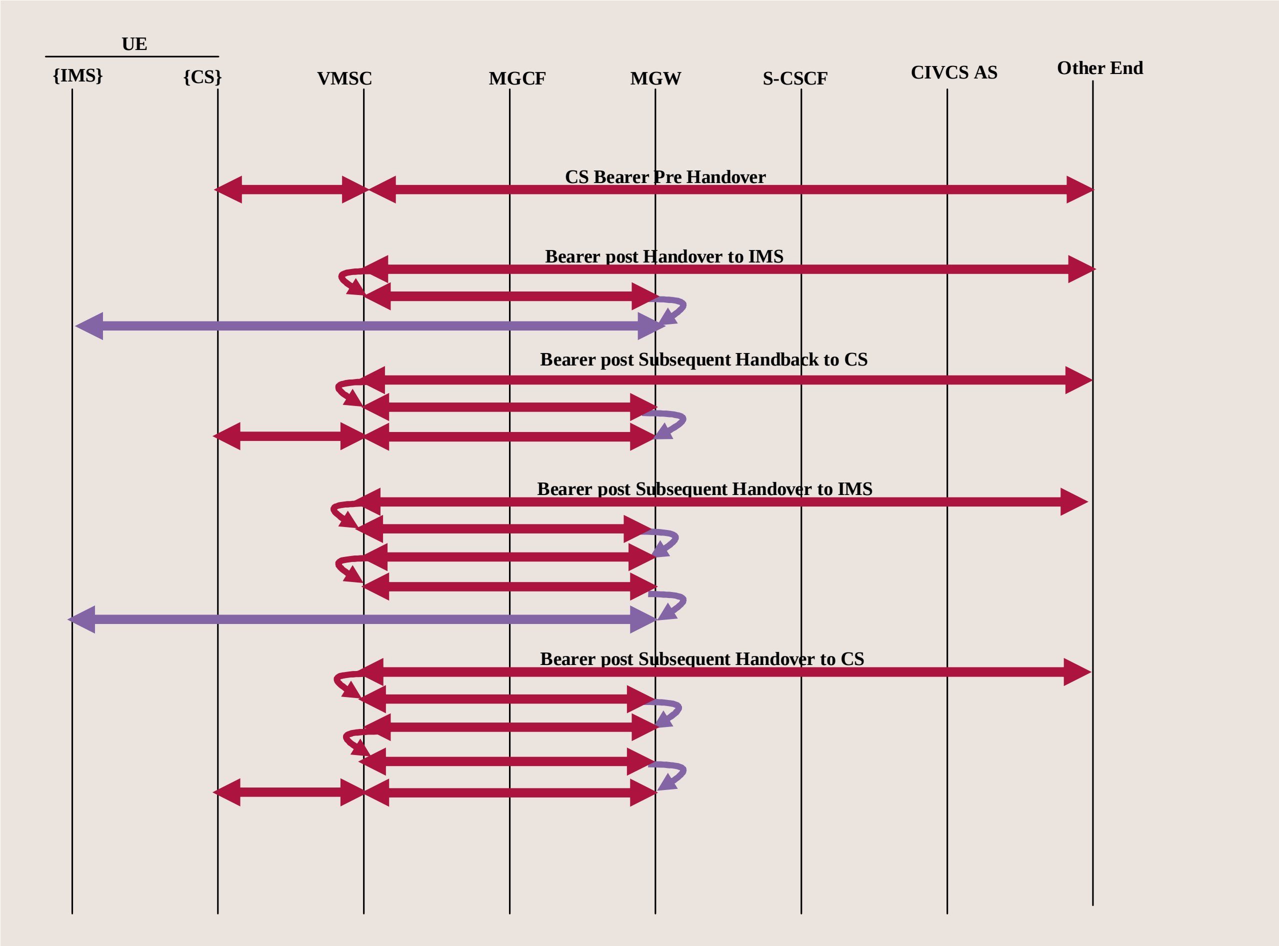


Figure 6.3.18: Undesirable Resource Daisy Chain with use of ECT for Subsequent Handovers

Once a session anchor has been established at CIVCS and the session identifiers have been communicated to the UE, the UE executes all Inter domain Handovers via CIVCS. This helps eliminate the daisy chain effect as shown in Figure 6.3.19 below, because the call remains anchored in the IMS MGW after the first CS to IMS Handover.

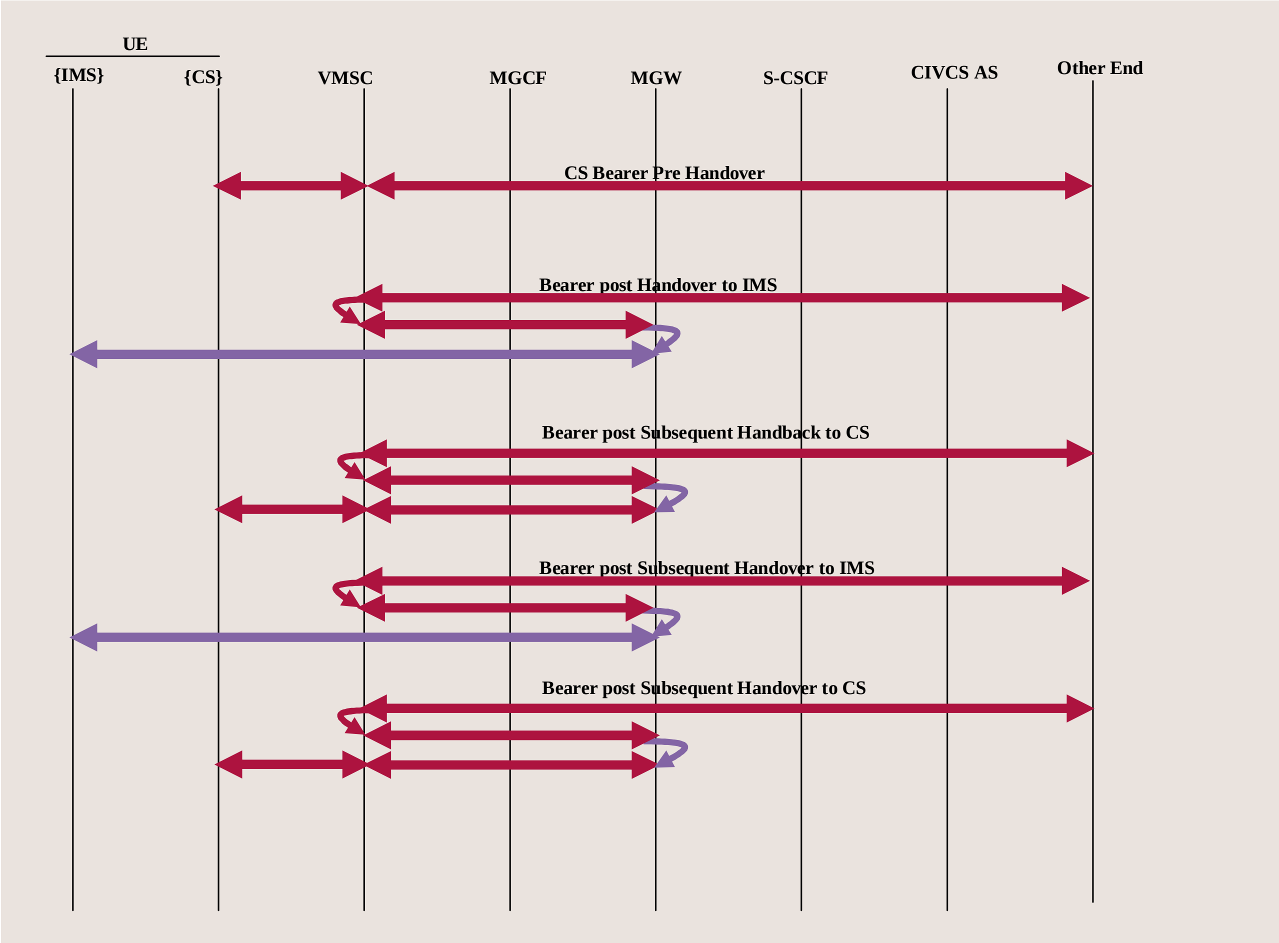


Figure 6.3.19: Resource optimization with use of CIVCS for subsequent Handovers

6.3.7 Impact on Supplementary Services

6.3.7.1 Voice Call Continuity for Multi-Session calls

CIVCS uses the Mobility Event package to communicate with the UE, session specific information that is required to perform handover 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 CIVCS via 3pcc Routing B2BUA function using static anchoring or dynamic anchoring techniques. Upon successful allocation of a B2BUA function for a particular user session, CIVCS assigns it a unique identifier along with and a SIP URI that is used to uniquely identify the session when requesting its handover to IMS or a unique CS Handover ID that can be used to uniquely identify the session when requesting its handover 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 CIVCS. Connected party address is used to enable Handovers 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 Handover in these conditions.

The fundamental principle of transferring the call control protocol state machine from the handing-out domain to the handing-in domain discussed elsewhere for single sessions 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 Multi-Session services support with Static Anchoring

6.3.7.1.1.1 Anchoring of IMS Held and Active sessions at CIVCS

Figures 6.3.20 and 6.3.21 below provides a walkthrough of a scenario in which CS IMS 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.

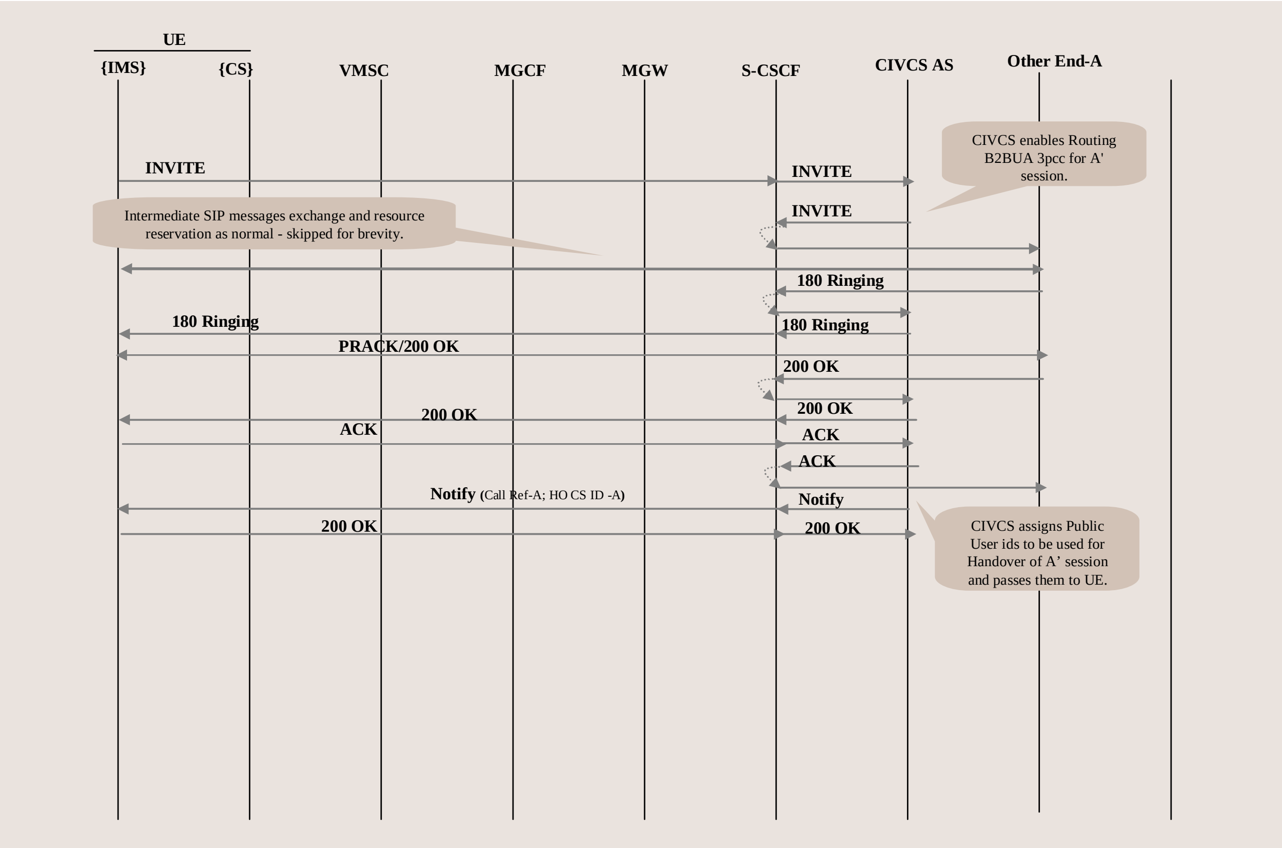


Figure 6.3.20: IMS session toward other end – A

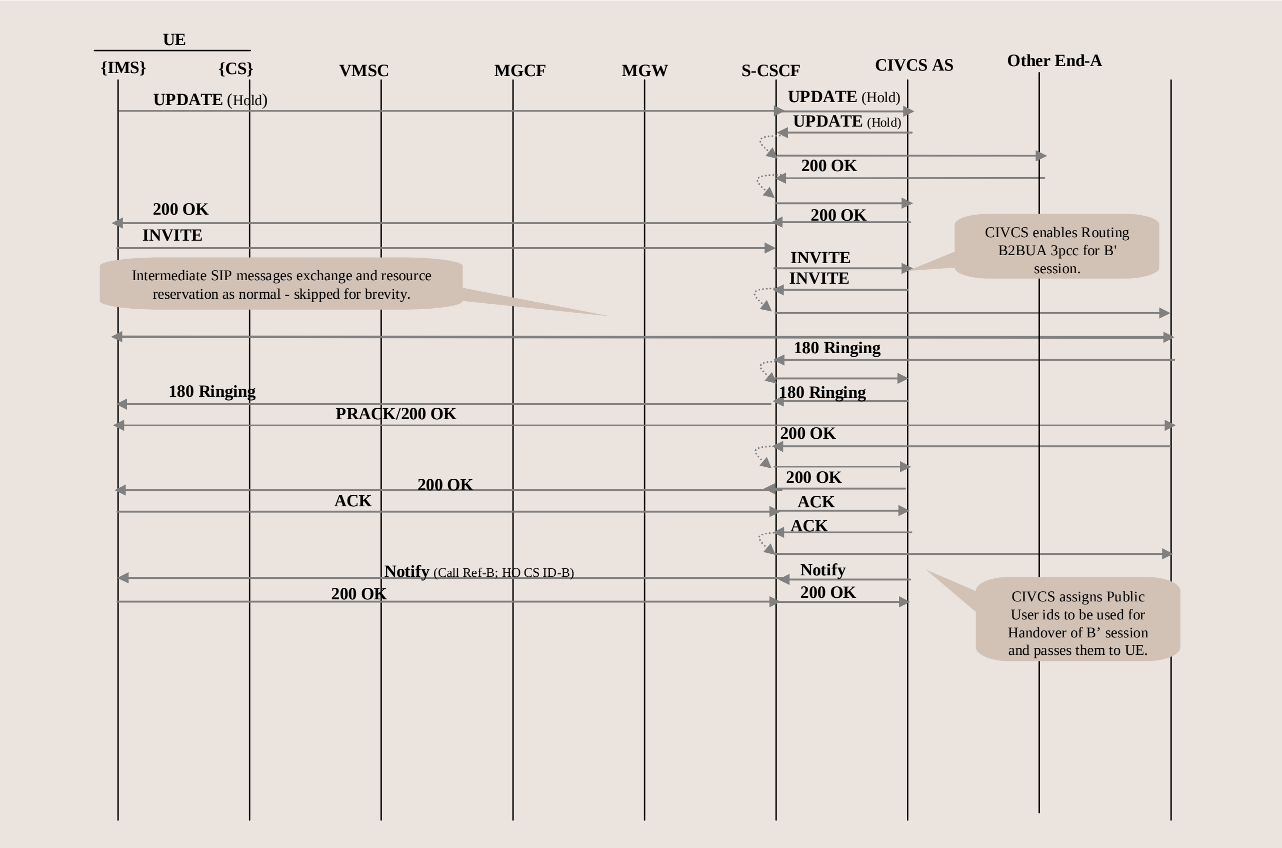


Figure 6.3.21: Hold A and originate an IMS session toward other end - B

- As part of IMS registration, CIVCS and the UE subscribe to each other for the Mobility Event package.
- Upon successful execution of a Routing B2BUA function for IMS session to the other end A at CIVCS, CIVCS assigns a unique call reference identifier to the session for identification of the session between the UE and CIVCS in subsequent dialogues. It also assigns a unique identifier which can be used for Handover of this session to CS Domain. The CS Handover 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 CIVCS DN when requesting Handover, the later 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.
- The CS IMS puts session toward the other end A on hold and originates a new session towards the other end B.
- Upon successful execution of a Routing B2BUA function for IMS session to the other end B at CIVCS, CIVCS assigns a unique call reference identifier to the session for identification of the session between the UE and CIVCS in subsequent dialogue. It also assigns a unique identifier which can be used for Handover of this session to CS Domain. These new identifiers are communicated to the UE via a Notify with Mobility Event package.

6.3.7.1.1.2 IMS to CS Handover of IMS Held and Active sessions

Figure 6.3.22 below provides a walkthrough of Handover of IMS Held and Active sessions established as described in previous section.

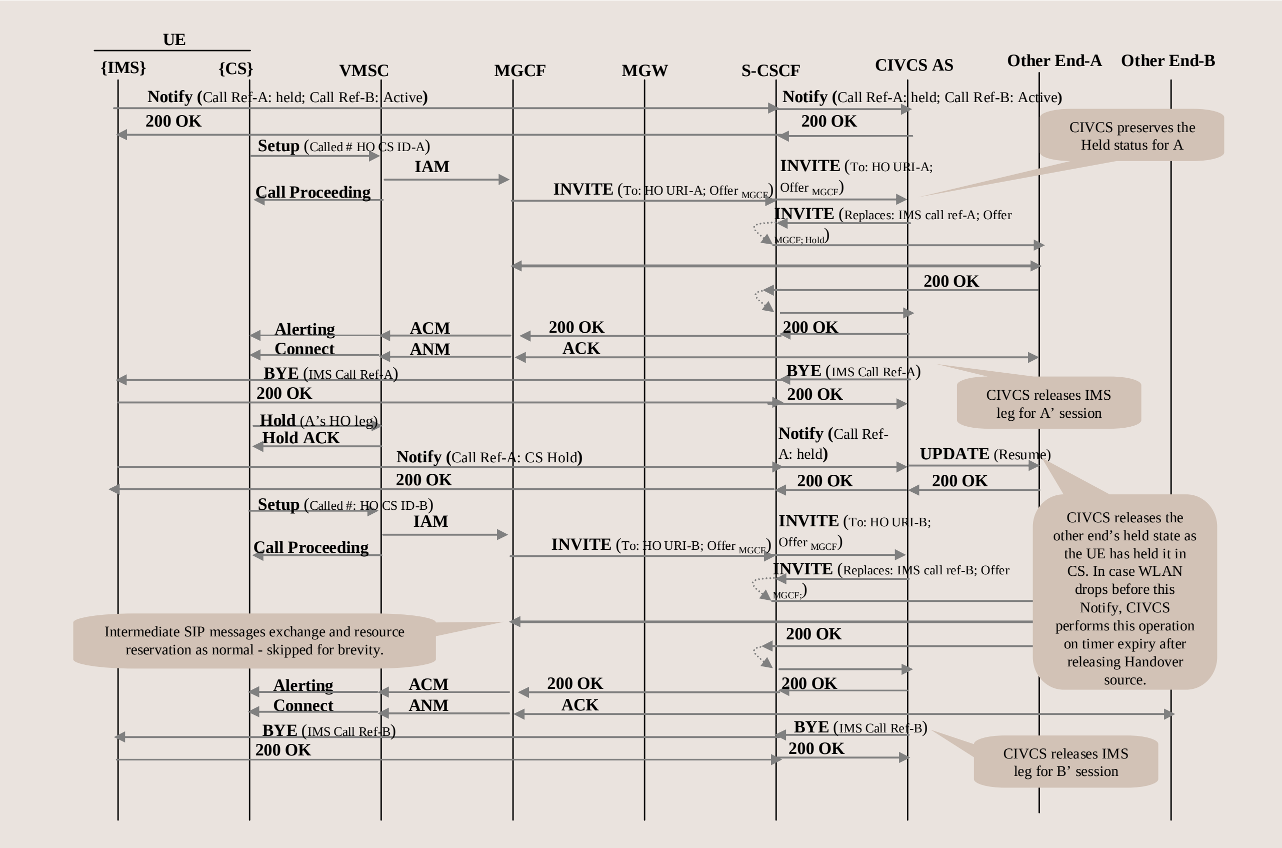


Figure 6.3.22: Handover of IMS Held and Active sessions

- Upon detection of border conditions, the UE updates CIVCS with the current session state information to be used during the Handover procedure. Session Hold, Active states are passed to CIVCS 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 CIVCS.

- It should be noted that this message exchange can be avoided if CIVCS 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.

- The UE performs handover of the held IMS leg to CS using IMS to CS Handover procedures described elsewhere. CIVCS ensures that the held status is maintained at the other end A when transferring the CS IMS user from IMS to CS.

- The UE holds the CS Handover leg for A's session at the MSC to re-establish the protocol state machine at the MSC. The UE sends a Notify to CIVCS 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 Handover procedures. CIVCS 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 handover of the Active session to the other B to CS using IMS to CS Handover procedures described elsewhere.

- 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 handing-in domain.

6.3.7.1.1.3 Anchoring of CS Held and Active sessions at CIVCS with active IMS Registration

Figure 6.3.23 below provides a walkthrough of a scenario in which CS IMS 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 has active IMS Registration at the time it establishes the CS calls for this walkthrough.

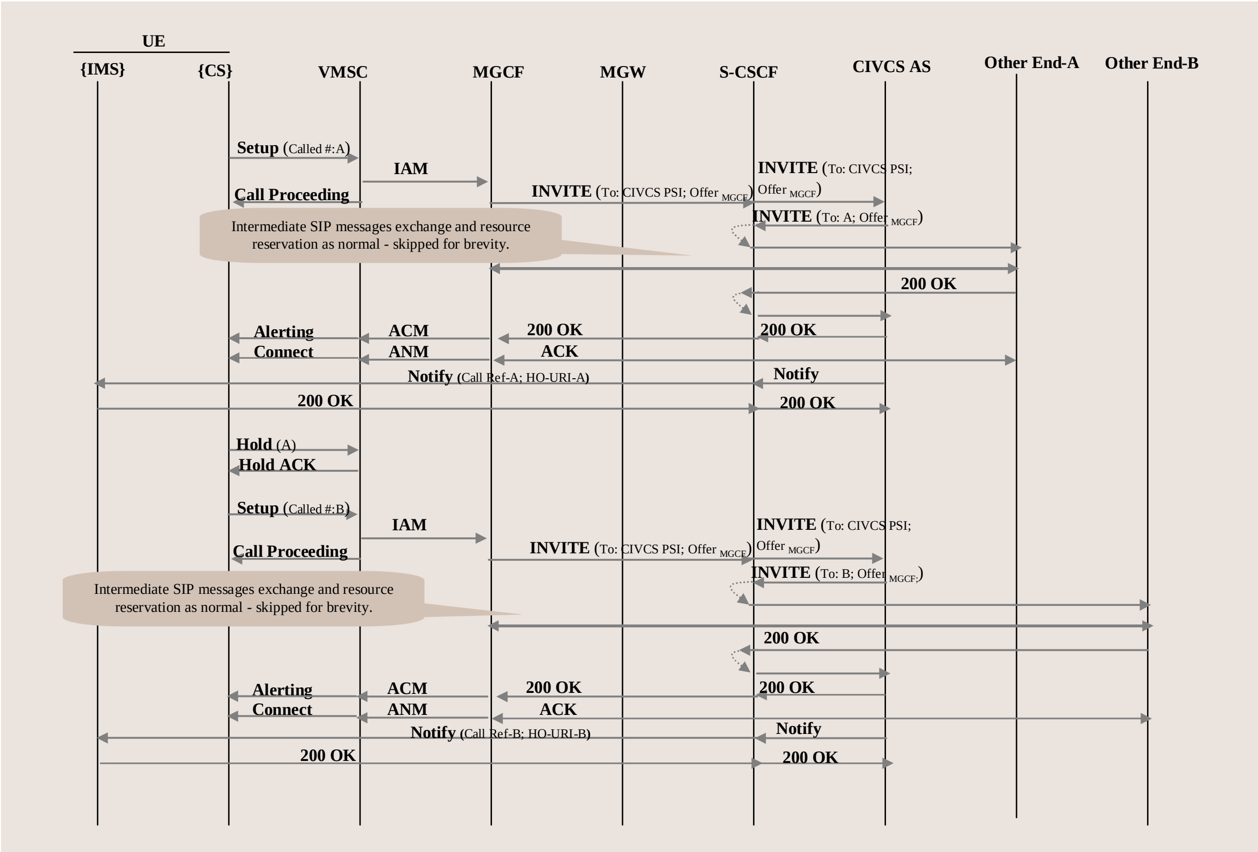


Figure 6.3.23: Anchoring of CS Held and Active sessions at CIVCS

- As part of IMS registration, CIVCS and the UE subscribe to each other for the Mobility Event package.
- Upon successful execution of a Routing B2BUA function for CS session to the other end A at CIVCS, CIVCS assigns a unique call reference identifier to the session for identification of the session between the UE and CIVCS in subsequent dialogue. It also assigns a SIP URI which can be used for Handover of this session to IMS. These new identifiers are communicated to the UE via a Notify with Mobility Event package.
- The CS IMS puts session toward the other end A on hold and originates a new CS session towards the other end B.
- Upon successful execution of a Routing B2BUA function for session to the other end B at CIVCS, CIVCS assigns a unique call reference identifier to the session for identification of the session between the UE and CIVCS in subsequent dialogue. It also assigns a SIP URI which can be used for Handover of this session to CS Domain. These new identifiers are communicated to the UE via a Notify with Mobility Event package.

6.3.7.1.1.4 CS to IMS Handover of CS Held and Active sessions; IMS active at the time of CS Anchoring

Figure 6.3.24 below provides a walkthrough of Handover of CS Held and Active sessions established as described in previous section.

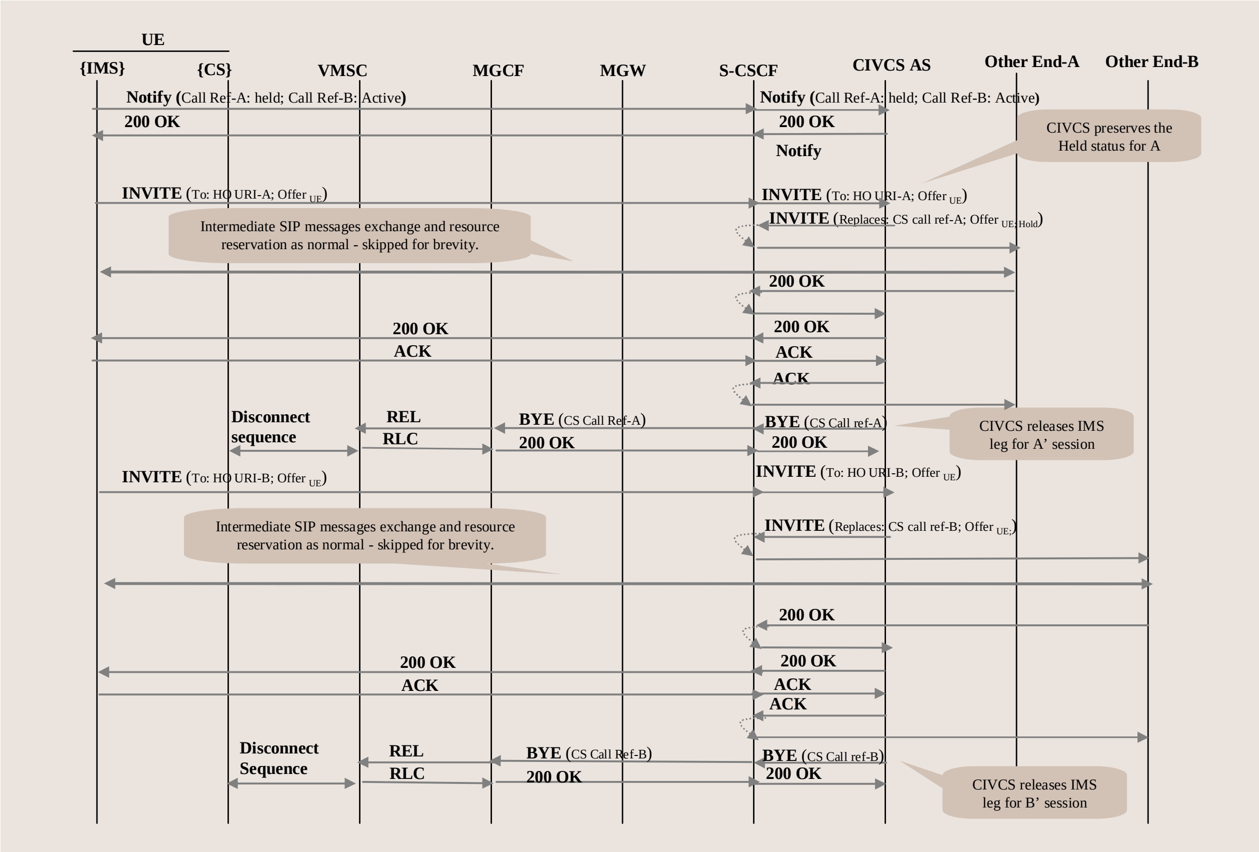


Figure 6.3.24: CS to IMS Handover of Held and Active sessions; IMS active at CS anchoring

- Upon detection of border conditions, the UE updates CIVCS with the current session state information to be used during the Handover procedure. Session Hold, Active states are passed to CIVCS 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 CIVCS.

- The UE performs handover of the held CS leg to IMS using CS to IMS Handover procedures described elsewhere. CIVCS ensures that the held status is maintained at the other end A when transferring the CS IMS user from IMS to CS.

- The UE subsequently performs handover of the Active session to the other party B to IMS using CS to IMS Handover procedures described elsewhere.

- 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 handing-in domain.

6.3.7.1.1.5 Anchoring of CS Held and Active sessions at CIVCS without IMS Registration

Figure 6.3.25 below provides a walkthrough of a scenario in which CS IMS 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.

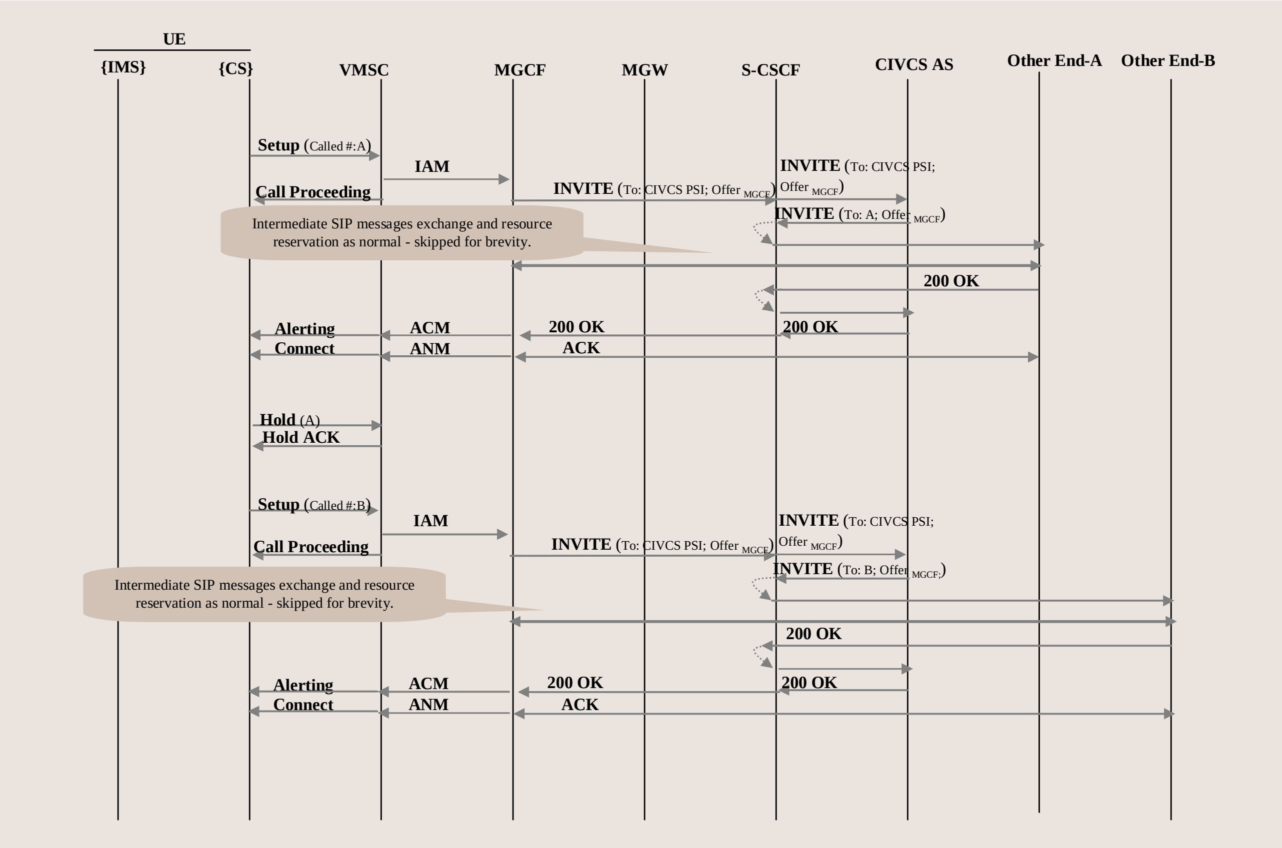


Figure 6.3.25: Anchoring of CS Held and Active sessions at CIVCS

- Since the user is not registered in IMS, exchange of session identifier is not possible with the UE. However, CIVCS 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.

6.3.7.1.1.6 CS to IMS Handover of CS Held and Active sessions; IMS not active at the time of CS Anchoring

Figure 6.3.26 below provides a walkthrough of Handover of CS Held and Active sessions established as described in previous section.

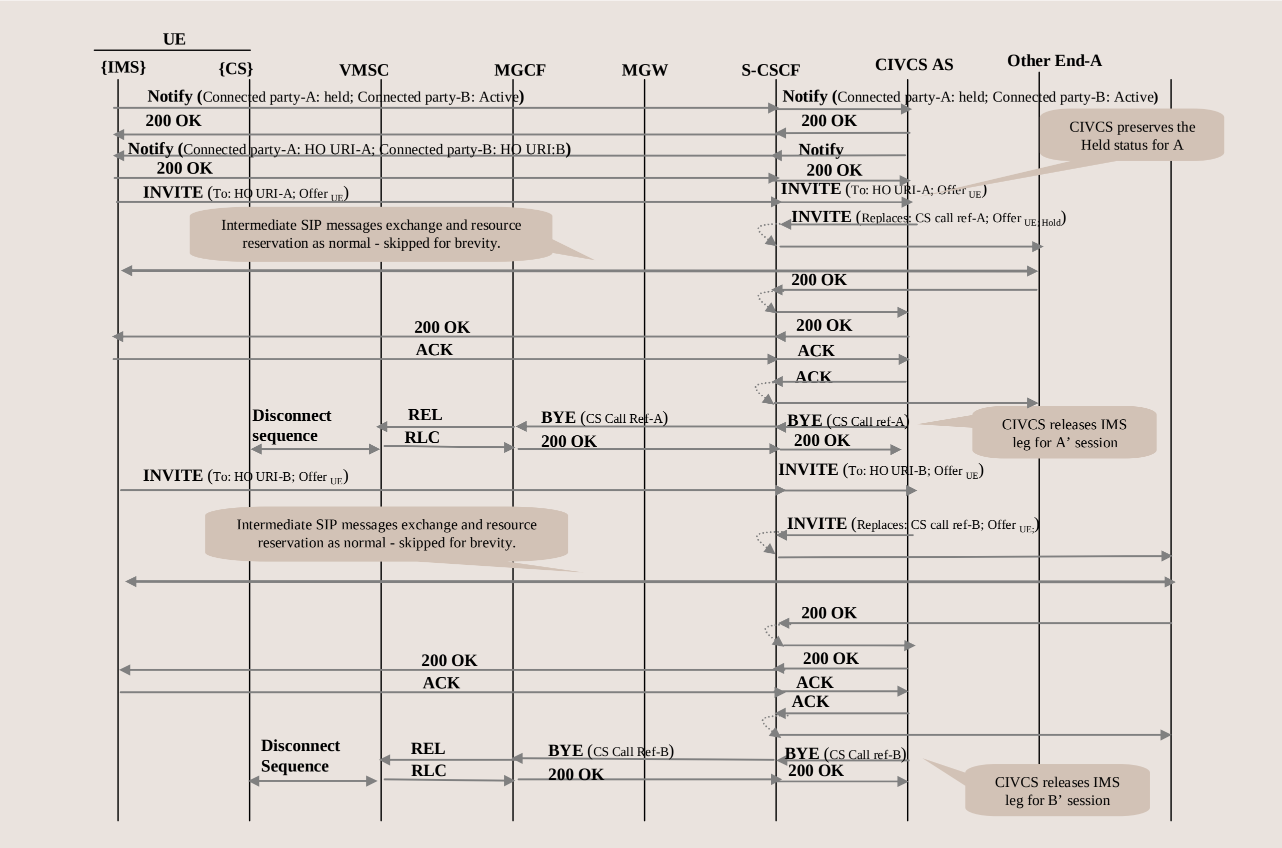


Figure 6.3.26: CS to IMS Handover of Held and Active sessions; IMS not active at CS anchoring

- Upon detection of border conditions, the UE performs IMS Registration and updates CIVCS with the current session state information to be used during the Handover procedure. Session Hold, Active states are passed to CIVCS 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 CIVCS.

- CIVCS communicates SIP URIs to be used for Handover 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 Handover of Held/Active sessions with IMS Registration.

6.3.7.1.2 Multi-Session services support with DACCI

6.3.7.1.2.1 General

DACCI uses techniques similar to the techniques described in a companion paper (ref [4]) to transfer a CS call with multiple sessions from CS domain to IMS.

6.3.7.1.2.2 Handover of Multi-Session services

Figure 6.3.27 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.

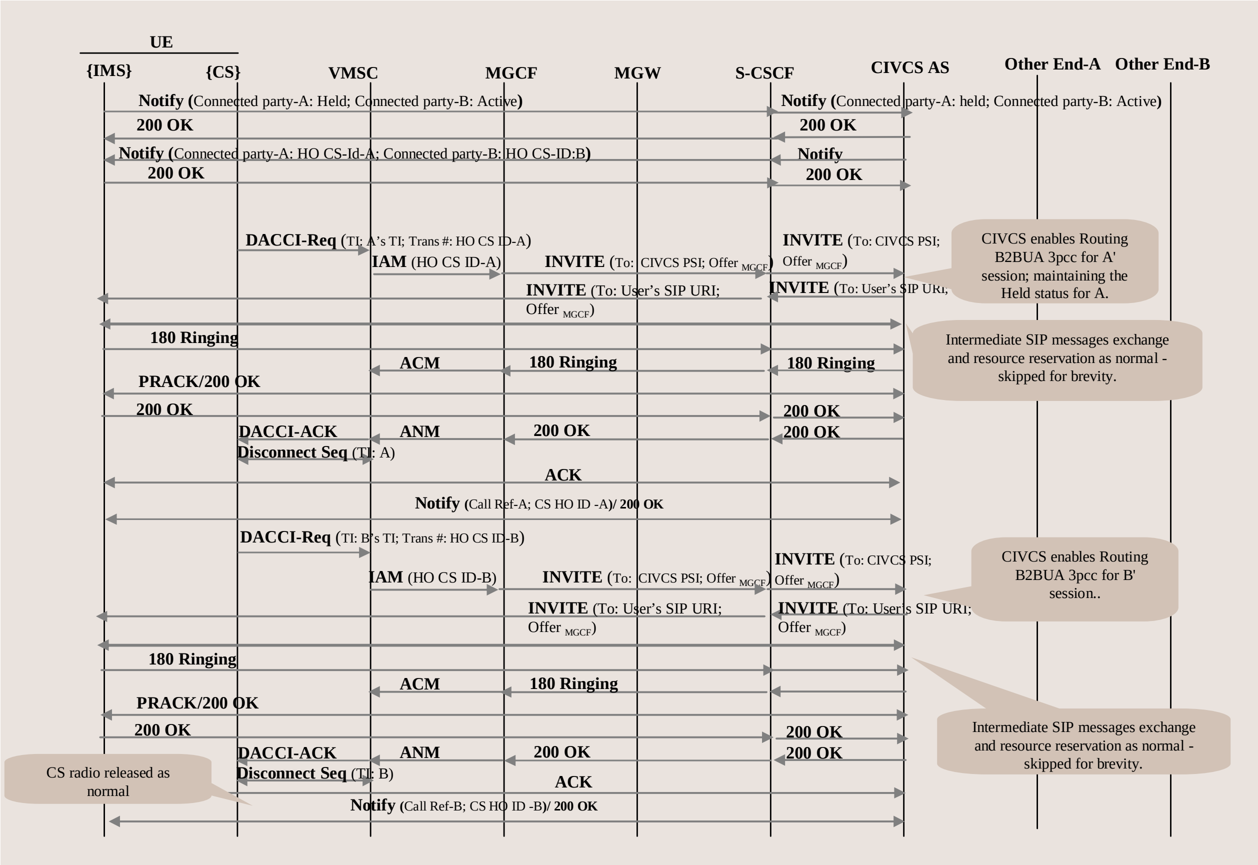


Figure 6.3.27: Handover of Held/Active CS Sessions via DACCI

- The UE detects a need for DACCI enabled Handover, registers with IMS, and provides session state information to CIVCS.
- CIVCS allocates CS HO Identifiers for A's and B's session and communicates to the UE.
- UE executes Handover 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.
- The MSC allocates a new circuit connection for A's session, extending it's bearer toward CIVCS as described in previous section.
- CIVCS 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. CIVCS establishes A's session in the Held state as suggested by session state information received from the UE.
- Upon completion of Handover of A's session, UE executes Handover of B's session.
- The MSC allocates a new circuit connection for B's session, extending it's bearer toward CIVCS as described in previous section.
- CIVCS 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.

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. The UE establishes all the sessions in the handing-in domain, followed by establishment of conference bridge in the handing-in domain, subsequently followed by release of the conference bridge and associated legs in the handing-out domain.

The bearer path interruption for the transfer of MPTY service could be larger than the bearer path interruption caused by transferring Held/Active sessions. Therefore, it's recommended that the UE informs the user via the MMI procedures that Handover of MPTY service is in progress.

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

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 handover 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 handover 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 Handover procedure will handover the between the CS and IMS domains. The Services AS may not need to be informed of the handover.

6.3.7.4 Supplementary Service Impact summary

Table 1 provides preliminary impact statements for commonly used supplementary services.

Table 1: Supplementary Service Impact of CIVCS

Supplementary Service	Impact statement
Home Network Call Forwarding (CFU, CFNRc-HLR detached)	CIVCS 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 CIVCS is unaware of user's availability.
Incoming Call Barring	No impact.
Outgoing Call Barring	Handover cannot be executed if certain flavours of outgoing call barring are enabled for the users.
Calling Line ID Presentation (CLIP)	CIVCS ensures delivery of the originating party's CLIP information to the CS-IMS user when CS incoming calls are anchored via CIVCS. CLIP presentation for target leg to be blocked by CIVCS with appropriate use of screening indicators. Interactions with CLIP Override are for further study.
Connected Line Identity Presentation (COLP)	CIVCS ensures delivery of the actual connected party's COLP information to the CS-IMS user when CS originating calls are anchored via CIVCS. COLP presentation to the user is required to be blocked for the target leg, preferably at UE.
Closed User Group	CIVCS PSI is required to be included in subscriber's Closed User Group profile.
Call Hold/Retrieve	No impact.
Call Wait	No impact.
Multi-party	Transfer of Multi-party service is possible as long as the handing-in domain supports it. 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.
Optimal Routing	Optimal routing for basic mobile to mobile calls to be disabled for CS-IMS users.

6.3.7.4.1 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. 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

Handover 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 CIVCS 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, handover 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, handover 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 CIVCS PSI must be an allowed Fixed Destination.

6.3.8 Evaluation of the model

6.3.8.1 General

This clause presents the evaluation of the service continuity solution against the set of criteria.

The benefits of this proposal include:

- The solution provides an access agnostic approach with cohesive techniques applied for Handovers in both directions, CS to IMS and IMS to CS.
- The subsequent Handovers and Handbacks are handled in an efficient manner. Such Handovers do not result in daisy chain effect that results from use of disjoint techniques applied in CS to IMS and IMS to CS directions.
- The CS call/IMS session is anchored at CIVCS for the life of the CS call/IMS session that enables comprehensive billing with complete Handover history.
- The solution does not impact the CS or PS Domain Core Network, or the Access Network for mobility between GERAN/UTRAN CS and I-WLAN.
- The solution is based on a service in user's home IMS network; therefore, CIVCS service delivery to the user is not impacted when roaming in non supporting networks.
- The solution applies to all roaming situations and has no restrictions on the location of the WLAN or CS MSC.

The drawbacks of this proposal include:

- None identified so far.

Editor's Note: The use of CIVCS for providing CS-IMS voice continuity when the IMS is accessed over UTRAN is for further study.

Alignment with the Reference Architecture to ensure correct terminology is for further study.

6.3.8.2 Techniques for enabling static anchoring for CS calls and IMS sessions at CIVCS

The benefits of this proposal include:

- Provides uniform solution for anchoring CS and IMS calls/sessions at CIVCS which is not restricted by any service interactions as some of the dynamic anchoring techniques.
- CIVCS is in control of the bearer path since the initial call setup, therefore, Handover execution 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 effect Handover execution when dynamic anchoring techniques are applied.
- Static anchoring expedites Handover execution time as the anchor is in place at the time of Handover and additional steps are not required to establish the anchor.

The drawbacks of this proposal include:

All CS originated calls made by CS-IMS users are routed via the user's home IMS network which results in some inefficiency in call setup delays and network resource usage when setting up calls for subscribers of CS-IMS Voice Continuity Service. However, it should be noted that since a MGW is used to anchor the CS bearer, the bearer path backhaul can be minimized by strategically locating such MGWs close to legacy networks.

6.3.8.3 Dynamic CS Anchoring for CIVCS using ECT

The benefits of this proposal include:

- This approach enables dynamic anchoring with CIVCS that alleviates concerns around resource usage with static anchoring.

The drawbacks of this proposal include:

- ECT cannot be used to enable Handover from CS to IMS if multi-session services like CW are active at the time of Handover.
- ECT cannot be used to dynamically anchor CS Emergency calls with CIVCS as ECT Supplementary Service is not applicable for Emergency calls.

6.3.8.4 Dynamic CS Anchoring for CIVCS using DACCI

The benefits of this proposal include:

- This approach enables dynamic anchoring with CIVCS 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.

6.3.8.5 Supplementary Services Support

The benefits of this proposal include:

- Approach consistent with the basic principle used for CIVCS that releases the call control PSM in handing-out domain and establishes a new PSM in handing-in domain.
- The transfer is seamless to the user as the user can continue to control the supplementary services after the Handover.
- Since the information required to complete inter domain transition is exchanged with CIVCS via the Notify with Mobility Event package at the beginning of the execution of the inter domain transition procedure, and since a completely new protocol state machine is established in the new domain, loss of radio/I-WLAN coverage in the old domain any time after the exchange of initial information exchange does not affect completion of the inter domain transition procedure.
- Converged services that are supported by both the CS domain and the IMS domain are supported during the handover 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.

The drawbacks of this proposal include:

- Exchange of Subscribe/Notify for Mobility Event package adds to signalling exchange required for IMS Registration which is exacerbated when combined with subscription of other events that may be required in addition to this package. It is recommended to consider signalling optimizations so that multiple events could be subscribed to during IMS Registration without significantly impacting the time required to complete IMS Registration.
- Handover 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 Handover procedure continues successfully even if coverage is lost in Handing-out domain during the Handover execution.

6.4 Service Continuity Model: Anchored Call Control Model

6.4.1 General Description

6.4.1.1 Architecture Model

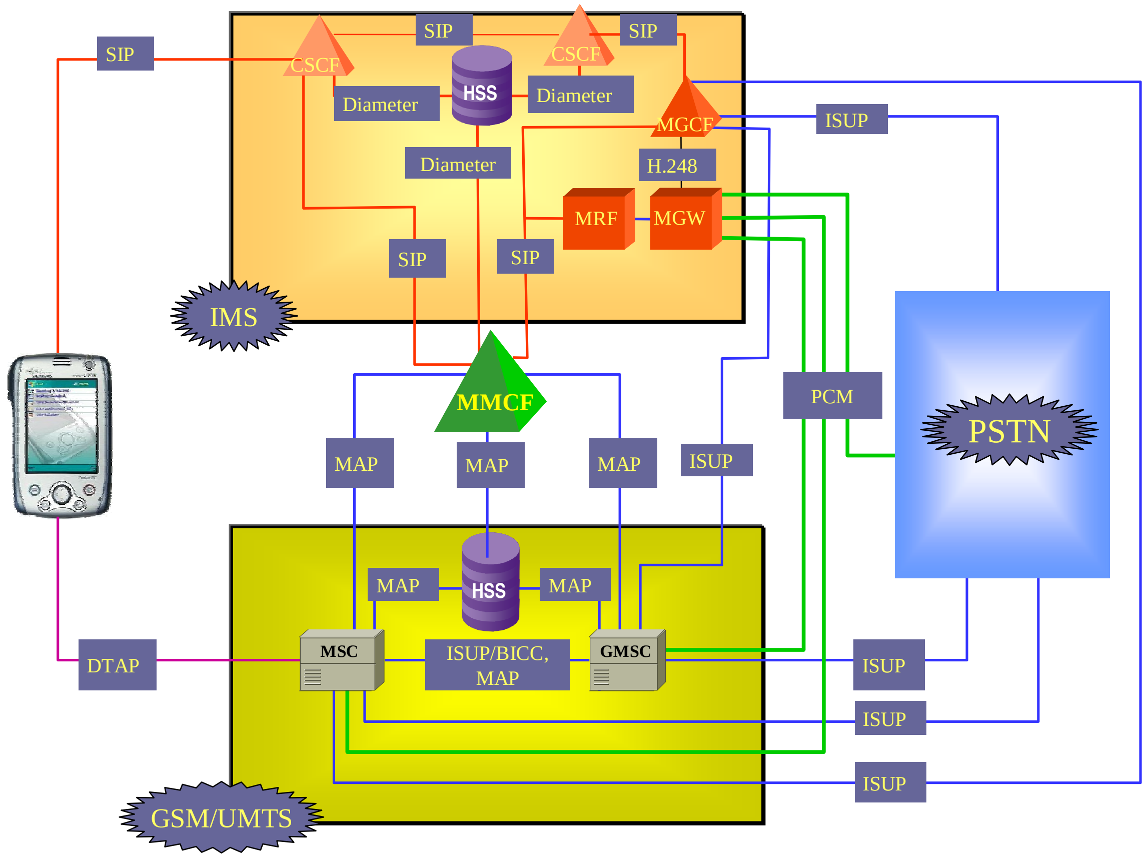


Figure 6.4.1: Network Architecture Diagram

The anchored call control model follows the existing GSM inter-MSC handover model as defined in TS23.009 Handover Procedures. To allow the HLR and MSC interworking, a Mobility Management Control Function (MMCF) is introduced into the architecture. The MMCF connects to the GSM CS domain via the MAP interface and to the IMS domain via SIP and Diameter.

Editor's note: Need to align the terminology of MMCF with contribution S2-051193.

The role of the Mobility Management Control Function (MMCF) is to provide a handoff control function between the IMS and CS domains. At the time of handoff from IMS to GSM, the MMCF appears as the originating MSC to the GSM MSC and it appears as the session transfer point for the IMS sessions that are being handed off to the GSM domain. When there are multiple voice sessions active in the dual mode handset (e.g. call waiting), the MMCF introduces the MRF to provide media stream control and/or mixing. At the time of handoff from GSM to IMS, the MMCF appears as the target MSC to the originating MSC and it appears as the link between the MGCF and the dual mode handset in the IMS domain.

In addition, it provides location update information when the dual mode handset "roams" between the two domains. When the E.164 number is owned by the CS domain, the MMCF acts as a VLR for the GSM phone when it roams into IMS coverage, thus allowing a telephony call to the E.164 number to route to the IMS location. When the E.164

number is owned by the IMS domain, the MMCF acts as an HLR for the IMS phone when it roams into GSM coverage, thus allowing a telephony call to the E.164 number to route to the GSM location.

6.4.1.2 Mobility Management Control Function

The Mobility Management Control Function (MMCF) Function manages the overall handoff process. It interfaces to the S-CSCF via SIP and has a role similar to an IMS Application Server. It interfaces to the MGCF and MRF acting as a CSCF. It interfaces to the 2G Network via the MAP protocol and acts as a VLR and HLR.

Editor's Note: the role of MMCF acts as VLR and HLR needs further elaboration.

The MMCF sub functions are:

MMCF Registrar Function:

- receives registration requests from the S-CSCF when the UE registers in the IMS domain

MMCF HO Procedure Function:

- SIP Event driven mobile interface
- may subscribe to a UE for event notifications or publish information to a UE after registration
- Target system determination function

MMCF HO Function:

- Pool of temporary E.164 numbers and tel URIs

MMCF HO Connection Function:

- Inserted in the call when the HO begins and juggles the legs of the call to switch the bearer path from source to target
- Remains in the call after the HO, acting as a B2BUA
- Call parking function
- Call rendezvous function
- Call transfer function
- SIP interface to BGCF and MGCF

MMCF MSC/VLR:

- Used to setup handover legs with MAP procedures
- Does update location / cancel location when the UE does a SIP register/deregister
- Assigns MSRN when HLR does a request
- Responds to MGCF request (via HO Connection Function) to resolve MSRN on receipt of incoming PSTN calls
- MAP interface to GSM MSC

Editor Note: The requirement of the location update and cancel location need further elaboration.

Editor Note: How to discover the 3G Cell ID or RNC ID information for IMS to CS handover need to be investigated.

Editor Note: Further elaboration on the SIP message encapsulation for IMS to CS handover.

6.4.2 Routing Selection Decision

6.4.3 Registration

There are 4 different registration scenarios to consider based on where the user's telephone number is provisioned. The provision of the user telephone number is based on one user subscription.

1. The telephone number is provisioned in the GSM HLR and the user is registering in the GSM domain. This is a normal GSM registration. The only change from the normal registration scenario is that the MMCF may have been acting as a VLR (see registration case 2) and thus the GSM registration will cause the HLR to send a cancel location to the MMCF to remove the user from the VLR.

2. The telephone number is provisioned in the GSM HLR and the user is registering in the IMS domain for voice calls(i.e., it wants to do voice via the IMS domain and not the CS domain). This is a normal IMS registration (flow line 1) with the following additional steps:

   a. The MMCF is informed of MS1 registration. This could be via proxy registration, MS1 publishing its registration status to the MMCF or some similar mechanism (flow line 2).

   b. The MMCF and dual mode handset subscribe to each other's mobility event package. As part of the subscription, the MMCF sends a handoff URI to the dual mode handset that will be used during the IMS to CS handover procedure (flow lines 3 – 18).

   c. If the dual mode handset's mobility event package indicates that the user is not in a call (i.e., this is an idle state registration), the MMCF creates a VLR and does an update location procedure with the GSM HLR (flow lines 19 – 24).

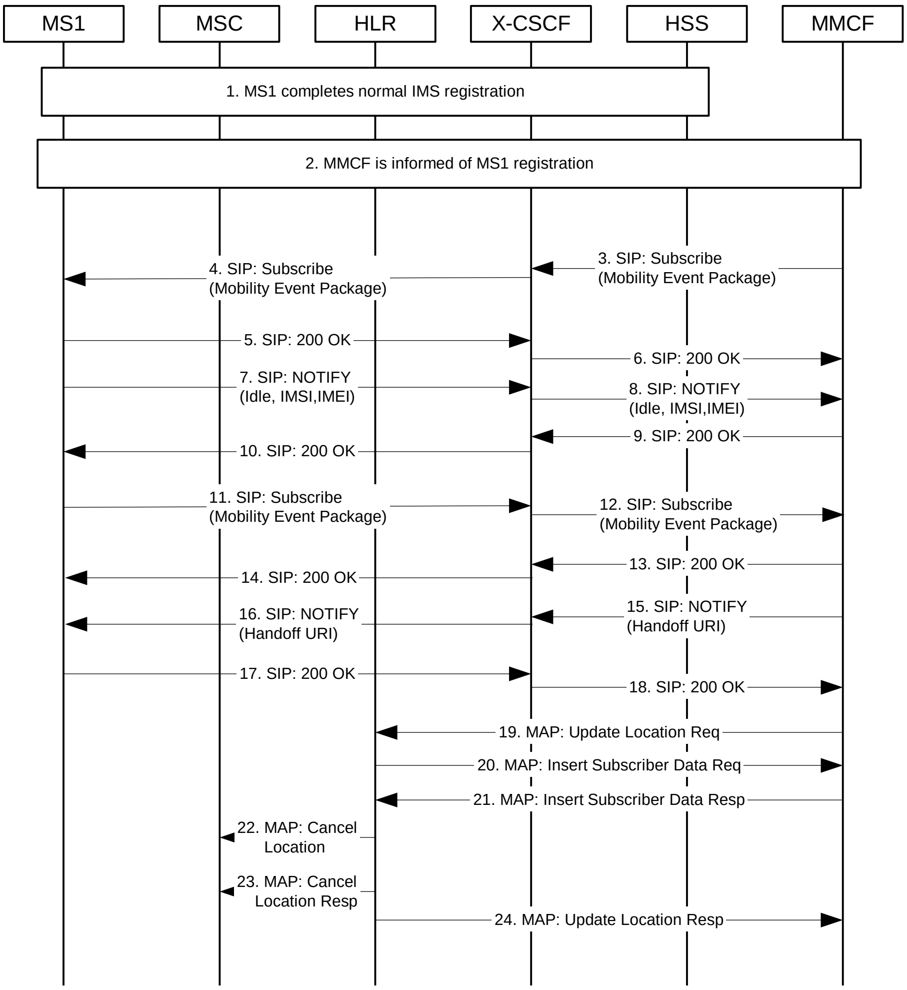


Figure 6.4.2

3. The telephone number is provisioned in the IMS HSS and the user is registering in the IMS domain for voice calls(i.e., it wants to do voice via the IMS domain and not the CS domain). This is a normal IMS registration with the following additional steps:

    a. The dual mode handset includes a trigger function that causes the S-CSCF to do a proxy registration with the MMCF.

    b. The MMCF and dual mode handset subscribe to each other's mobility event package. As part of the subscription, the MMCF sends a handoff URI to the dual mode handset that will be used during the IMS to CS handover procedure.

    c. If the dual mode handset's mobility event package indicates that the user is not in a call (i.e., this is an idle state registration), the MMCF will send a cancel location to the previous MSC if applicable.

4. The telephone number is provisioned in the IMS HSS and the user is registering in the GSM domain. This is a normal roaming registration in the GSM domain, with the following differences described below and shown in the attached flow diagram:

    a. When the visited MSC creates the VLR, it communicates with the MMCF, which appears like an HLR to the MSC (flow lines 3 – 5, 7).

    b. The MMCF will use the HSS to validate the GSM credentials and then will create a registration in the HSS that causes the S-CSCF to treat the MMCF as one of the registered contact addresses for the public userid of the dual mode handset. The contact address provided by the MMCF should be sufficient for the MMCF to be able to do a provide roaming number request when requested by the S-CSCF (flow line 6).

    c. The MMCF will send a cancel location to a previous MSC if applicable (flow lines 9 – 10).

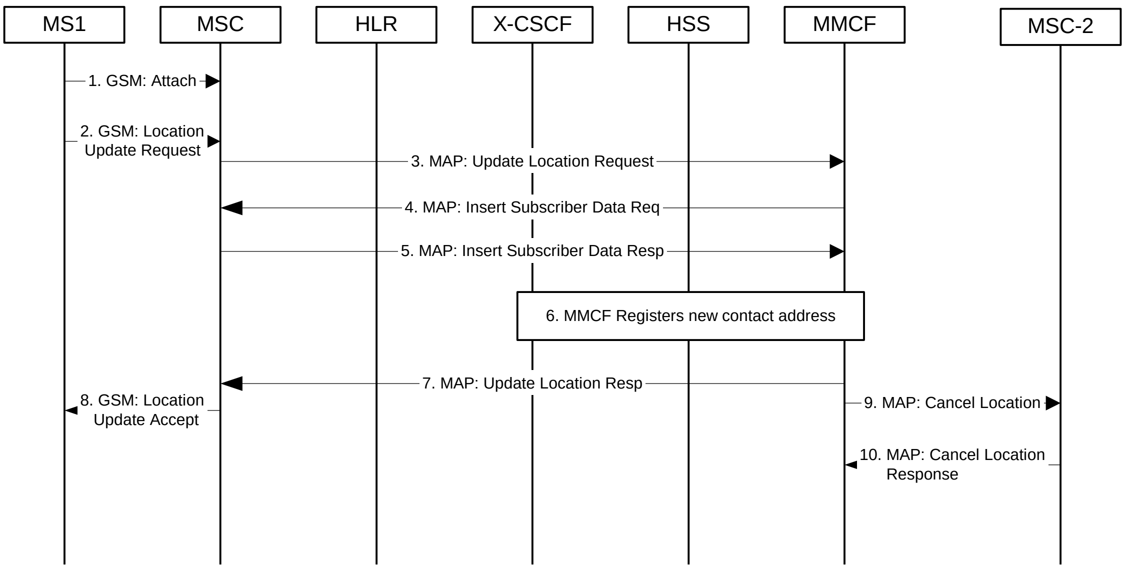


Figure 6.4.3

Editor's Note: For the above figure step 3, consider to terminate at HSS instead of MMCF as shown.

Editor's Note: How to address VLR neighboring configuration by MMCF.

Editor's Note: Need to elaborate the inter VLR communication by MMCF.

Editor's Note: Whether study is required into e.g. terminating SMS the behavior, when the subscriber has CAMEL trigger. Consider new location update procedure?

6.4.4 Origination

6.4.4.1 IMS origination

When a user who is registered in IMS initiates a voice session via a SIP INVITE to another telephone number or IMS public user id, normal IMS domain call processing occurs. There is no special processing because this is a dual mode handset.

6.4.4.2 GSM/UMTS CS origination

When a user who is registered in GSM/UMTS initiates a voice session to another telephone number normal CS domain call processing occurs. There is no special processing because this is a dual mode handset.

6.4.5 Termination

6.4.5.1 IMS termination

The anchored call control model does not assume that the GSM operator has an IMS domain or that the IMS operator has a GSM domain. Thus there is no assumption that there is a shared HSS/HLR between the 2 domains.

There are two cases depending on where the dual mode handset is registered:

1. dual mode handset is registered in IMS domain; IMS owned telephone number
    a. Prior to the call, the mobile powers up in the WLAN or moves back into the WLAN while idle, then registers with IMS. This registration includes the subscription for the Mobility Event Package. The MMCF acts as an HLR and cancels any prior GSM registration (Cancel Location).
    b. Normal IMS call termination procedures are followed for calls from the PSTN to the dual mode handset.
2. Dual mode handset is registered in GSM domain; IMS owned telephone number
    a. Prior to the call, the mobile powers up in the GSM domain or moves back into the GSM domain while idle, then it accesses the control channels and registers in GSM. Since the MMCF is acting as the HLR for this telephone number, the MSC sends an Update Location Request to it and the MMCF registers itself as the contact address for the dual mode handset (flow line 1).
    b. Normal call termination procedures are now followed for calls from the PSTN to the dual mode handset (flow lines 2 – 3)
    c. The S-CSCF determines that the MMCF is the current contact address for dual mode handset URI and sends the INVITE to the MMCF, who then gets the MSRN and redirects the INVITE to the MSRN (flow lines 4 – 8).
    d. The MGCF sets up a call to the GMSC (flow lines 9 - 14).

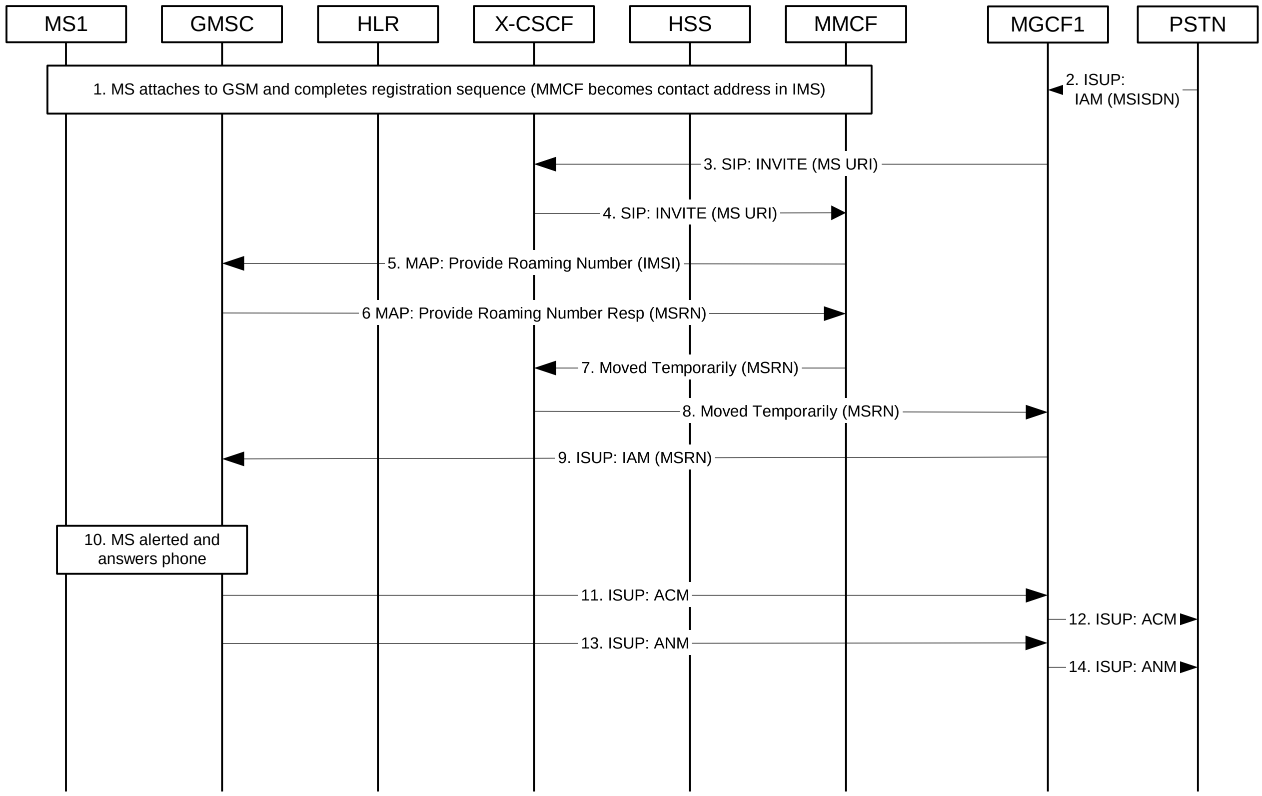


Figure 6.4.4

6.4.5.2 GSM/UMTS CS termination

The anchored call control model does not assume that the GSM operator has an IMS domain or that the IMS operator has a GSM domain. Thus there is no assumption that there is a shared HSS/HLR between the 2 domains.

There are two cases depending on where the dual mode handset is registered:

1. Dual mode handset is registered in GSM domain; GSM owned telephone number

   a. Prior to the call, the mobile powers up in the GSM domain or moves back into the GSM domain while idle, then it accesses the control channels and registers. If the dual mode handset had previously been registered in the IMS domain, the HLR will send a Cancel location to MMCF. The MMCF will send a deregistration to the S-CSCF to cancel the IMS registration for that contact.

   b. A normal mobile termination from the PSTN happens. This is the normal GSM terminating call flow. There are no special steps because this is a dual mode dual mode handset.

2. Dual mode handset is registered in IMS domain; GSM owned telephone number

   a. Prior to the call, the mobile powers up in the WLAN or roams into the WLAN while idle, then registers with IMS. This registration includes the subscription for the Mobility Event Package. Because the mobile is Idle, a VLR record is established by the MMCF and it sends a location update to the HLR. This procedure may start immediately after the initial NOTIFY from the mobile. The HLR will inform the last known serving MSC to delete the mobile's VLR record (flow line 1).

   b. A normal mobile termination from the PSTN happens. The GMSC connects to the HLR to get routing information and the HLR requests a roaming number from the MMCF (flow lines 2 – 4).

   c. The MMCF issues a temporary dual mode handset roaming number. This number will cause a call to be routed to an MGCF (flow lines 5 – 7).

   d. The MGCF does an ENUM query or some other means to determine that the roaming number belongs to MMCF and then forwards the call to MMCF via an INVITE. MMCF converts the roaming number to the dual mode handset URI and sends a moved temporarily response back to the MGCF. The MGCF routes to the correct I-SCSCF (flow lines 8 – 10).

e. Normal call termination procedures are now followed for calls from the PSTN to the dual mode handset (flow lines 11 – 16).

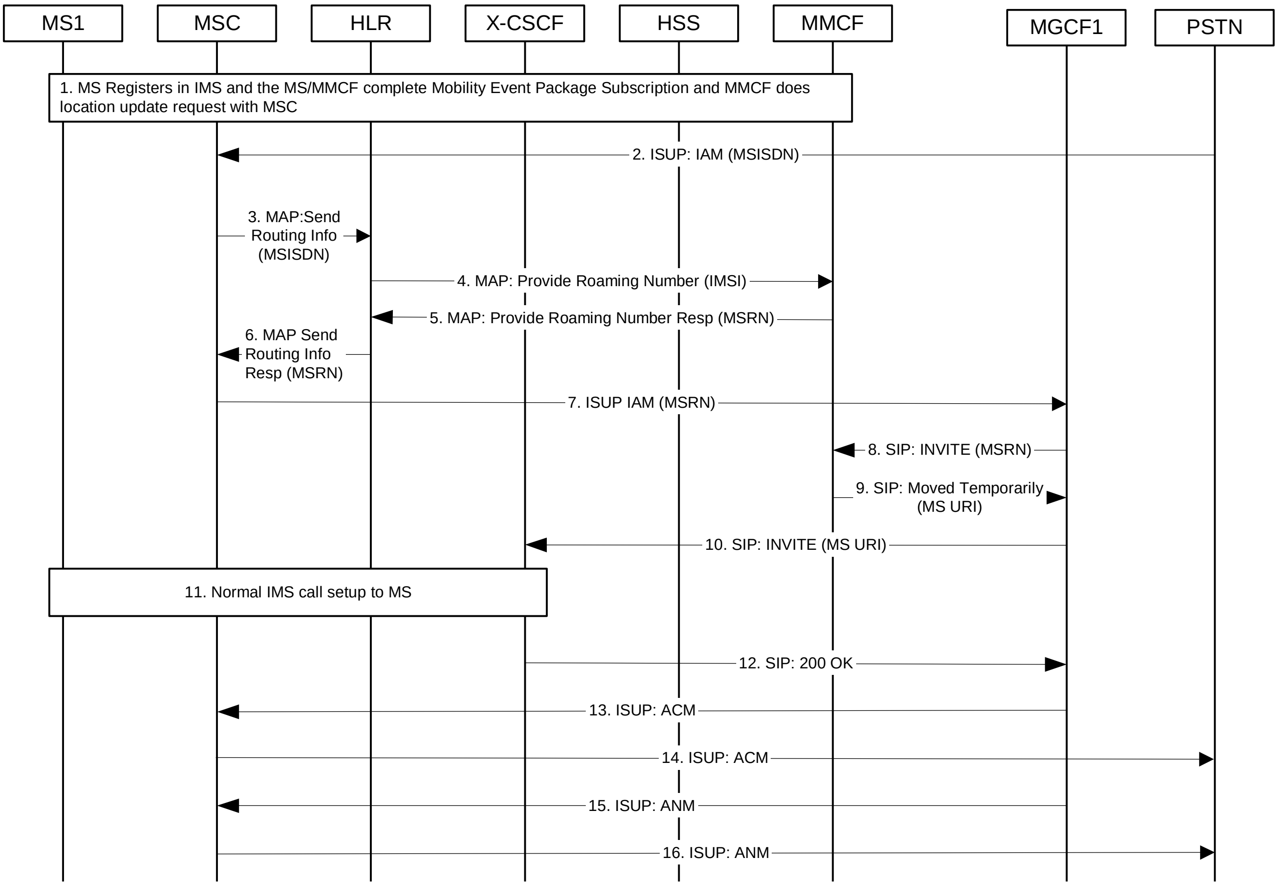


Figure 6.4.5

Editor note: investigate how to satisfy operator policy for domain selection.

Editor note: identify the difference between this solution and Siemens' anchor solution.

Editor note: Elaborate the role of MMCF acting as HLR function.

Editor note: replace GSM with CS domain.

6.4.6 Handover Scenarios

6.4.6.1 CS UE to CS UE call

This Use Case illustrates the architecture used for handing off a dual mode dual mode handset on a circuit voice call on a GSM/UMTS system to a VoIP call on a WLAN/IMS system.

In Figure 6.4.6, the dual mode handset is operating in the GSM domain and has two active GSM calls. One session is to a PSTN telephone (which could be a wireline or wireless phone) . The other session is to a GSM dual mode handset connected to the same MSC. Since these are GSM calls, the MSC manages both call legs for services such as call waiting or 3-way calling. Any mixing of the media streams is managed internal to the MSC.

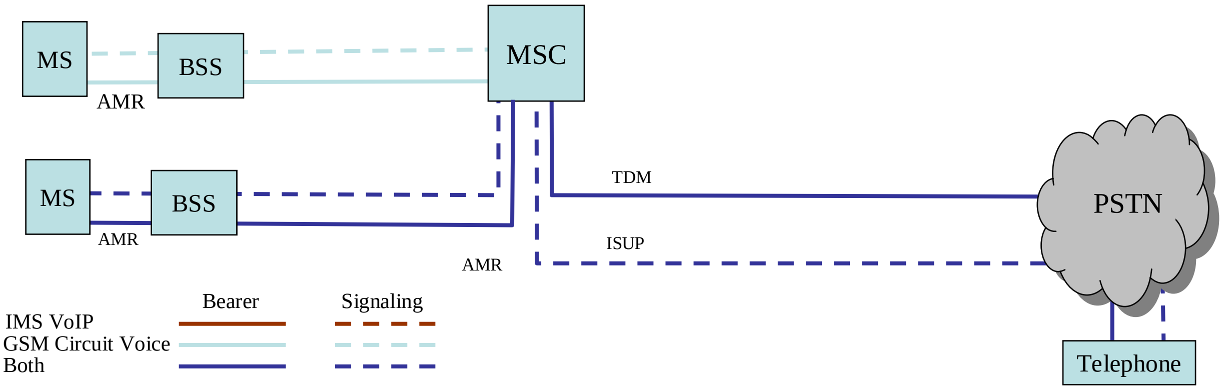


Figure 6.4.6: Initial State GSM - dual mode handset has 1 GSM call and 1 PSTN Call active (either 3-way or call waiting)

Figure 6.4.7 shows the state of the calls after the dual mode handset had handed over to the IMS domain. Since the GSM MSC remains in the call as the anchor MSC, only 1 call leg is handed over. The GSM MSC, as anchor MSC still manages the call and the various call legs. The dual mode handset sends and receives call state change to the MMCF as DTAP messages embedded in SIP NOTIFY messages (SIP INFO messages could be used as an alternative). The MMCF acts as a pass thru point to the GSM MSC for these DTAP messages, as is the normal procedure with GSM handovers.

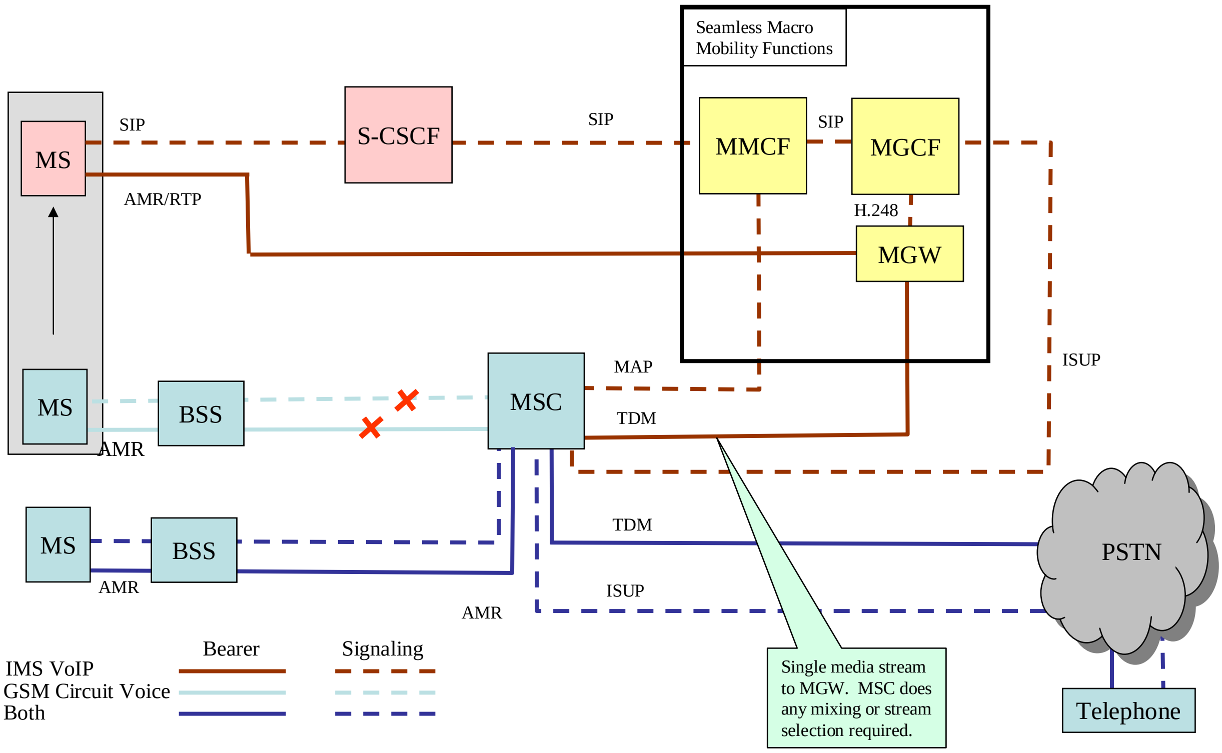


Figure 6.4.7: Handover - GSM Circuit Voice to IMS VoIP

The steps in the procedure for handoff of a GSM circuit voice call to a WLAN VoIP to are as follows.

The dual mode handset is in a call with two parties, one is another GSM MS served by the MSC and the other is a PSTN party which could be an IMS based user. This could be a 3-way call or it could be a call waiting call.

1. The dual mode WLAN/GSM mobile, upon entry into a WLAN coverage area, must register with the IMS (IP Multimedia Subsystem) network and a Macro-Mobility Control Function (MMCF). The S-CSCF does a proxy register with the MMCF. The MMCF will use a SIP SUBSCRIBE/NOTIFY method to subscribe to a newly defined Mobility Event Package with the mobile, and the mobile will likewise subscribe to the same event package with the MMCF. The dual mode handset notifies the MMCF that it is in a GSM call and provides its IMSI. In this case, the MMCF sends a null notification to the dual mode handset (flow line 1).

2. On the GSM circuit voice network, the mobile initiates the GSM handover procedure via the MSC and the mobile is instructed to tune its WLAN radio to the WLAN-IMS (flow lines 2 – 8).

    - dual mode handset now initiates a normal GSM handoff, with the target MSC pointing to the MMCF. Even though there are multiple connections, only 1 connection is setup from the MSC to the IMS network. The MSC anchors the call and will continue to manage both the call leg to MS2 and the PSTN.

    - The MMCF provides a handover number to the MSC and the MSC initiates a call to the handover number.

    - When the MGCF receives the call setup message, it uses ENUM or some other means to translate the handover number to a handover number URI that routes to MMCF. It then initiates a call to MMCF.

    - On the GSM side, the dual mode handset receives a handover command with the reference number and the fake traffic channel info.

3. On the WLAN-IMS network, the mobile uses the SIP NOTIFY to inform the MMCF that it is handing off and includes the reference number that it received (flow lines 9 – 12).

4. The MMCF then establishes a SIP session with the mobile and the mobile now switches to its WLAN radio (flow lines 13 – 17).

   - MMCF sends an invite to dual mode handset with MGW1 as the endpoint. It is able to do this since the reference number allows it to correlate the dual mode handset with the incoming call to the handover number URI from MGCF1.

   - After dual mode handset accepts the call, MMCF can respond to the call from the MGCF.

   - The MGCF uses the dual mode handset endpoint information and connects the MGW to the dual mode handset.

5. The GSM network is informed the handoff is complete and the GSM core network and radio access network resources are released (flow lines 18 – 20).

   - The MGCF provides an answer message to the MSC.

   - The MSC now clears the connection to the dual mode handset.

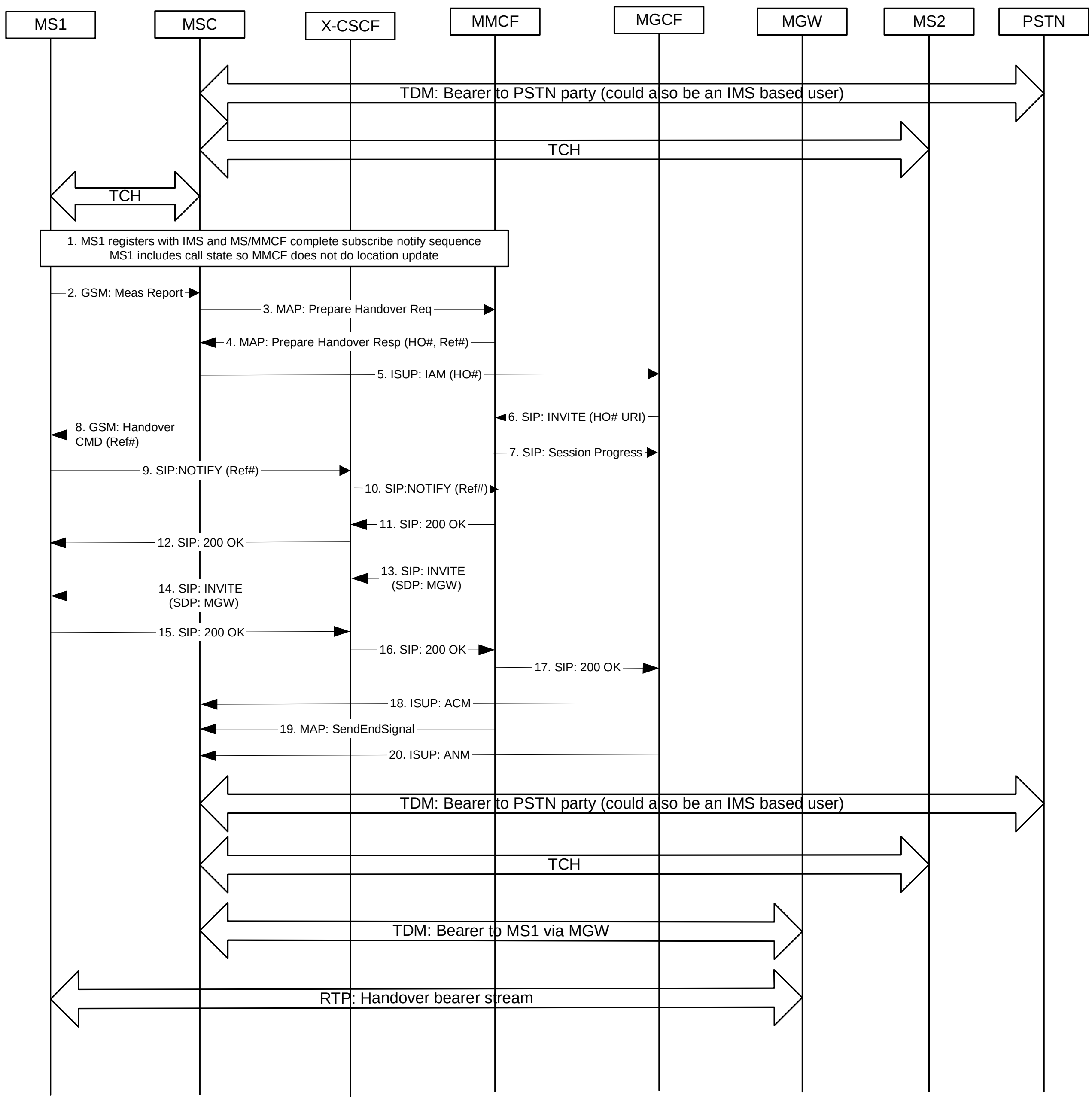


Figure 6.4.8

6.4.6.2 CS UE to IMS UE call

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

Editor note: Elaborate on the simultaneous registration for CS and PS domain,

6.4.6.3 IMS UE to IMS UE call

This Use Case illustrates the architecture used for handing off a dual mode dual mode handset on a VoIP call from a WLAN/IMS system to a circuit voice call on a GSM/UMTS MSC.

In Figure 6.4.9, the dual mode handset is operating in the WLAN domain and has two active IMS call sessions. One session is to a PSTN telephone (which could be a wireline or wireless phone) . The other session is to an IMS dual mode handset. Since these are IMS sessions, the dual mode handset manages both sessions for services such as call waiting or 3-way calling. Any mixing of the media streams is managed internal to the dual mode handset.

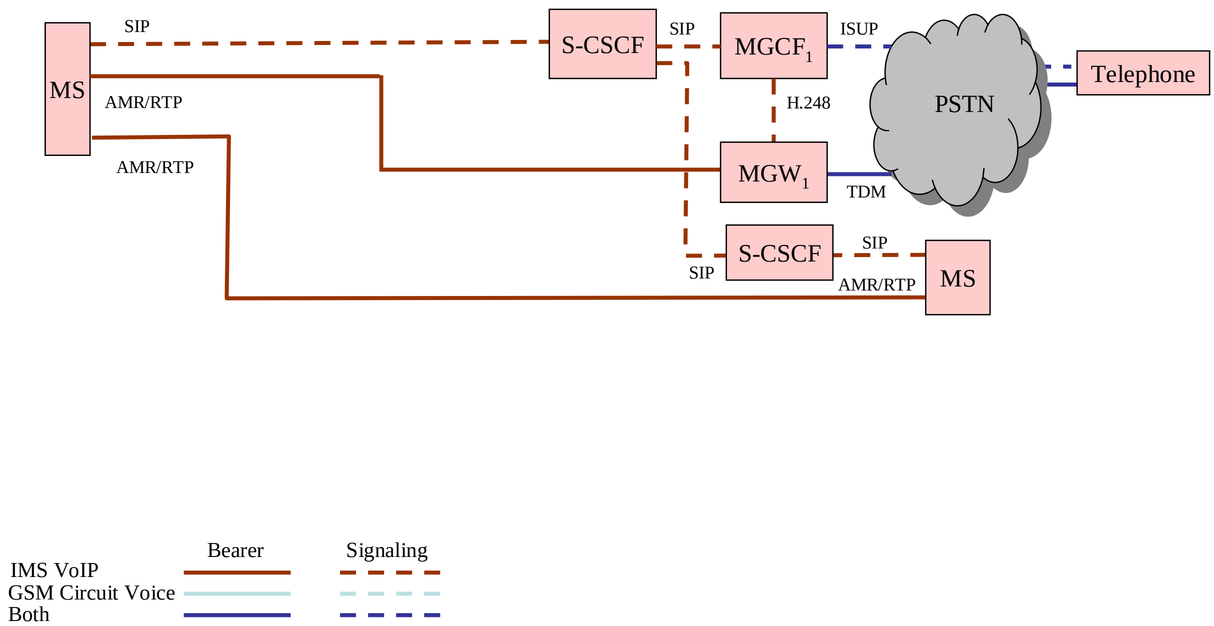


Figure 6.4.9: Initial State IMS - dual mode handset has 1 IMS session and 1 PSTN Call Active (either 3-way or call waiting)

Figure 6.4.10 shows the state of the sessions after the dual mode handset has handed over to the GSM domain. Since the GSM domain only supports 1 media stream to the dual mode handset, the MMCF has introduced an MRF in the IMS domain to terminate the IMS sessions that have been transferred to the MMCF. The MMCF appears as the anchor MSC to the GSM MSC after handover and still manages the call and the various call legs. The dual mode handset sends and receives session state change messages to and from the MMCF via DTAP messages, as is the normal procedure with GSM handovers.

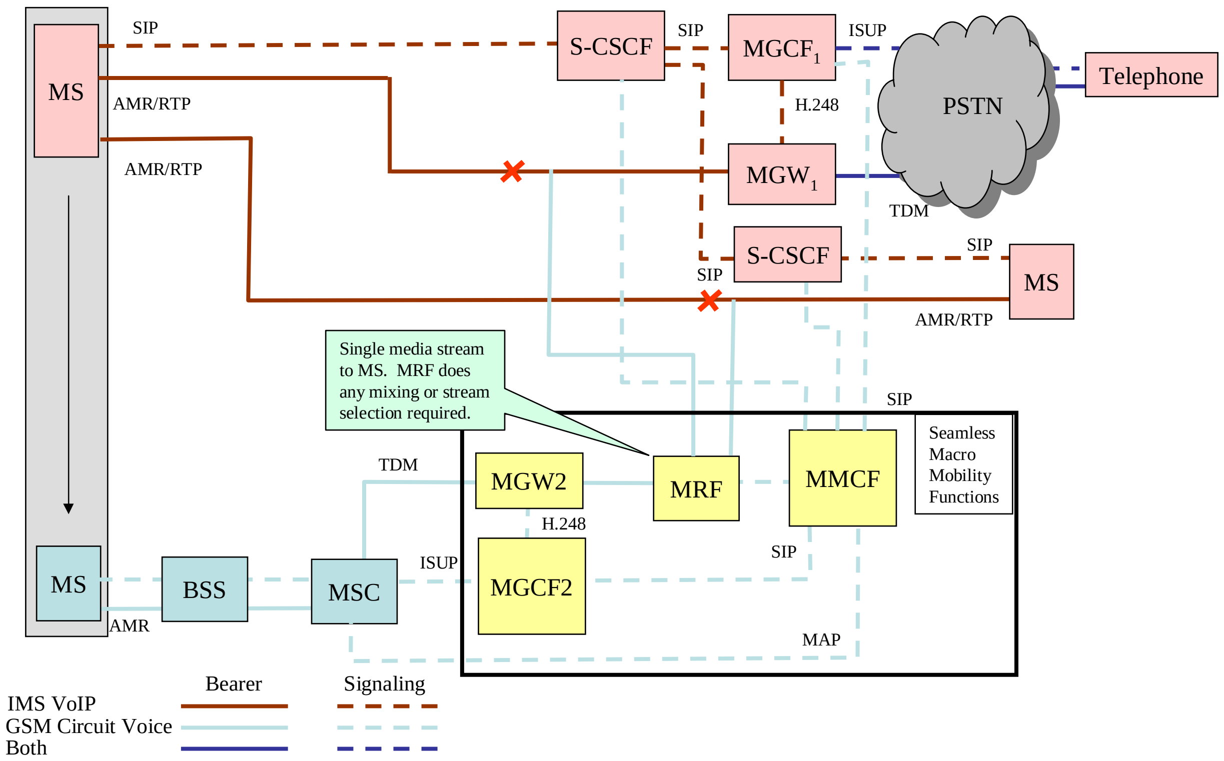


Figure 6.4.10: Handover - MMCF and MRF replace dual mode handset for IMS session and PSTN call

The steps in the procedure for handoff of a WLAN VoIP to a GSM circuit voice call and a subsequent hand back to WLAN are as follows.

1. STEP 1: The dual mode WLAN/GSM mobile, upon entry into a WLAN coverage area, must register with the IMS (IP Multimedia Subsystem) network and a Macro-Mobility Control Function (MMCF). The MMCF will use a SIP SUBSCRIBE/NOTIFY method to subscribe to a newly defined Mobility Event Package with the mobile, and the mobile will likewise subscribe to the same event package with the MMCF (flow line 1).

    - Standard IMS registration occurs, with the dual mode handset providing an indications that this is a dual mode dual mode handset.

    - The S-CSCF does a proxy register with the MMCF

    - The MMCF and dual mode handset subscribe to each others mobility events. The dual mode handset notifies the MMCF that it is idle and provides its IMSI. The MMCF notifies the dual mode handset of a handoff URI to use when a handoff is required. The handoff URI will uniquely identify the session as one being handed off for this dual mode handset.

2. The Mobile sets up the PSTN call and IMS dual mode handset VoIP session. In IMS, call waiting and 3-way call may be client features and not network features.

    - In the example flow, the dual mode handset has 2 sessions active

    - The first session is a call to the PSTN

    - The second session is an IMS VoIP call to another IMS dual mode handset.

    - In this example, both media streams are connected to the dual mode handset and the dual mode handset will do any media stream selection (e.g. for call waiting) or media stream mixing (e.g. for 3-way calling)

3. When the mobile is engaged in a call with another party on the PSTN (or with another mobile) and it detects that it is leaving WLAN coverage and entering the coverage area of the GSM network, the mobile makes the decision to hand over the call. It initiates 3 different steps in parallel:

    a. It notifies the MMCF that it wants to handoff .

    b. It initiates IMS session transfer procedures with the active PSTN session to transfer it to the MMCF.

    c. It initiates IMS session transfer procedures with the active MS2 session to transfer it to the MMCF.

4. STEP a: The dual mode handset will notify the MMCF and pass it parameters that identify the target GSM system and cell ID. This marks the start of the handover sequence (a.1 – a.18).

    - The dual mode handset sends a notify to the MMCF with target information as well as information on the state of its existing IMS call sessions (e.g. there are 2 sessions, actively talking to MS2, but the PSTN session is on-hold in call waiting)

    - The MMCF will use GSM Handoff MAP signaling procedures to communicate with the GSM network and a circuit switched MSC.

        - Standard inter-MSC handoff MAP messages are sent and the MMCF receives back the target traffic channel

        - The MMCF notifies the dual mode handset which traffic channel to switch to.

        - When the dual mode handset switches to the traffic channel, the MSC completes the signaling with the MMCF to bring up the path from the dual mode handset to the MMCF via the GSM network. This results in connecting the GSM media stream to the MRF via MGW2.

5. STEP b: Using IMS session transfer procedures, MS1 transfers the existing session with the PSTN via MGCF1/MGW1 to the MMCF/MRF. MS1 uses the Handoff URI to inform MGCF1 of the destination. The MMCF had supplied this Handoff URI to MS1 in a NOTIFY at registration time (flow line b.1).

6. STEP c: Using IMS session transfer procedures, MS1 transfers the existing session with MS2 to the MMCF/MRF. MS1 uses the same Handoff URI to inform MS2 of the destination (flow line b.2).

With the completion of these steps, the handover has been completed. The bearer stream now flows to the MRF from MS2 and MGW1 on the IMS side. Media is combined or selected by the MRF and sent to MGW2 and then to the MSC and out to the dual mode handset.

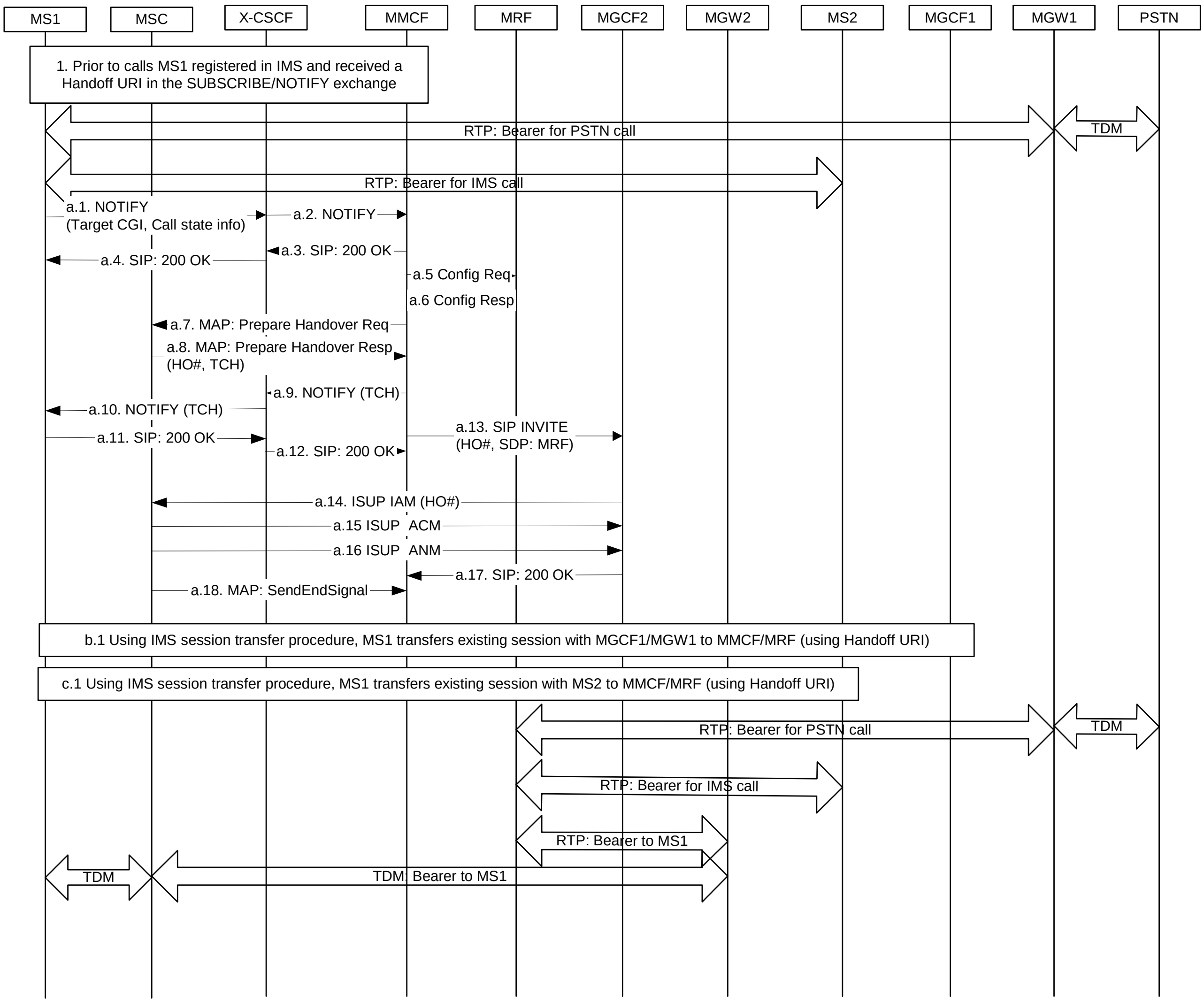


Figure 6.4.11

6.4.6.4 IMS UE to CS UE call

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

Editor note: investigate the starting of the DTAP state machine in response for the need of handover.

Editor note: elaborate the encapsulation of the SIP message in DTAP.

Editor note: elaborate the Iur communication is required.

Editor note: consider point to point call scenario (e.g., IMS to IMS).

Editor note: fixed the call flow for the bearer to terminate at MS2 instead of MGCF1.

6.4.7 Impact on Supplementary Services

6.4.8 Evaluation of the model

This clause presents the evaluation of the service continuity solution against the set of criteria

6.5 Service Continuity Model: CS to IMS Voice Call Continuity – MS assisted

6.5.1 General Description

In this approach the MS initiates the appropriate sequence towards the IMS domain once it detects that a handoff/transition is required.

After the handoff/transition (in this document we use the terms handoff, handover and transition synonymously) is completed, Leg C would have replaced leg B as shown in figure 6.5.1 below.

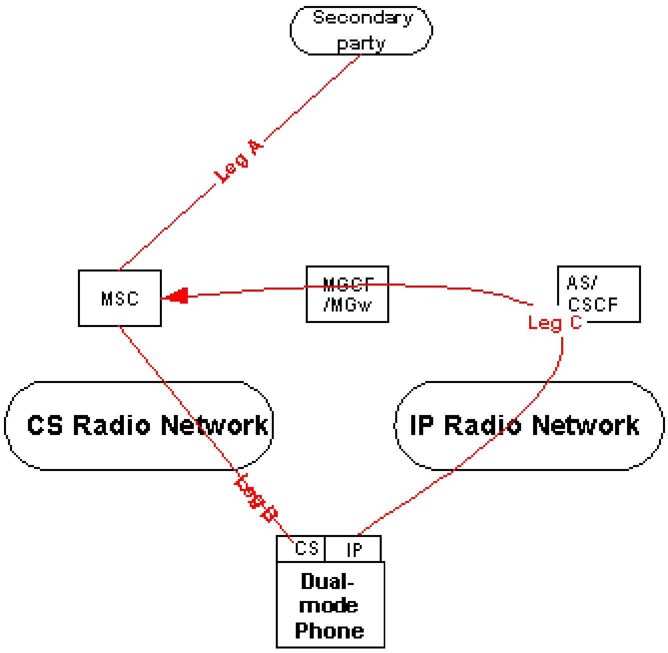


Figure 6.5.1: CS to IMS Voice Call Continuity – MS assisted (high level view)

This approach does not use an anchor point for the handoff until one is actually needed. As such, it employs an anchor point dynamically once a handoff is to occur.

6.5.2 Routing Selection Decision

6.5.3 Registration

6.5.4 Origination

6.5.4.1 IMS origination

6.5.4.2 GSM/UMTS CS origination

6.5.5 Termination

6.5.5.1 IMS termination

6.5.5.2 GSM/UMTS CS termination

6.5.6 Handover Scenarios

6.5.6.1 CS UE to CS UE call

The call flow to accomplish the above is illustrated below.

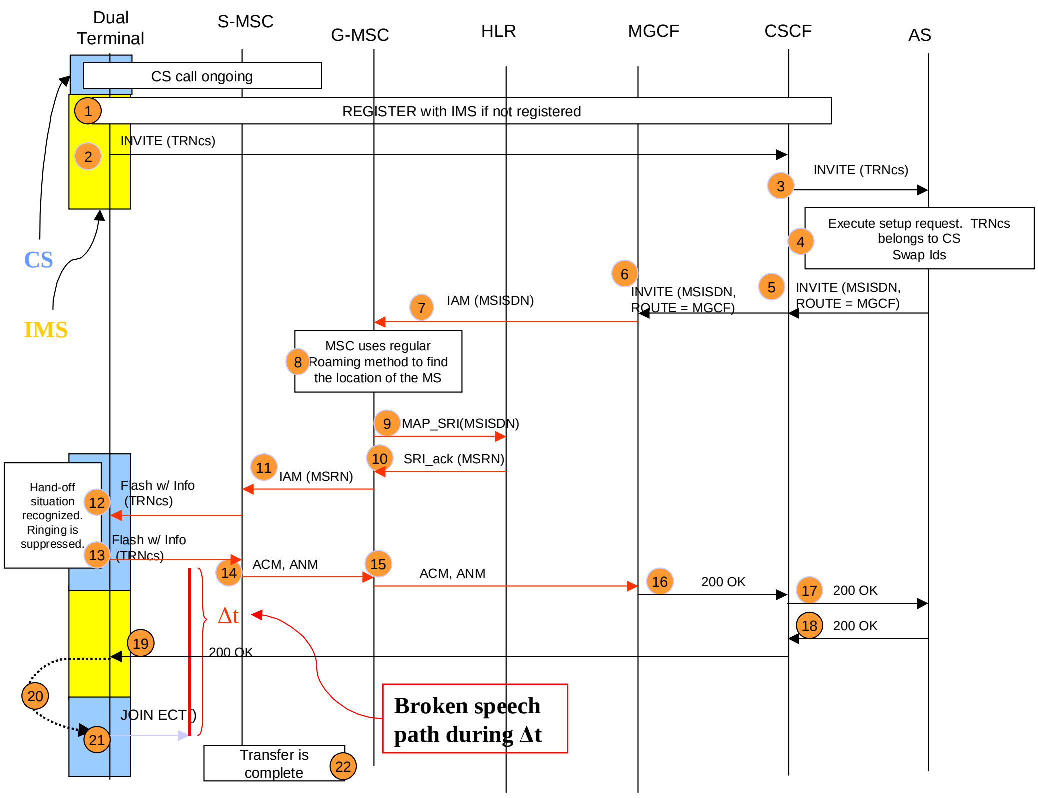


Figure 6.5.2: Call flow for CS to IMS Voice Call Continuity – MS assisted

1. The MS recognizes that a new access is available and makes the decision to handoff to this access. The MS will register the contact information and associated network access.

2. To start the setup of the IMS leg (Leg C), the terminal initiates a new call over IMS by sending an INVITE to the CSCF. A special transfer number (TRNcs) is used as an address uniquely identifying the call as a handoff call. The UE own MSISDN is included as calling party number (CgPN).

3. The INVITE is sent to the Application Server (AS) associated with the user, as result of an activated trigger for the originating INVITEs.

4. The AS analyses the call and identify the request as a handoff of voice service (based on the TRNcs) and that the call shall be connected to the CS domain. The AS swaps the MSISDN with the TRNcs to start a new call leg.

5. The AS sends an INVITE to the CSCF with destination MS address in the CS domain (Called Party Number - CdPN = MSISDN, CgPN = TRNcs, etc.) and ROUTE = MGCF.

6. The CSCF forwards the INVITE to MGCF.

7. The MGCF send an IAM with CdPN = MSISDN, CgPN = TRNcs to the GMSC.

8. The GMSC uses regular methods to request routing information from the HLR.

9. The GMSC sends a MAP_SRI to the HLR.

10. The HLR returns a routing number to the G-MSC.

11. The call is forwarded to the Serving MSC.

12. The MSC finds that the MS is busy in a call, and sends a Call Waiting indication to the MS.

13. The ringing in the MS is suppressed. Note: The MS can recognize that the CgPN is this specific TRNcs.

14. The MS automatically accepts the incoming. The ongoing call (Leg A + Leg B) is put on hold.

15. The Serving MSC sends ACM/ANM to the MGCF to indicate that the call has been accepted.

16-18. The MGCF sends a SIP 200 OK to indicate that the call shall be setup. The message is routed through the CSCF, AS and to the MS (IMS Part). The call media path is setup between the IP and the CS part of the MS.

19. The MS receives the SIP 200 OK and triggers the Explicit Call Transfer.

20-21. The MS send an ECT JOIN to the serving MSC over leg B. This connects the A party with the IMS side of the MS (Leg A + Leg C). The Leg over the CS radio is disconnected (Leg B).

22. Handoff is complete.

6.5.6.2 CS UE to IMS UE call

6.5.6.3 IMS UE to IMS UE call

6.5.6.4 IMS UE to CS UE call

6.5.7 Impact on Supplementary Services

6.5.8 Evaluation of the model

The call will have a short voice interruption time, as indicated in figure 6.5.1.

This approach has the advantage that no changes are required to the CS domain nodes.

The AS and the UE implement all the needed functionality to achieve the desired behaviour.

Given the fact that CS coverage is quite adequate, the risk of losing CS coverage before the handoff/transition is completed is very slim, there are no impacts on nodes in CS domain, it is recommended to adopt that alternative for CS-IMS handoff/transition.

It is also important to note that there is a need in this approach to ensure that service interaction is properly handled to ensure that the handoff call is successfully delivered. To that extent, unconditional transfer services and all transfer services in general must become deactivated prior to handoff. In addition call waiting and CLIP must be activated. Following the handoff all of the above services must be restored to their original condition.

Finally in this approach, the handoff cannot succeed if the UE is already engaged in a 2-way call prior to the handoff or the call is initiated as an emergency call.

The figure below presents the user media plane topologies before and after the handover procedures.

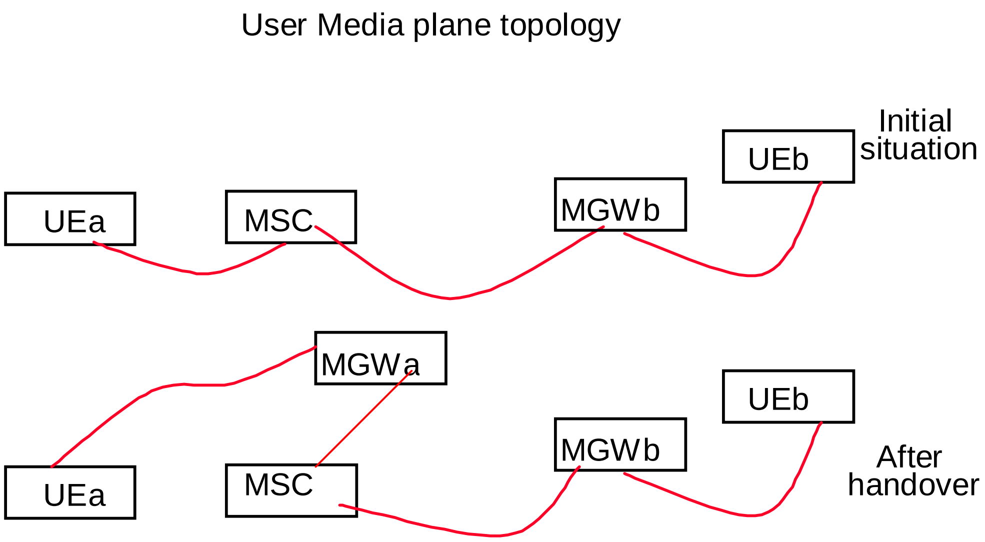


Figure 6.5.3: User media path from CS to IMS, UEb in IMS domain

6.6 Service Continuity Model: IMS to CS Voice Call Continuity – MS assisted

6.6.1 General Description

In this approach, there is a voice anchor point within the IMS domain to handle the call leg toward the MS via the CS domain.

After the handoff/transition is completed, Leg C would have replaced leg B as shown in the figure below.

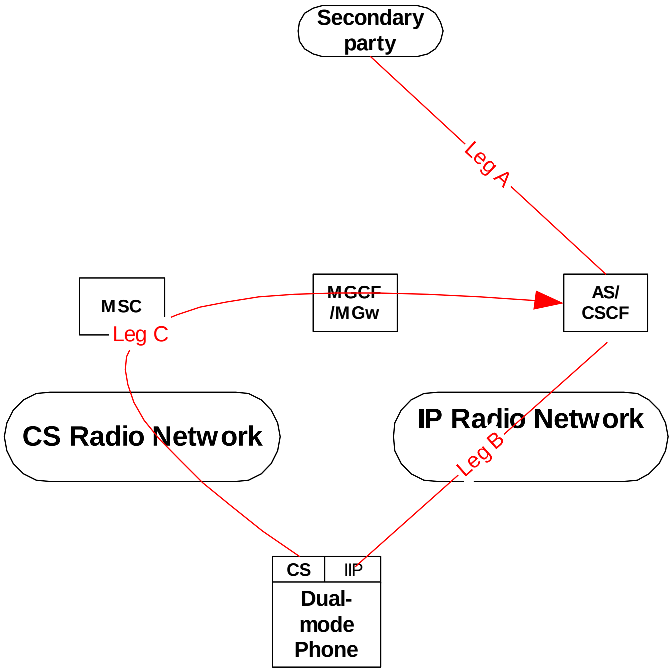


Figure 6.6.1: IMS to CS Voice Call Continuity – Anchoring model (high level view)

There are two approaches described here, one as static anchoring and the other as dynamic anchoring. Static anchoring means an AS will always be involved in every IMS call, acting as a B2BUA. Dynamic anchoring means the AS will only be involved when voice continuity to CS domain is needed.

6.6.2 Routing Selection Decision

6.6.3 Registration

6.6.4 Origination

6.6.4.1 IMS origination

6.6.4.2 GSM/UMTS CS origination

6.6.5 Termination

6.6.5.1 IMS termination

6.6.5.2 GSM/UMTS CS termination

6.6.6 Handover Scenarios

6.6.6.1 CS UE to CS UE call

6.6.6.2 CS UE to IMS UE call

6.6.6.3 IMS UE to IMS UE call

6.6.6.3.1 Static Anchoring Model

The call flow to accomplish the Static Anchoring Model is illustrated below.

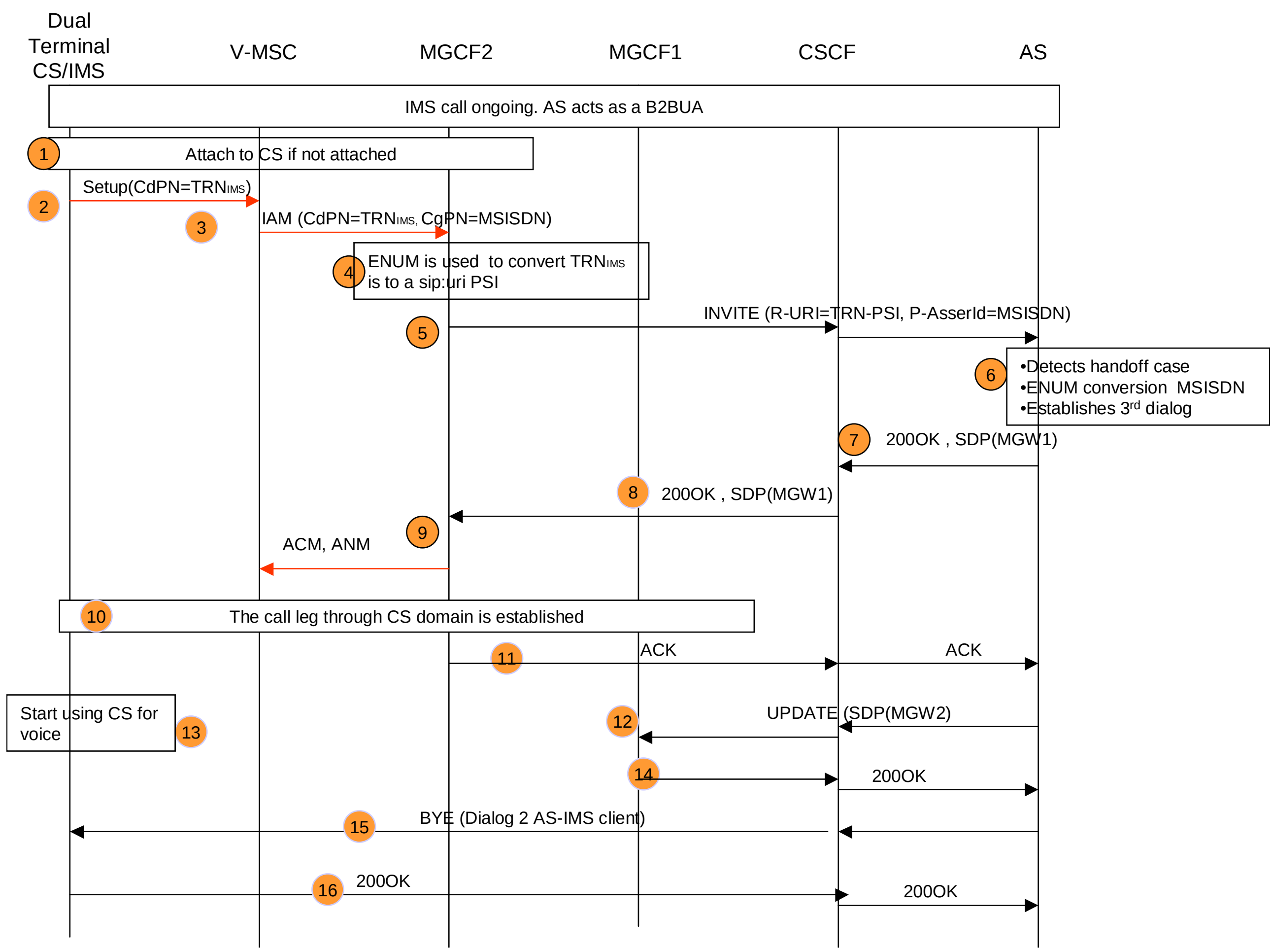


Figure 6.6.2: Call flow for IMS to CS Voice Call Continuity – Static Anchoring Model

1. The MS registers and attaches to the CS domain.
2. To start the new leg (Leg C) the MS initiates a call in the CS domain. The address used is the TRNims uniquely identifying the call as a handoff call.

3. The Serving MSC adds the CgPN = MSISDN to the IAM and forwards the call to MGCF2 (depending on the actual network configuration, the IAM might transit the G-MSC).

4. MGCF2 (or an I-CSCF/S-CSCF which are not shown, but which might have been contacted by MGCF) interrogates ENUM database and converts the TRNims to a sip:uri Public Service Identifier (PSI). The information from the CgPN is copied in the P-Asserted ID SIP header.

5. By performing normal SIP routing procedures the INVITE is send to an AS hosting the hand-off service, identified by the PSI.

6. The AS detects the handoff case, performs ENUM conversion on the MSISDN, then establishes a third dialog.

7. The AS sends a 200 OK establishing the third dialog.

8. The S-CSCF forwards the 200 OK to MGCF2.

9. MGCF2 sends an ANM to the V_MSC.

10. Voice path is through connected, but no voice is yet transmitted from the CS part of the client. The IMS part of the client is the one receiving/transmitting voice at this moment.

11. MGCF2 sends an ACK confirming the Dialog 3 establishment.

12. On reception of ACK, AS changes Dialog 1, by sending an UPDATE that informs MGCF1 to start sending/receiving media from MGW2 (associated with MGCF2).

13. The client which is aware of the hand-off situation, starts using the CS bearers for sending/receiving the voice.

14. MGCF1 confirms the update. At this moment voice is connected between MGW1 and MGW2.

15. At the same time as 12, AS releases the Dialog 2, towards the IMS client, by sending a BYE.

16. On the reception of BYE, the client responds with a 200 OK and releasing Dialog 2.

NOTE: Step 16 may not be fully completed if the IMS coverage is lost, but that should not impact the robustness of the procedure. The client design should be built with that assumption in mind.

6.6.6.3.2 Dynamic Anchoring Model

The call flow to accomplish the Dynamic Anchoring Model is illustrated below.

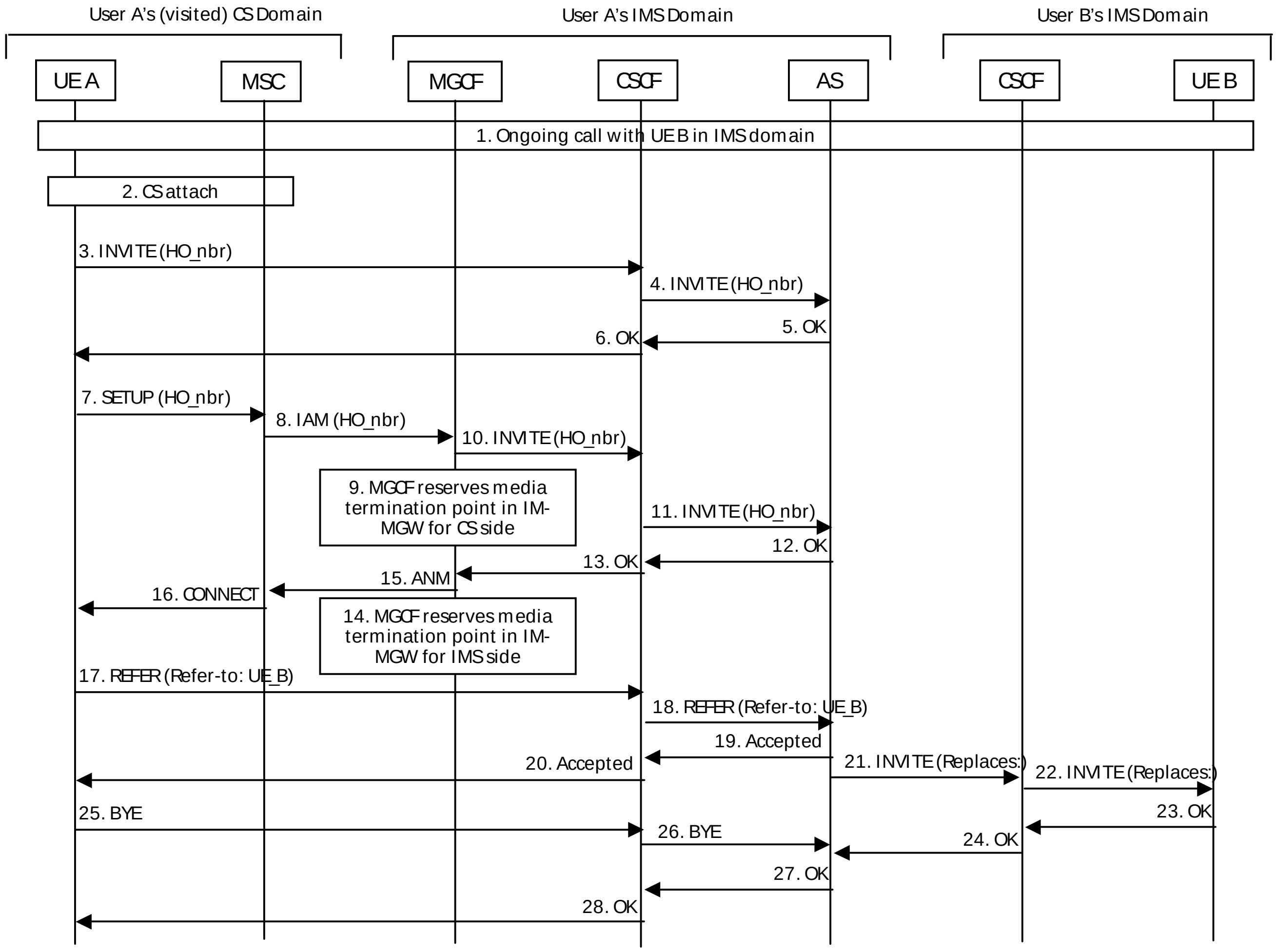


Figure 6.6.3: Call flow for IMS to CS Voice Call Continuity – Dynamic Anchoring Model

1. UE A in IMS domain has an ongoing voice session with UE B in IMS domain.
2. UE A compares the cellular and WLAN radio coverages, and decides to perform a handover. UE A performs a CS Attach procedure.
3. UE A sends an INVITE request towards a provisioned E.164 Handover number (HO_nbr) that points to the Application Server. The Handover number may be pre-provisioned during registration or it may be learned during IMS registration procedures.
4. S-CSCF of UE A forwards the request towards the Application Server based on the Request-URI.
5. AS acknowledges the session establishment.
6. Acknowledgement is forwarded to UE A
7. UE A initiates a normal CS call to the Handover number.
8. The Visited MSC routes the call towards the IMS domain via MGCF using normal CS-IMS interworking procedures.
9. MGCF reserves a media termination point for the CS side.
10. MGCF creates a INVITE request and sends it towards the AS.
11. S-CSCF forwards the INVITE towards the AS based in the Request-URI.

12. OK response to the INVITE is forwarded towards the S-CSCF.
13. OK response to the INVITE is forwarded towards the MGFC.
14. MGCF reserves a media termination point in the IM-MGW for the IMS side of the call.
15. MGCF responds with ANM to the Visited MSC.
16. CONNECT response is forwarded to the UE
17. UE A creates a REFER request that instructs the AS to invite UE B to the handover call.
18. S-CSCF forwards the request towards the AS.
19. AS acknowledges the REFER request.
20. Acknowledgement is forwarded to UE A.
21. AS invites UE B to the handover call. The INVITE request has a Replaces header that instructs the UE B to replace the ongoing call with UE A with the new call.
22. S-CSCF of UE B forwards the request to UE B.
23. UE B sends an OK reply to the AS. UE B modifies the ongoing session with UE A so that the media is now routed towards the MGW.
24. S-CSCF of UE B forwards the OK to the AS.
25. UE A releases the ongoing call with UE B.

Editor note: How supplementary services are handled with Dynamic model is FFS.

6.6.6.4 IMS UE to CS UE call

6.6.7 Impact on Supplementary Services

6.6.8 Evaluation of the model

For Static Anchoring Method, the MS initiates the setup of the leg C in order to minimize the interruption in the voice call, due to paging or traffic channel setup times, while the IMS domain is in control of the handoff process.

Hence, an AS will be involved in every IMS call, acting as a B2BUA. Before the hand-off occurs the AS controls two SIP dialogs:

1. Dialog1 - MGCF1 to AS
2. Dialog2 - AS to the terminal (IMS client)

The entire handoff procedure should take around 1.3 seconds including the speech switch over.

It is to be noted that even if the IMS coverage is lost during the time it takes to establish the circuit switched leg of the call (Leg C), only one of the IMS legs will be lost for a very brief period < 1.3 seconds.

That brief period is insignificant to the extent that the switch over to the CS will be completed before the end user (at the remote end) perceives any interruption.

For Dynamic Anchoring Model, this alternative is similar to Static Anchoring model with the exception that there is no AS linked in the call at the beginning of the call. The AS is linked in dynamically only when a handoff is required, and is achieved by sending an INVITE to the AS.

Once the AS takes over the handoff process, and if the IMS coverage is lost thereafter during the time it takes to establish the circuit switched leg of the call (Leg C), only one of the IMS legs will be lost for a very brief period <1.3 seconds.

That brief period is insignificant to the extent that the switch over to the CS will be completed before the end user (at the remote end) perceives any interruption.

The switch over and the usage of the CS leg of the call (Leg C) is independent from the clearing of the IMS leg of the call (Leg B) which would not be graceful. The client design should be as such that the switch over to the CS will occur regardless if the IMS leg is gracefully terminated or not due to coverage problems)

Note that if the IMS coverage is lost before the AS takes over then the call will be lost, and the handoff procedure will fail.

This will happen if the INVITE, or the ACK never makes it to the AS. This is a key difference from Static Anchoring Model, which of course required an anchor point at all times.

6.7 Service Continuity Model: Mobility Management AS with Anchoring HO Model

6.7.1 General Description

Mobility Management Control Function as described in ref [4] consists of mainly Registrar, HO, and VMSC/VLR functions, and it is similar to an IMS Application Server. This proposal adopts its Registrar and the VMSC/VLR functions excluding the HO related parts to manage the IMS registration and CS side location update, to synchronise the registration state between the two networks, and to manage the routing of terminating calls between domains. There may be additional functions described in this alternative that are in addition to the ones defined in ref [4].

The registration synchronisation between domains may be affected by both the network and user preferences. Such operator policies may be (pre-) configured to the terminal, or they may be available dynamically. User preferences may also be (pre-) configured to the terminal, or they may be based on user input. This also determines the routing toward the terminating domains for terminating call.

The HO mechanism from CS to IMS is based on ECT (see ref [1,3]) when call is first established on the CS side and it has not been anchored in the IMS domain. Ref [1] indicates how ECT is used to transfer the call to IMS domain and ref [3] is used to indicate that the subsequent transfer due to "ping-pong" HO would not require further ECT usage to avoid further unnecessary call leg establishment between the domains. The HO mechanism from IMS to CS is based on anchoring model (see ref [2]) in which the logics is resided in Application server that has this voice anchoring function.

6.7.2 Routing Selection Decision

The following cases and sub-scenarios are analysed with this architecture for network domain termination.

Case 1: the user can only be reached via WLAN access with IMS

Scenario 1A: user's E.164 MSISDN is assigned by the IMS and subscription is stored in the HSS.

No impact. In this case, the PSTN routes the called party E.164 number toward the IMS network. Normal IMS routing procedure toward the user takes place and no new procedure is needed in this TR.

Scenario 1B: user's E.164 MSISDN is assigned by the CS Domain and subscription is stored in the HSS/HLR.

During IMS registration, the MMCF acts as a VMSC an update the HSS/HLR with its address. For call termination, the PSTN routes the called party E.164 number toward the CS Domain (GMSC). GMSC queries the HSS/HLR which quires the MMCF for the MSRN for routing toward the IMS network. Procedure is defined in ref [4] to handle this.

Scenario 1C: user has 2 E.164 MSISDNs, one is assigned by IMS and is stored in the HSS. The other MSISDN is assigned by CS domain and is stored in the HSS/HLR.

This case is considered that the user has 2 separate subscriptions. If the dialled number is a CS domain assigned E.164 number then the routing process is based on scenario 1B above. If the dialled number is an IMS assigned E.164 number then the routing process is based on scenario 1A above.

Case 2: user can be reached only via Cellular access with CS Domain

Scenario 2A: user's E.164 MSISDN is assigned by the IMS and subscription is stored in the HSS.

During Location Update procedure in the CS Domain, it contacts the subscriber HPLMN via MAP_UPDATE_LOCATION. The PSTN routes the called party E.164 number toward the IMS network. The Mobile termination call procedure related to unregistered PSI is used and the MMCF acting as an application server forwards the call toward the VPLMN using the MSRN received from the VMSC. Procedure is defined in ref [4] to handle this.

Scenario 2B: user's E.164 MSISDN is assigned by the CS Domain and subscription is stored in the HSS/HLR.

No impact. In this case, the PSTN routes the called party E.164 number toward the CS Domain (GMSC). Normal GSM terminating call routing is used in this case. No new procedure is needed in this TR.

Scenario 2C: user has 2 E.164 MSISDNs, one is assigned by IMS and is stored in the HSS. The other MSISDN is assigned by CS domain and is stored in the HSS/HLR.

This case is considered that the user has 2 separate subscriptions. The USIM from the CS domain is used for registering to the CS domain. After that, this user can only be reached via the CS domain assigned E.164 number. It should be possible that the IMS domain deploys a type of call forwarding service so that all terminated call toward the IMS assigned E.164 could also be forwarded to the CS assigned E.164 number during unregistered state.

Case 3: user can be reached by both the Cellular or WLAN access and is registered to both domains.

Scenario 3A: user's E.164 MSISDN is assigned by the IMS and subscription is stored in the HSS.

During IMS registration, the UE needs to indicate to the MMCF that it has also registered to the CS domain. MMCF receive IMS registration and MAP_UPDATE_LOCATION from the VMSC. The PSTN routes the called party E.164 number toward the IMS network. The S-CSCF invokes the MMCF AS to determine whether the call should be continue routed using the IMS domain or divert the routing toward the VPLMN using the CS access. The decision could be based on e.g., static rule such as use the IMS if an ongoing VoIP session is in place, or based on user's preference or operator's preference.

Scenario 3B: user's E.164 MSISDN is assigned by the CS Domain and subscription is stored in the HSS/HLR.

During IMS registration, the UE needs to indicate to the MMCF that it has also registered to the CS domain. The IMSI indicates to the MMCF that the CS Domain registration is with the CS assigned E.164 number. The logic resided in the MMCF could invoke a MAP_Location_update toward the HSS/HLR to force all the call to be handled via the IMS side. If this is done, it also has to indicate to the UE that it should not perform Update_location procedure on the CS side. Otherwise, it will cancel the MMCF's location update. The decision for whether the MMCF's location update is invoked or not may depended on operator's or the user's preference. If MMCF location update is not performed then the terminating call will be handled via the CS domain as they are today. The user may have a choice to originate the call via CS or IMS, subject to the UE capabilities (e.g., UE may only not allow multiple voice call/sessions simultaneously on both the CS and IMS side).

Scenario 3C: user has 2 E.164 MSISDNs, one is assigned by IMS and is stored in the HSS. The other MSISDN is assigned by CS domain and is stored in the HSS/HLR.

This case is considered that the user has 2 separate subscriptions. During IMS registration, the UE needs to indicate to the MMCF that it has also registered to the CS domain. The IMSI indicates to the MMCF that the CS Domain registration is with the CS assigned E.164 number. In order to terminate CS assigned E.164 number in IMS domain, the logic resided in the MMCF would have to invoke a MAP_Location_update toward the HSS/HLR to force all the CS assigned E.164 call to be handled via the IMS side. In order to terminate IMS assigned E.164 number in the CS side, the logic resided in the MMCF would have to invoke call forwarding service to the CS assigned E.164 number. These logics must be coordinated and could be based on operator's or the user's preference.

Editor note: CS domain subscriber can contain multiple MSISDNs for different services in the subscription; the impact toward the services beside voice (i.e., fax, data, UDI) with MAP_UPDATE_LOCATION from IMS side. This is FFS!

Editor note: In IMS, multiple devices in addition to the dual mode UE can be registered with the same public user identity, how to ensure that the routing of terminating call to the dual mode UE which is now being served in the CS domain will not affect the other IMS devices that are registered using the same public user identity? This is FFS!

Editor note: Assuming the IMS and CS subscriptions are coming from separate domain, the assumption is also that both of these subscriptions are used within the same device. The MMCF has the role of registration synchronisation between domains. If the subscription can be separated, i.e., one device uses IMS, other physically separated device uses CS subscription, will there be any interaction issue to the service provided to each device respectively by each domain. This is FFS!

6.7.3 Registration

6.7.4 Origination

6.7.4.1 IMS origination

6.7.4.2 GSM/UMTS CS origination

6.7.5 Termination

6.7.5.1 IMS termination

6.7.5.2 GSM/UMTS CS termination

6.7.6 Handover Scenarios

6.7.6.1 CS UE to CS UE call

6.7.6.2 CS UE to IMS UE call

6.7.6.3 IMS UE to IMS UE call

6.7.6.4 IMS UE to CS UE call

6.7.7 Impact on Supplementary Services

6.7.8 Evaluation of the model

This clause presents the evaluation of the service continuity solution against the set of criteria.

6.8 Service Continuity Model: Mobility Management Application Server

6.8.1 General Description

The Mobility Management Application Server (MM-AS) defines a logical entity that enables roaming and handover functionality between the CS domain and the IMS.

The MM-AS acts as a "virtual" MSC/VLR to the GSM/UMTS domain, when a UE is registered in IMS.

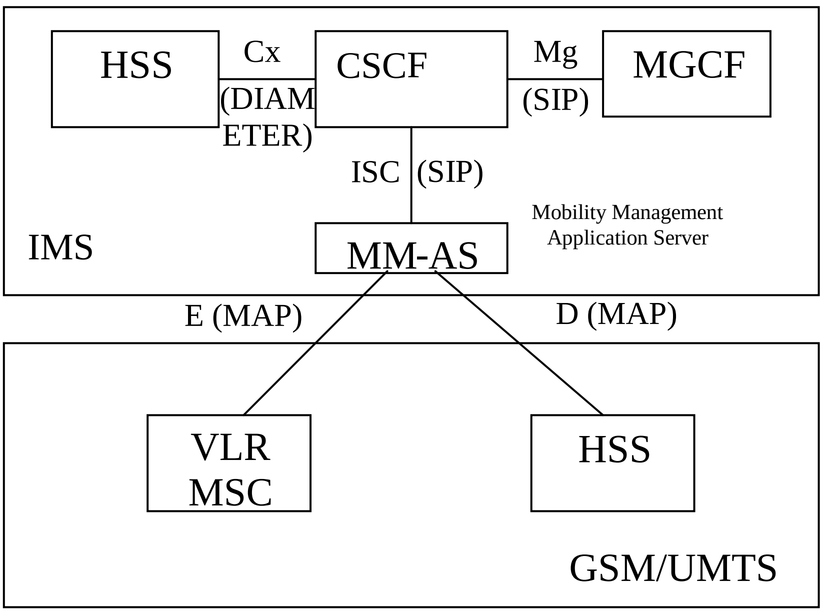


Figure 6.8.1: IMS-GSM/UMTS Inter-working Network Architecture

The proposed alternative concept does not require any changes to deployed CS domain infrastructure (HLR, MSC, etc.).

Editor's Note: Alignment with the reference architecture model required.

6.8.2 Routing Selection Decision

6.8.3 Registration

6.8.3.1 General

When a dual mode terminal is switched on, it attempts to register with the GSM/UMTS network (including authentication). After GSM/UMTS registration is successful or if certain network selection settings within the terminal indicate to search for I-WLAN access first, the terminal searches for I-WLAN and attempts for IMS registration.

6.8.3.2 Roaming between GSM/UMTS and IMS

A dual mode terminal periodically polls for I-WLAN signal and attempts to register with I-WLAN, if all selection criteria are fulfilled. When I-WLAN access registration is successful, the dual mode terminal attempts IMS registration.

The MM-AS is configured in the HSS user profile to receive REGISTER requests through the ISC interface.

The Figure 6.8.2 shows the roaming scenario for a dual mode handset moving from GSM/UMTS access to I-WLAN access.

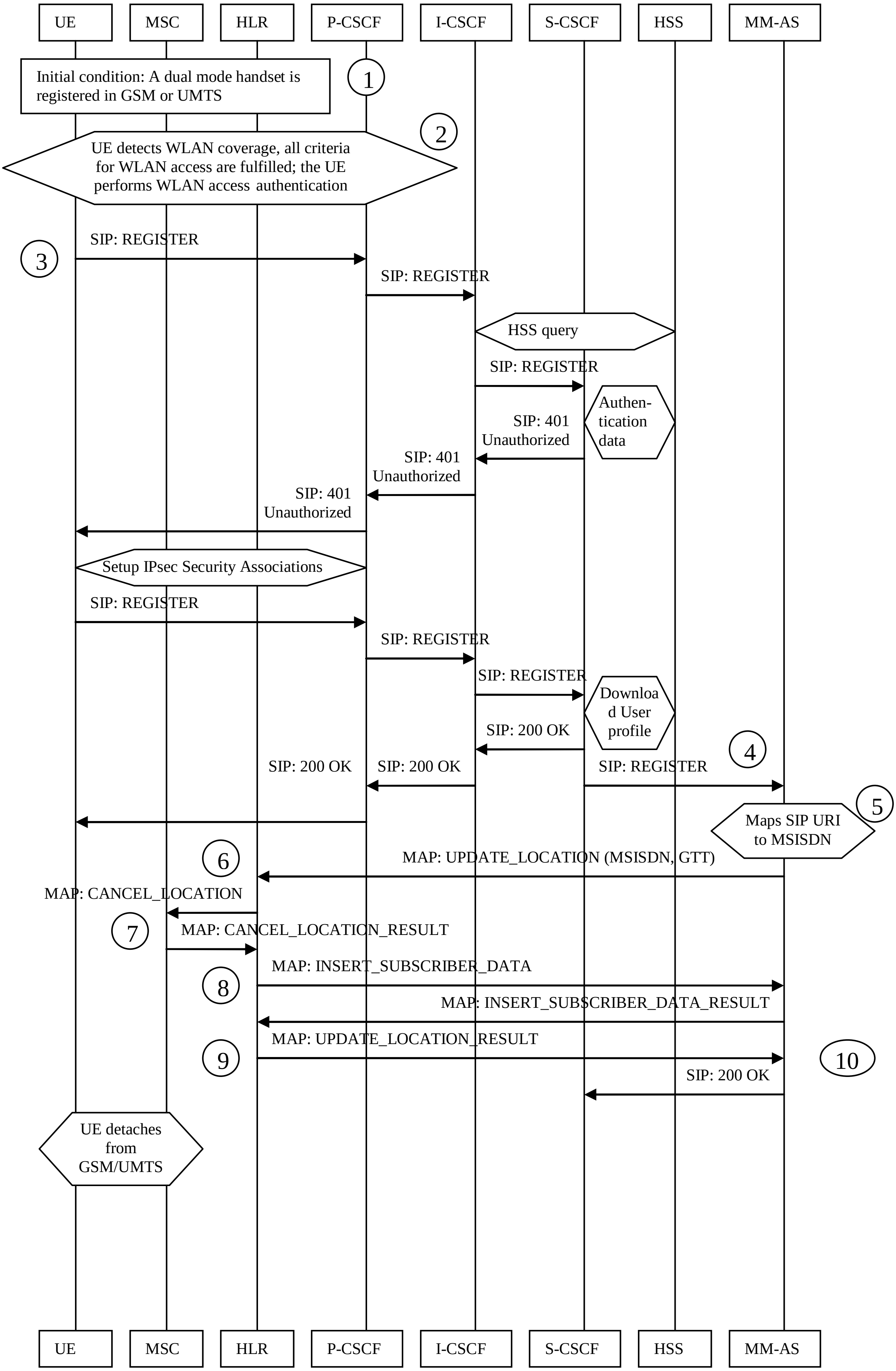


Figure 6.8.2: Roaming from GSM/UMTS to IMS

1. Initial condition: A dual mode UE is attached to the CS domain in GSM/UMTS
2. A dual mode UE enters a WLAN coverage area. The selection criteria for WLAN access are fulfilled. The UE performs WLAN access authentication and authorization and establishes a connection to the serving network (e.g. tunnel to the PDG in the HPLMN).
3. The UE sends a REGISTER request to the user's home IMS network to perform SIP registration (SIP: +49-89-636-12345@siemens.com, user = phone).
4. The MM-AS receives the REGISTER request.
5. The MM-AS maps the user's SIP URI (received in the REGISTER request) to MSISDN from its User Profile Database.
6. The MM-AS performs Global Title Translation on the MSISDN to determine the HLR and sends a MAP: UPDATE_LOCATION to the HLR.
7. The HLR sends MAP: CANCEL_LOCATION to the previous attached VLR/MSC.

Editor's note: It is for further study if the UE can be CS attached too.

8. The HLR sends the user's profile data through MAP: INSERT_SUBSCRIBER_DATA request(s).
9. The HLR acknowledges the MAP: UPDATE_LOCATION to the MM-AS.
10. Finally, the MM-AS accepts the registration with a 200 OK.

Now the user is registered successfully in the IMS and detaches from GSM/UMTS.

6.8.3.3 Roaming from IMS to GSM/UMTS

When a dual mode terminal is registered in IMS, it periodically checks the WLAN signal strength. When the UE senses drop of WLAN signal or because of certain network selection settings, it attempts to register to GSM/UMTS.

The figure 6.8.3 shows the roaming scenario for a dual mode handset moving from WLAN (IMS) access to GSM/UMTS access to support voice call continuity.

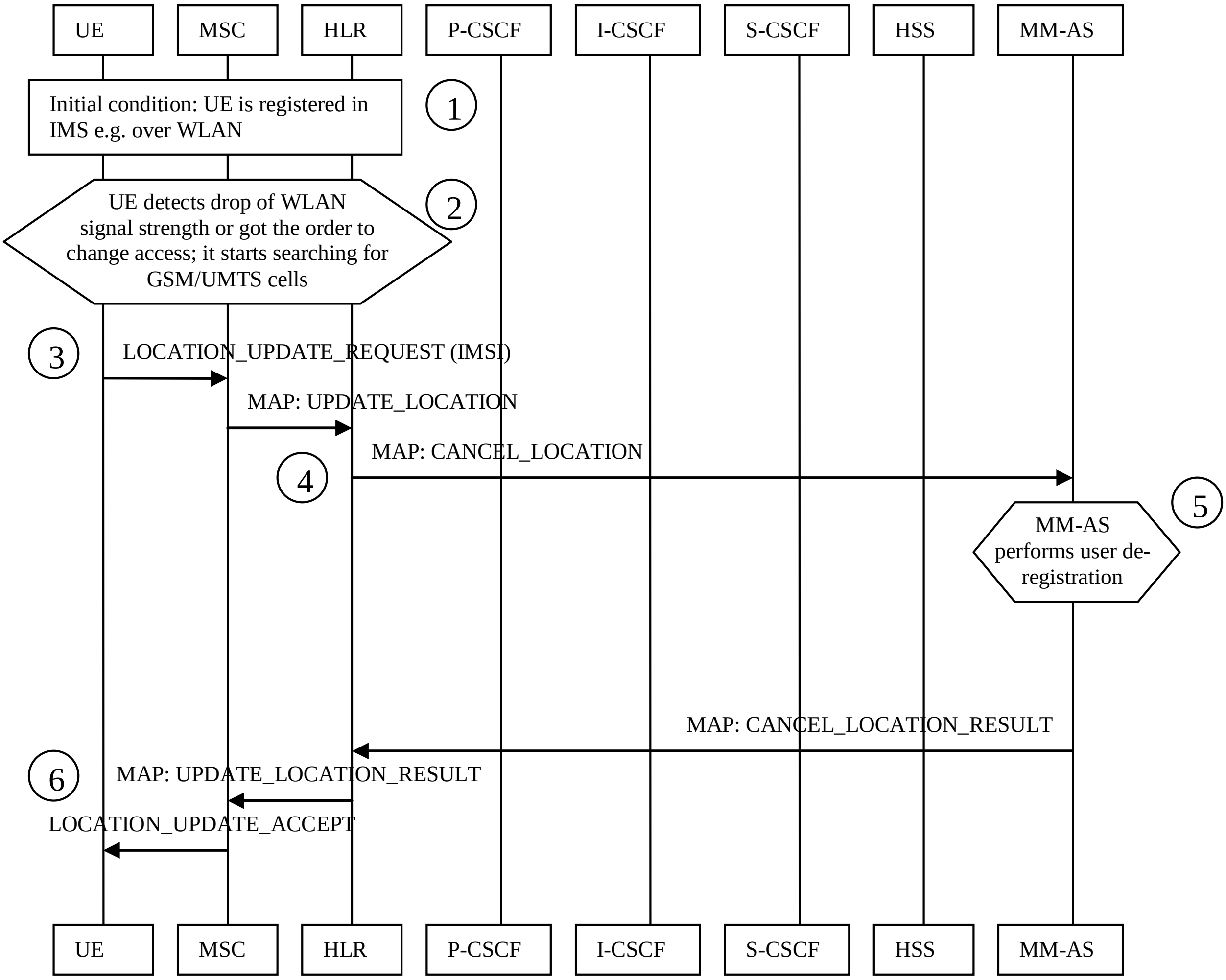


Figure 6.8.3: Roaming from IMS to GSM/UMTS

1. Initial condition: A dual mode UE is registered in IMS e.g. over WLAN.
2. UE detects drop of WLAN signal strength or got the order to change access; it starts searching for GSM/UMTS cells.
3. If the UE finds a suitable GSM/UMTS cell, it performs IMSI attach by sending a LOCATION_UPDATE_REQUEST.
4. The HLR informs the MM-AS about the IMSI attach by sending a MAP: UPDATE_LOCATION towards the MM-AS.
5. The MM-AS initiates IMS de-registration for this user. During this "Network Initiated De-registration by Service Platform" procedure, the UE is informed by S-CSCF about the De-registration.
6. The MM-AS acknowledges the HLR by a MAP: CANCEL_LOCATION_RESULT. The user receives the LOCATION_UPDATE_ACCEPT.
7. UE performs IMS de-registration if WLAN signal is still available.

6.8.4 Origination

6.8.4.1 IMS origination

6.8.4.1.1 Mobile Originating Call from IMS to IMS

The MM-AS acts as a "virtual" MSC/VLR to the GSM/UMTS domain, when a UE is registered in IMS.

1. Initial condition: A dual mode terminal is registered in IMS
2. UE1 initiates a VoIP call to UE2, which is also IMS registered.
3. The MM-AS is contacted on the basis of initial filter criteria, which were downloaded by the S-CSCF from HSS during UE1 registration.
4. The MM-AS is contacted on the basis of initial filter criteria, which were downloaded by the S-CSCF from HSS during UE2 registration, i.e. UE2 is also a dual mode terminal.
5. UE2 receives the INVITE and responds with a 200 OK.
6. UE1 receives 200 OK from UE2 and responds with the final ACK.
7. UE2 receives the final ACK, which establishes the call.

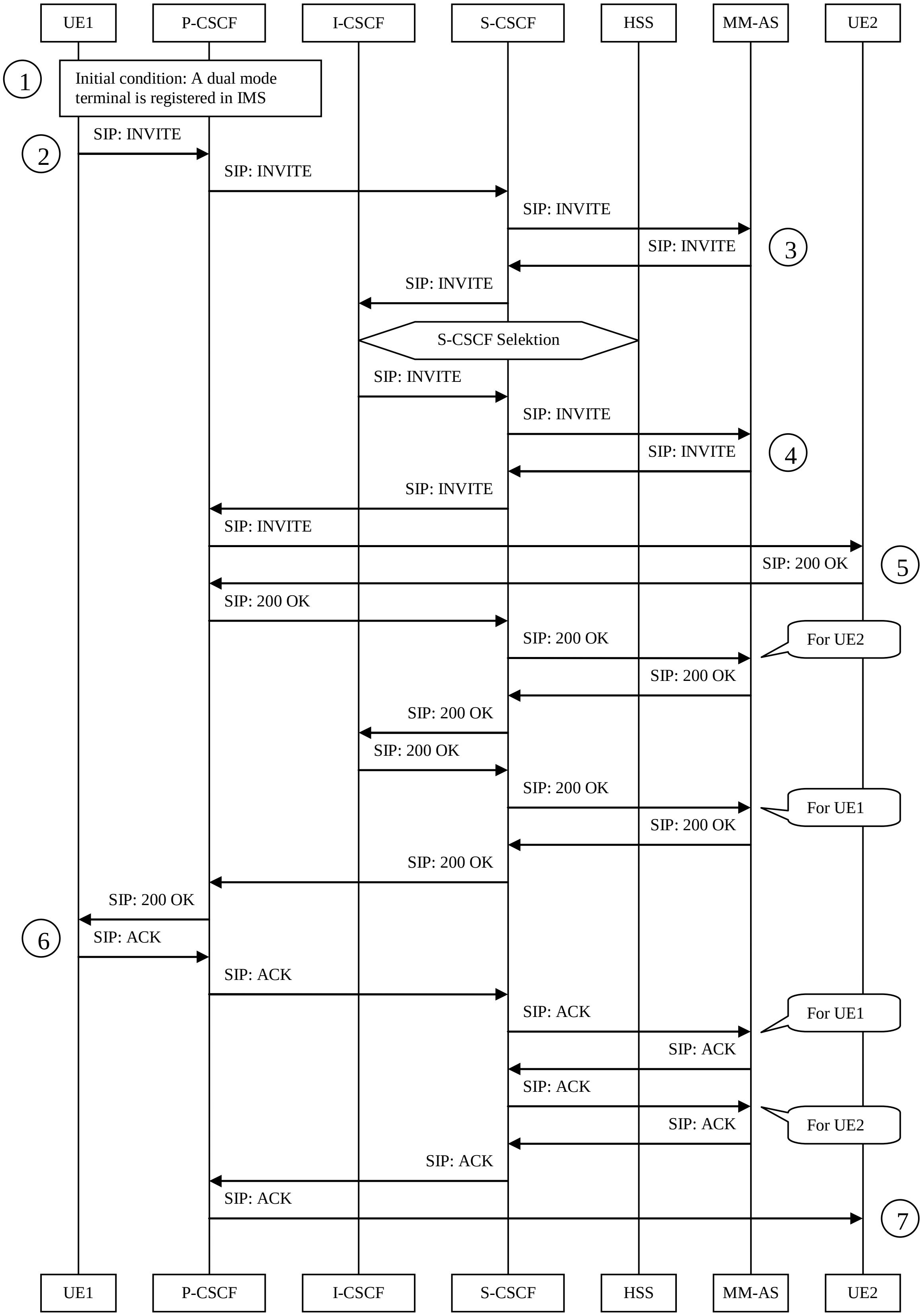


Figure 6.8.4: Mobile Originating Call from IMS to IMS

6.8.4.1.2 Mobile Originating Call from IMS to GSM/PSTN

The MM-AS acts as a "virtual" MSC/VLR to the GSM/UMTS domain, when a UE is registered in IMS.

1. Initial condition: A dual mode terminal is registered in IMS

2. The UE initiates a call to a user in GSM or PSTN.
3. The MM-AS is contacted on the basis of initial filter criteria, which were downloaded by the S-CSCF from HSS during UE registration.
4. The MGCF receives the INVITE and sends an ISUP: INITIAL_ADDRESS_MESSAGE (IAM) to the GMSC and PSTN.
5. The GSMC/PSTN sends an ISUP: ADDRESS_COMPLETE_MESSAGE to the MGCF, which sends a SIP: 180 Ringing towards the UE.
6. The ISUP: ANSWER_MESSAGE indicates that the call is confirmed.
7. The UE receives a 200 OK and responds with the ACK, the call is now established.
8. The S-CSCF forwards the ACK to the MM-AS.

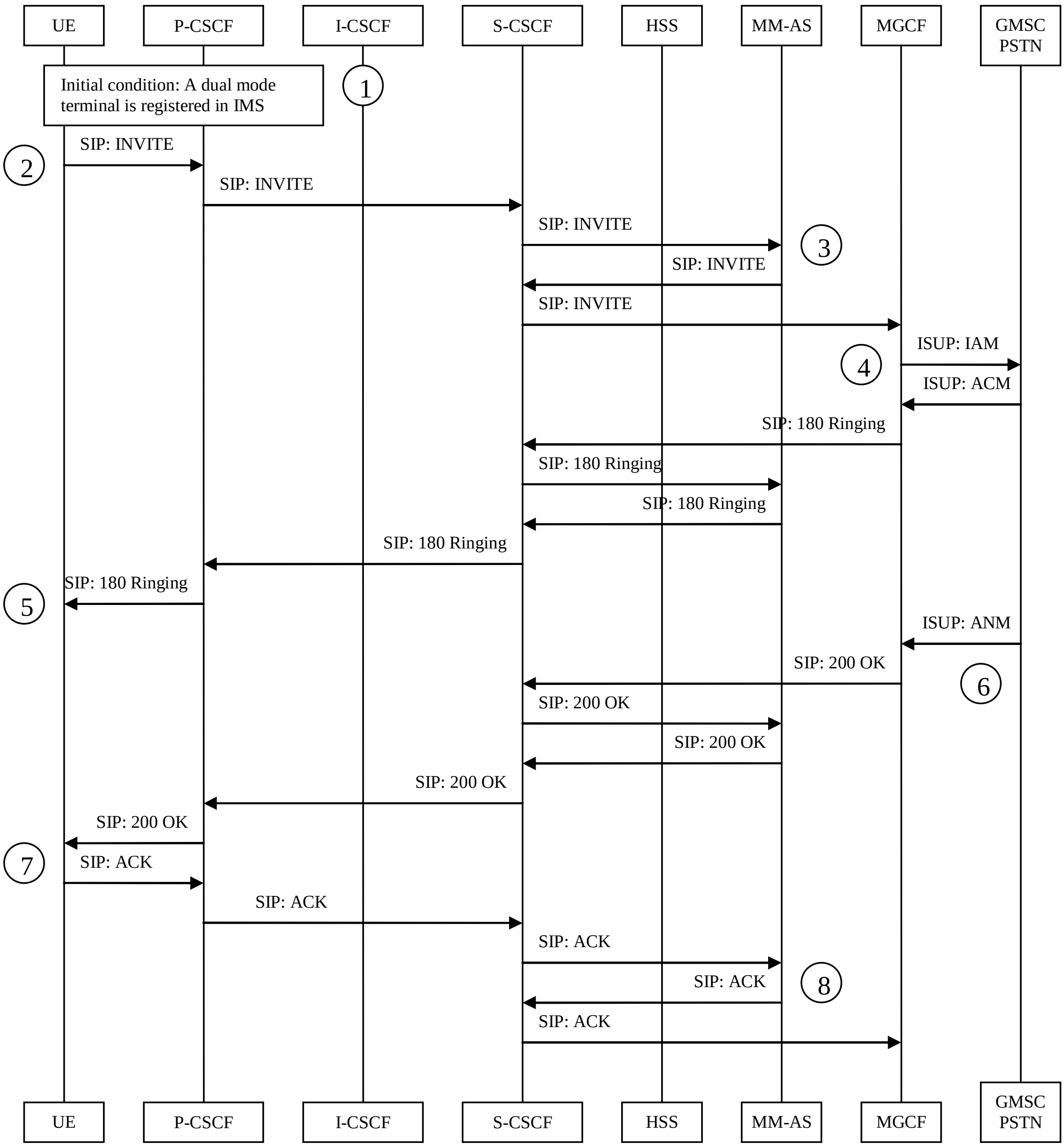


Figure 6.8.5: Mobile Originating Call from IMS to GSM/PSTN

6.8.4.2 GSM/UMTS CS origination

6.8.5 Termination

6.8.5.1 IMS termination

The MM-AS acts as a "virtual" MSC/VLR to the GSM/UMTS domain, when a UE is registered in IMS.

1. When the MTC is received, the call is routed to the GMSC (since the number is owned by the GSM domain).
2. GMSC queries the HLR to retrieve a roaming number from the serving MSC/VLR through MAP: SEND_ROUTING_INFO.
3. HLR requests the MM-AS to generate a Roaming Number (MSRN) for the UE through MAP: PROVIDE_ROAMING_NUMBER procedure.
4. If the user is still registered in IMS, the MM-AS generates a MSRN, which allows the GMSC to route to a MGCF in the roaming network. Furthermore, the MSRN should be coded in such a way that the MSISDN can be derived from it.
5. The MM-AS sends the MSRN within the MAP: ROVIDE_ROAMING_NUMBER_ACK to the HLR, the HLR to the GMSC. The GMSC sends an ISUP IAM, which terminates at the MGCF.
6. The MGCF uses the MSRN to generate a SIP: INVITE containing the MSISDN of the user as a tel URL towards the I-CSCF.

Editor's Note: It is FFS, if number translation or ENUM query to derive the tel URL.

7. The MM-AS is contacted on the basis of initial filter criteria, which were downloaded by the S-CSCF from HSS during registration.
8. The UE receives the terminating call, starts ringing (not shown in figure 1) and sends the 200 OK.
9. (and 9') The MM-AS is involved in the whole SIP message flow.
10. The MGCF maps the 200 OK to an ISUP: ANM and sends it towards the GMSC.
11. The UE receives the final ACK, which finally establishes the terminating call.

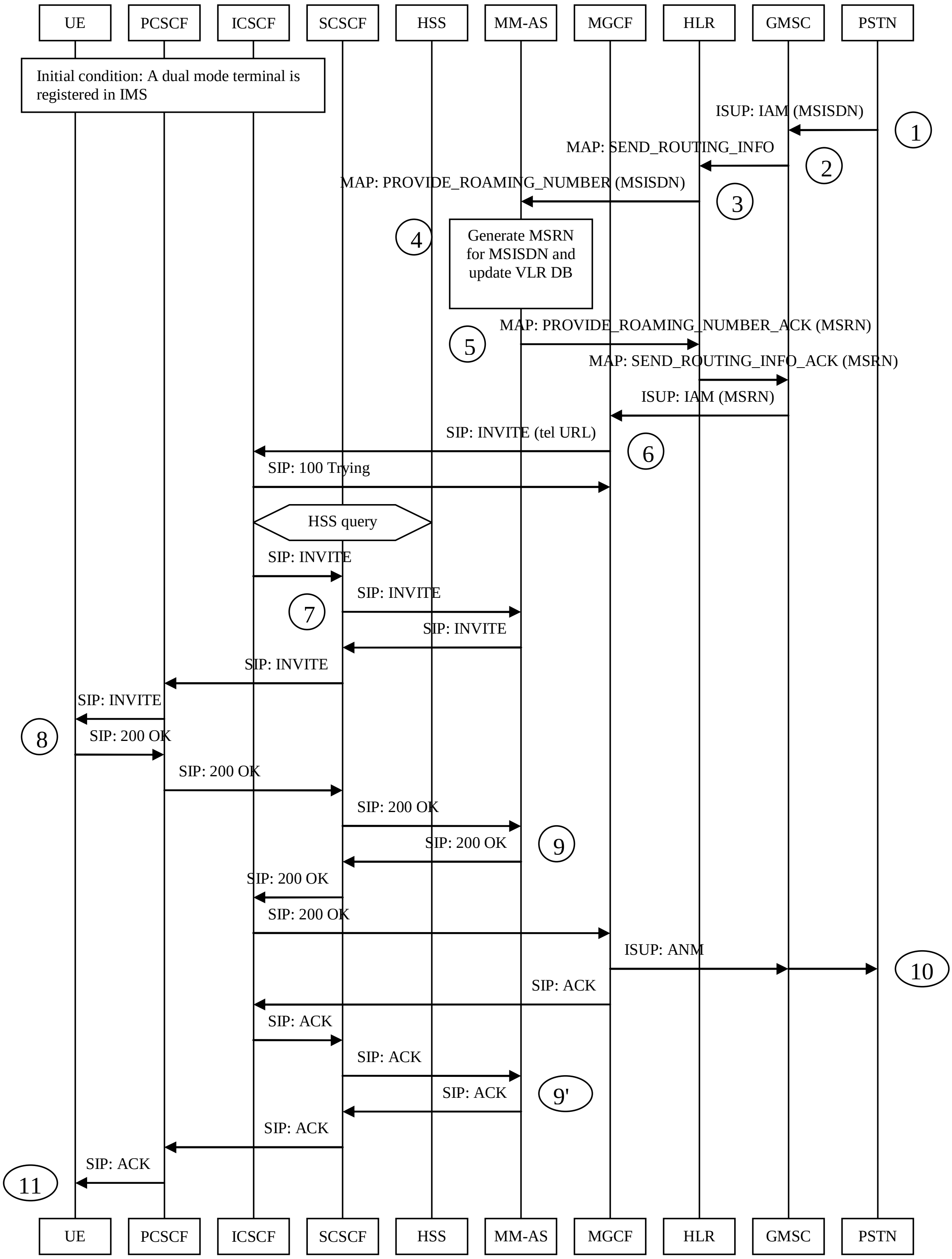


Figure 6.8.6: Terminating Call when UE is registered in IMS

6.8.5.2 GSM/UMTS CS termination

6.8.6 Handover Scenarios

6.8.6.1 CS UE to CS UE call

6.8.6.2 CS UE to IMS UE call

6.8.6.2.1 Bearer Path: Established call with a PSTN subscriber

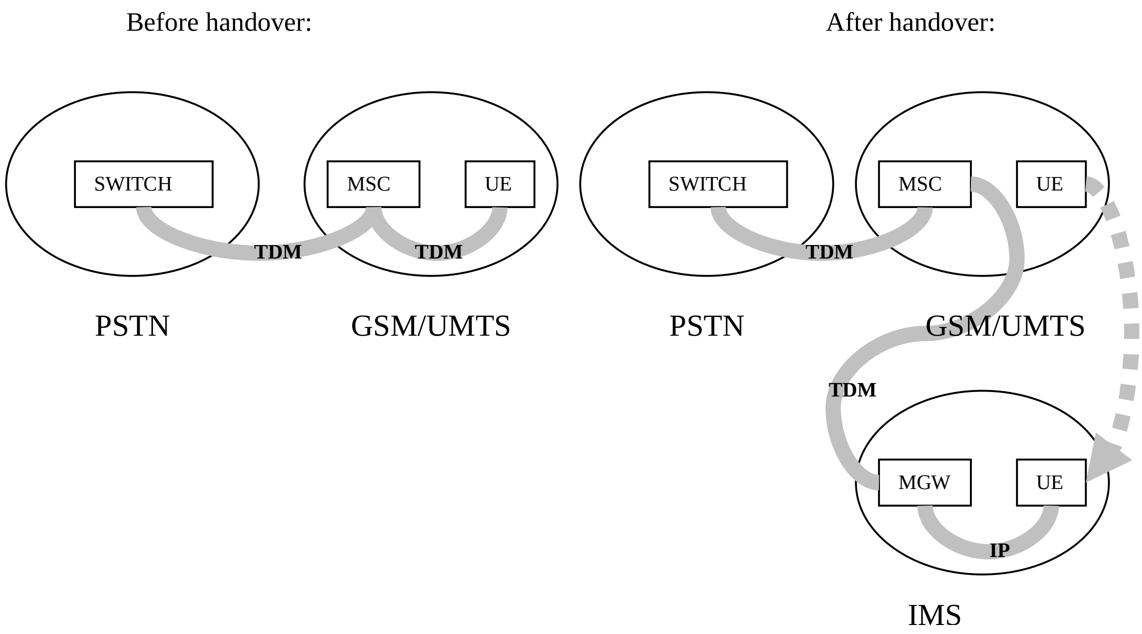


Figure 6.8.7

6.8.6.2.2 Bearer Path: Established call with GSM or UMTS subscriber

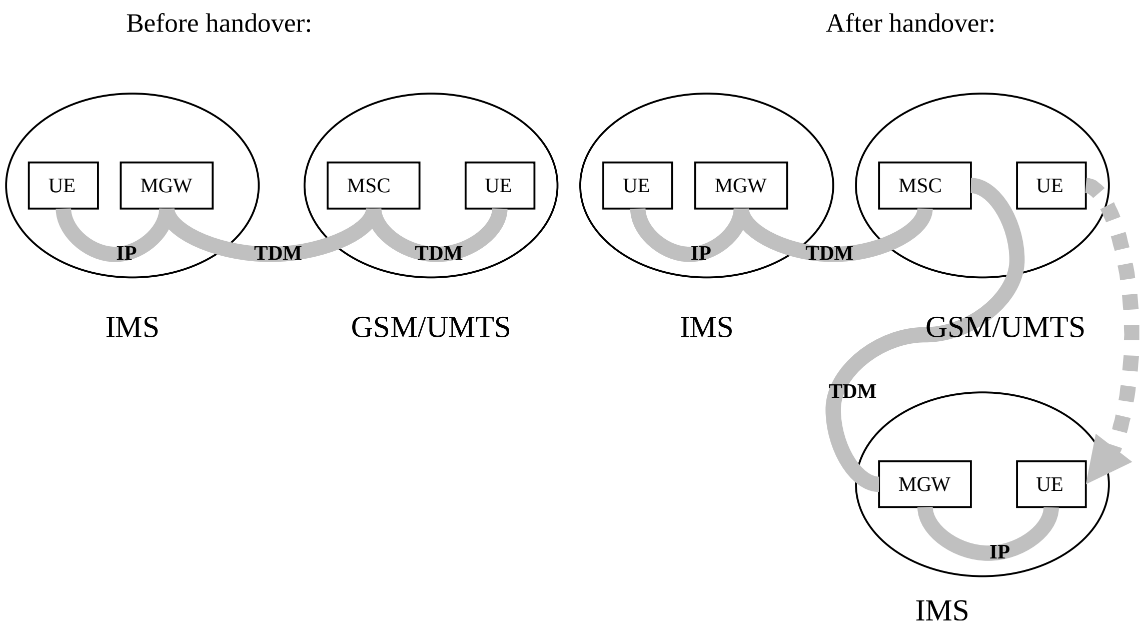


Figure 6.8.8

6.3.1.3 Message Flow: From GSM/UMTS to IMS

In this scenario, the dual mode UE has an active call to a PSTN subscriber. Periodically, the UE scans the WLAN signal and when it detects WLAN with sufficient signal strength, it may attempt to attach WLAN.

1. After the UE has performed WLAN access and service authentication, it performs IMS registration. The REGISTER request may contain already an indication to prepare handover. Alternatively, a subsequent SIP message exchange could indicate the handover.

2. The MM-AS is notified about the IMS registration and is informed about the used IP-CAN-CGI. This CGI enables later the MSC to route MAP HO requests and ISUP IAM to the MM-AS.

3. The UE may force handover for the GSM/UMTS call with RRC Measurement Reports indicating the IP-CAN-CGI with the highest signal level. This should trigger the BSC/RNC to request handover to the IP-CAN-CGI by sending an HO Request message to the MSC.

Editor's Note: The neighbour cell list of the BSC/RNC has to be configured in such a way to support a proper handling of the IP-CAN-CGI. enable the correct routing to the MSC.

4. The MSC sends a MAP Prepare Handover Request message to the MM-AS by using the IP-CAN-CGI.

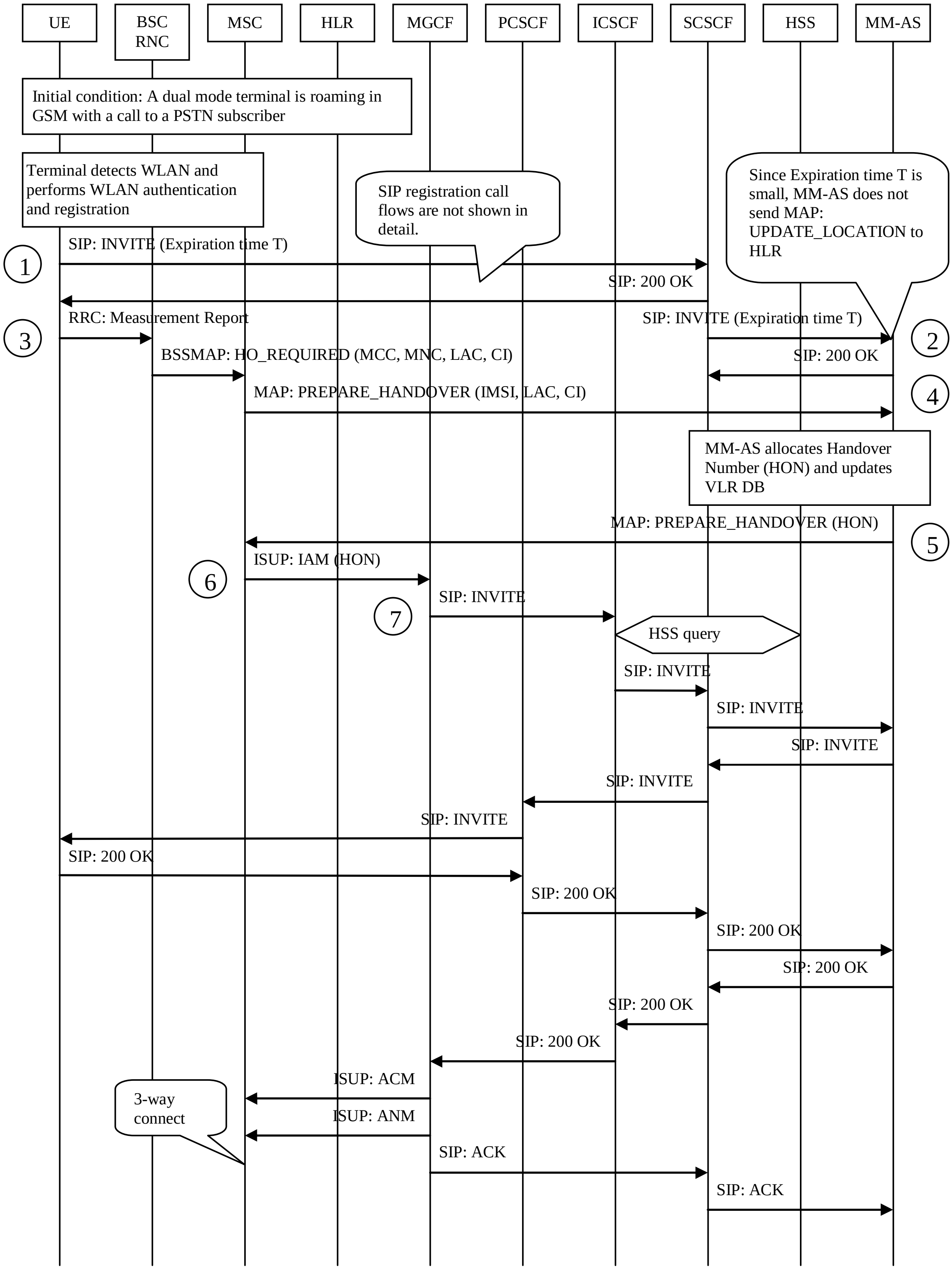
5. The MM-AS returns a MAP Prepare Handover Response containing an E.164 Handover number.

6. The serving MSC establishes an ISUP call to the handover number generated in the previous step. The call is terminated in the MGCF, which routes the SIP INVITE to the I-CSCF based on the SIP URI or Tel URI.

7. Call setup continues similar to the Mobile Terminating Call procedure.

8. After the call is successfully established, the MM-AS sends the MAP: SEND_END_SIGNAL procedure to the MSC that causes the GSM/UMTS resources to be released. Now the call has been handed over to IMS (WLAN).

9. After call is terminated, the UE performs an IMS registration. This would cause the MM-AS to update the location data in the HLR.


UE
BSC RNC
MSC
HLR
MGCF
PCSCF
ICSCF
SCSCF
HSS
MM-AS
Initial condition: A dual mode terminal is roaming in GSM with a call to a PSTN subscriber
Terminal detects WLAN and performs WLAN authentication and registration
SIP registration call flows are not shown in detail.
Since Expiration time T is small, MM-AS does not send MAP: UPDATE_LOCATION to HLR
1
SIP: INVITE (Expiration time T)
SIP: 200 OK
3
RRC: Measurement Report
SIP: INVITE (Expiration time T)
2
BSSMAP: HO_REQUIRED (MCC, MNC, LAC, CI)
SIP: 200 OK
4
MAP: PREPARE_HANDOVER (IMSI, LAC, CI)
MM-AS allocates Handover Number (HON) and updates VLR DB
MAP: PREPARE_HANDOVER (HON)
5
6
ISUP: IAM (HON)
7
SIP: INVITE
HSS query
SIP: INVITE
SIP: INVITE
SIP: INVITE
SIP: INVITE
SIP: INVITE
SIP: 200 OK
SIP: 200 OK
SIP: 200 OK
SIP: 200 OK
SIP: 200 OK
SIP: 200 OK
ISUP: ACM
3-way connect
ISUP: ANM
SIP: ACK
SIP: ACK

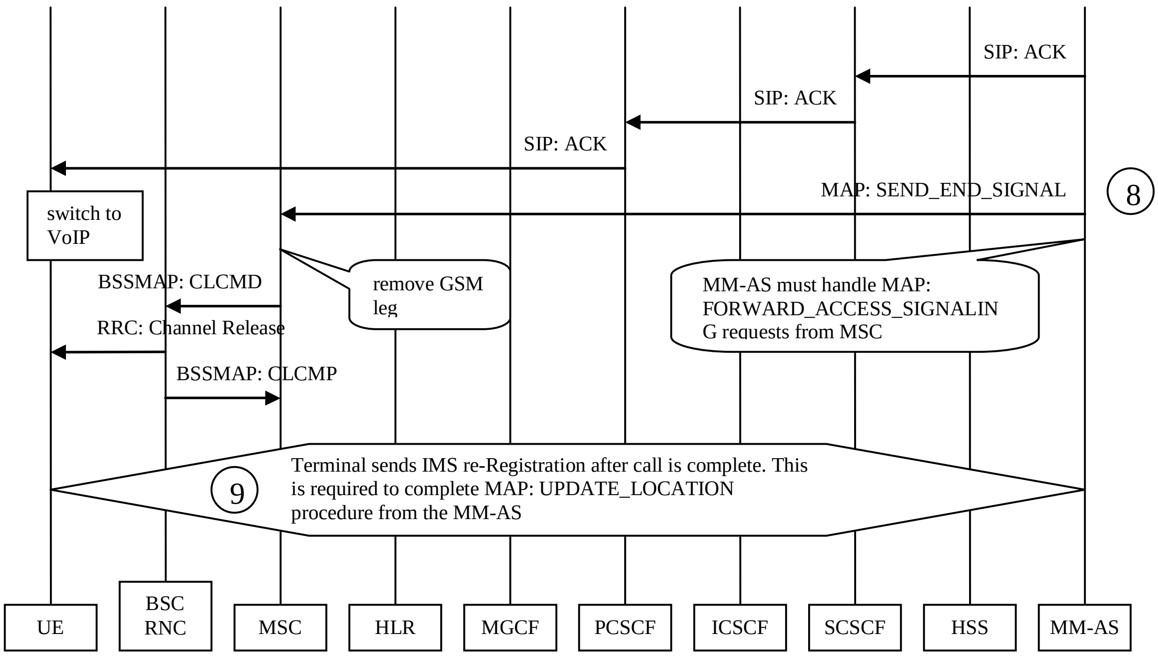


Figure 6.8.9: Handover from GSM/UMTS to IMS

Editor's Note: The message flow does not consider the support of supplementary services.

6.8.6.3 IMS UE to IMS UE call

6.8.6.4 IMS UE to CS UE call

6.8.6.4.1 Bearer Path: Established call with a PSTN subscriber

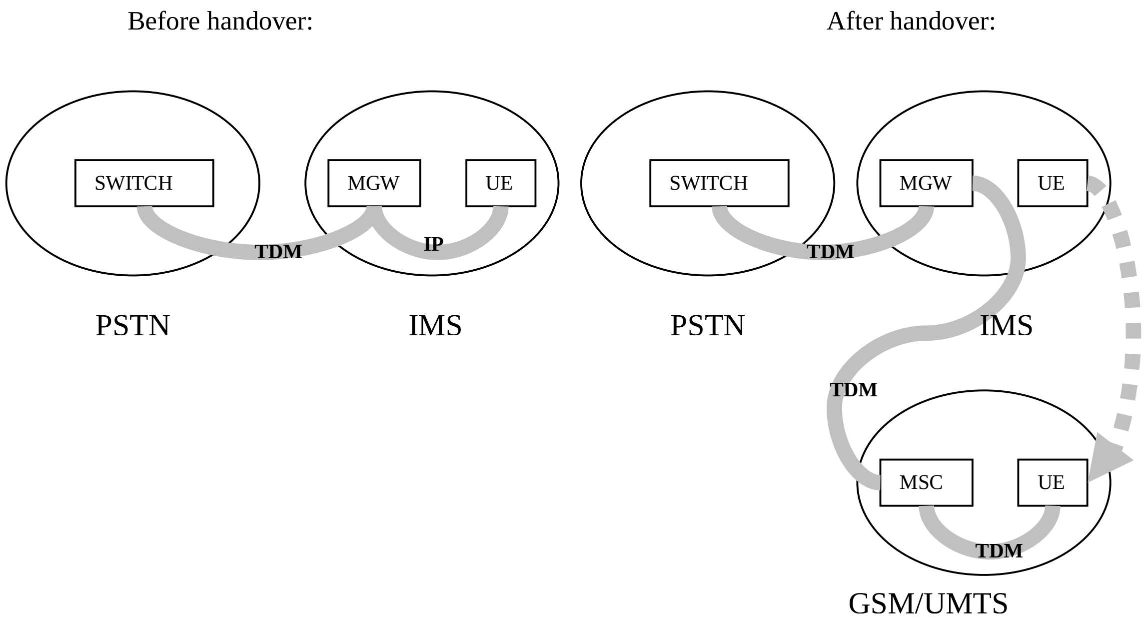


Figure 6.8.10

6.8.6.4.2 Bearer Path: Established call with a IMS subscriber

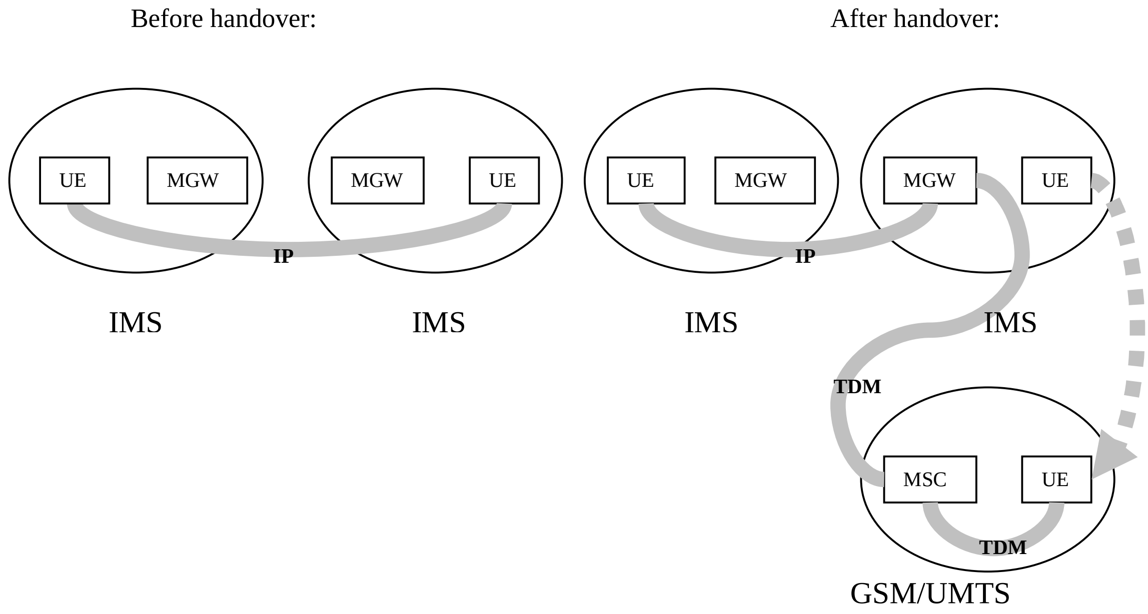


Figure 6.8.11

6.8.6.4.3 Message Flow: From IMS to GSM/UMTS

The initial condition is an active call with a PSTN or IMS subscriber, the UE has an active IMS session over I-WLAN. The S-CSCF forwards SIP messages to the MM-AS based on filter criteria:

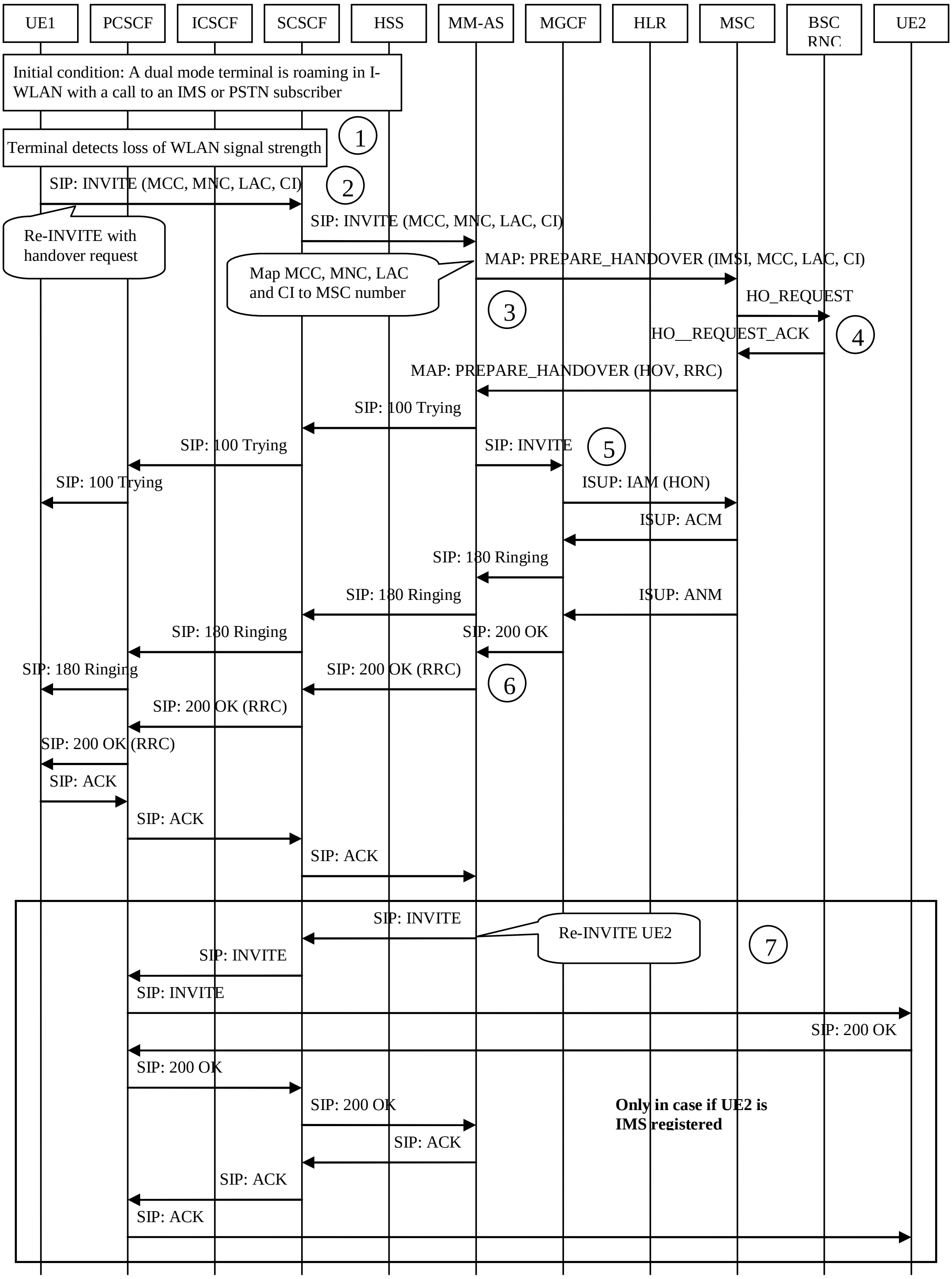
1. The UE periodically polls the WLAN signal strength to decide, if a handover to GSM/UMTS is required.
2. UE requests handover to GSM/UMTS through a re-INVITE in the context of the existing SIP dialog. The re-INVITE contains the desired GSM/UMTS parameters like MCC, MNC, LAC, and CI.

Editor's Note: In UMTS it is FFS how the MM-AS gets the RNC-id.

3. MM-AS sends a MAP: PREPARE_HANDOVER request to the MSC based on the CGI.
4. The MSC sends a HANDOVER_REQUEST to the serving BSC/RNC. After the BSC/RNC acknowledges the request, the MSC responds back to the MM-AS with a Handover number (HON) and the necessary RRC information for the target cell.
5. MM-AS request the MGCF to establish an outgoing ISUP call to the MSC with the Handover Number (HON) by sending a SIP: INVITE. This establishes the bearer towards the target MSC.

Editor's Note: Clarify if the proposed handover scenario is different from existing handover procedures.

6. MM-AS transfers the RRC information to the UE within the SIP: 200 OK.
7. Only in the case if UE2 is registered in IMS: The MM-AS re-INVITEs UE2 to establish a direct UE connection.
8. After ISUP call establishment is completed, the UE tries to access the target GSM/UMTS cell.
9. When handover to GSM/UMTS is complete, the BSC/RNC sends HO-Complete to the MSC. The MSC sends a MAP: Send End Signal to the MM-AS.
10. Only in the case if UE2 is in the PSTN: The MM-AS issues a SIP: REFER (Call Transfer) command to the MGCF to join the two ISUP call legs (with MSC and PSTN). The voice path in the UE will be interrupted for the time needed by the MGCF to perform this operation.
11. The MM-AS releases the IMS connection towards UE1.


UE1
PCSCF
ICSCF
SCSCF
HSS
MM-AS
MGCF
HLR
MSC
BSC RNC
UE2
Initial condition: A dual mode terminal is roaming in I-WLAN with a call to an IMS or PSTN subscriber
1
Terminal detects loss of WLAN signal strength
SIP: INVITE (MCC, MNC, LAC, CI)
2
Re-INVITE with handover request
SIP: INVITE (MCC, MNC, LAC, CI)
MAP: PREPARE_HANDOVER (IMSI, MCC, LAC, CI)
Map MCC, MNC, LAC and CI to MSC number
3
HO_REQUEST
HO__REQUEST_ACK
4
MAP: PREPARE_HANDOVER (HOV, RRC)
SIP: 100 Trying
SIP: 100 Trying
SIP: INVITE
5
SIP: 100 Trying
ISUP: IAM (HON)
ISUP: ACM
SIP: 180 Ringing
SIP: 180 Ringing
ISUP: ANM
SIP: 180 Ringing
SIP: 200 OK
SIP: 180 Ringing
SIP: 200 OK (RRC)
6
SIP: 200 OK (RRC)
SIP: 200 OK (RRC)
SIP: ACK
SIP: ACK
SIP: ACK
SIP: INVITE
Re-INVITE UE2
7
SIP: INVITE
SIP: INVITE
SIP: 200 OK
SIP: 200 OK
SIP: 200 OK
Only in case if UE2 is IMS registered
SIP: ACK
SIP: ACK
SIP: ACK

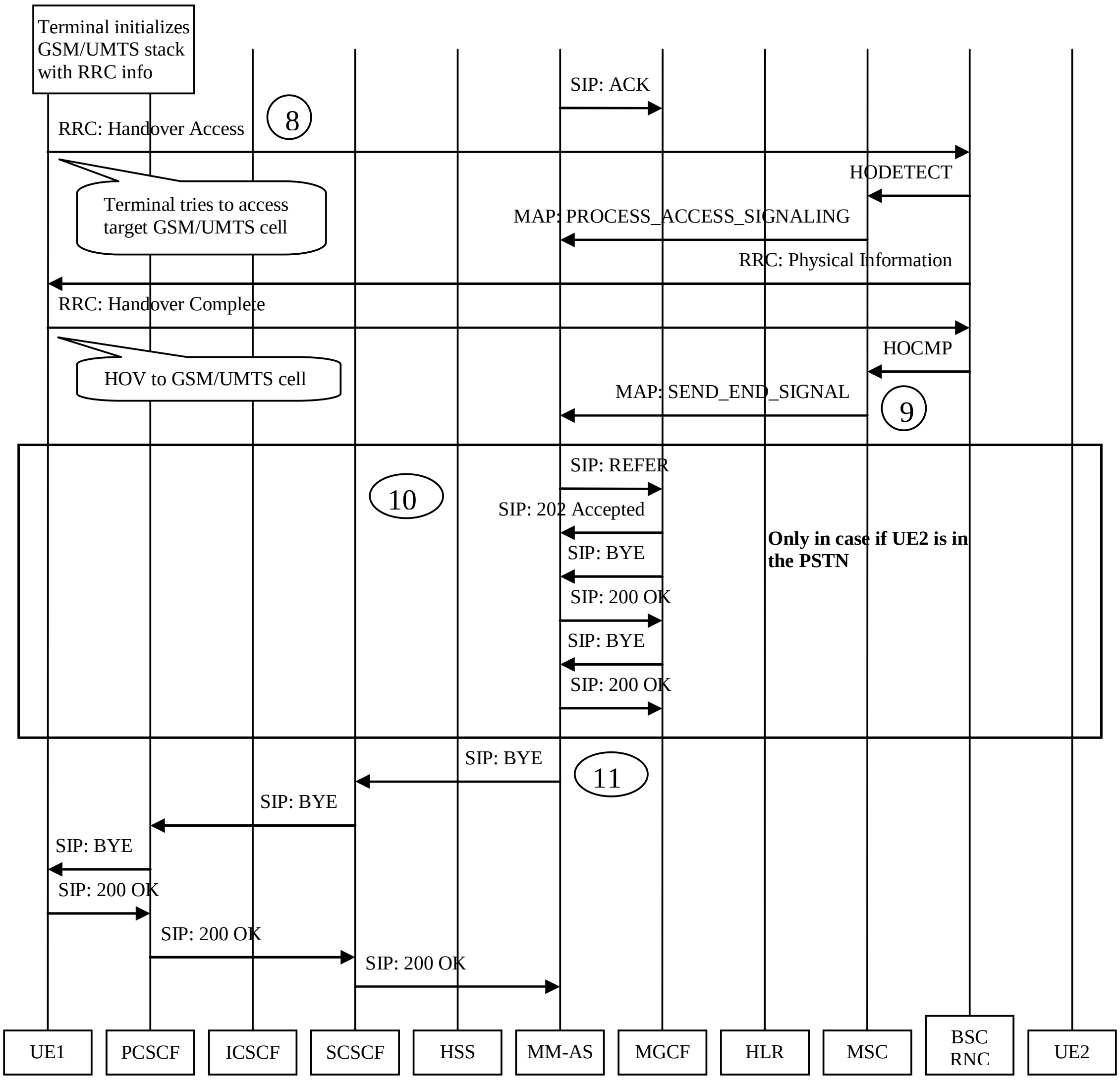


Figure 6.8.12: Handover from IMS to GSM/UMTS

6.8.7 Impact on Supplementary Services

6.8.8 Evaluation of the model

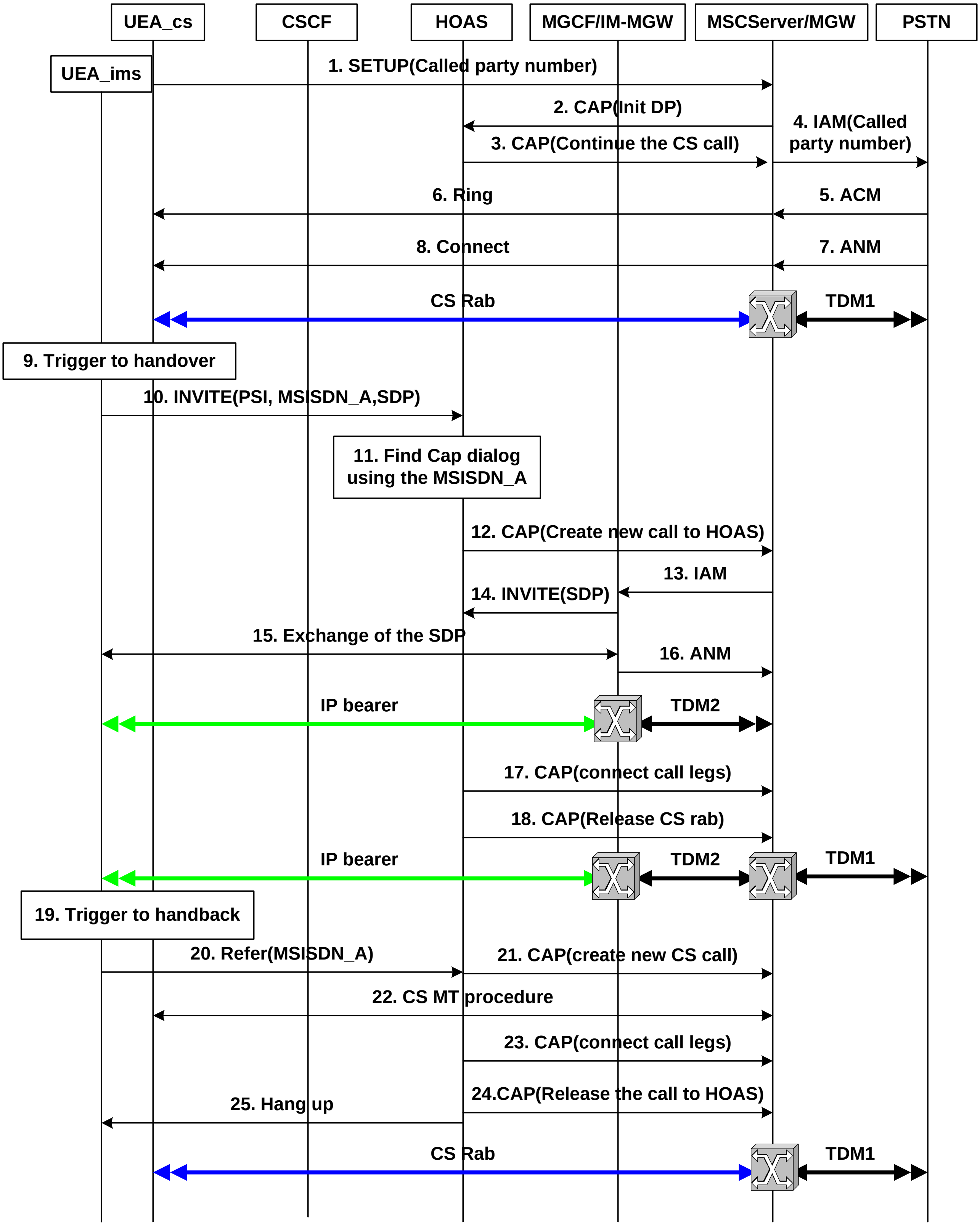
This clause presents the evaluation of the service continuity solution against the set of criteria

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

6.9.1 General Description

This section describes CS-IMS subscriber handover from the CS domain to the IMS. This section focuses on the scenario that a CS-IMS subscriber has an active call in the CS domain to a PSTN subscriber and the CS-IMS subscriber decide to handover from the CS domain to the IMS. The scenario which the called party is a IMS subscriber is very similar with this scenario.

The following steps show the handover procedure from CS domain to IMS.


UEA_cs
CSCF
HOAS
MGCF/IM-MGW
MSCServer/MGW
PSTN
UEA_ims
1. SETUP(Called party number)
2. CAP(Init DP)
3. CAP(Continue the CS call)
4. IAM(Called party number)
6. Ring
5. ACM
8. Connect
7. ANM
CS Rab
TDM1
9. Trigger to handover
10. INVITE(PSI, MSISDN_A,SDP)
11. Find Cap dialog using the MSISDN_A
12. CAP(Create new call to HOAS)
13. IAM
14. INVITE(SDP)
15. Exchange of the SDP
16. ANM
IP bearer
TDM2
17. CAP(connect call legs)
18. CAP(Release CS rab)
IP bearer
TDM2
TDM1
19. Trigger to handback
20. Refer(MSISDN_A)
21. CAP(create new CS call)
22. CS MT procedure
23. CAP(connect call legs)
24.CAP(Release the call to HOAS)
25. Hang up
CS Rab
TDM1

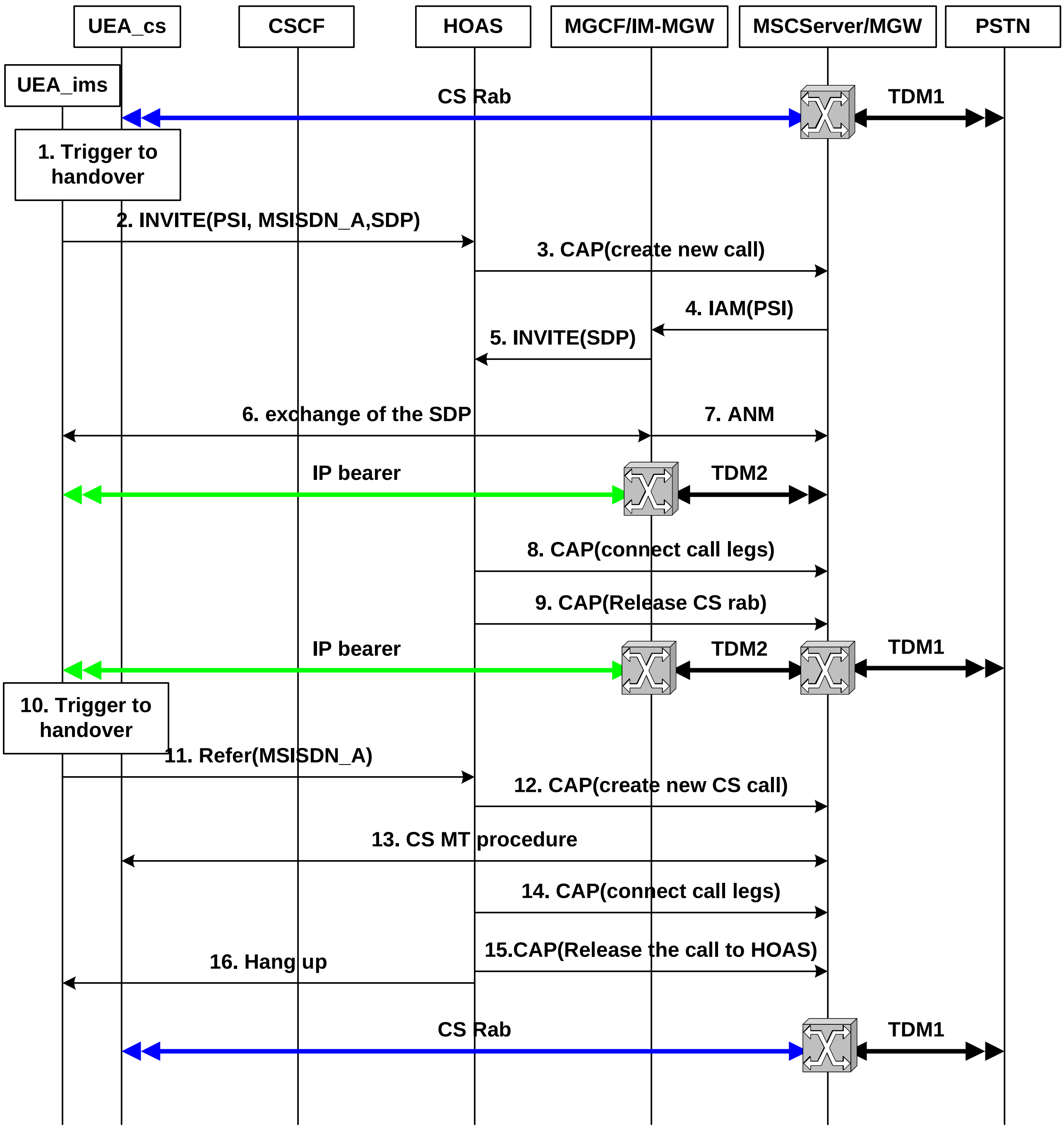


Figure 6.8.13

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 dialog to HOAS according to the camel criteria in the MSC Server.
3. The HOAS indicated the MSC Server to continue the CS call to the PSTN user.
4. The MSC Server send a IAM to the PSTN user.
5. The PSTN user send a ACM to the MSC Server to indicate that the PSTN user is ring.
6. After receiving the ACM then the MSC Server send a Ring to the user A.
7. When the PSTN user answer the CS call it shall send ANM to the MSC Server.
8. The MSC Server send a Connect to the user A and connect the bearer paths

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 DTM between the MSC Server/MGW and PSTN, as described in Figure 1. During the CS call procedure the MSC Server establishes a CAP dialog to HOAS.

9. According to the radio condition and/or user preference, the user A decides to handover from CS domain to IMS. User A register in IMS firstly.

NOTE: The registration in IM domain may increase the duration of handover. It is FFS how to handle this issue.

10. After the registration, user A send a INVITE request to the IMS core network(PSI of the HOAS, 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 HOAS. The PSI of the HOAS can be automatically transferred in the registration or statically configured in the user A.

11. The HOAS store the initial SDP included in the INVITE request. Using the MSISDN of user A, the HOAS find the CAP dialog established in the CS domain.

12. After find the CAP dialog the HOAS shall indicate the MSC Server through this CAP dialog to create a CS call leg to the HOAS□This message include the Tel number of HOAS.

13. Once receive the indication, the MSC select a proper MGCF and send an IAM to this MGCF, the called number is the PSItel number of HOAS, as indicated by the HOAS.

14. After receive the IAM the MGCF shall translate the tel number of HOAS to a tel: URI,which is the PSI of the HOAS. Then the HOAS shall initiate a INVITE request(PSI of the HOAS, MSISDN of user A, SDP). The INVITE request is then routed to the HOAS.

15. When receive the INVITE request, the HOAS shall find the session originated by the user A in step 1 according to the MSISDN of user A. The HOAS shall exchange the SDP between the user A and the MGCF. After the exchange the bearer path between the user A and the IM-MGW controlled by the MGCF is established.

16. After establishment of the bearer path(IP bearer) between the user A and the IM-MGW, the MGCF send a ANM to the MSC Server. So the bearer path(DTM2) between the MSC Server/MGW and IM-MGW is established.

17. The HOAS shall indicate the MSC through the CAP dialog 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

18. The HOAS shall indicate the MSC through the CAP dialog to release the CS connection between the MSC Server/MGW and user A in CS domain. When the user A receive the release message it shall change it's voice channel to IM domain.

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

19. According the radio condition and user preference, user A decide to handback to CS domain.

20. User A will send a REFER to HOAS using the existing session in IMS, containing "Refer-To" the MSISDN of user A.

21. When the HOAS receive the REFER message, the HOAS shall look for whether the CAP dialog of the user A have already existed. If it find the CAP dialog then the HOAS shall indicate through the dialog the MSC to originate a CS call leg to user A in CS domain. If it can not find the CAP dialog it shall reject the handover request.

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

NOTE: This step which including the paging procedure may increase the duration of handover. 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 HOAS shall indicate the MSC 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 HOAS shall indicate the MSC Server to release the call leg from MSC Server to HOAS.

25. the HOAS shall release the IMS session from user A to HOAS. 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.

6.9.8 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 handover 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 handover. After the handover from CS domain to IM domain, the IM service in IM domain can also provide to the user. The Camel dialog exist during the whole call so it is possible that the camel logic in HOAS can also control this voice call even after the handover.

The drawbacks of this solution are:

1. The MSC Server must support Camel 4 service.
2. The user which want to handover between the CS domain and IMS must have camel 4 subscription.

7 Security

8 Charging

9 Comparison of the Architecture Model

10 Conclusion

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 handover procedure was described and discussed based on it.

A.1.1 Logical Model Introduction

For the flexibilities from IMS network, handover 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 handover is proposed shown in the figure A.1.

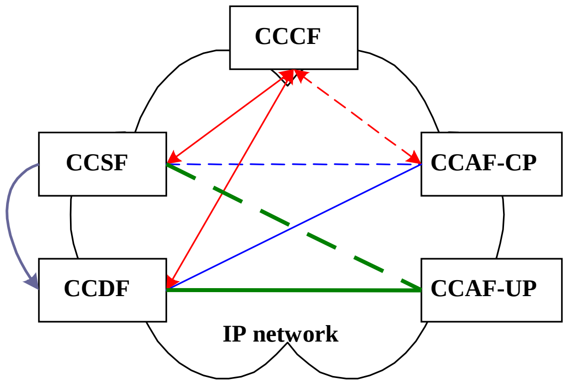


Figure A.1: system model

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 handover, 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 handover user establishes session to the remote UE side after handover, 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 handover.

CCAF (Call Continuity Anchor Function): The CCAF acts as an anchor in handover procedure. It establishes the connection with CCSF before handover and with CCDF after handover. 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 handover procedure, establishes control connection at control plane with CCSF and CCDF separately before/after handover. During the course of handover, 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 handover procedure, establishes media connection at user plane with CCSF and CCDF separately before/after handover. During the course of handover, under the control of CCAF-CP, the old media connection can be replaced with new-established media connection.

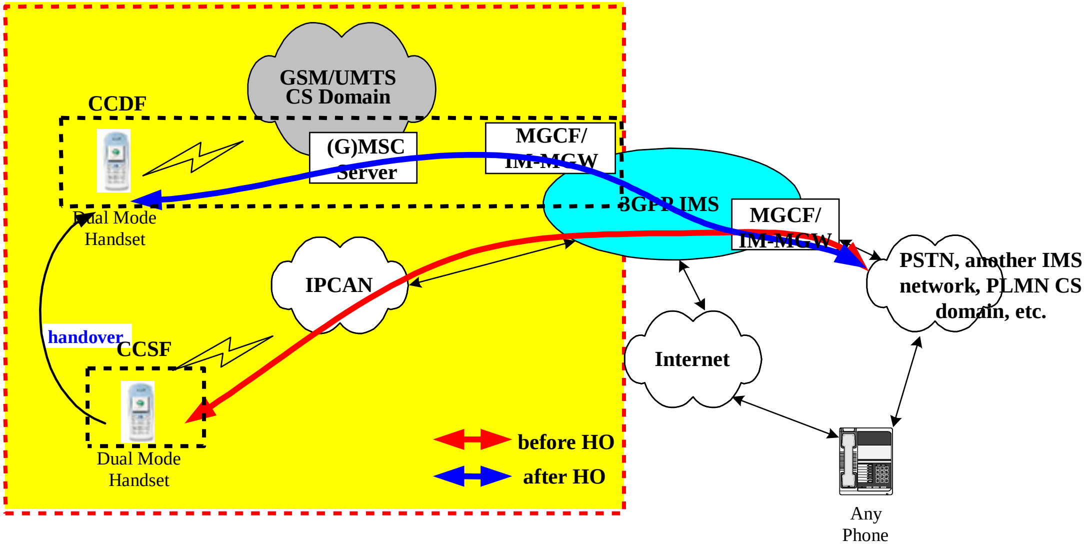
CCCF (Call Continuity Control Function): CCCF is an optional unit and is to enable:

- Authenticating handover request
- Allocating and Controlling CCAF-CP

As we discuss on IMS based control modes, it was benefiting to think the handover 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 handover. The same remote SIP UA (i.e. CCAF) connects with another local SIP UA (i.e. CCDF) by real line after handover.

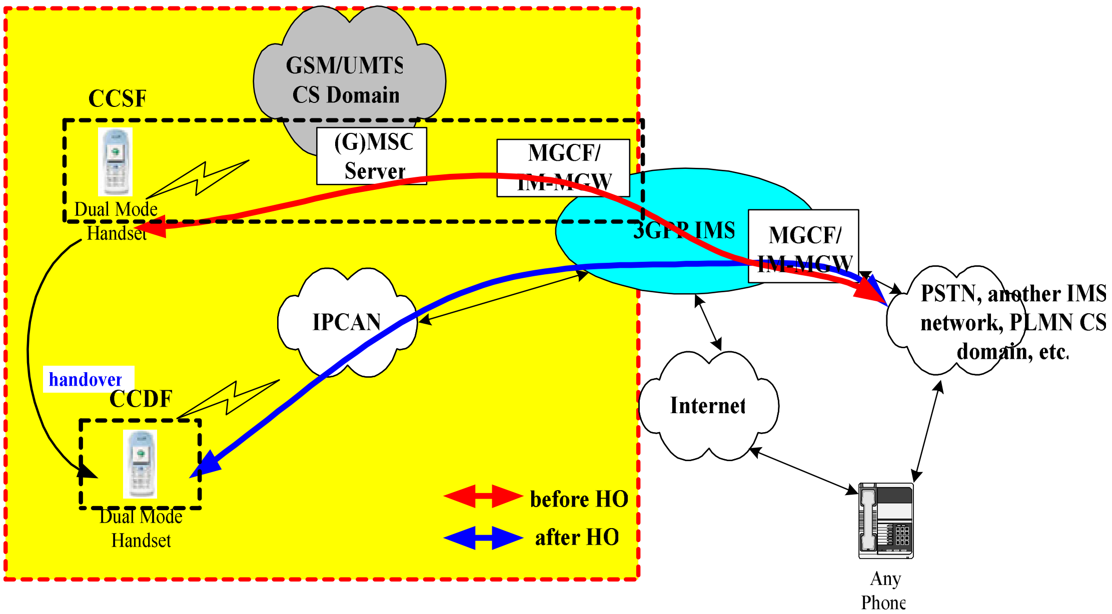
Figures A.2 and A.3 below show the session connection scenarios when user equipment switches between IMS-controlled VoIP call and CS call.

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: Handover 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 handover.

Figure A.3: Handover 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 CP&UP-segmented (control-plane & user-plane segmented) mode

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

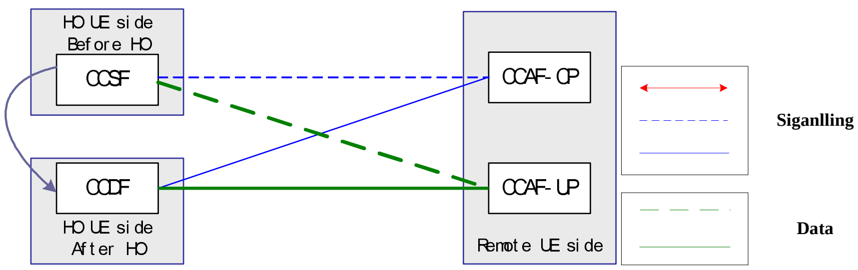


Figure A.4: Terminal-controlled Mode

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.
- Handover procedure, including handover detection, handover initiation and handover execution.
  - Handover detection is executed by CCSF

- Handover initiation: CCSF indicates to CCDF or CCAF to perform handover

- Handover execution: either CCDF or CCAF may be the master controller of handover, 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 handover 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

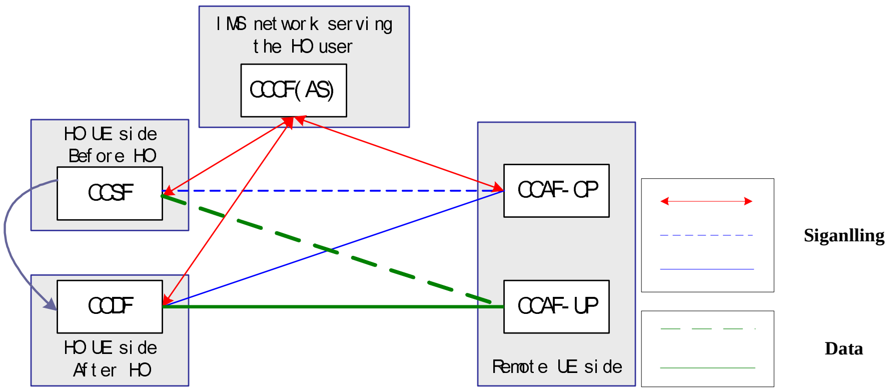


Figure A.5: Network-controlled Mode

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.

- Handover procedure, including handover detection, handover authentication, handover initiation and handover execution.

    - Handover detection is executed by CCSF

    - Handover authentication: CCSF sends handover request to CCCF and CCCF authenticates the request.

- Handover initiation: After completion of handover authentication, CCCF may either indicate directly to CCAF/CCDF to perform handover, or return authentication acknowledgement to CCSF and then CCSF sends handover indication.

- Handover 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 handover 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

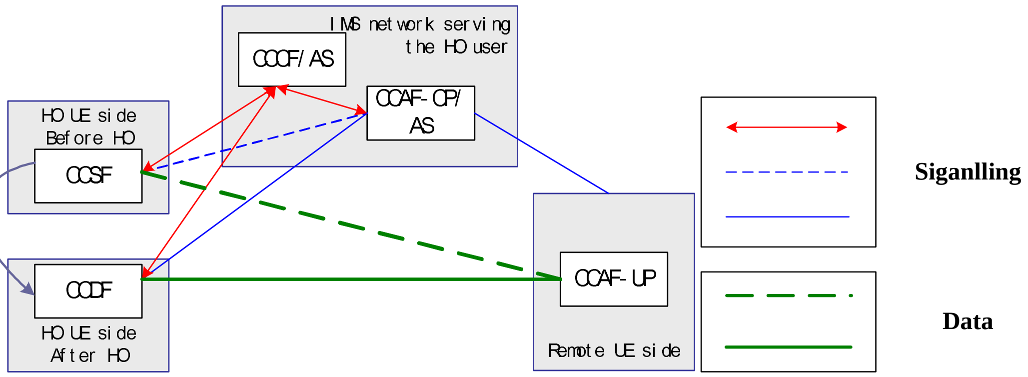


Figure A.6: CP-segmented Mode

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.

- Handover procedure, including handover detection, handover authentication, handover initiation and handover execution.
  - Handover detection, handover authentication, handover initiation and handover execution procedure is the same as that in End-to-End/Network-controlled mode.
  - During the handover 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 handover

Pros.

i. During handover 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 CP&UP-Segmented Mode

Cons.

i. More complicated and 3PCC capability need to be supported by AS comparing with End-to-end Mode

4. CP&UP-segmented Mode

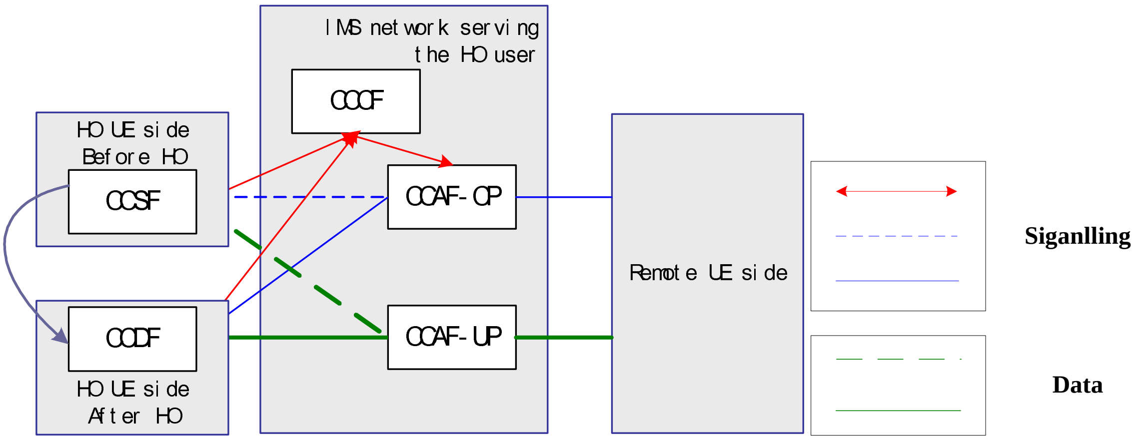


Figure A.7: CP&UP-segmented Mode

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.

- Handover procedure, including handover detection, handover authentication, handover initiation and handover execution.

  - Handover detection, handover authentication, handover initiation and handover 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 handover 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 handover..

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 handover 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 handover 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 handover from IMS to CS, or directly from IMS in case of handover 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 handover:

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 handover 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 handover:

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 handover 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 handover 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.

A.2 Conclusion and Recommendations

Annex B: Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2005-04					First version created	0.0.0	0.0.1
2005-04	SA2#45				Agreed documents from SA2#45 incorporated	0.0.1	0.1.0
2005-05	SA2#46				Agreed documents from SA2#46 incorporated	0.1.0	0.2.0
2005-05					Editorial updates based on email comments after SA2#46	0.2.0	0.2.1
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

3GPP TR 23.~~8bc~~ 806 V0.~~1~~2.~~0~~ 1 (2005-04)

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

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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, Technical Specification 23.234 (3GPP system to Wireless Local Area Network (WLAN) interworking: System description) [2] 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 IPCANs, 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 IPCANs.

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 Tele phone (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*.

[<seq>] <doctype> <#>[([up to and including]{yyyy[-mm]|V<a[.b[.c]]>}[onwards])]: "<Title>".

[1] 3GPP TR 41.001: "GSM Release specifications".

[2] 3GPP TS 23.234: "3GPP system to Wireless Local Area Network (WLAN) interworking"

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the [following] terms and definitions [given in ... and the following] apply.

***<defined term>**: <definition>*.

example: text used to clarify abstract rules by applying them literally.

3.2 Symbols

For the purposes of the present document, the following symbols apply:

<symbol> <Explanation>

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

I-WLAN Interworking WLAN
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 IP Connectivity Access Network
- 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, 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, TS 33.234) shall be reused.
- The UE will be capable of transmitting and receiving simultaneously in the CS domain on GERAN/UTRAN, and on the IP Connectivity Access Network.
- 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 inter domain constraints.
- When a user is attached to both the CS and IMS domains, the network has the responsibility for selecting the terminating service domain, depending on operator policy and possibly user preference.
- Support for Voice call and Emergency call handover shall be provided, if the target domain supports it.
- Use of available QoS mechanisms need to be considered, however, the Impact on the QoS mechanism is out of the scope.

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

Editor Note: The CS domain in roaming should be supported.

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 select the appropriate access Network (such as GSM/UMTS radio or IP Connectivity Access Network). The selection may be based, e.g., on operator policy for real-time voice service and user's preference.
- 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.
- While not in CS or IMS voice call, the UE shall be able to detect and automatically select the appropriate access Network (such as GSM/UMTS radio or IP Connectivity Access Network). The selection may be based, e.g., on operator policy for real-time voice service and user's preference.
- 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.
- Handoff should be provided such that from the end user's perspective minimal service disruption is perceived. Handoff 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
- ~~Use of available QoS mechanisms need to be considered, however, the Impact on the QoS mechanism is out of the scope.~~

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 handover occurs

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 Voice 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: this section illustrates the reference model to support service continuity

6.2.1 Call Continuity Control Function (CCCF)

CCCF provides functions for service 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.

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 Terminating Domain Selection

The ability to select the correct domain in which to terminate the call is required. While, 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.

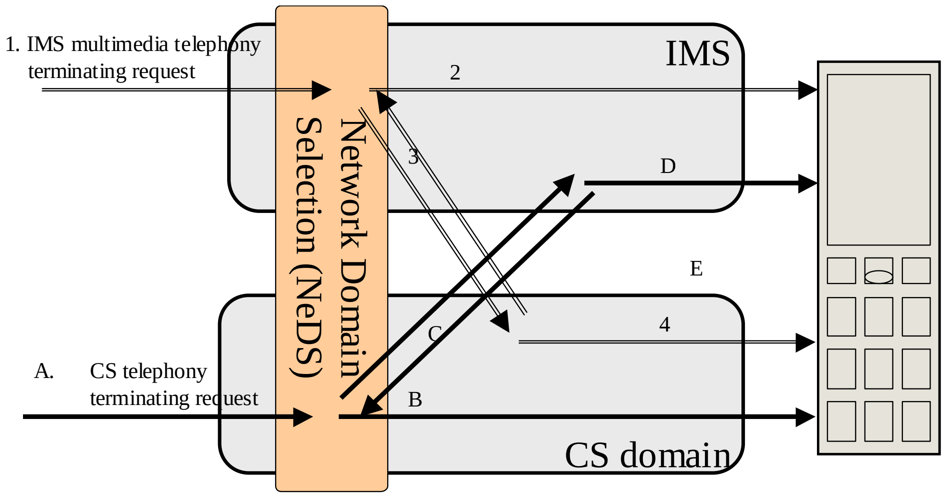


Figure 6.2.1: Logical view of Terminating Network Domain Selection Functionality

Figure 6.2.1 above depicts a logical representation of the network domain selection functionality. This could be one or several logical entities.

A telephony call to the CS domain (A) could be directly routed to the UE via the CS domain (B), or via the IMS domain (C, D) for terminating treatment.

An IMS terminating multimedia call (1) could be terminated on the IMS (2) or via the CS domain (3).

There are cases where the network domain selection may require to be re-invoked (indicated by the return arrows (3) and (C).

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.

There are a number of potential solutions for network domain selection. In order to generalise the discussion and understand the requirements before delving into protocol details, this contributions proposes a general approach to the problem.

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

- The Terminating NeDS function needs to be 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 Terminating NeDS function needs to be aware of whether the terminal is attached to the CS domain.

The Terminating NeDS functional can make a decision as to the appropriate terminating domain, taking into account the operator, user and service preferences.

6.3 Service Continuity Model: IMS Controlled Alternative

6.3.1 General Description

In this alternative the CS-IMS continuity is solved by employing CS IMS Voice Continuity Service (CIVCS) in the user's home IMS network.

CIVCS allows for Handing over, Subsequent handing over and Handing back of Call Control functions of a CS-IMS capable UE's voice call for active mode roaming across CS Domain and IM Subsystem with seamless user experience.

CIVCS is a home IMS network service that can be enabled either via static or dynamic anchoring techniques. Fundamental principles of CIVCS as applicable to static and dynamic anchoring are listed below:

- CIVCS 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 DTAP in SIP for Handovers from GSM/UMTS CS to IMS over I-WLAN, significant changes to GSM/UMTS Core Network and Access Network nodes are required for SIP encapsulation in BSSAP for Handover from IMS to GSM CS, and SIP encapsulation in RANAP for Handover from IMS to UMTS CS; it is therefore not feasible to maintain the same anchor control model with Handovers involving I-WLAN.

- CIVCS anchors the bearer for CS originations and terminations at a MGW controlled by an MGCF in user's home IMS network. Although the MGCF belongs to user's home IMS network, the MGW used to hairpin the bearer is located in close proximity with the CS network to save TDM backhaul cost.

- CIVCS is realized by using IETF Third Party Call Control (3pcc) function.

- CIVCS provides the Handover Control Functions for CS-IMS voice continuity. All CS-IMS Handovers are executed and controlled by CIVCS upon UE's request.

- Since Handovers are executed across CS Domain and IM Subsystem with different call control protocols, DTAP is used in CS Domain whereas SIP is used in IMS for call control procedures; the handover procedures execute 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. As a result, Handover execution can only be guaranteed in the active state of the CS call or IMS session.

- CIVCS provides cohesive billing with a complete Handover 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 inter domain transition are captured as Handover 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.

- CIVCS 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.

- 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 transition.

CIVCS does not influence supplementary service execution in the serving network node prior to or post handover. 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.

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

CIVCS 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 CIVCS to facilitate control of the bearer path upon Handover 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.

6.3.1.2 Dynamic CS Anchoring for CIVCS using DACCI

DACCI is a new CS domain service that provides dynamic anchoring of CS calls at CIVCS.

Enablement of first CS to IMS voice 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 transition in order to enable subsequent transitions between CS and IMS via CIVCS. This helps eliminate the requirement for static anchoring of CS calls whereby calls are anchored with CIVCS at initial call setup resulting in resource inefficiencies.

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.

DACCI uses the Mobility Event package to communicate with CIVCS, session information required to perform transition from CS to IMS.

DACCI is applicable to all voice teleservices including the Emergency Call Service.

6.3.1.3 Dynamic CS Anchoring for CIVCS using ECT

6.3.1.3.1 General

Although ECT has significant drawbacks when used as the only solution for Handovers from CS domain to IM Subsystem, it may be used as a means for dynamically anchoring CS calls at CIVCS 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 transition. CIVCS is used for all subsequent Handover and Handbacks from CS to IMS and from IMS to CS.

6.3.1.3.2 Supplementary Service control for ECT enablement

Certain supplementary services like Call Forward Unconditional and Incoming Call Barring can prevent ECT enabled Handovers 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 Handover procedure and re-enable these services just before the CS radio is released upon Handover.

6.3.2 Routing Selection Decision

6.3.2.1 Alternative A

6.3.2.1.1 Inter-Domain Routing Policy Definition

The Inter-domain routing policy is that set of service logic functions available to the MSC, GMSC, and IMS CSCF servers which examine the current state of registration within the domains, and, based on that knowledge and operator and subscriber preferences, route the call for completion in the appropriate domain.

6.3.2.1.2 Scenarios

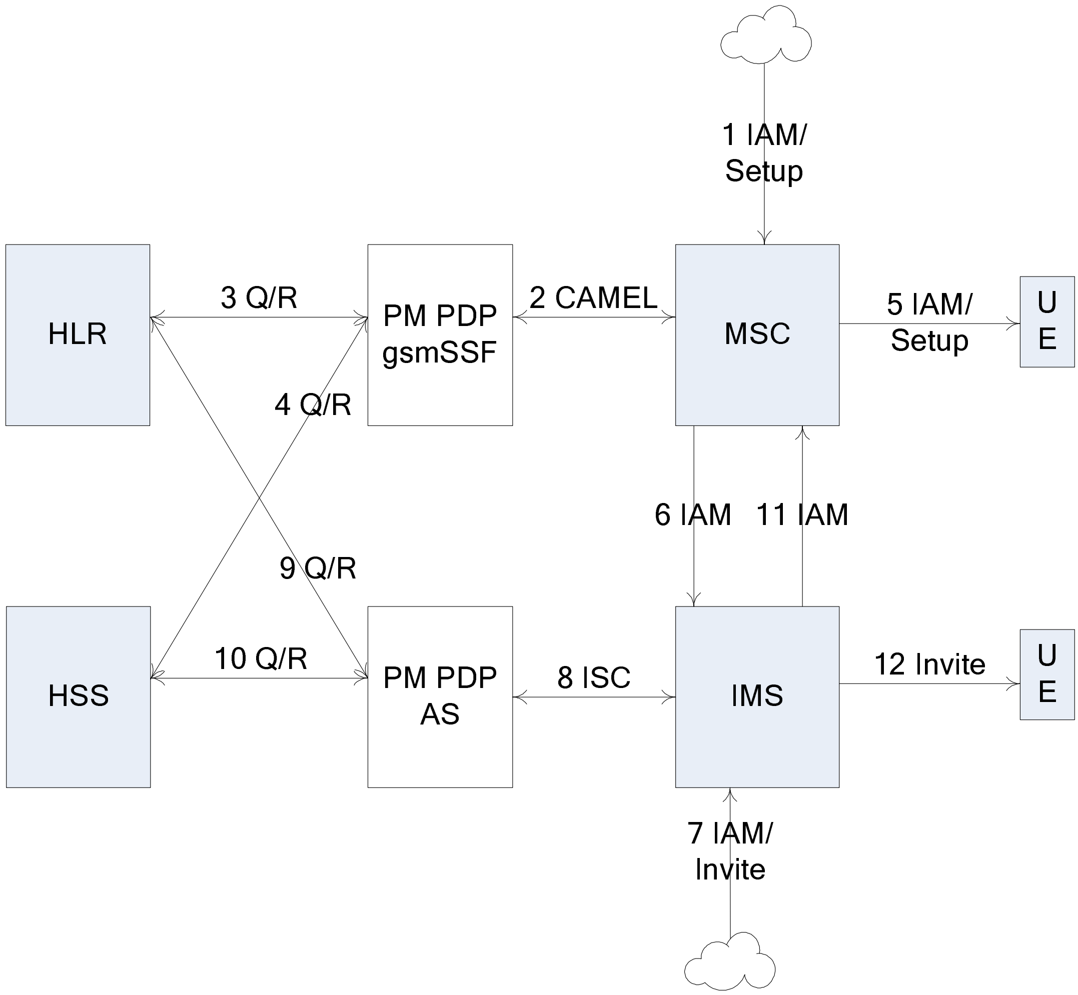
The scenarios to be supported by this architecture are:

- Registered in the Home CS Network
- Registered in a visited CS Network
- Registered on a non-3GPP Access Network capable of supporting IMS VoIP

The UE may or may not be simultaneously registered in the CS and IMS networks.

6.3.2.1.3 Logical Architecture

The following diagram depicts the architecture for implementing the routing policy.



In the architecture above, the service logic is executed as follows:

1. An IAM or call setup arrives at an MSC ar a GMSC from an external network or via a originating subscriber in the GSM network.

2. Service Logic is invoked via a termination trigger

3 and 4. The service logic causes queries to both the HLR and the HSS to obtain the current registration states for both CS and IMS domains.

5 or 6. Depending upon the registration states and the operator policy, the MSC routes the call either towards the IMS system, or towards the UE in the CS domain.

Similarly, in steps 7 through 12, an IAM or an invite entering the IMS system receives similar treatment.

6.3.2.1.4 Routing Policy Execution in CS Domain

In the CS domain, an SCP serves as the Policy Decision Point. The Policy Enforcement Point is the GMSC function, which envokes CAMEL triggers to the Policy Decision Point using existing standardized CS triggers to query the Policy Decicion Point. A CS termination will cause the gsmSSF acting as the Policy Decision Point to execute the queries and determine the routing based on operator policy. The response to the CAMEL trigger returns the routing information to the GMSC, which as the Policy Enforcement Point routes the call.

The change to the existing CS network is the addition of a gsmSSF that can query both the HLR and HSS and determines policu based upon the query responses, operator preferences, and subscriber preferences.

6.3.2.1.5 Routing Policy Execution in IMS Domain

In the IMS domain, the Serving CSCF (S-CSCF) queries the AS PDP to determine routing. The Policy Decision Point executes the queries to the HSS and HLR and determines the routing based upon the query responses and operator policy. The interface between the S-CSCF and the Policy Decision Point is the ISC interface.

Based upon the response from the Policy Decision Point, the S-CSCF routes either to the CS network via the BGCF or through the IMS network to the subscriber's current Proxy CSCF.

6.3.2.1.6 Conclusion

Sufficient flexibility exists within already standardized capabilities such that it is not necessary to define an architecture that attempts to combine or synchronize possibly separate HLRs and HSSs with new protocol for the purposes of flexible routing control. Note that this architecture makes no special distinctions about either HLR/HSS "ownership". All that is required is that cooperating networks allow access. Likewise, the architecture would support the PDP for both networks residing on a single platform.

Editor's Note: The following are open issues:

- Error Scenarios "IMS registered but UE not reachable" , "CS attached but UE not reachable" , "Registered in IMS but not with 802xx voice(e.g., GPRS)" need to be considered.
- Correctly label the query/response interfaces.

6.3.2.2 Alternative B

6.3.2.2.1 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: The term "Reachable" is used throughout this document to designate that the user has registered in the domain, and no corresponding incoming call barring service been activated.

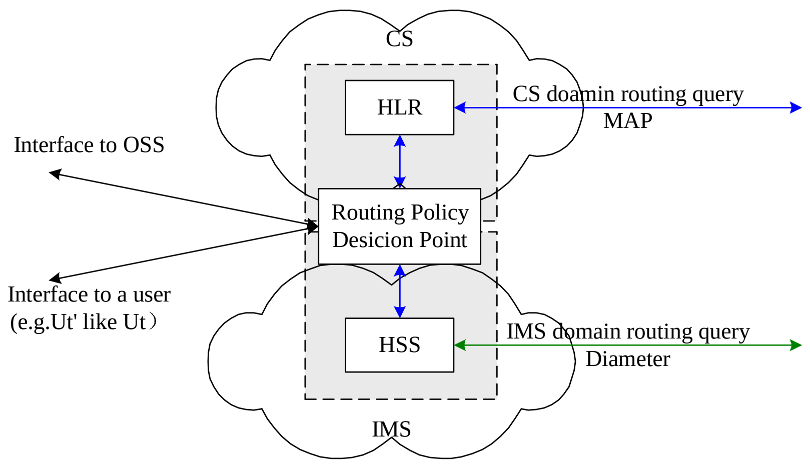
The control can be summarised as follows:

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

6.3.2.2.2 Architecture

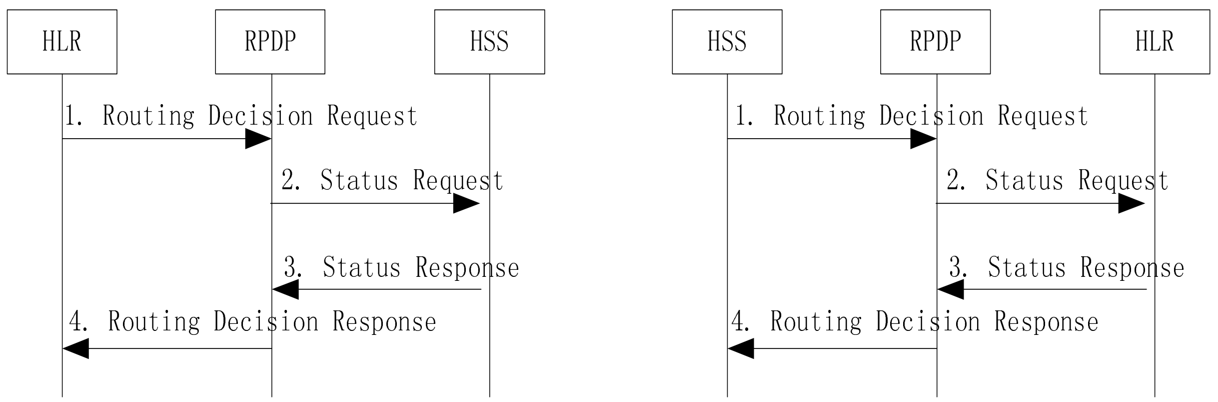
According to the above discussion, different scenarios have different input parameters or different output results, and a user's preferences and an operator's configurations are also needed to be taken into account. Then, a new functional module is introduced, named Routing Policy Decision Point (RPDP),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



A Routing Policy Decision Point (RPDP) is a newly-added logical functional entity for the enhancement of inter-domain routing control, which can be a separate physical entity, or be a logical entity embedded in HLR or HSS. A RPDP stores routing policy of a user and provides routing policy decision to the HLR/HSS that initiates routing decision query. In addition, A RPDP 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 HLR/HSS can make routing decision based on the information owned by them, they will not quire the RPDP for the routing policy. A typical procedure shown as follows:

NOTE 1: The detailed enhancement on HLR/HSS, i.e. interface and procedure between HLR/HSS and RPDP, is FFS.

NOTE 2: From Release 5, there is only a HSS, which consists of the original HLR/AUC functionality required by the CS/PS Domain, and so HLR is no longer a separate network entity as described in TS 23.002. If operator only owns the HSS for IMS and CS, the RPDP was embedded on the HSS and the interaction shown below was all performed internal.



1. Once receiving the routing information query from GMSC/I-CSCF, the HLR/HSS sends a Routing Decision Request (User's reachable status, User Identifier) to the RPDP. User's reachable statuses consist of the Registration status i.e. CS attached for HLR or IMS registered for HSS, and status of incoming call barring service. As for User Identifier identified the user, if queried from HSS, it may be a Public User ID. As for HLR, it may be a MSISDN.

2. According to the current routing policy, the RPDP determine whether the user's status in the other domain is needed for routing decision. If user's status is necessary, RPDP sends a Status Request (User Identifier) to HSS/HLR. Otherwise, the RPDP perform step 4.

3. After received Status Request form RPDP, the HSS/HLR response a Status Response (User status) to RPDP. Including the User's Registration status i.e. CS attached for HLR or IMS registered for HSS.

4. Based on the combination of routing policy and the Status information, the RPDP determines the network through which the terminating call should be routed and sends a Routing Decision Response (Routing Decision) to HLR/HSS. Then the HLR/HSS can get the routing information based on this routing decision and responds the routing information to GMSC/I-CSCF.

Therefore, a RPDP is also to enable to:

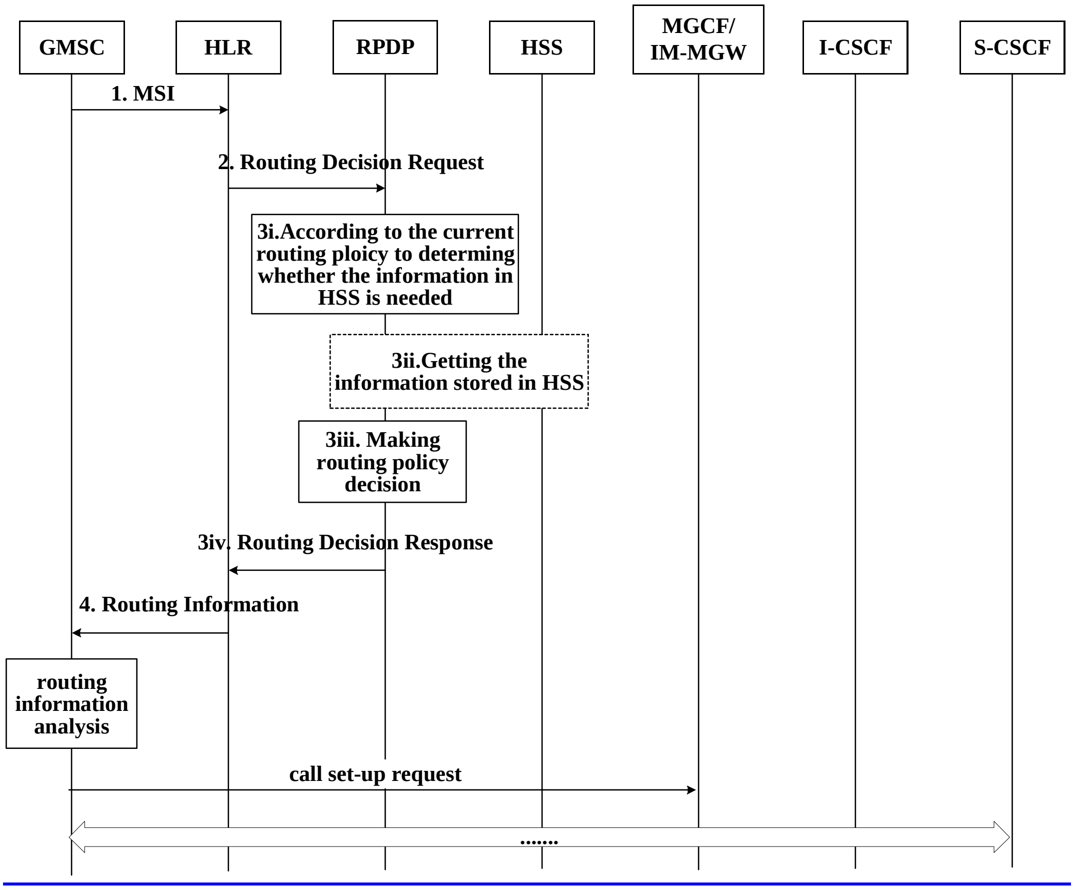
- provide interfaces to HLR/HSS through which the HLR/HSS can retrieve the routing information and routing decision based on the current routing policy, and the RPDP can retrieve the user's subscription information in CS/IMS domain, status information about registration, location update and location information. This interface may be implemented by using MAP or other appropriate protocols and related extended protocols;

- provide interfaces to OSS. Operators can utilise this interface to configure routing policy flexibly based on the operation policy;

provide interfaces to a user to configure routing policy based on the user's preference. This interface may be implemented by means of, e.g. like Ut interface in IMS.

6.3.2.2.3 An Example of Routing Policy Decision

Here is an example of routing policy decision. In this case, when a terminating call from PTSN arrives at a GMSC, the GMSC interrogates a HLR which the called user is subscribed to get the routing information. At receiving the query request, the HLR communicates with a RPDP to get routing policy decision. The RPDP makes a decision based on the current configured routing policy and called user's status information (e.g. reachable flag) stored in HLR and HSS. In this case, the result of decision is that the call should be routed through IMS domain, and then the HLR return the routing information to the GMSC. Detailed behaviour is shown as follows:



1. The GMSC receiving a terminating call from PSTN initiates a routing information query to the HLR using MSI message (MAP_SEND_ROUTING_INFORMATION).
2. The HLR sends a Routing Decision Request to the RPDP querying routing policy decision. In this request, the user's status information in CS domain should also be delivered to the RPDP.
3. After received the Routing Request, the RPDP makes routing policy decision and return the result of decision to the HLR, including some sub-steps:
   i. According to the current routing policy, the RPDP determine whether the user's status is needed for routing decision. If user's status is necessary, continue the following step. Otherwise, perform step iii.;
   ii. The RPDP interacts with HSS to get the user's status in IMS domain;
   iii. Based on the combination of routing policy, and the status information, the RPDP determines the network through which the terminating call should be routed (i.e. routing policy decision);
   iv. The RPDP returns the result of routing policy decision to the HLR and indicates the incoming call should be routed through the IMS domain. In the result, an identifier of MGCF is included for further call process.
4. The HLR returns the Routing Information (the MSRN getting from VLR based on the routing policy decision) to the GMSC and the GMSC routes the call to the appointed network and related entity, e.g. a MGCF.

The routing through IMS domain is similar to the above case and not to be listed here.

6.3.3 Registration

The UE and CIVCS subscribe to each other for the Mobility Event package upon UE Registration with IMS to enable exchange of session information that is necessary to perform Inter domain Voice Call Continuity for a CS IMS user engaged in multiple sessions.

Figure 1 below describes a sequence for UE Registration with IMS to enable Voice Call Continuity using CIVCS.

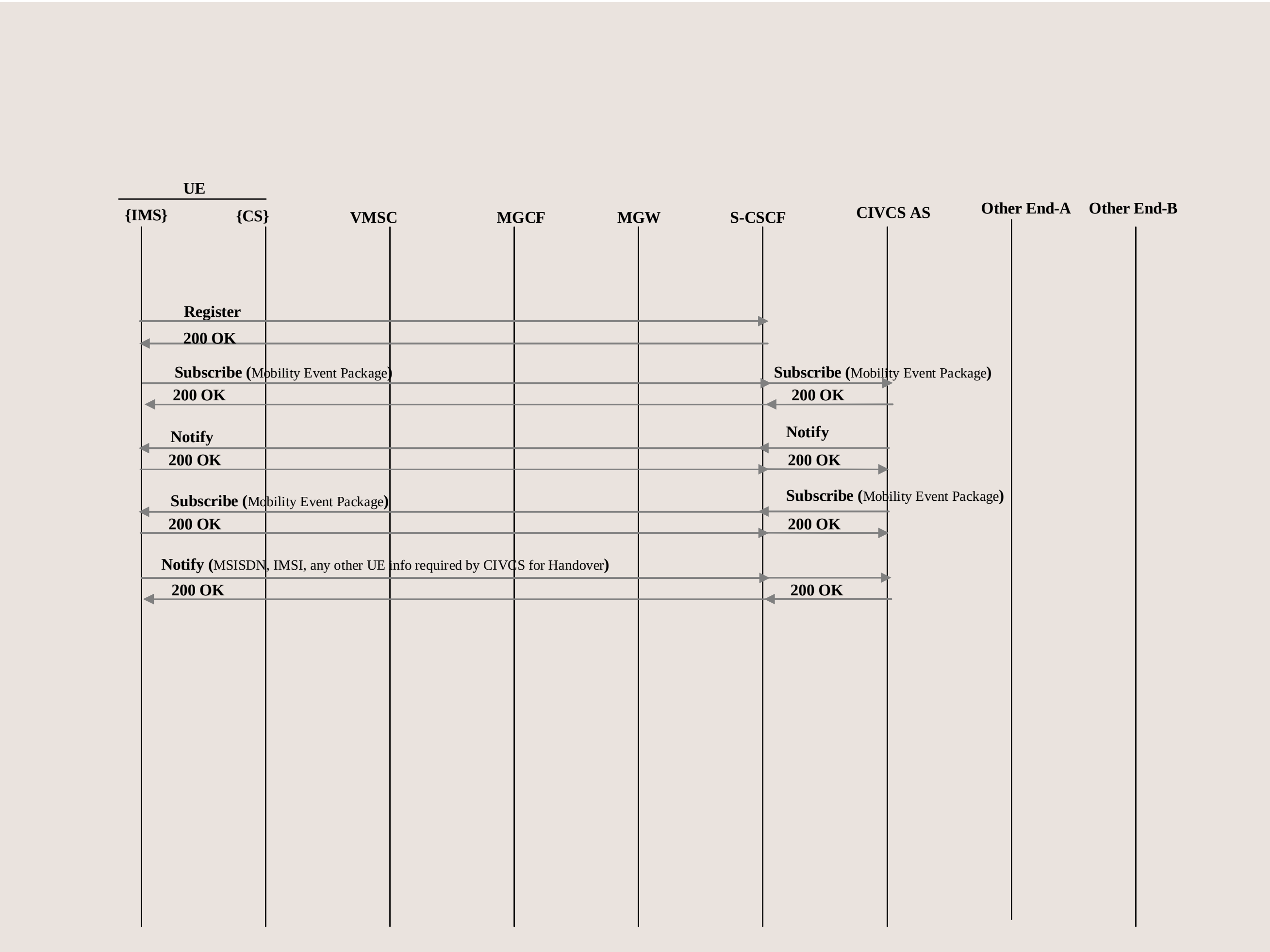


Figure 1: IMS Registration to enable CIVCS-UE information exchange

- The filter criteria set for the CS IMS user on Subscribe with Mobility Event directs the Subscribe for Mobility Event initiated by the UE post IMS Registration to the CIVCS AS assigned to the registered user.

- The CIVCS AS replies with a 200 OK and a Notify followed by a Subscribe for the Mobility Event toward the UE.

- The UE responds with a Notify consisting of all user identities available at the UE along with any other information that may be required by CIVCS for execution of session anchoring and enablement of Voice Call Continuity procedures.

6.3.4 Origination

6.3.4.1 IMS origination

6.3.4.1.1 Static Anchoring: CIVCS controlled IMS originating sessions

6.3.4.1.1.1 General

Filter criteria associated with CIVCS are stored in the HSS as part of the CS-IMS user's service subscription profile. The CIVCS filter criteria are downloaded to the currently assigned S-CSCF as part of Initial Filter Criteria at the time of subscriber's registration with IMS network. Initial Filter Criteria are executed at the S-CSCF upon IMS session initiation from the CS-IMS user that results in routing of the user's session to CIVCS. CIVCS enables a Routing B2BUA 3pcc function to control the bearer path for the session.

In order to maximise network efficiencies with the use of CIVCS, it is recommended that subscriber's current location be used at the S-CSCF or at CIVCS such that it results in invocation of 3pcc function at CIVCS only when the user is being served in the areas with overlapping coverage or with borders between the two domains.

6.3.4.1.1.2 IMS originations walkthrough

Figures 3a and 3b describe how signalling and bearer paths are established for IMS originations from CS-IMS users.

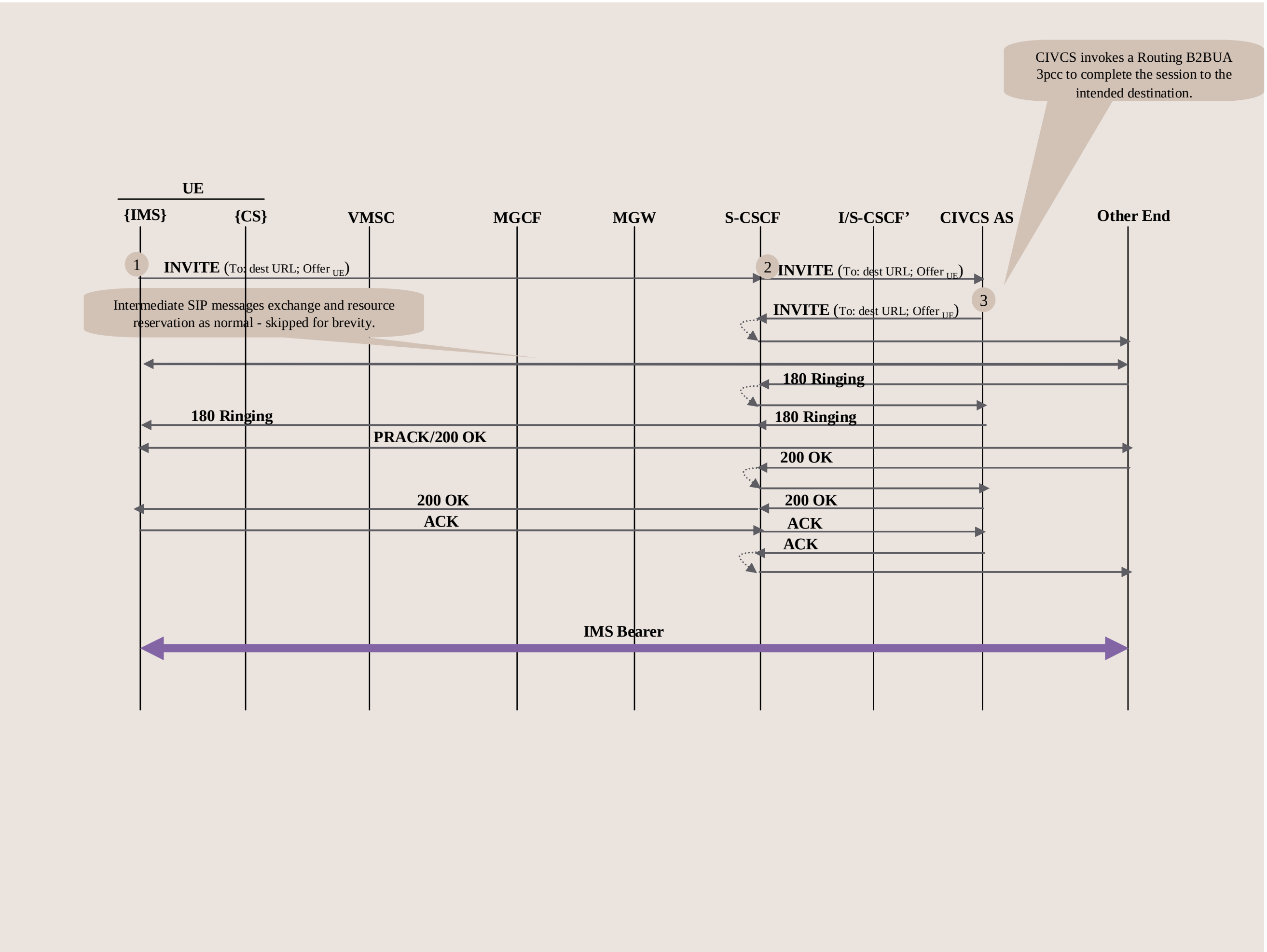


Figure 3a: IMS Origination controlled at CIVCS walk-through

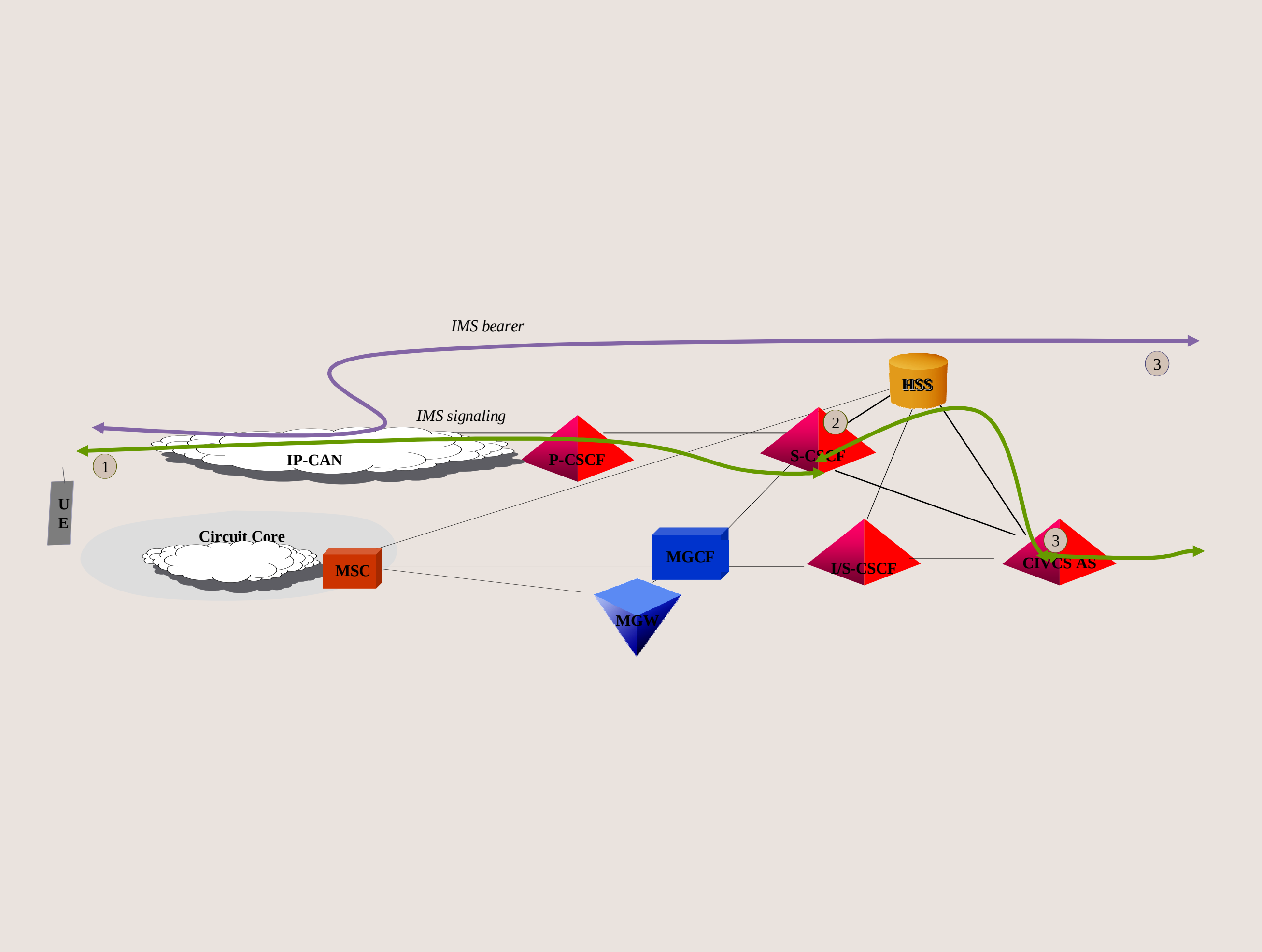


Figure 3b: IMS Origination controlled at CIVCS walk-through

1. The UE registers with IMS resulting in an assignment of S-CSCF. It subsequently initiates an originating session as normal.

2. Filter criteria at the S-CSCF result in forwarding of the INVITE to the CIVCS application server.

CIVCS completes the session establishment by setting up the terminating leg of the session. It enables 3pcc function to maintain session states for control of the originating and terminating leg of the CS-IMS user's session in order to control bearer upon Handover requests from the UE.

6.3.4.2 GSM/UMTS CS origination

6.3.4.2.1 Static Anchoring: Originating call anchoring at CIVCS for users roaming in CS Domain

6.3.4.2.1.1 General

Special routing techniques are established at the Visited MSC in areas with overlapping coverage and with borders between the two domains so that CS originations for CS-IMS users are routed via CIVCS in user's home IMS network. The Visited MSC routes the CS originating call to CIVCS which enables a Routing B2BUA 3pcc function to control the bearer path of the call. The original called number is passed to the CIVCS application so that it can perform final translations to route the call to the originally intended destination.

This routing function can be realized by using appropriate translation techniques or using a CAMEL service at the Visited MSC that steer the calls made by CS-IMS Voice Continuity service subscribers via the user's home IMS network. A few potential options to realize this function are specified below:

- Use CAMEL service or special translation techniques at the Visited MSC to prefix steering digits to the called party number in the outgoing IAM message. Note that special translation techniques may not be available at the networks owned by roaming partners, whereas a standardized R99 CAMEL trigger is the only requirement for enablement of the CAMEL service.
- Use CAMEL service to manipulate the outpulsed digits such that the CIVCS PSI is sent as the called party number and the original called number is communicated in another, to be determined, ISUP parameter to the IMS network.

6.3.4.2.1.2 CIVCS Anchored CS Origination walkthrough

Figures 1a and 1b describe 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.

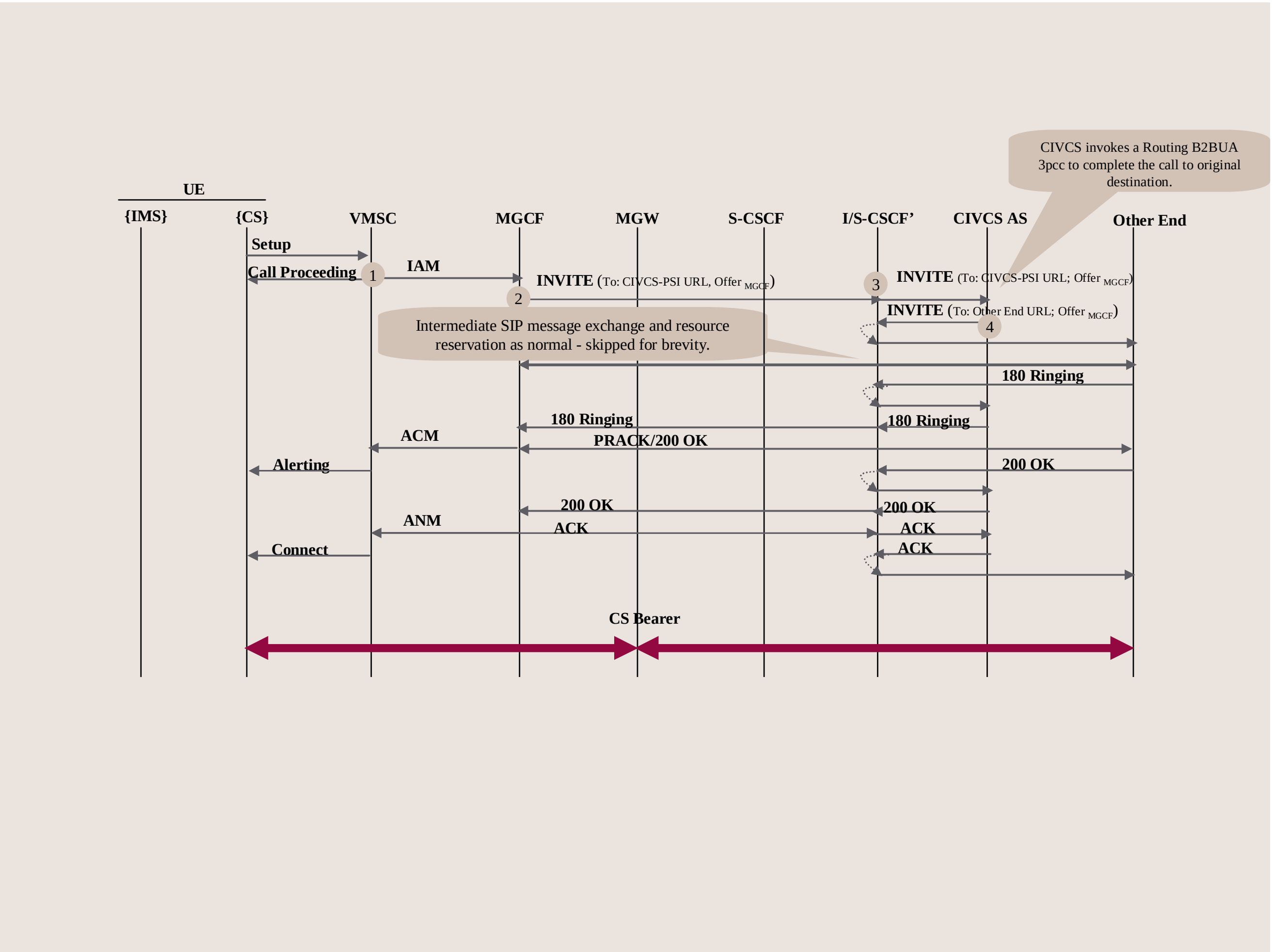


Figure 1a: CS Origination Anchored at CIVCS walk-through

NOTE: The CS bearer indicates the bearer path for the user when being served in CS Domain, whereas the IMS bearer shows bearer path for the user when being served in IMS.

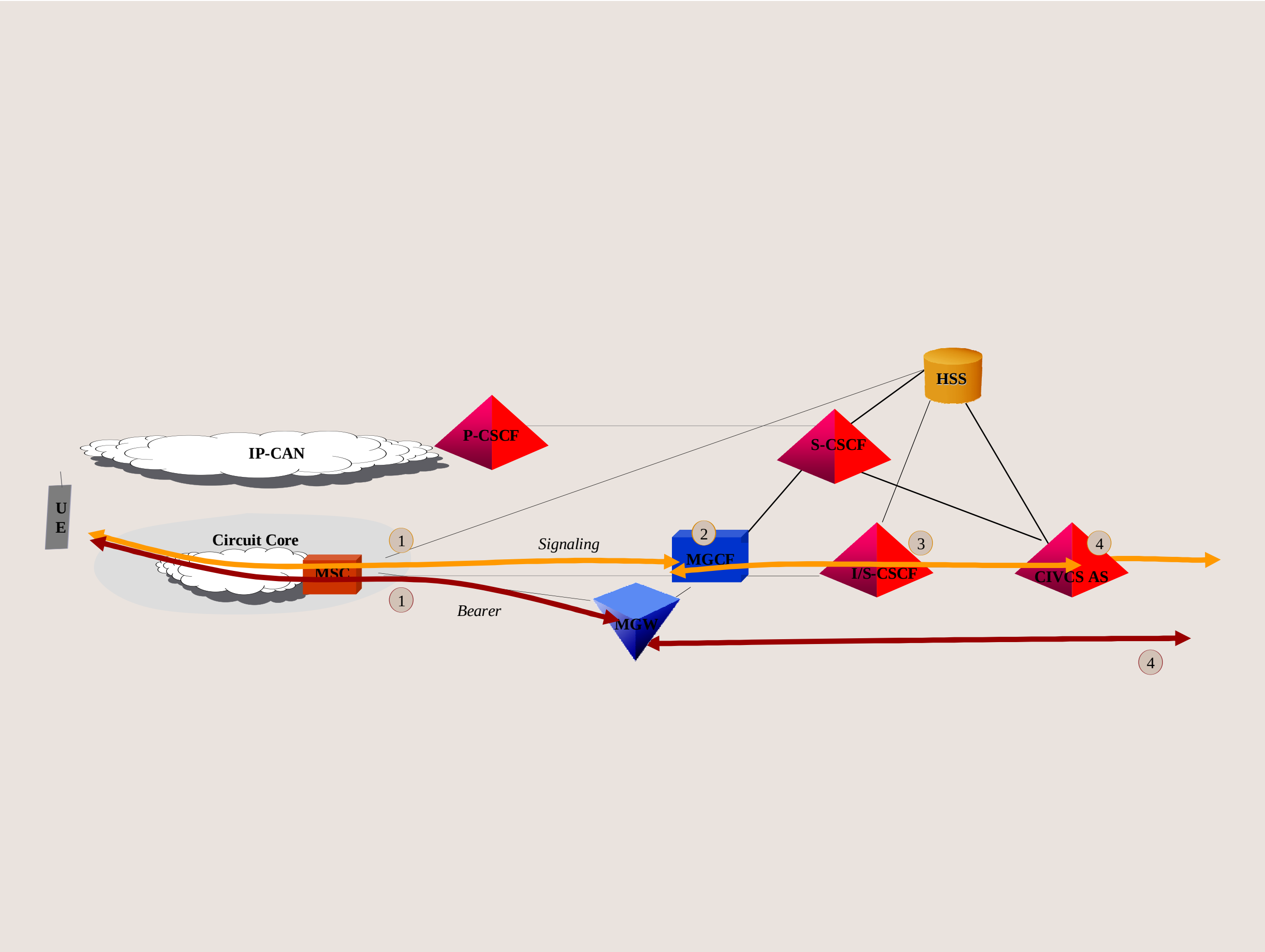


Figure 1b: CS Origination Anchored at CIVCS walk-through

1. The user originates a call after registering with the MSC. Techniques described previously in this section are used to route the call via the user's home IMS network.

2. The MGCF acts as a user agent on behalf of the CS user initiating an INVITE towards CIVCS. The original called number is passed to CIVCS for routing to the terminating party.

3. The MGCF or the I-CSCF discovers the CIVCS PSI using information received from the originating network. An INVITE addressed to CIVCS PSI is routed to an S-CSCF assigned

CIVCS completes the call by routing it 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. CIVCS 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.

6.3.5 Termination

6.3.5.1 IMS termination

6.3.5.1.1 Static Anchoring: CIVCS controlled IMS terminating sessions

6.3.5.1.1.1 General

Filter criteria associated with CIVCS are stored in the HSS as part of the CS-IMS user's service subscription profile. The CIVCS filter criteria are downloaded to the currently assigned S-CSCF as part of Initial Filter Criteria at the time of subscriber's registration with IMS network. Initial Filter Criteria are executed at the S-CSCF upon incoming IMS session delivery toward the user that result in routing of the user's session to CIVCS. CIVCS enables a Routing B2BUA 3pcc function to control the bearer path for the session.

In order to maximise network efficiencies with the use of CIVCS, it is recommended that subscriber's current location be used at the S-CSCF or at CIVCS such that it results in invocation of 3pcc function at CIVCS only when the user is being served in the areas with overlapping coverage or with borders between the two domains. IMS Originations and Terminations

6.3.5.1.1.2 IMS terminations walkthrough

Figures 4a and 4b describe how signalling and bearer paths are established for IMS terminations from CS-IMS users.

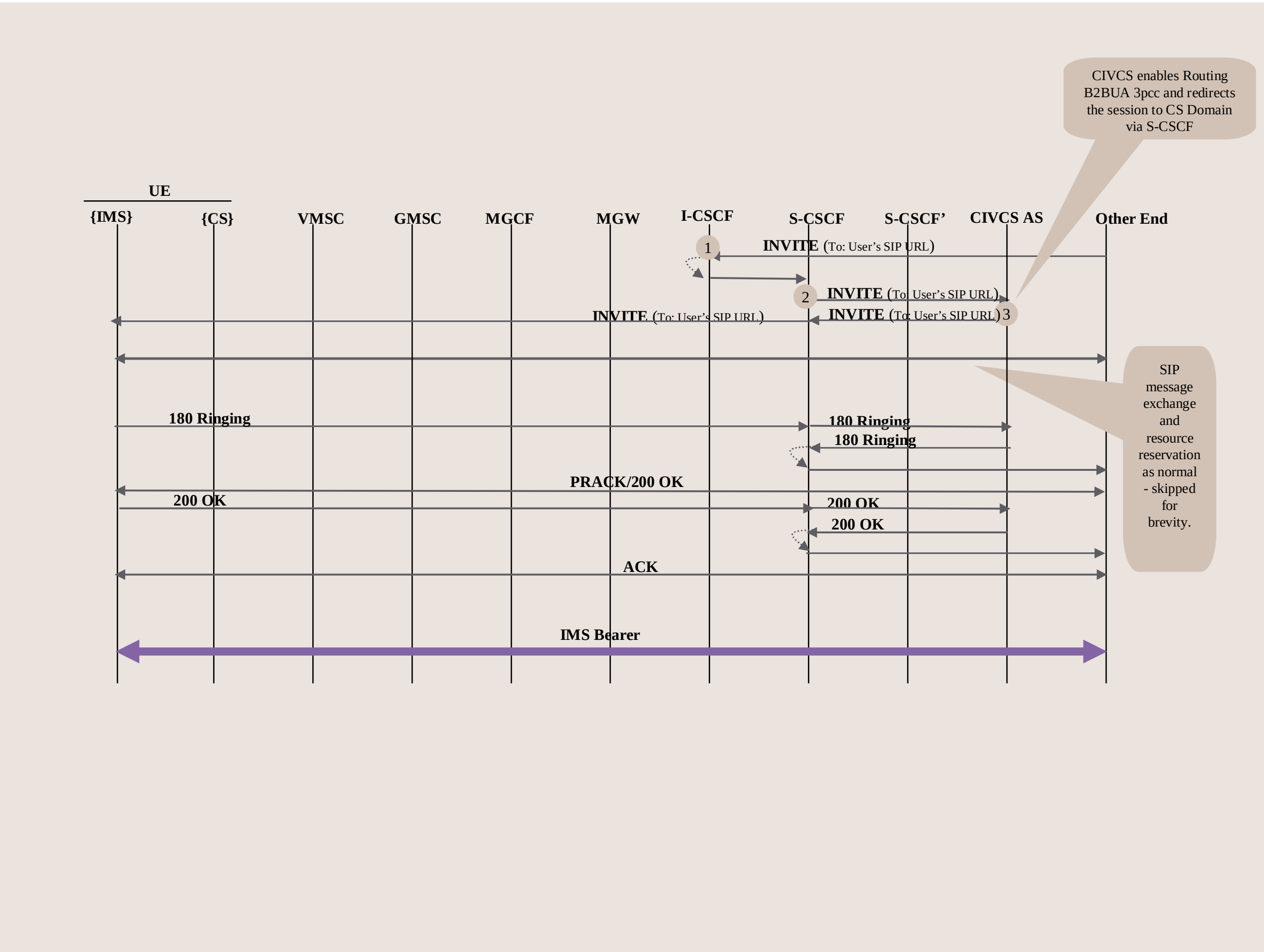


Figure 4a: IMS Termination controlled at CIVCS walk-through

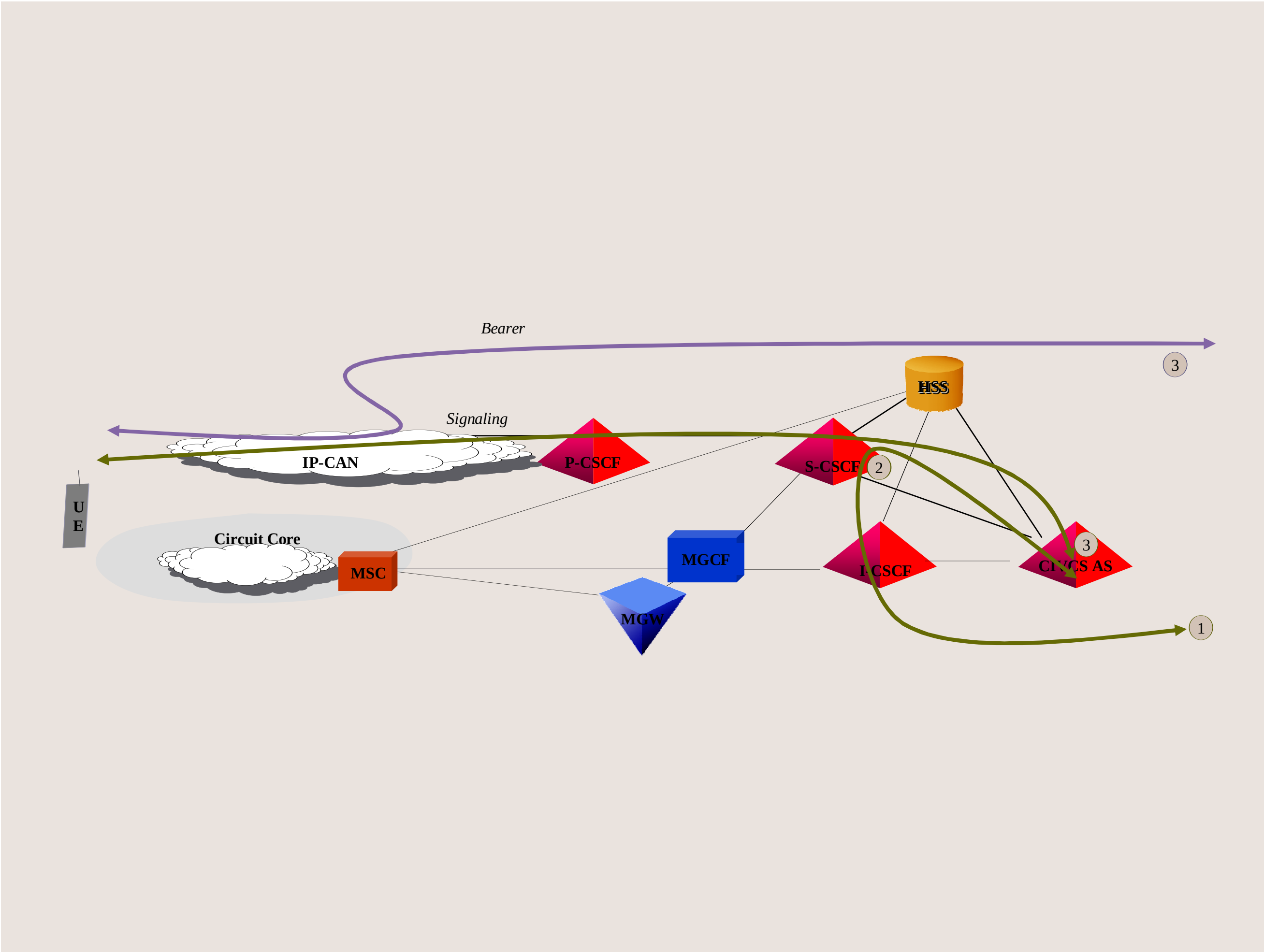


Figure 4b: IMS Termination controlled at CIVCS walk-through

1) A session originated in IMS is delivered at the user's home I-CSCF function. A call originated in CS Domain and/or PSTN is delivered to the MGCF function in the user's home IMS network. Location Query at I-CSCF results in forwarding of the INVITE to S-CSCF assigned to the user upon IMS Registration. Filter criteria at the S-CSCF

2) Filter criteria at the S-CSCF result in forwarding of the INVITE to CIVCS.

3) CIVCS completes the session establishment by setting up the terminating leg of the session. It enables 3pcc function to maintain session states for control of the originating and terminating leg of the CS-IMS user's session in order to control bearer upon Handover requests from the UE.

6.3.5.2 GSM/UMTS CS termination

6.3.5.2.1 Static Anchoring: Terminating call anchoring at CIVCS for users roaming in CS Domain

6.3.5.2.1.1 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 CIVCS. Incoming calls originated in the CS domain network, PSTN or other IMS networks, which are destined for CS-IMS users can be anchored at CIVCS by setting up routing functions at the originating nodes such that the incoming calls for CS IMS users are delivered to the user's home IMS network. CIVCS 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.

6.3.5.2.1.2 CIVCS Anchored CS Terminations walkthrough

Figures 2a and 2b describe how signalling and bearer paths are established for calls terminating to CS-IMS users when roaming in CS Domain. The walk-through assumes that the user is not registered in IMS at the time of incoming call delivery.

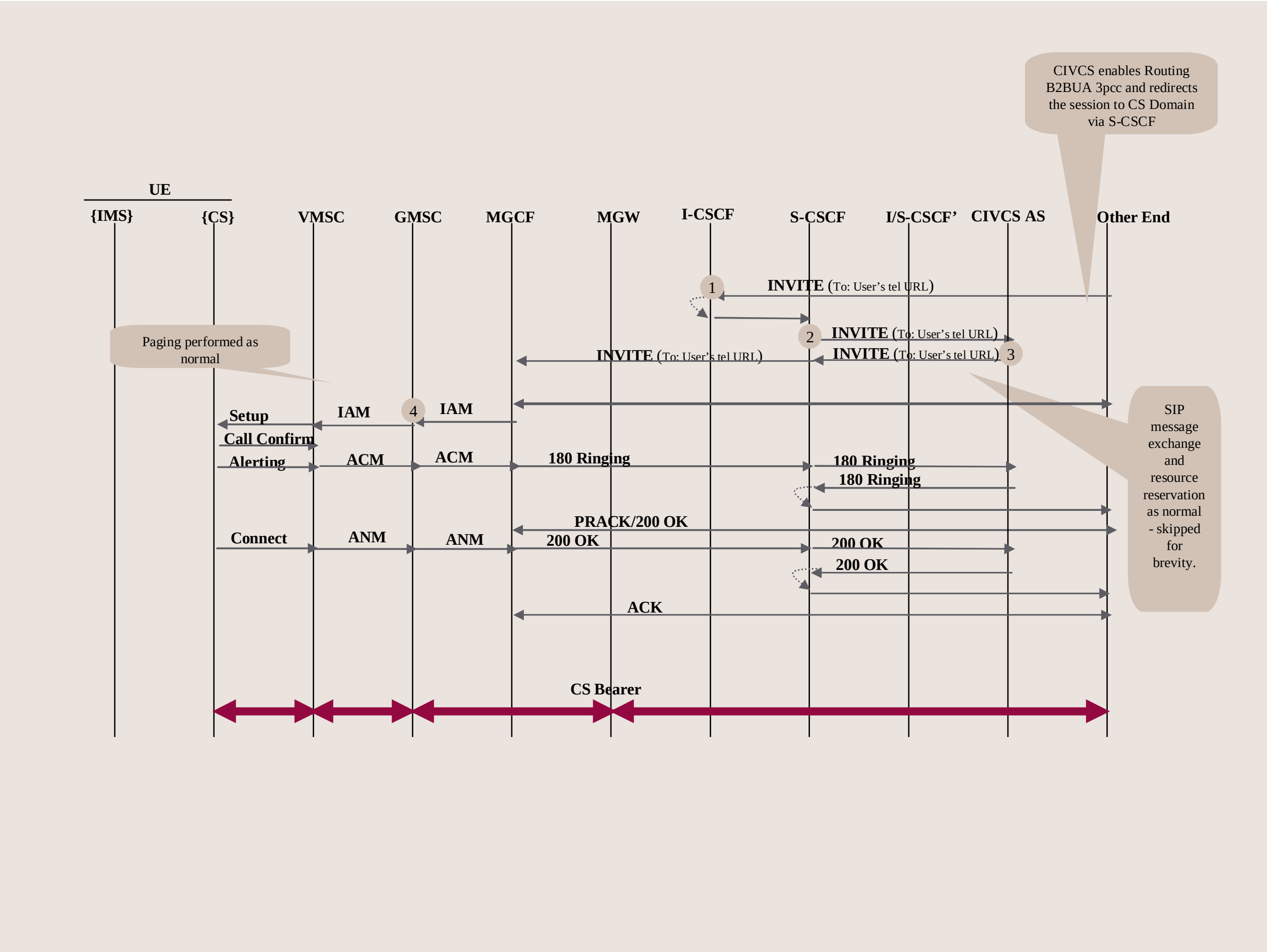


Figure 2a: CS Termination Anchored at CIVCS walk-through

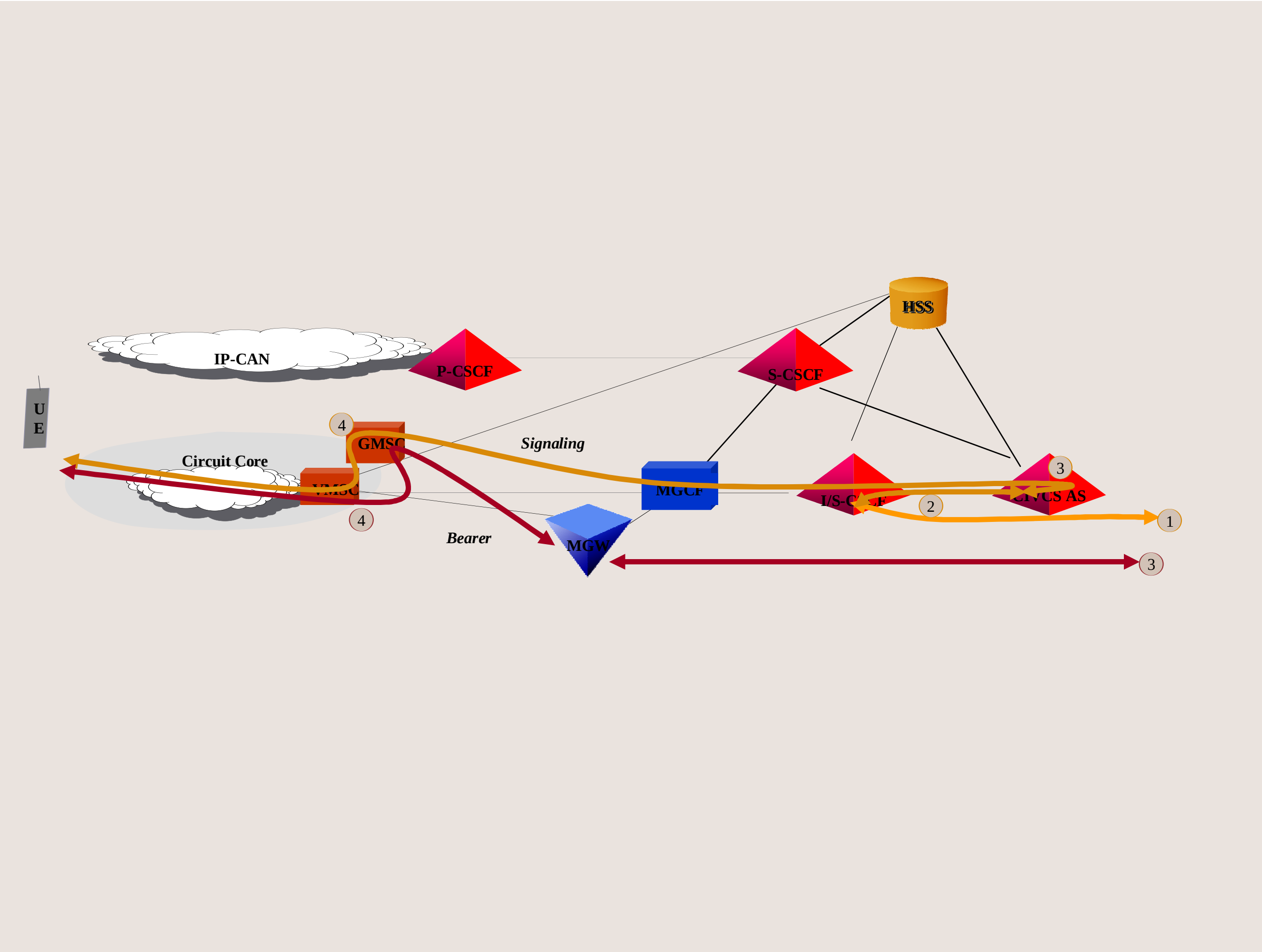


Figure 2b: CS Termination Anchored at CIVCS walk-through

1. A call originated in IMS is delivered at the user's home I-CSCF function. A call originated in CS Domain and/or PSTN is delivered to the MGCF function in the user's home IMS network. "Mobile Terminating call procedures to unregistered IMS Public User Identity that has services related to unregistered state" as specified by 3GPP TS 23.228, "IP Multimedia Systems, Stage 2" are applied to route the call to a temporary S-CSCF. Procedures for S-CSCF assignment when user is registered in IMS are applied when the user is registered in IMS at the time of incoming call delivery.

2. Filter Criteria at temporary S-CSCF result in forwarding of the INVITE to the CIVCS application server.

3. CIVCS enables a 3pcc function to maintain session states for the originating and terminating legs of the call in order to control bearer upon Handover requests from the UE. "Mobile termination, CS Domain Roaming procedures" as described in 3GPP TS 23.228, "Stage 2 IMS specification" and "Routing Sessions from the IMS to the CS domain" as described in 3GPP TS 23.221, Architecture requirements, are subsequently applied to route the terminating leg to the GMSC in the CS Domain via an MGCF.

4. GMSC performs normal GMSC procedures to route the call to the user via the Visited MSC that the user is currently registered at.

NOTE: Direct application of 3GPP procedures referenced in this section requires a direct connection of GMSC with the home IMS network which may not always be possible. Use of these procedures will result in a circular routing loop between the PSTN network and the IMS network if a direct connection is not possible. Furthermore, it results in two HSS/HLR dips, one at the I-CSCF in the user's home IMS network and the other at the GMSC in user's home CS network. It is therefore required that "Mobile Terminating call procedures to unregistered IMS Public User Identity that has services related to unregistered state" and "Mobile termination, CS Domain Roaming procedures" as specified by 3GPP TS 23.228, "IP

Multimedia Systems, Stage 2", be reviewed to evaluate possible corrections and optimizations of the incoming call delivery for users with subscriptions in both domain which are roaming in CS Domain.

6.3.6 Handover Scenarios

6.3.6.1 CS UE to CS UE call

6.3.6.2 CS UE to IMS UE call

6.3.6.2.1 CS to IMS Handovers

Figures1a and 1b describes how signalling and bearer paths are established for execution of Handover of CS originations from CS Domain to IM Subsystem. 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.

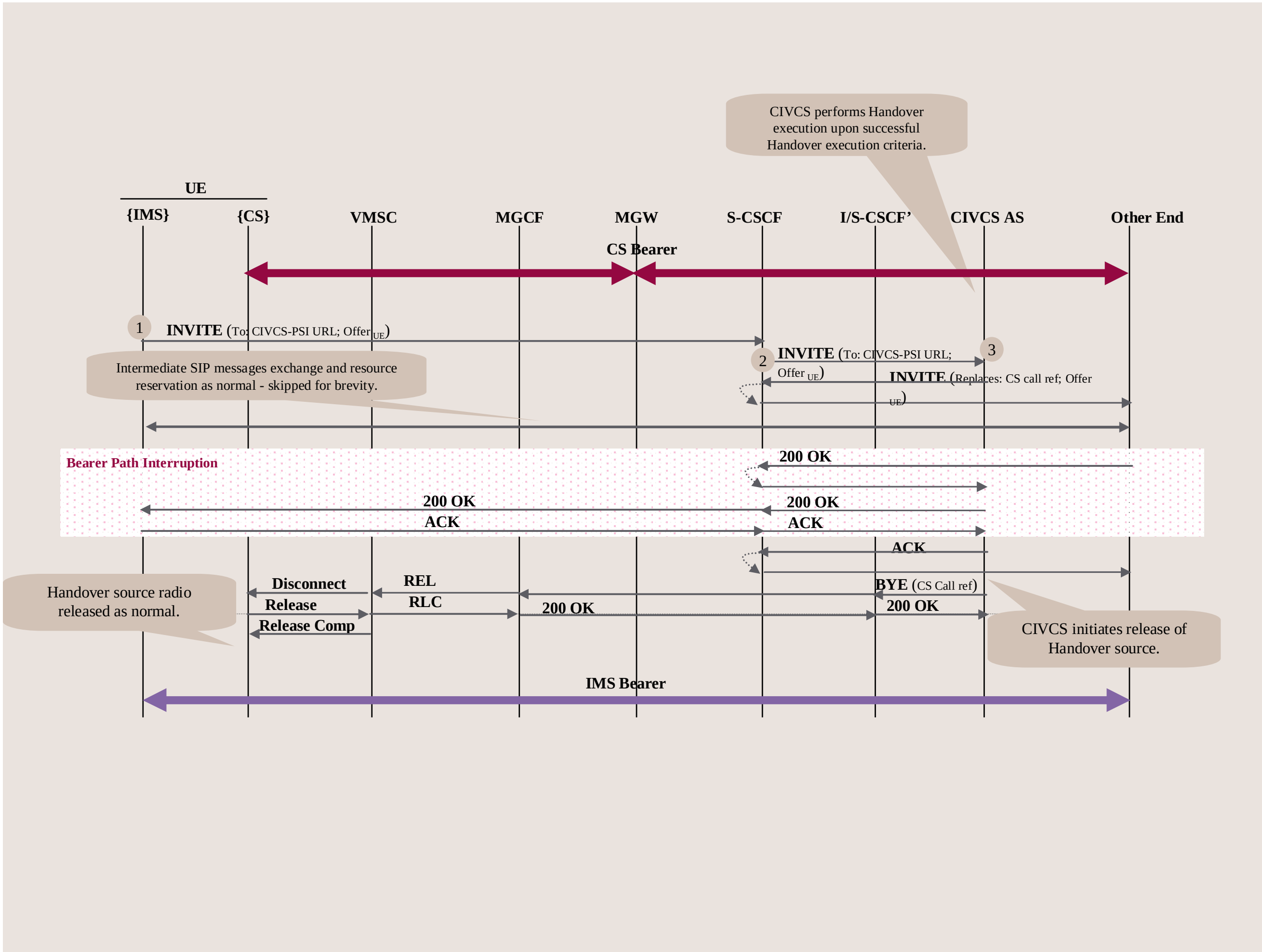


Figure 1a: CS to IMS Handover walk-through

NOTE: The CS bearer indicates the bearer path for the user when being served in CS Domain, whereas the IMS bearer shows bearer path for the user when being served in IMS.

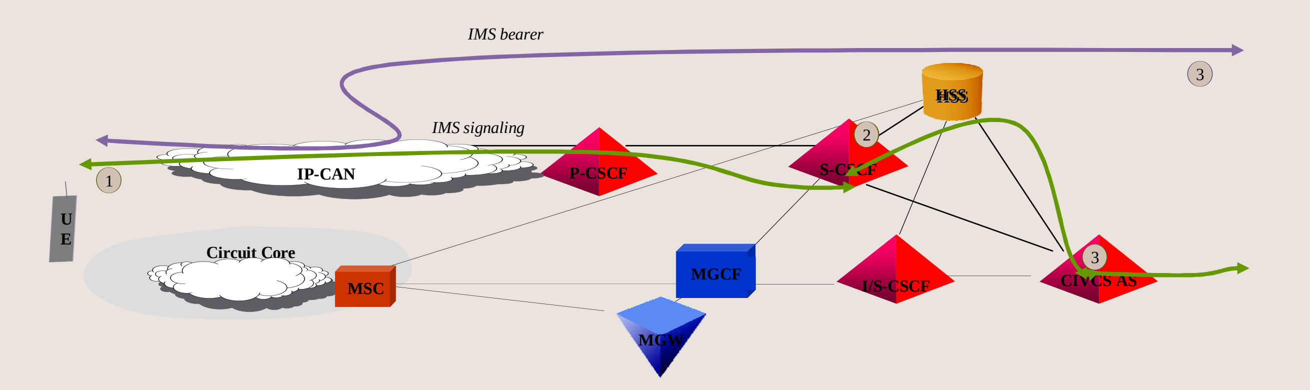


Figure 1b: CS to IMS Handover walk-through

1 If the user is not registered with IMS at the time when the UE determines a need for Handover to IMS, the UE initiates Registration with IMS. It subsequently sends an INVITE to CIVCS using CIVCS PSI requesting it to perform Handover of the active CS call to IM Subsystem.

2 User's S-CSCF routes the INVITE to CIVCS service instance assigned to the user upon execution of filter criteria.

3 CIVCS 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 INVITE with Replaces header consisting of the SDP of the IMS leg is illustrated here; however, other options such as SIP REFER and UPDATES methods 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. The CS bearer and signalling legs are released upon successful execution of SIP Transfer.

Editor's Note: IETF enhancements to enable SIP Session Transfer with "make before break" sequence which will result in quality of user experience similar to Handovers within GSM/UMTS CS are for further study.

6.3.6.2.2 Subsequent Handback to CS

Figure 2a and 2b describes how signalling and bearer paths are established for execution of subsequent Handback to CS Domain.

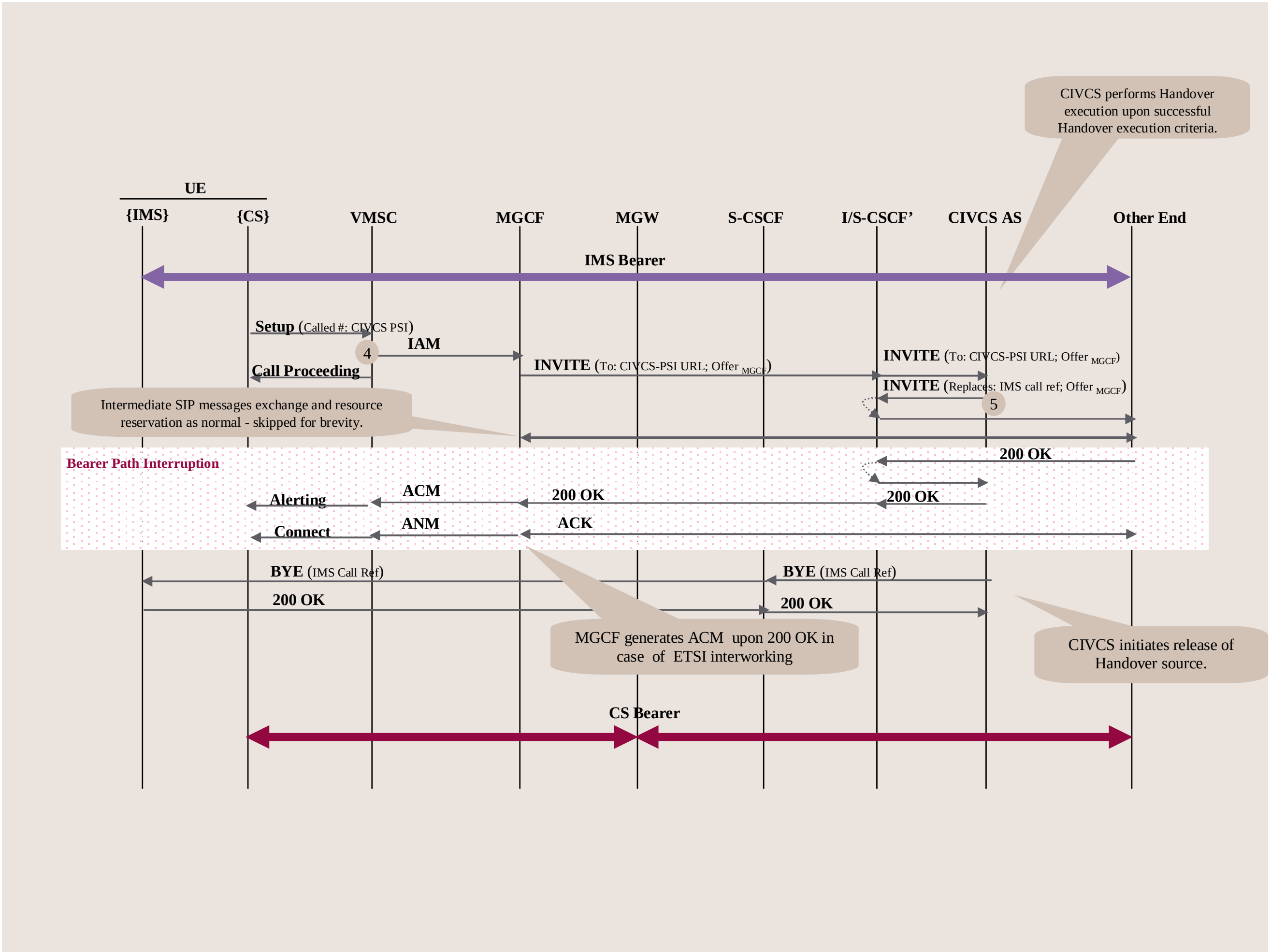


Figure 2a: Subsequent Handback to CS walk-through

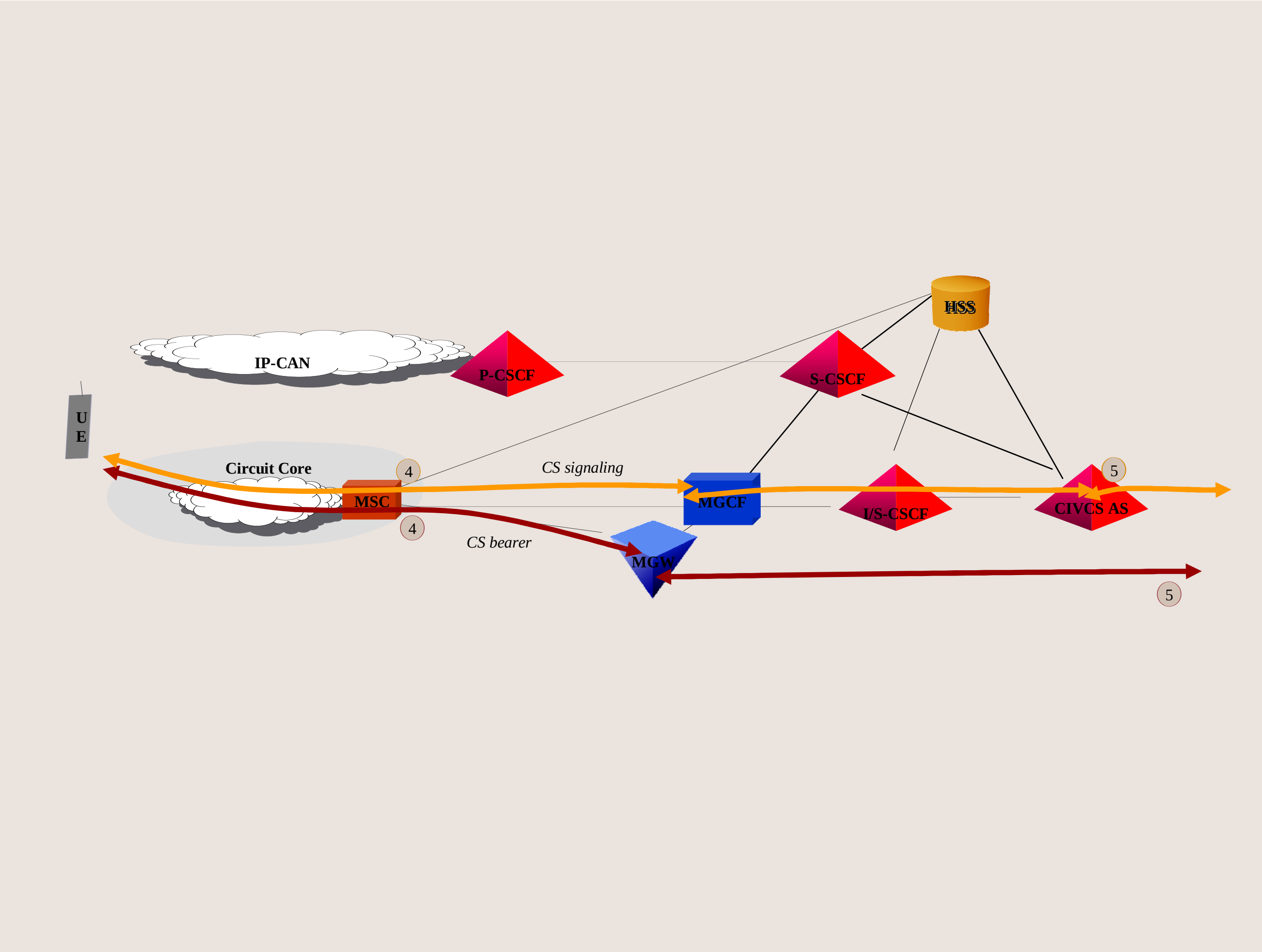


Figure 2b: Subsequent Handback to CS walk-through

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

5 CIVCS 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 Handover walk-through. The IMS bearer and signalling legs are released upon successful execution of SIP Transfer.

6.3.6.2.3 AS discovery in CS to IMS Handovers

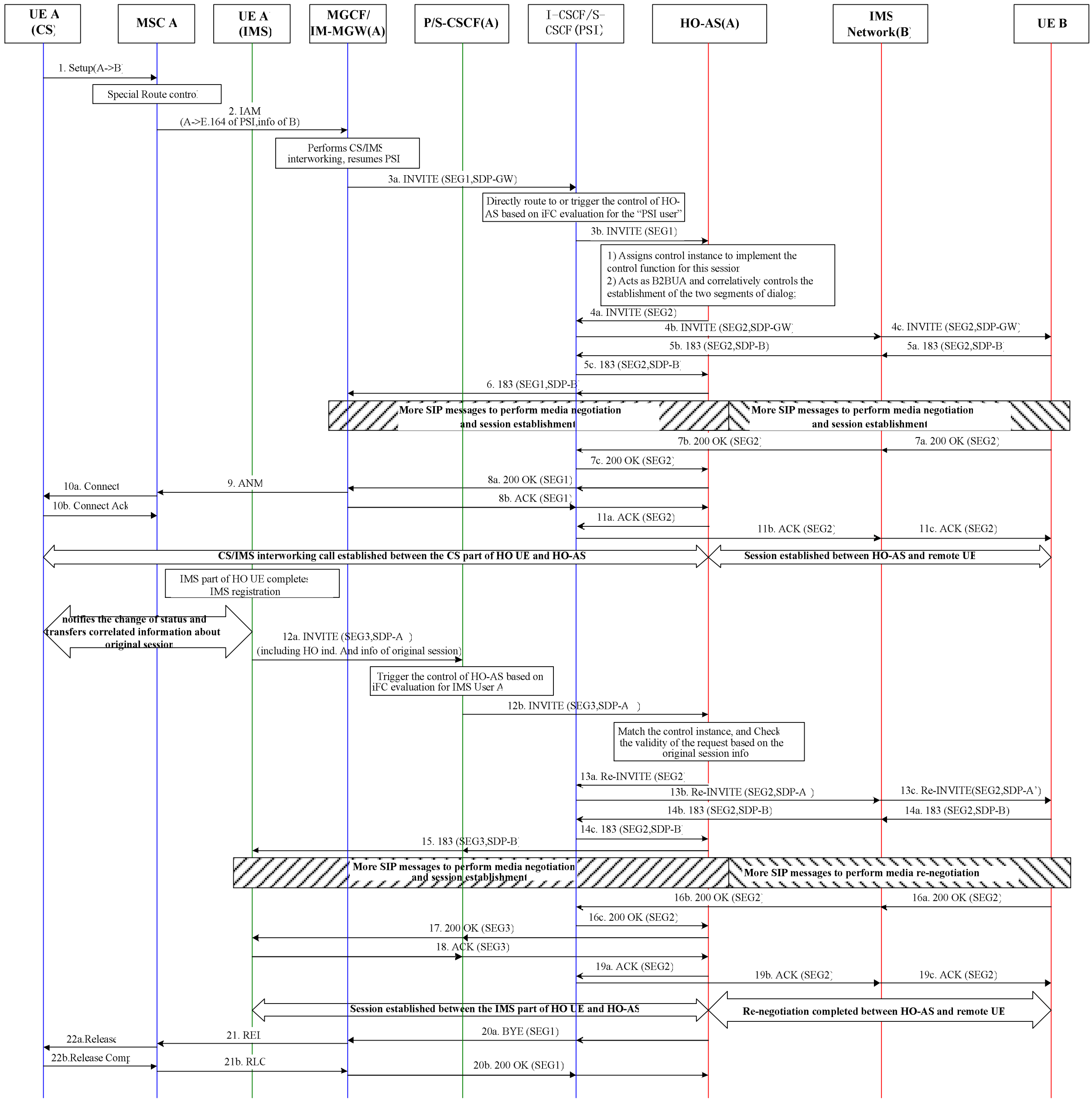


Figure 1: CS handover to IMS (Calling Party handover for example)

1. When user A uses his dual-mode UE to establish a communication with remote user B from CS domain, the corresponding MSC (MSC-A) shall perform special routing control based on the user's subscription data or based on a special prefix the user dialled. Then the called party number is changed to an E.164 number or with a service prefix corresponding to a HO-PSI, and the originating CS call is routed to a CS-IMS interworking gateway (MGCF/IM-MGW). The IAM message sent to MGCF should also include the original dialled number of user B, how to do it is FFS. (1~2)
2. The MGCF/IM-MGW performs CS-IMS interworking and resume the HO-PSI based on its local configuration or inquiry of ENUM server, then forwards the INVITE message to an AS hosting the HO-PSI according to the standard

procedure of "PSIs on the terminating side" described in TS 23.228. (3)

Note1: In case of that there are more than one HO AS in a domain to serve a huge amount of users, more PSIs may be needed to distinguish the different ASs and to ensure the sessions correlating with the same user can be triggered to the same AS.

Note2: There provide two ways to route towards the AS hosting the HO-PSI in TS 23.228, one is directly from I-CSCF to AS based on a HSS query, the other one is from I-CSCF and a S-CSCF assigned for the "PSI user". The former one is recommended here since it is more efficient.

3. HO-AS assigns control instance to implement the control function for this session, acts as B2BUA and correlatively controls the establishment of the two segments of dialog, respectively with the CS part of UE A via the MGCF (dialog 1) and the remote UE (dialog 2); The original dialled number of user B is then used to route the session to the remote UE. These two segments of dialog finally establish the media exchange between IM-MGW and remote UE. (4~11)
4. When the dual-mode UE A decides to initiate a handover, the CS part and IMS part interact with each other to notify the change of status and/or exchange information about the original session. The information about the original session may be Call-ID or a serial number assigned by the HO-AS, and the CS part of the dual-mode UE A may get it during the establishment of original session. How to create it, how to transfer it in SIP/CS signalling is FFS.
5. The IMS part of the UE A then initiates an INVITE to the remote UE or HO PSI to establish a dialog 3, including HO indication and original session information. The so-called HO indication may be an implicit indication such as the appearance of HO-PSI or the original session information, which can be understanded by the HO-AS. The INVITE message is transferred to the S-CSCF assigned for user A during his IMS registration, and then triggered to the same HO-AS based on the user's iFC downloaded from HSS. (12)

   Note: In case of that there are more than one HO AS in a domain and different PSIs are used to distinguish these different ASs, it should keep the configuration in user's iFC consistent with configuration in the user's UEs to ensure the sessions correlating with the same user can be triggered to the same AS.
6. The procedure of signalling and bearer paths establishment for execution of CS to IMS Handover, refer to handover procedure in [1]. Finally, the HO-AS initiates the release of its dialog 1, the CS-IMS interworking call with the CS part of UE A via MGCF/IM-MGW, and completes the procedure of handover from CS to IMS. (13-20)

Note: It is also possible for the UE A to initiate the release of the original connection after handover, but the process of HO-AS shall not relay on it since the UE may be unable to do so as a result of having lost of access in the handling-out domain.

6.3.6.2.4 CS to IMS Handover using ECT

ECT can be used to enable first CS to IMS Handover and establish the anchor at CIVCS so that subsequent CS to IMS and IMS to CS Handovers are executed via CIVCS. Figure 1 below provides a walkthrough of first CS to IMS Handover using ECT.

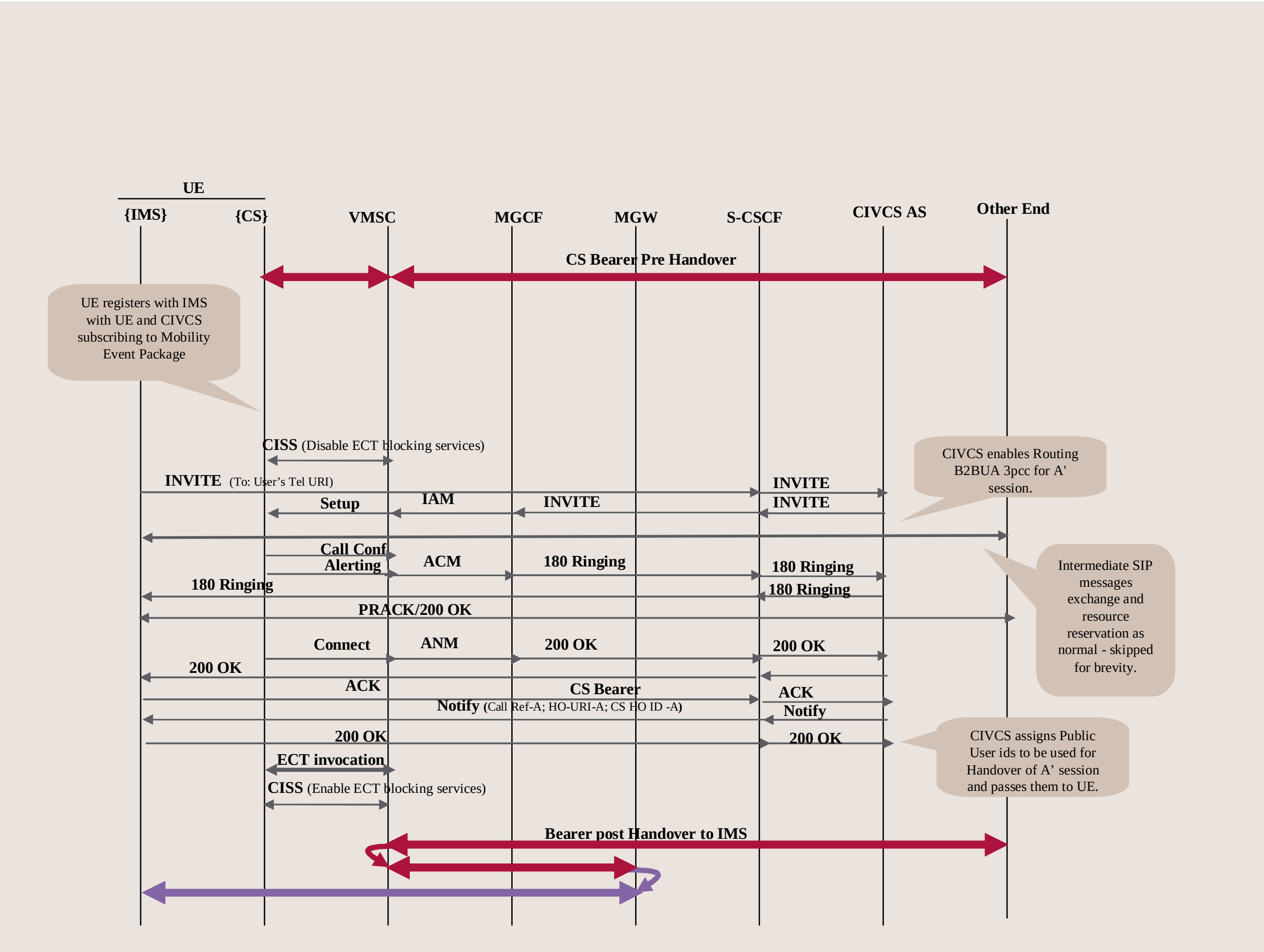


Figure 1: CIVCS anchoring via ECT

- UE registers with IMS as it detects border conditions requiring handover to IMS. If a CIVCS call anchor reference is available, the UE executes CS to IMS Handover via CIVCS as described in a companion paper (ref [1]). If a CIVCS call anchor reference is not available at the UE, then it enables ECT as a mechanism to execute Handover 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.

- CIVCS inserts a routing B2BUA function of completion of the call toward the CS Domain as a result of filter criteria execution at S-CSCF.

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

The UE re-enables the supplementary services disabled previously upon successfully receiving the incoming CS call.

6.3.6.2.5 First CS to IMS Handover using DACCI

Figure 1 below provides a walk through of a first CS to IMS transition enabled by DACCI.

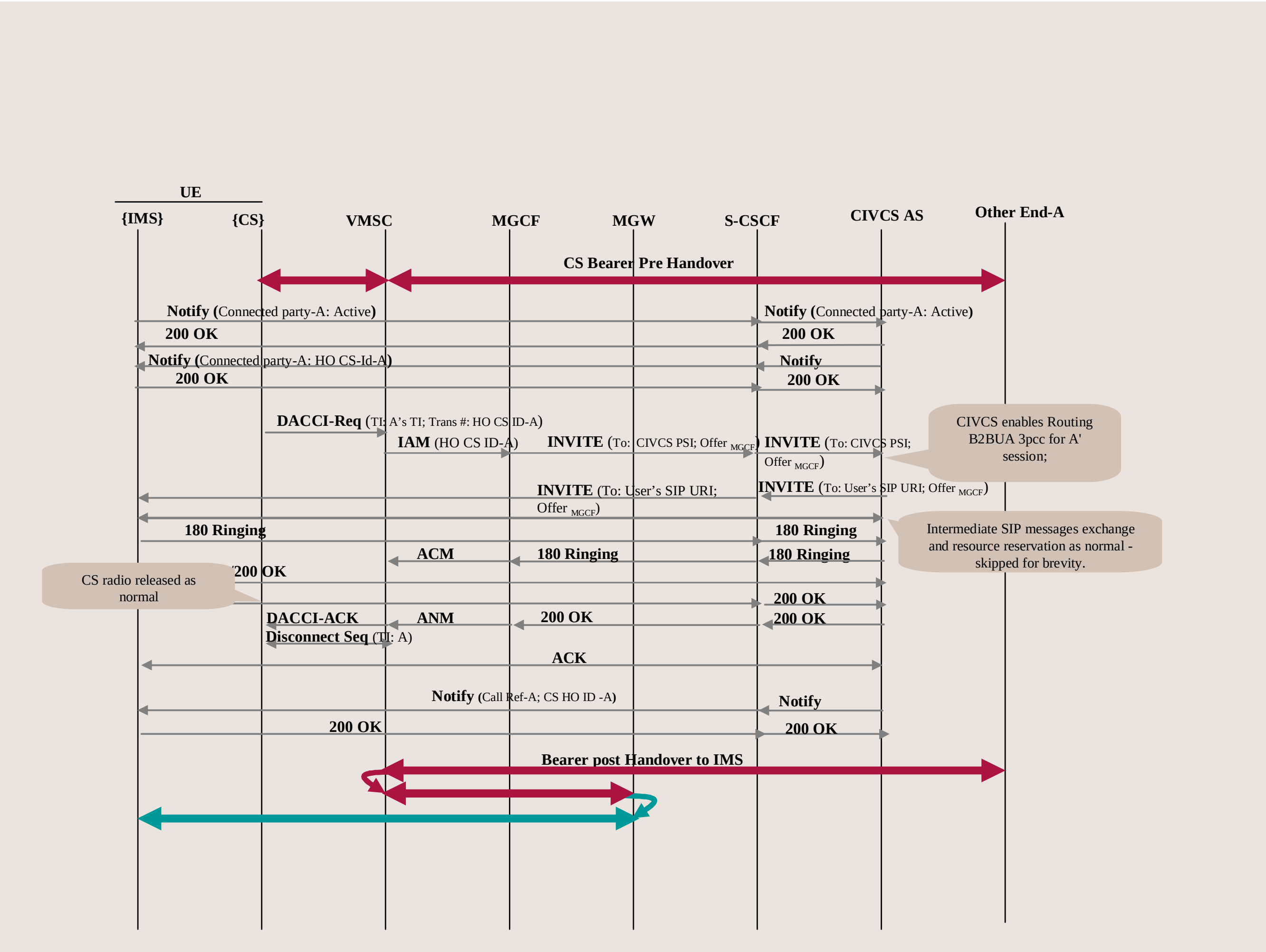


Figure 1: First CS to IMS Handover using DACCI

- The UE registers with IMS as it detects border conditions requiring handover to IMS. If a CIVCS call anchor reference is available, the UE executes CS to IMS Handover via CIVCS. If a CIVCS call anchor reference is not available at the UE, then it uses DACCI to execute Handover to IMS.

- The UE communicates connected party information with an intent to execute DACCI enabled CS to IMS Handover to CIVCS in a Notify. The connected party information is communicated to CIVCS for use in subsequent Handover 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 Handback to CS. CIVCS assigns a unique Handover CS Identifier to be used for the first CS to IMS transition using DACCI and communicates it to the UE in a Notify.

- UE invokes DACCI at the CS Core Network, providing a routing number derived from the CS HO Identifier to set up the Handover leg towards IMS. It also provides the CC Transaction ID of the session to be handed over.

- The MSC uses the routing number to establish an ISUP circuit connection toward CIVCS 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.

- CIVCS 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 Handback to CS, upon successful setup of the IMS session.

- Upon sending 200 OK in response to the INVITE extended from CIVCS, 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 CIVCS and the bearer is anchored at IM-MGW.

6.3.6.3 IMS UE to IMS UE call

6.3.6.3 IMS UE to CS UE call

6.3.6.3.1 IMS to CS handovers

Figure 3a and 3b describes how signalling and bearer paths are established for execution of IMS to CS Handover.

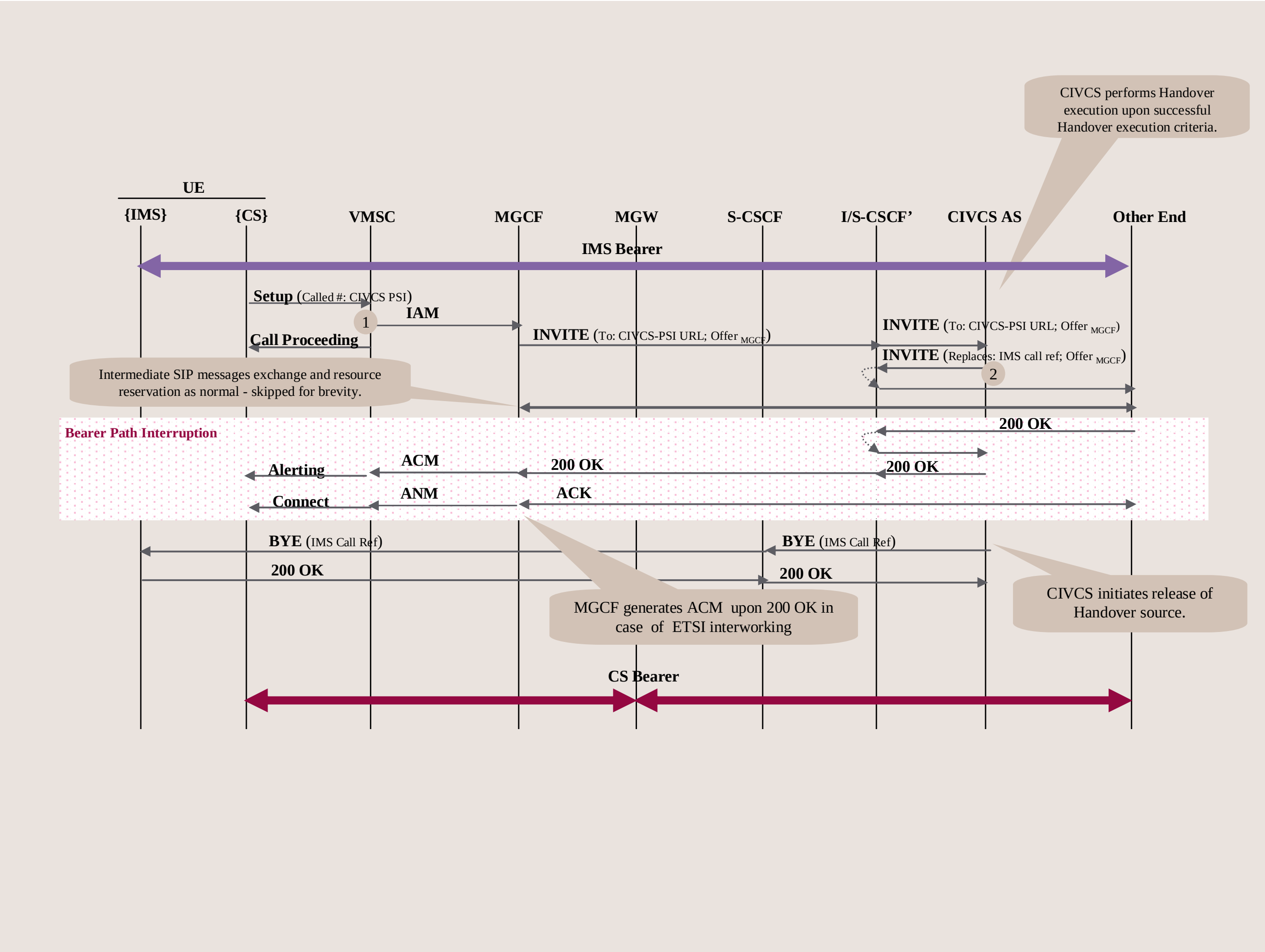


Figure 3a: IMS to CS Handover walk-through

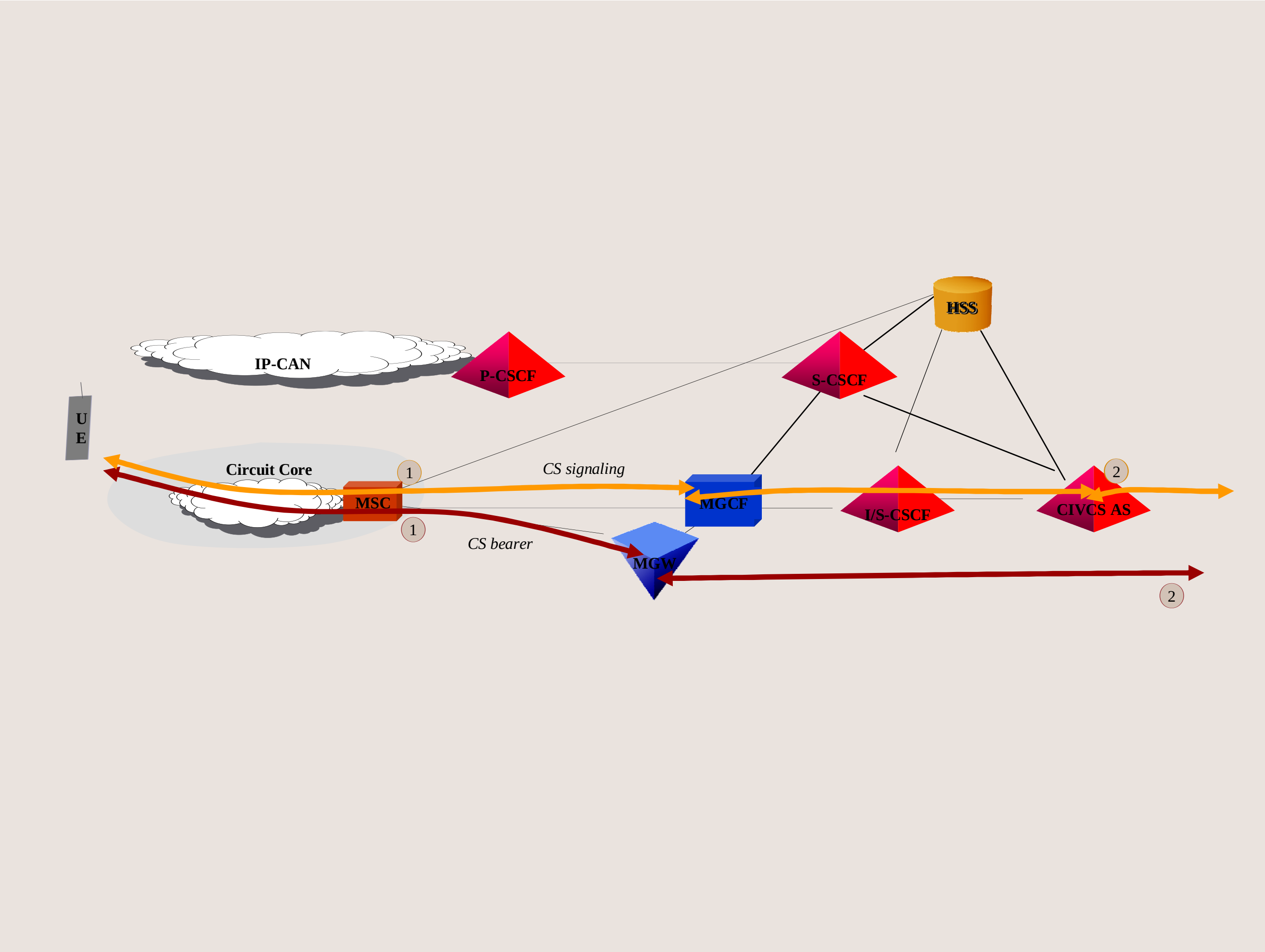


Figure 3b: IMS to CS Handover walk-through

Note that the Handover procedures executed in steps 1 and 2 are the same as Handover procedures executed in steps 4 and 5 in Figures 2a and 2b as the Handover procedure between CS Domain and IMS is agnostic of the previous Handover history.

6.3.6.3.2 AS discovery in IMS to CS Handovers

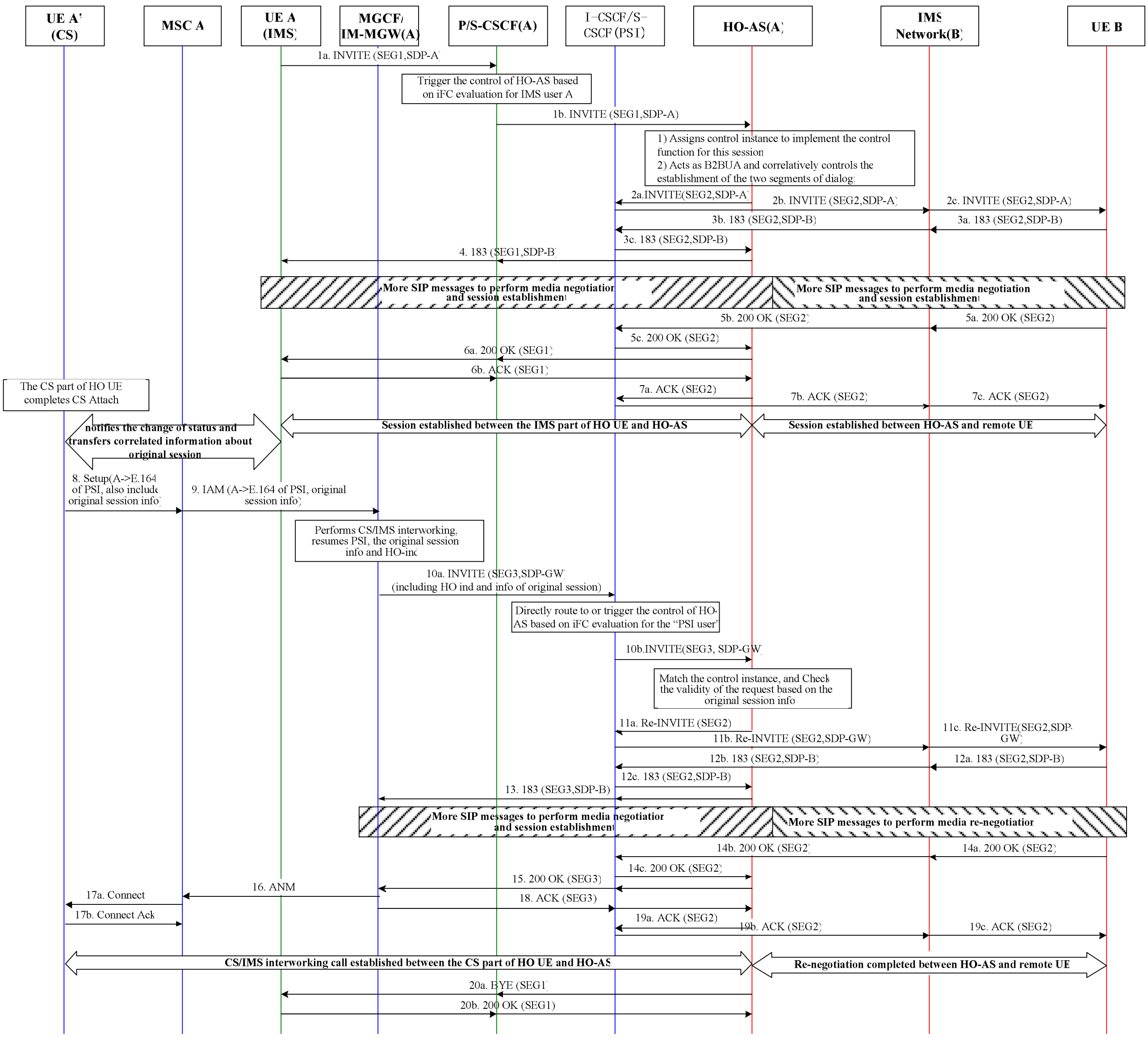


Figure 2: IMS handover to CS (Calling Party handover for example)

1. When user A uses his dual-mode UE to establish a communication with remote user B from IMS domain, the IMS part of UE A initiates an INVITE to the remote UE. The INVITE message is transferred to the S-CSCF assigned for user A during his IMS registration, and then triggered to a HO-AS based on the user's iFC downloaded from HSS. (1)
2. The HO-AS assigns control instance to implement the control function for this session, acts as B2BUA and correlatively controls the establishment of the two segments of dialog, respectively with the IMS part of UE A (dialog 1) and the remote UE (dialog 2); These two segments of dialog finally establish the media exchange between the IMS part of UE A and remote UE. (4~7)
3. When the dual-mode UE A decides to initiate a handover, the CS part and IMS part interact with each other to notify the change of status and/or exchange information about the original session.
4. The CS part of the UE A then initiates a setup request to the corresponding MSC (MSC-A), while the called party number is the E.164 number corresponding to the HO-PSI, and the corresponding MSC (MSC-A) then routes the CS call to a CS-IMS interworking gateway (MGCF/IM-MGW) based on the analysis of the E.164 number

corresponding to the HO-PSI. The Setup and IAM message should include the original session information, which will be used to find the original control instance in the HO-AS. The information about the original session may be Call-ID or a serial number assigned by the HO-AS, and the IMS part of the dual-mode UE A may get it during the establishment of original session. How to create it, how to transfer it in SIP/CS signalling is FFS. (8~9)

5. The MGCF/IM-MGW performs CS-IMS interworking, resumes the HO-PSI based on its local configuration and/or inquiry of ENUM server, the original session information and HO-indication, then forwards the INVITE message to the HO-AS hosting the HO-PSI to establish a dialog 3, the routing of AS hosting a PSI is performed according to the standard procedure of "PSIs on the terminating side" described in TS 23.228. The so-called HO indication may be an implicit indication such as the appearance of HO-PSI or the original session information, which can be understanded by the HO-AS. (10)

   Note 1: In case of that there are more than one HO AS in a domain to serve a huge amount of users, more PSIs may be needed to distinguish the different ASs, and should keep the configuration in user's iFC consistent with configuration in the user's UEs to ensure the sessions correlating with the same user can be triggered to the same AS.

   Note2: There provide two ways to route towards the AS hosting the HO-PSI in TS 23.228, one is directly from I-CSCF to AS based on a HSS query, the other one is from I-CSCF and a S-CSCF assigned for the "PSI user". The former one is recommended here since it is more efficient.

   Note 3: Another choice is that, the CS part of UE A initiates a Setup request while the called party number is its own MSISDN with a special prefix, used to indicate that the call should be routed to the IMS domain. IMS domain entity then transforms this number to the UE A's IMPU and send the INVITE to the S-CSCF assigned for user A, then trigger to the HO-AS based on iFC evaluation for user A. This will ensure the INVITE of dialog 3 be sent to the same S-CSCF which control the original session.

6. The procedure of signalling and bearer paths establishment for execution of IMS to CS Handover, refer to handover procedure in [1]. Finally, the HO-AS initiates the release of its dialog 1 with the IMS part of UE A, and completes the procedure of handover from IMS to CS (11-20)

Note: It is also possible for the UE A to initiate the release of the original connection after handover, but the process of HO-AS shall not relay on it since the UE may be unable to do so as a result of having lost of access in the handling-out domain.

6.3.6.5.1 Subsequent Handovers and Handbacks with DACCI enabled anchoring

Once a session anchor has been established at CIVCS and the session identifiers have been communicated to the UE, the UE executes all Inter domain Handovers via CIVCS using Handover procedures described elsewhere. Subsequent Handovers 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 transition as shown in Figure 2 below.

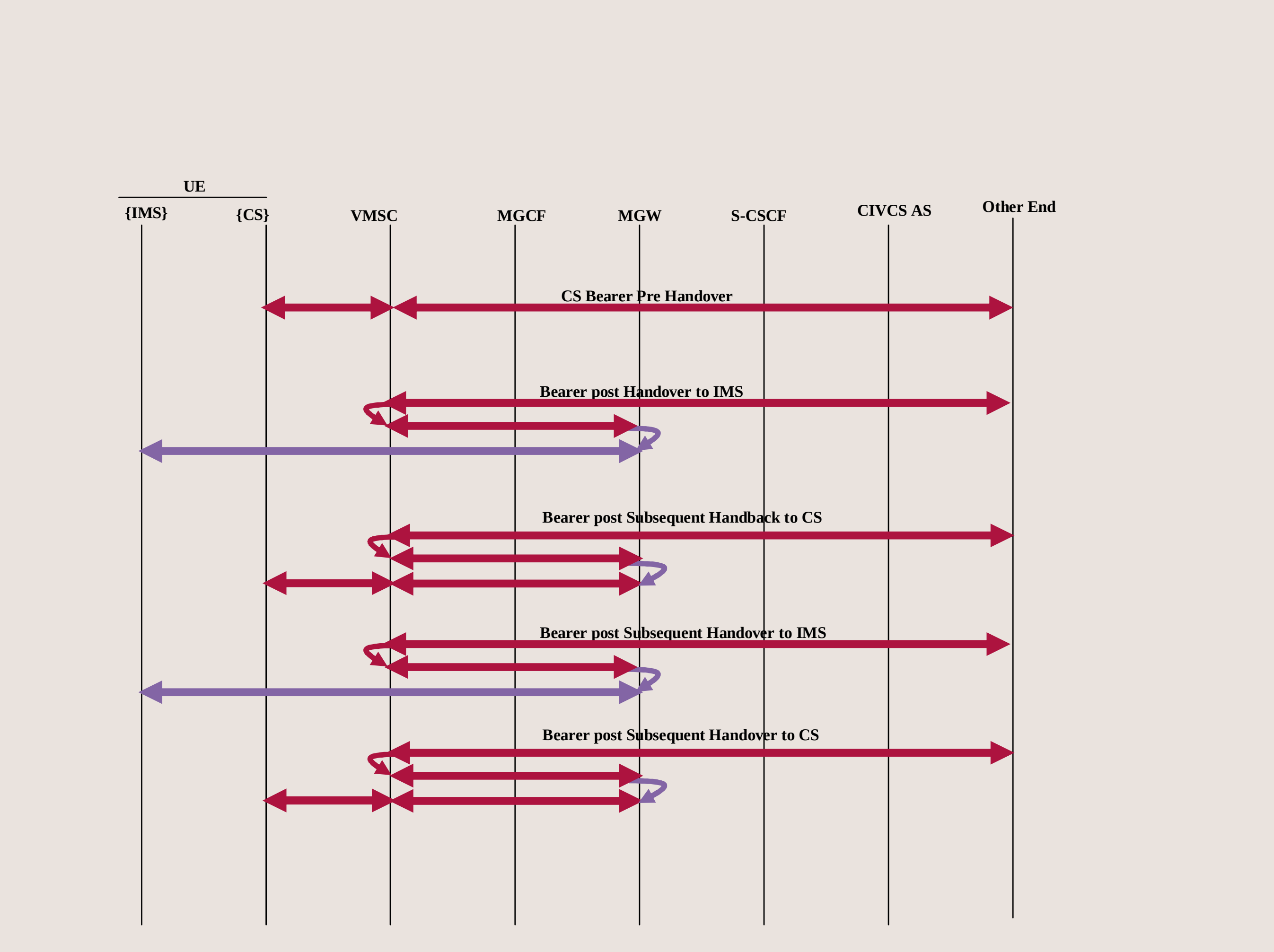


Figure 2: Use of CIVCS for subsequent Handovers

6.3.6.5.3 Subsequent Handovers and Handbacks

Use of ECT for subsequent Handovers and Handbacks results in a daisy chain effect as shown in Figure 2 below, due to the fact that the call anchor moves between CS and IMS upon each Handover.

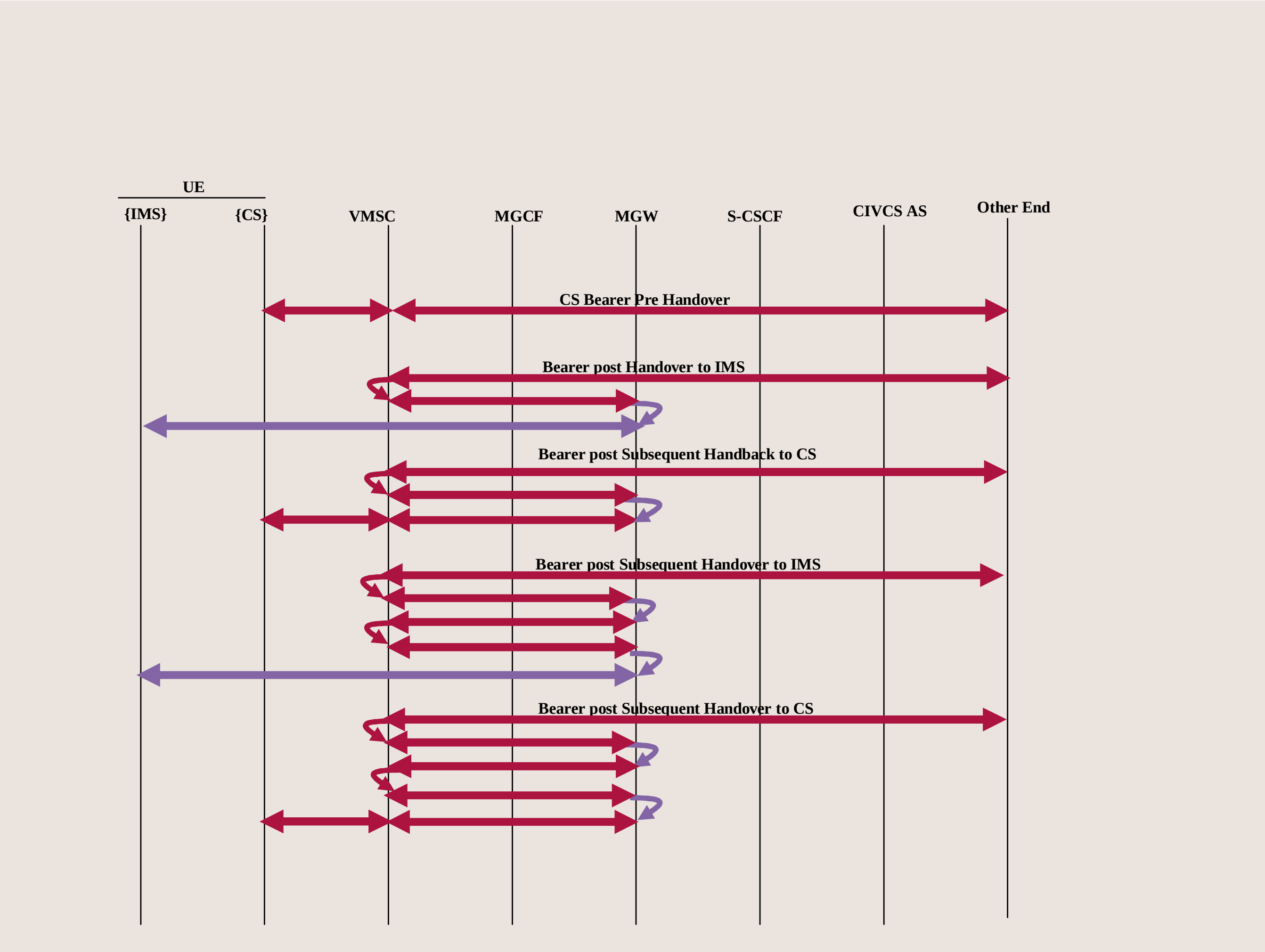


Figure 2: Undesirable Resource Daisy Chain with use of ECT for Subsequent Handovers

Once a session anchor has been established at CIVCS and the session identifiers have been communicated to the UE, the UE executes all Inter domain Handovers via CIVCS. This helps eliminate the daisy chain effect as shown in Figure 3 below, because the call remains anchored in the IMS MGW after the first CS to IMS Handover.

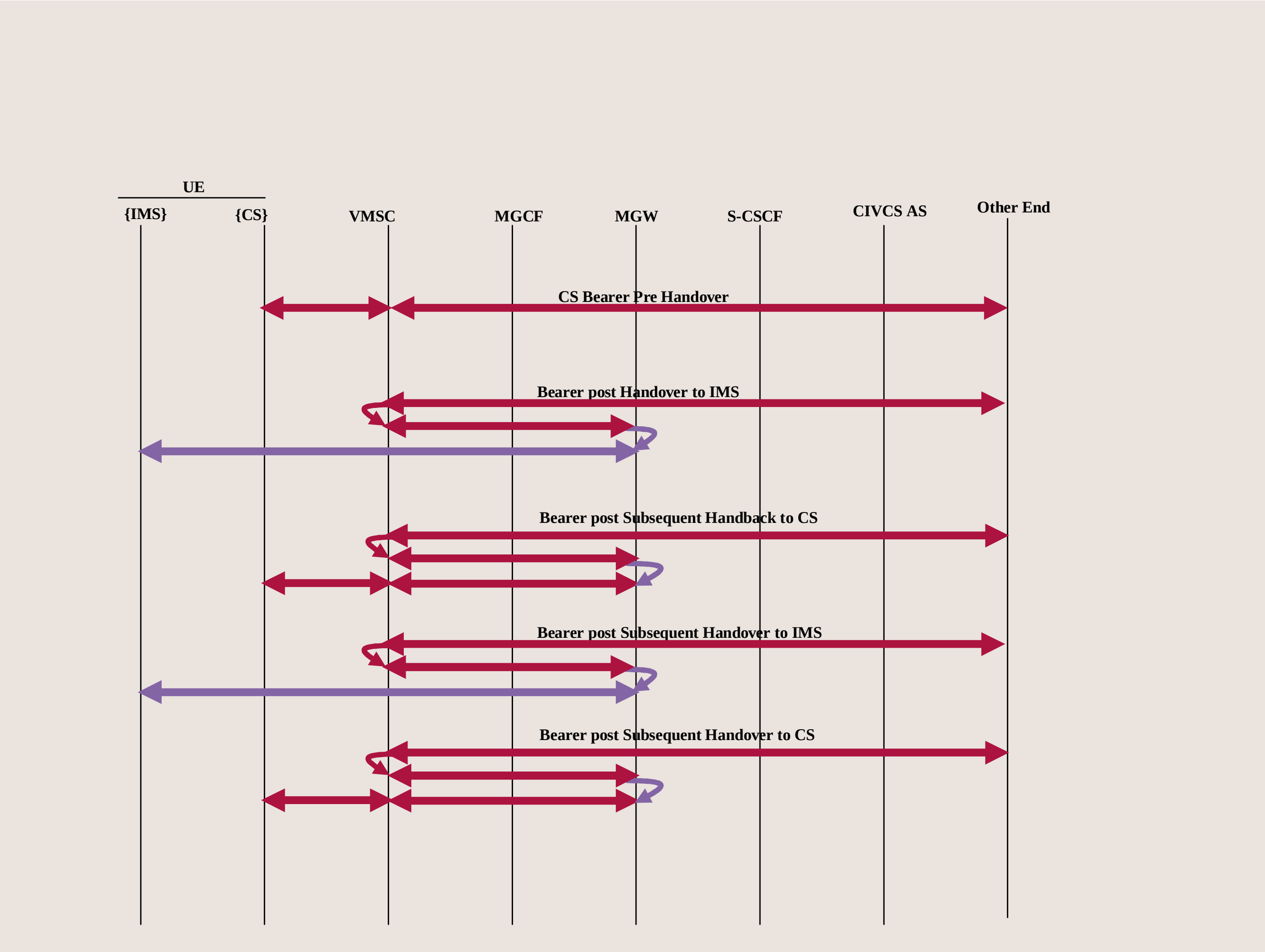


Figure 3: Resource optimization with use of CIVCS for subsequent Handovers

6.3.7 Impact on Supplementary Services

6.3.7.1 Voice Call Continuity for Multi-Session calls

CIVCS uses the Mobility Event package to communicate with the UE, session specific information that is required to perform handover 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 CIVCS via 3pcc Routing B2BUA function using static anchoring or dynamic anchoring techniques. Upon successful allocation of a B2BUA function for a particular user session, CIVCS assigns it a unique identifier along with and a SIP URI that is used to uniquely identify the session when requesting its handover to IMS or a unique CS Handover ID that can be used to uniquely identify the session when requesting its handover 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 CIVCS. Connected party address is used to enable Handovers 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 Handover in these conditions.

The fundamental principle of transferring the call control protocol state machine from the handing-out domain to the handing-in domain discussed elsewhere for single sessions 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 Multi-Session services support with Static Anchoring

6.3.7.1.1.1 Anchoring of IMS Held and Active sessions at CIVCS

Figures 2 & 3 below provides a walkthrough of a scenario in which CS IMS 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.

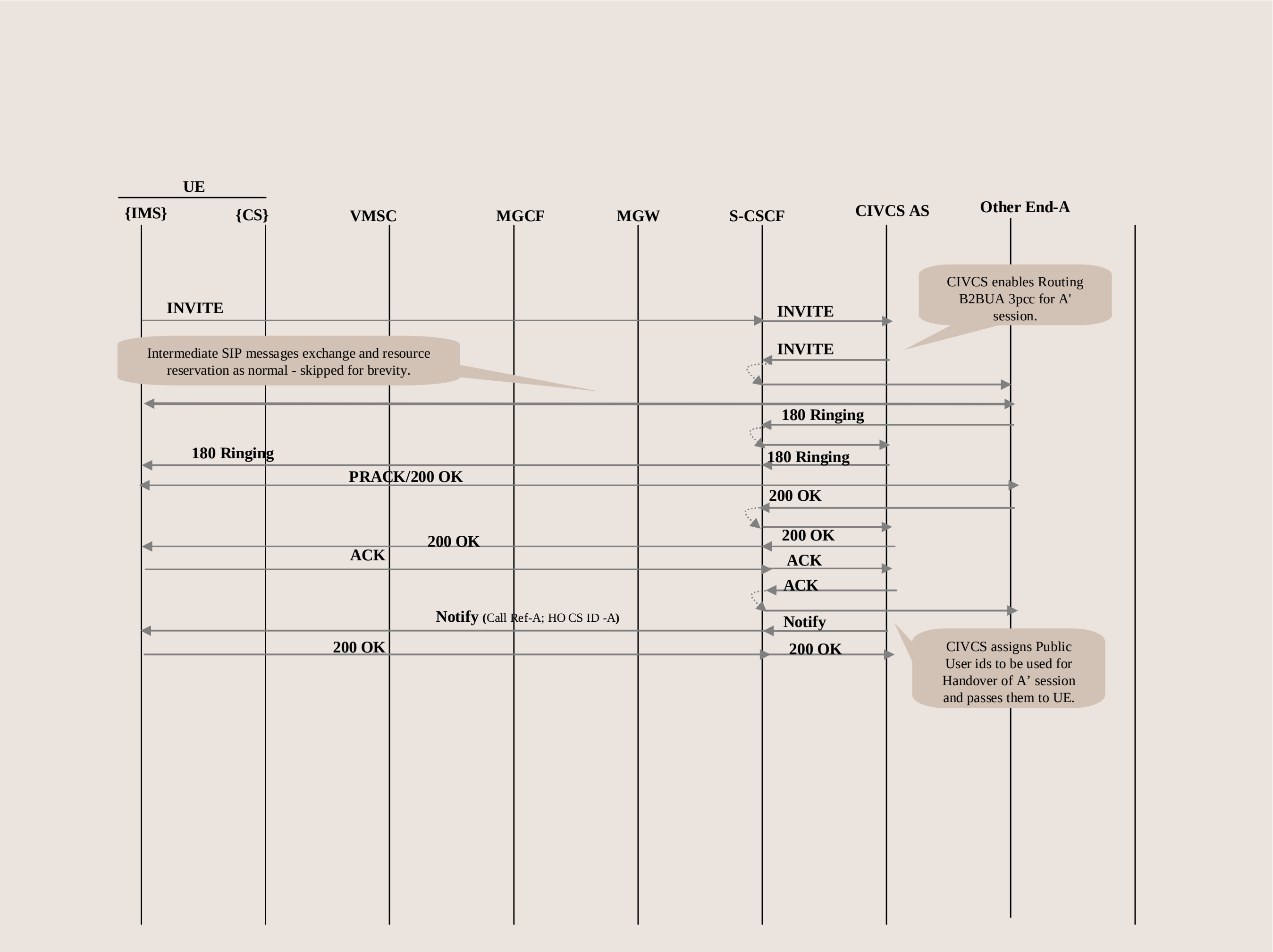


Figure 2: IMS session toward other end – A

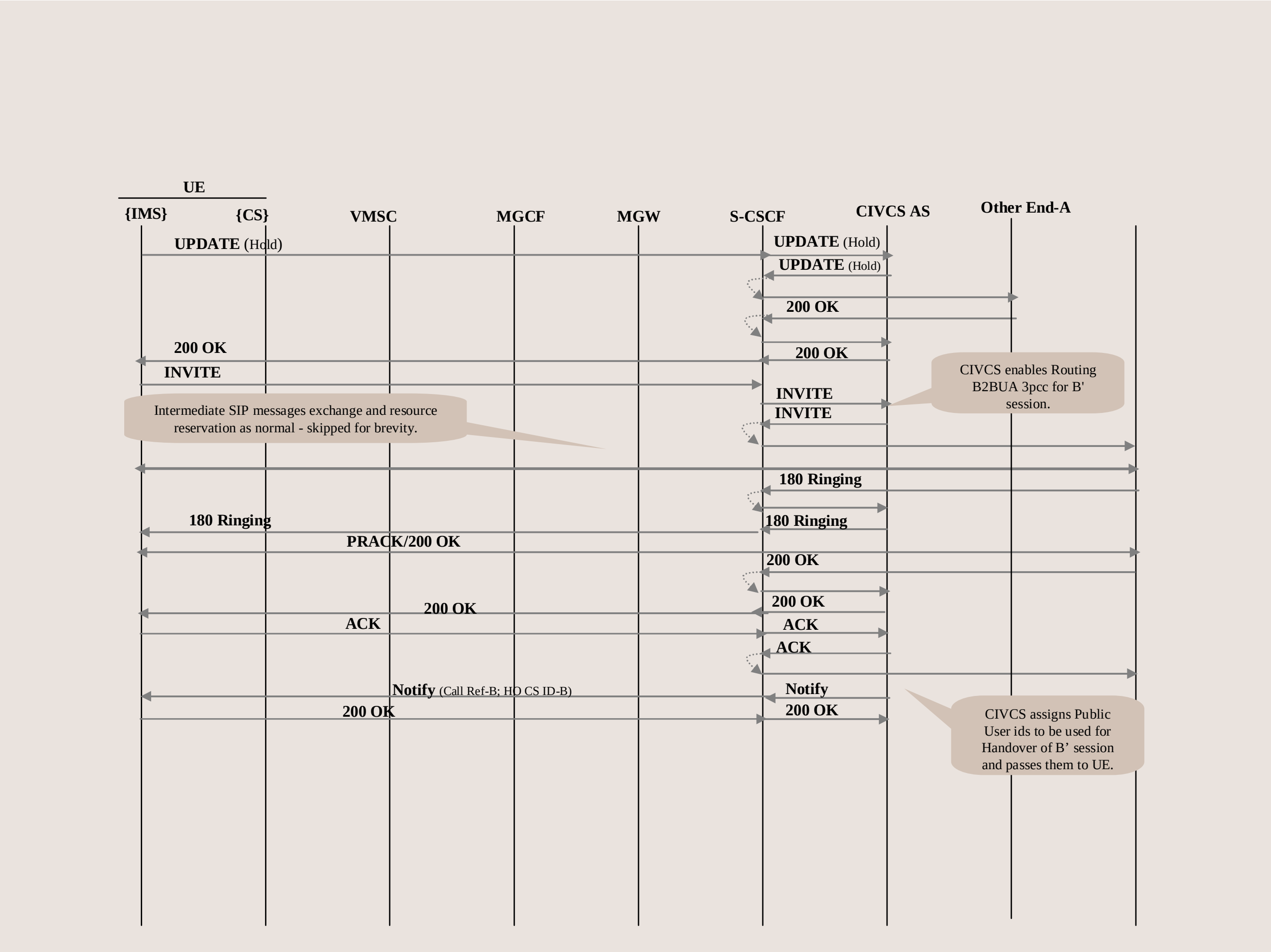


Figure 3: Hold A and originate an IMS session toward other end - B

- As part of IMS registration, CIVCS and the UE subscribe to each other for the Mobility Event package.

- Upon successful execution of a Routing B2BUA function for IMS session to the other end A at CIVCS, CIVCS assigns a unique call reference identifier to the session for identification of the session between the UE and CIVCS in subsequent dialogues. It also assigns a unique identifier which can be used for Handover of this session to CS Domain. The CS Handover 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 CIVCS DN when requesting Handover, the later 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.

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

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

6.3.7.1.1.2 IMS to CS Handover of IMS Held and Active sessions

Figure 4 below provides a walkthrough of Handover of IMS Held and Active sessions established as described in previous section.

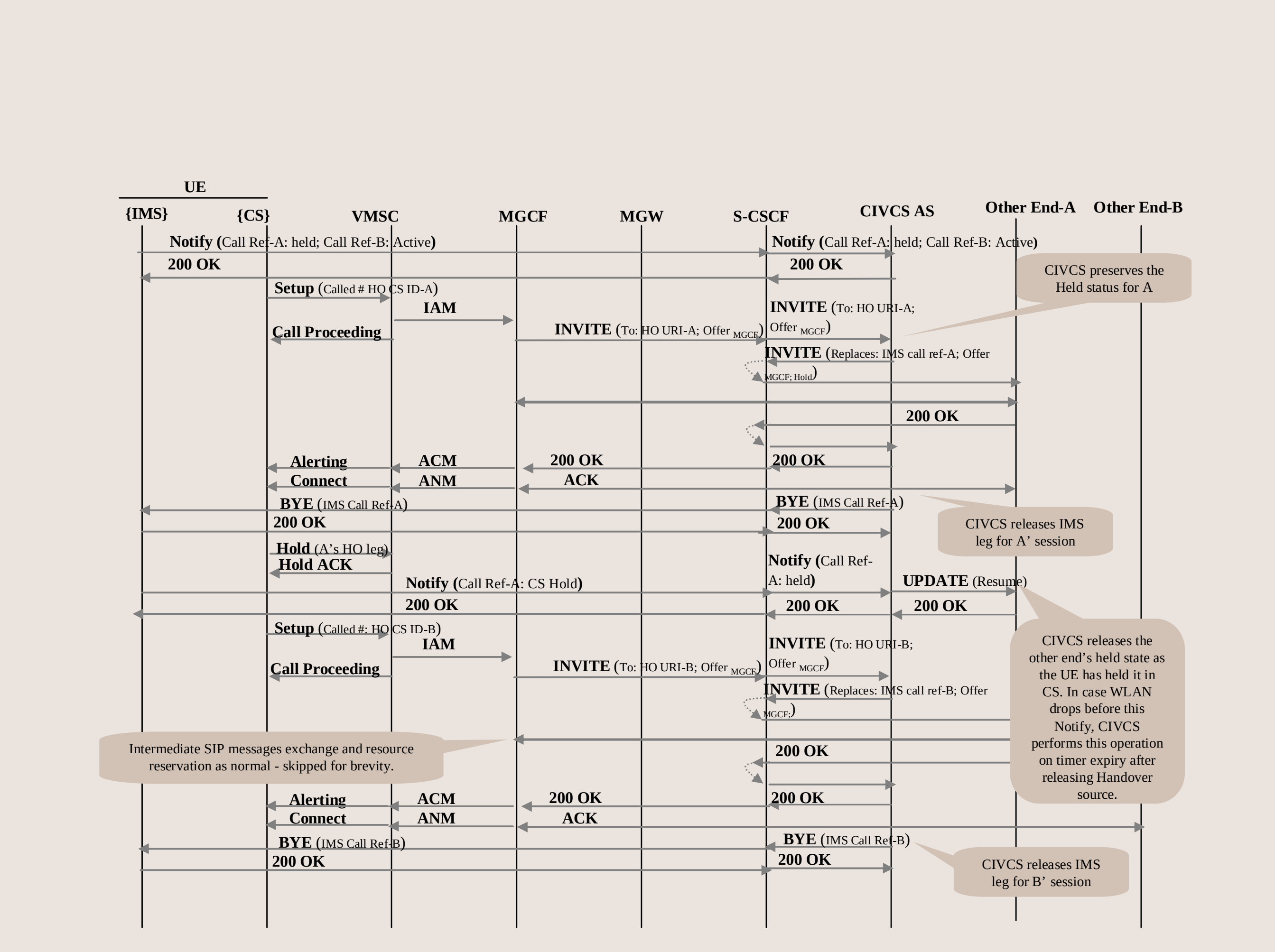


Figure 4: Handover of IMS Held and Active sessions

- Upon detection of border conditions, the UE updates CIVCS with the current session state information to be used during the Handover procedure. Session Hold, Active states are passed to CIVCS 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 CIVCS.

- It should be noted that this message exchange can be avoided if CIVCS 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.

- The UE performs handover of the held IMS leg to CS using IMS to CS Handover procedures described elsewhere. CIVCS ensures that the held status is maintained at the other end A when transferring the CS IMS user from IMS to CS.

- The UE holds the CS Handover leg for A's session at the MSC to re-establish the protocol state machine at the MSC. The UE sends a Notify to CIVCS 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 Handover procedures. CIVCS 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 handover of the Active session to the other B to CS using IMS to CS Handover procedures described elsewhere.

- 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 handing-in domain.

6.3.7.1.1.3 Anchoring of CS Held and Active sessions at CIVCS with active IMS Registration

Figure 5 below provides a walkthrough of a scenario in which CS IMS 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 has active IMS Registration at the time it establishes the CS calls for this walkthrough.

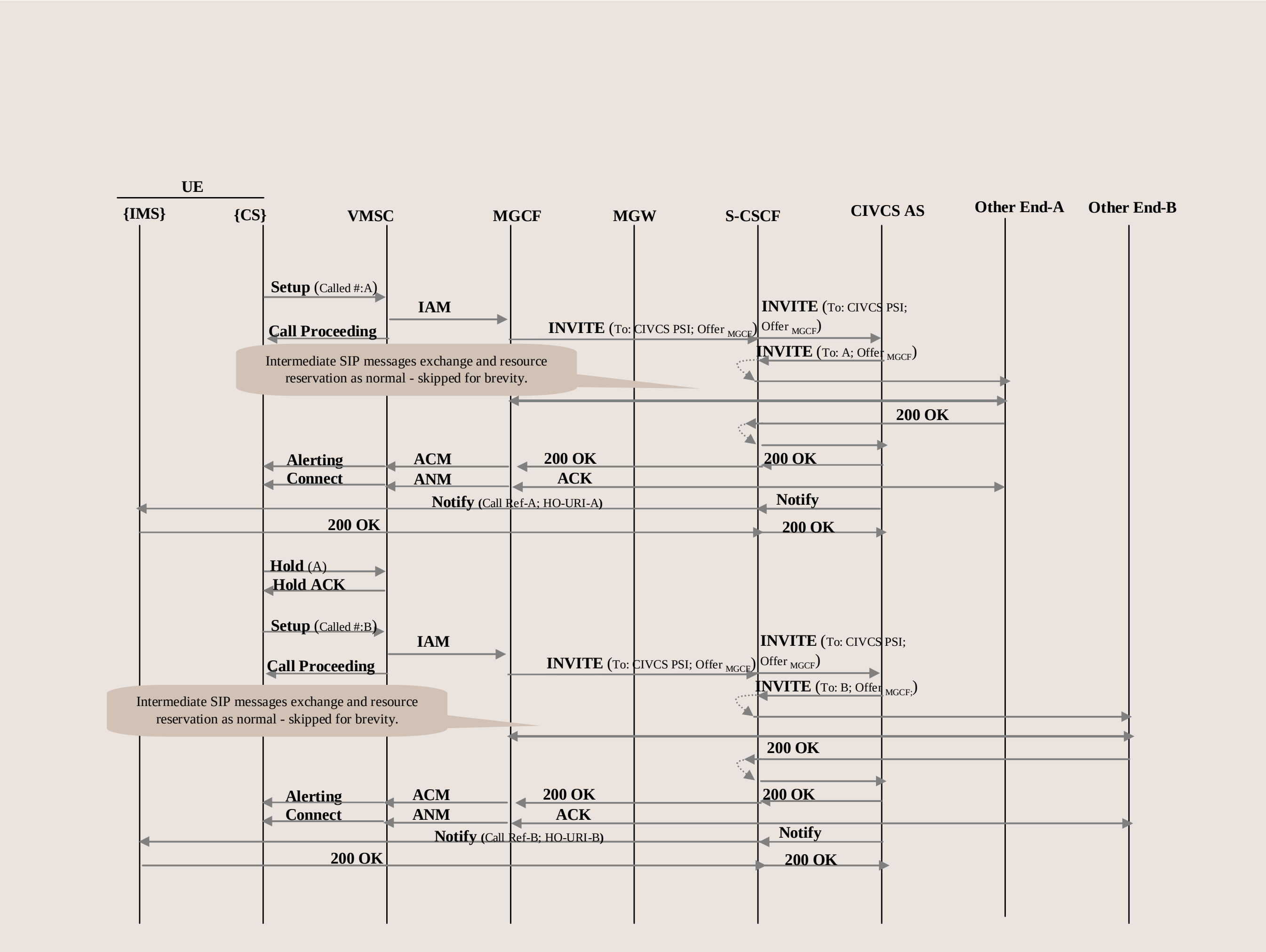


Figure 5: Anchoring of CS Held and Active sessions at CIVCS

- As part of IMS registration, CIVCS and the UE subscribe to each other for the Mobility Event package.

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

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

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

6.3.7.1.1.4 CS to IMS Handover of CS Held and Active sessions; IMS active at the time of CS Anchoring

Figure 6 below provides a walkthrough of Handover of CS Held and Active sessions established as described in previous section.

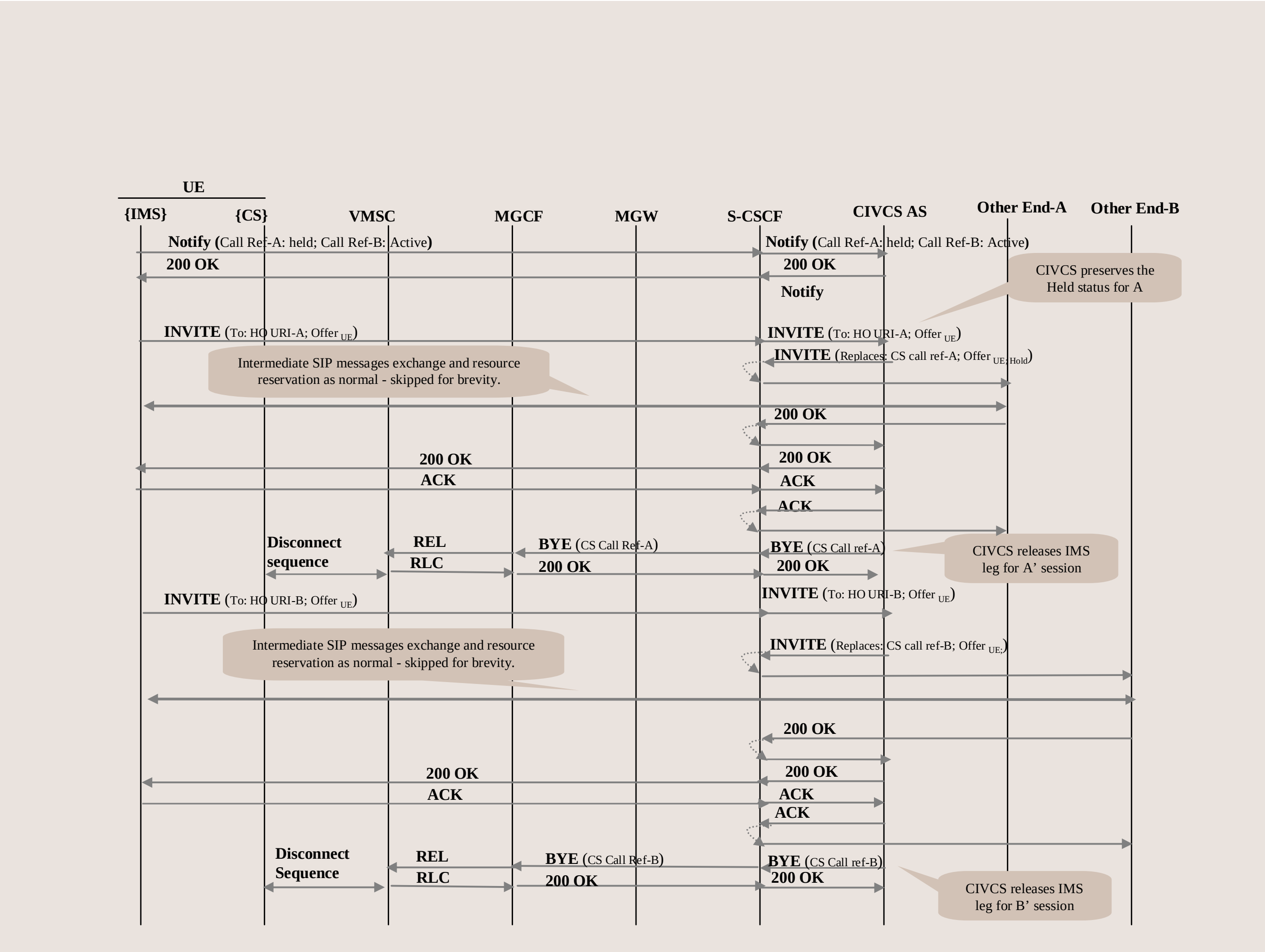


Figure 6: CS to IMS Handover of Held and Active sessions; IMS active at CS anchoring

- Upon detection of border conditions, the UE updates CIVCS with the current session state information to be used during the Handover procedure. Session Hold, Active states are passed to CIVCS 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 CIVCS.

- The UE performs handover of the held CS leg to IMS using CS to IMS Handover procedures described elsewhere. CIVCS ensures that the held status is maintained at the other end A when transferring the CS IMS user from IMS to CS.

- The UE subsequently performs handover of the Active session to the other party B to IMS using CS to IMS Handover procedures described elsewhere.

- 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 handing-in domain.

6.3.7.1.1.5 Anchoring of CS Held and Active sessions at CIVCS without IMS Registration

Figure 7 below provides a walkthrough of a scenario in which CS IMS 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.

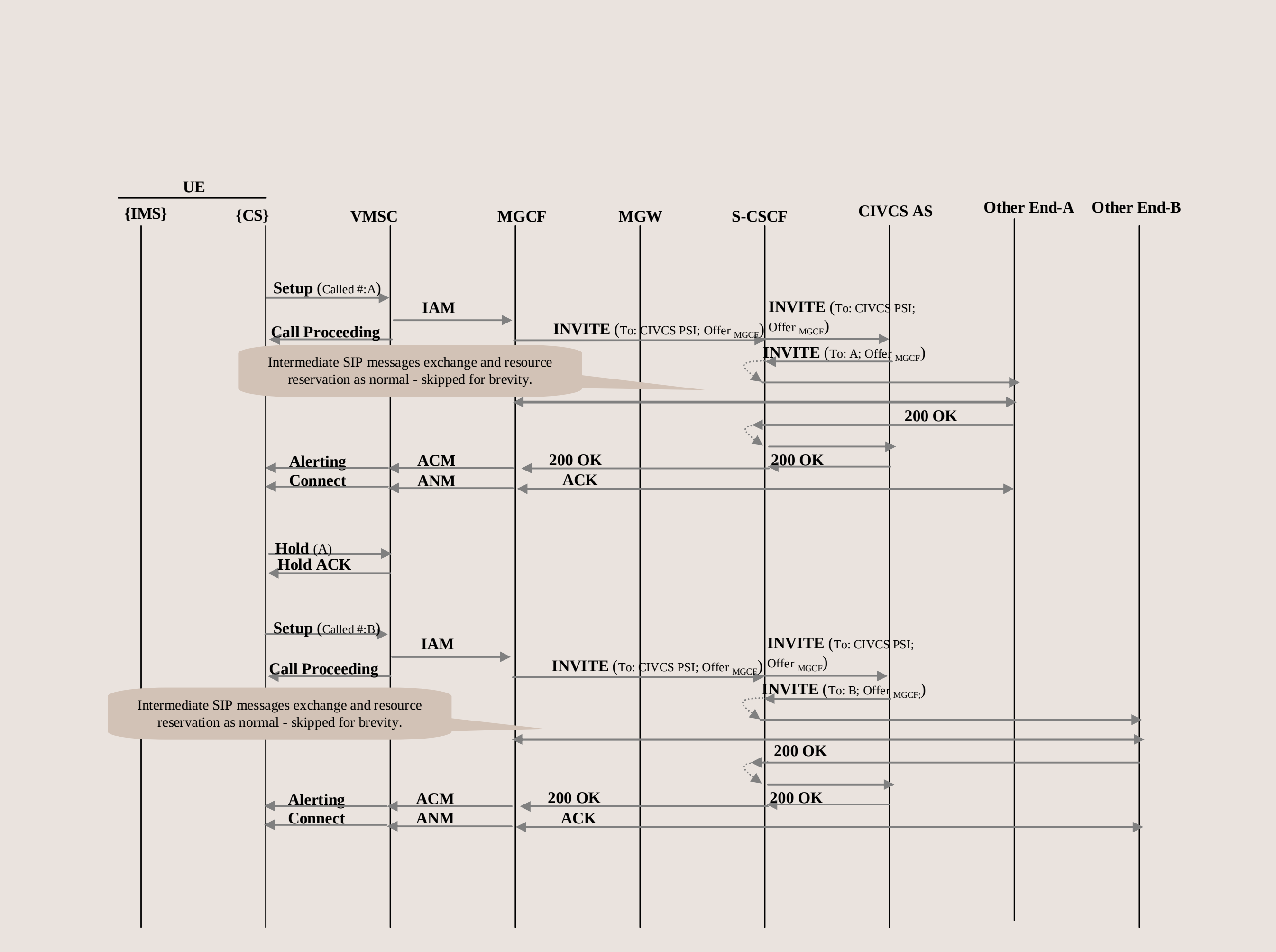


Figure 7: Anchoring of CS Held and Active sessions at CIVCS

- Since the user is not registered in IMS, exchange of session identifier is not possible with the UE. However, CIVCS 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.

6.3.7.1.1.6 CS to IMS Handover of CS Held and Active sessions; IMS not active at the time of CS Anchoring

Figure 8 below provides a walkthrough of Handover of CS Held and Active sessions established as described in previous section.

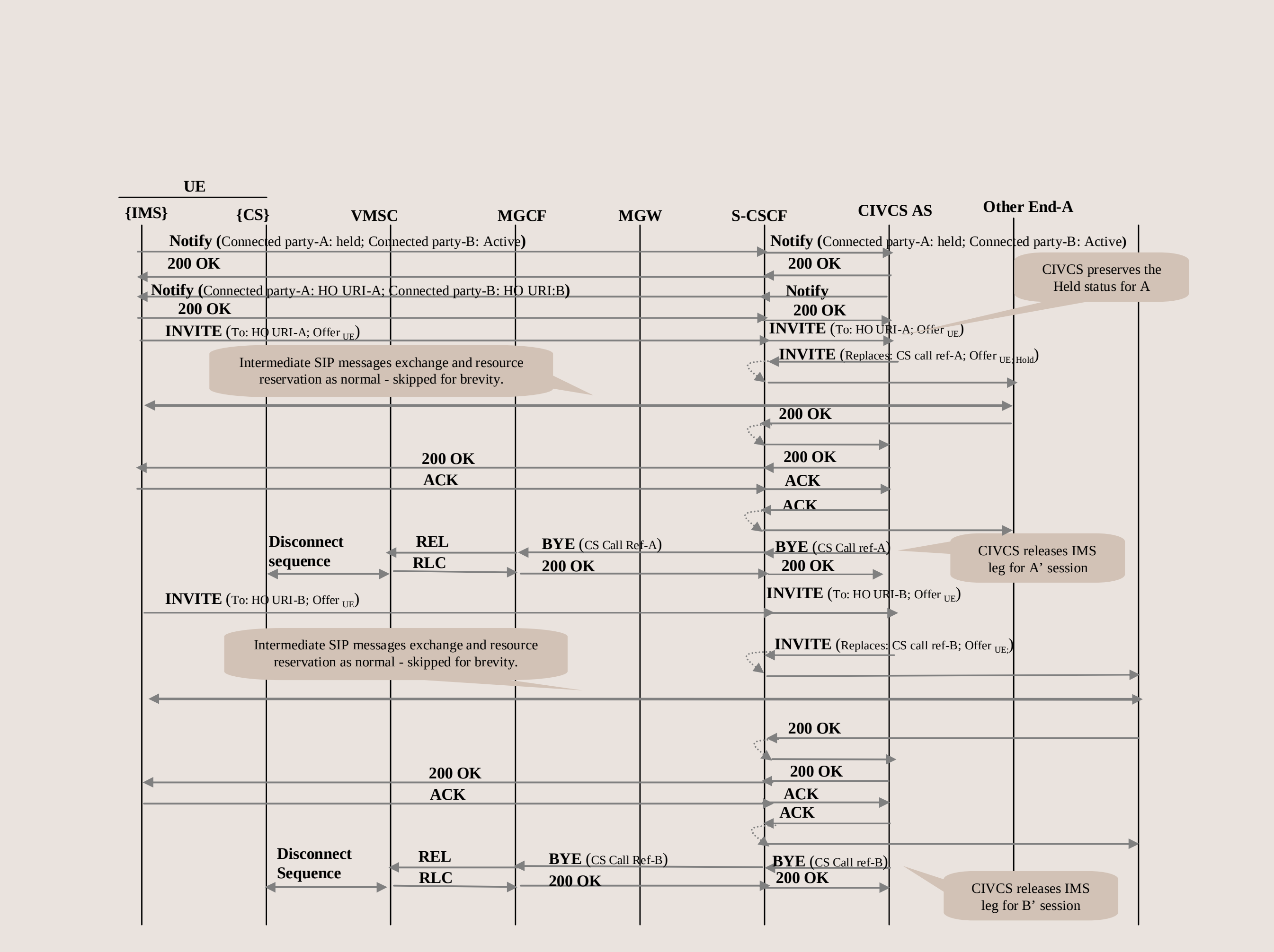


Figure 8: CS to IMS Handover of Held and Active sessions; IMS not active at CS anchoring

- Upon detection of border conditions, the UE performs IMS Registration and updates CIVCS with the current session state information to be used during the Handover procedure. Session Hold, Active states are passed to CIVCS 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 CIVCS.
- CIVCS communicates SIP URIs to be used for Handover 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 Handover of Held/Active sessions with IMS Registration.

6.3.7.1.2 Multi-Session services support with DACCI

6.3.7.1.2.1 General

DACCI uses techniques similar to the techniques described in a companion paper (ref [4]) to transfer a CS call with multiple sessions from CS domain to IMS.

6.3.7.1.2.2 Handover of Multi-Session services

Figure 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.

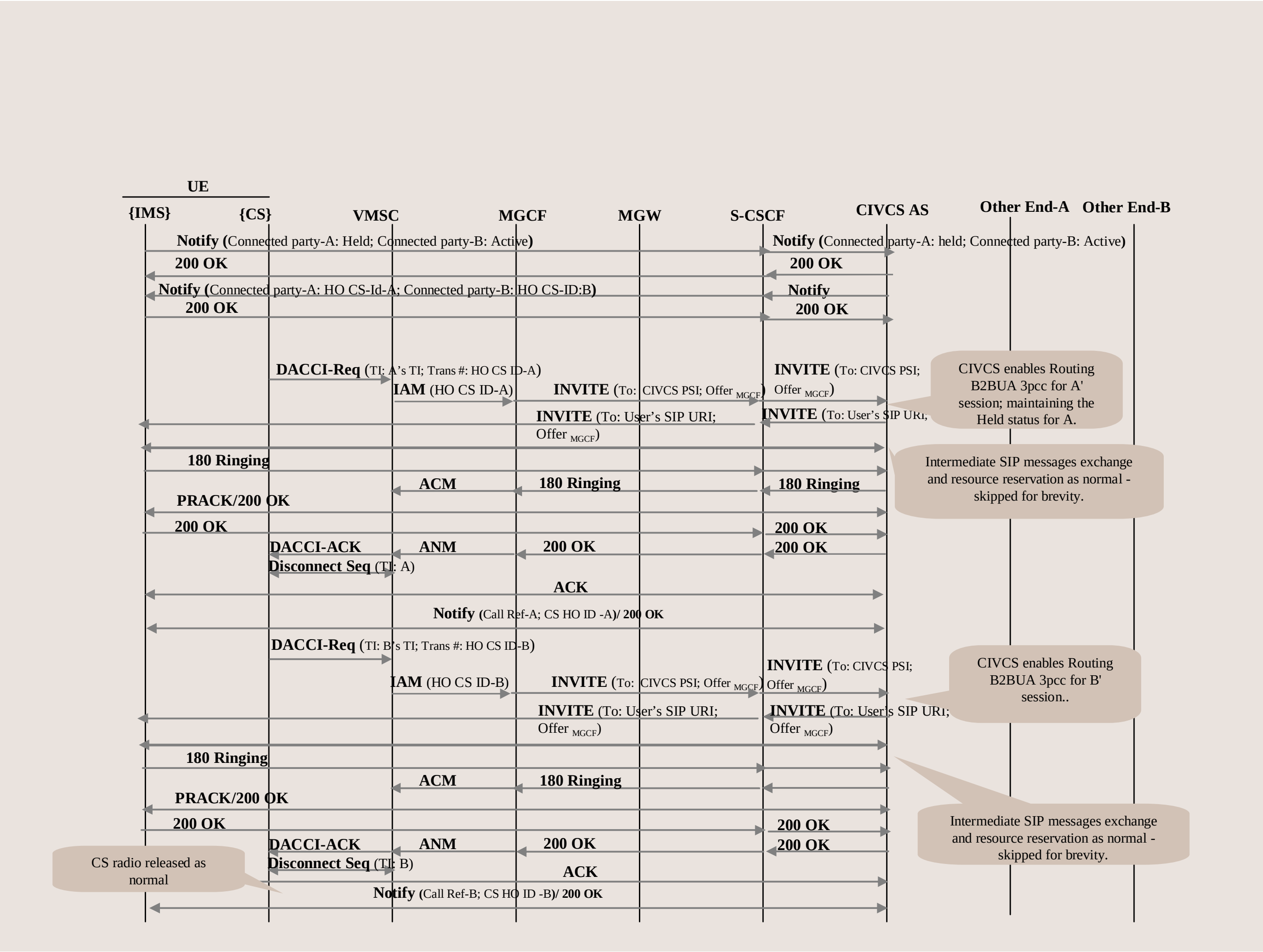


Figure 3: Handover of Held/Active CS Sessions via DACCI

- The UE detects a need for DACCI enabled Handover, registers with IMS, and provides session state information to CIVCS.

- CIVCS allocates CS HO Identifiers for A's and B's session and communicates to the UE.

- UE executes Handover 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.

- The MSC allocates a new circuit connection for A's session, extending it's bearer toward CIVCS as described in previous section.

- CIVCS 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. CIVCS establishes A's session in the Held state as suggested by session state information received from the UE.

- Upon completion of Handover of A's session, UE executes Handover of B's session.

- The MSC allocates a new circuit connection for B's session, extending it's bearer toward CIVCS as described in previous section.

- CIVCS 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.

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. The UE establishes all the sessions in the handing-in domain, followed by establishment of conference bridge in the handing-in domain, subsequently followed by release of the conference bridge and associated legs in the handing-out domain.

The bearer path interruption for the transfer of MPTY service could be larger than the bearer path interruption caused by transferring Held/Active sessions. Therefore, it's recommended that the UE informs the user via the MMI procedures that Handover of MPTY service is in progress.

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

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 handover 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 handover 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 Handover procedure will handover the between the CS and IMS domains. The Services AS may not need to be informed of the handover.

6.3.7.4 Supplementary Service Impact summary

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

Supplementary Service	Impact statement
Home Network Call Forwarding (CFU, CFNRc-HLR detached)	CIVCS 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 CIVCS is unaware of user's availability.
Incoming Call Barring	No impact.
Outgoing Call Barring	Handover cannot be executed if certain flavours of outgoing call barring are enabled for the users.
Calling Line ID Presentation (CLIP)	CIVCS ensures delivery of the originating party's CLIP information to the CS-IMS user when CS incoming calls are anchored via CIVCS. CLIP presentation for target leg to be blocked by CIVCS with appropriate use of screening indicators. Interactions with CLIP Override are for further study.
Connected Line Identity Presentation (COLP)	CIVCS ensures delivery of the actual connected party's COLP information to the CS-IMS user when CS originating calls are anchored via CIVCS. COLP presentation to the user is required to be blocked for the target leg, preferably at UE.
Closed User Group	CIVCS PSI is required to be included in subscriber's Closed User Group profile.
Call Hold/Retrieve	No impact.

Call Wait	No impact.
Multi-party	Transfer of Multi-party service is possible as long as the handing-in domain supports it. 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.
Optimal Routing	Optimal routing for basic mobile to mobile calls to be disabled for CS-IMS users.

Table 1: Supplementary Service Impact of CIVCS

6.3.7.4.1 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. 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

Handover 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 CIVCS 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, handover 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, handover 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 CIVCS PSI must be an allowed Fixed Destination.

6.3.8 Evaluation of the model

6.3.8.1 General

This clause presents the evaluation of the service continuity solution against the set of criteria

The benefits of this proposal include:

- The solution provides an access agnostic approach with cohesive techniques applied for Handovers in both directions, CS to IMS and IMS to CS.
- The subsequent Handovers and Handbacks are handled in an efficient manner. Such Handovers do not result in daisy chain effect that results from use of disjoint techniques applied in CS to IMS and IMS to CS directions.
- The CS call/IMS session is anchored at CIVCS for the life of the CS call/IMS session that enables comprehensive billing with complete Handover history.
- The solution does not impact the CS or PS Domain Core Network, or the Access Network for mobility between GERAN/UTRAN CS and I-WLAN.
- The solution is based on a service in user's home IMS network; therefore, CIVCS service delivery to the user is not impacted when roaming in non supporting networks.
- The solution applies to all roaming situations and has no restrictions on the location of the WLAN or CS MSC.

The drawbacks of this proposal include:

- None identified so far.

Editor's Note: The use of CIVCS for providing CS-IMS voice continuity when the IMS is accessed over UTRAN is for further study.

Alignment with the Reference Architecture to ensure correct terminology is for further study.

6.3.8.2 Techniques for enabling static anchoring for CS calls and IMS sessions at CIVCS

The benefits of this proposal include:

- Provides uniform solution for anchoring CS and IMS calls/sessions at CIVCS which is not restricted by any service interactions as some of the dynamic anchoring techniques.
- CIVCS is in control of the bearer path since the initial call setup, therefore, Handover execution 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 effect Handover execution when dynamic anchoring techniques are applied.
- Static anchoring expedites Handover execution time as the anchor is in place at the time of Handover and additional steps are not required to establish the anchor.

The drawbacks of this proposal include:

All CS originated calls made by CS-IMS users are routed via the user's home IMS network which results in some inefficiency in call setup delays and network resource usage when setting up calls for subscribers of CS-IMS Voice Continuity Service. However, it should be noted that since a MGW is used to anchor the CS bearer, the bearer path backhaul can be minimized by strategically locating such MGWs close to legacy networks.

6.3.8.3 Dynamic CS Anchoring for CIVCS using ECT

The benefits of this proposal include:

- This approach enables dynamic anchoring with CIVCS that alleviates concerns around resource usage with static anchoring.

The drawbacks of this proposal include:

- ECT cannot be used to enable Handover from CS to IMS if multi-session services like CW are active at the time of Handover.

- ECT cannot be used to dynamically anchor CS Emergency calls with CIVCS as ECT Supplementary Service is not applicable for Emergency calls.

6.3.8.4 Dynamic CS Anchoring for CIVCS using DACCI

The benefits of this proposal include:

- This approach enables dynamic anchoring with CIVCS 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.

6.3.8.5 Supplementary Services Support

The benefits of this proposal include:

- Approach consistent with the basic principle used for CIVCS that releases the call control PSM in handing-out domain and establishes a new PSM in handing-in domain.

- The transfer is seamless to the user as the user can continue to control the supplementary services after the Handover.

- Since the information required to complete inter domain transition is exchanged with CIVCS via the Notify with Mobility Event package at the beginning of the execution of the inter domain transition procedure, and since a completely new protocol state machine is established in the new domain, loss of radio/I-WLAN coverage in the old domain any time after the exchange of initial information exchange does not affect completion of the inter domain transition procedure.

- Converged services that are supported by both the CS domain and the IMS domain are supported during the handover 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.

The drawbacks of this proposal include:

- Exchange of Subscribe/Notify for Mobility Event package adds to signalling exchange required for IMS Registration which is exacerbated when combined with subscription of other events that may be required in addition to this package. It is recommended to consider signalling optimizations so that multiple events could be subscribed to during IMS Registration without significantly impacting the time required to complete IMS Registration.

- Handover 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 Handover procedure continues successfully even if coverage is lost in Handing-out domain during the Handover execution.

6.4 Service Continuity Model: Anchored Call Control Model

6.4.1 General Description

6.4.1.1 Architecture Model

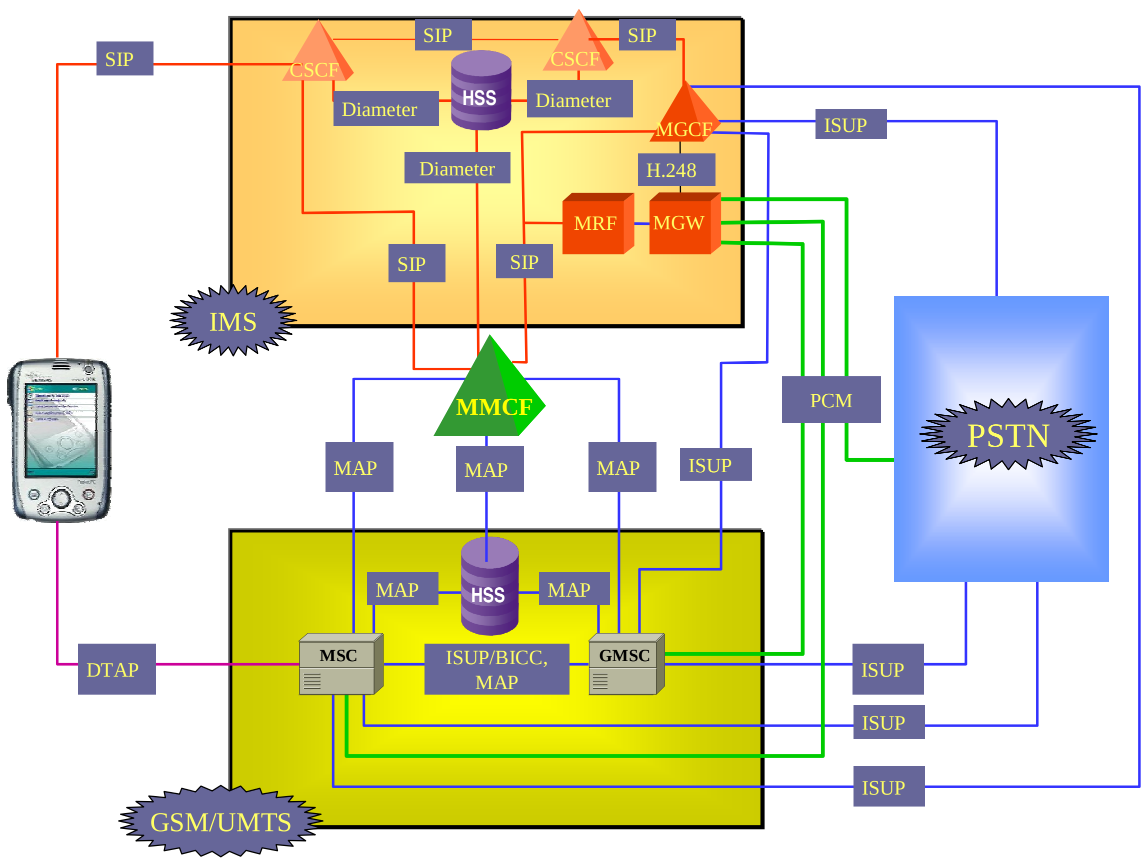


Figure 6.4.1.1-1 Network Architecture Diagram

The anchored call control model follows the existing GSM inter-MSC handover model as defined in TS23.009 Handover Procedures. To allow the HLR and MSC interworking, a Mobility Management Control Function (MMCF) is introduced into the architecture. The MMCF connects to the GSM CS domain via the MAP interface and to the IMS domain via SIP and Diameter.

Editor's note: Need to align the terminology of MMCF with contribution S2-051193.

The role of the Mobility Management Control Function (MMCF) is to provide a handoff control function between the IMS and CS domains. At the time of handoff from IMS to GSM, the MMCF appears as the originating MSC to the GSM MSC and it appears as the session transfer point for the IMS sessions that are being handed off to the GSM domain. When there are multiple voice sessions active in the dual mode handset (e.g. call waiting), the MMCF introduces the MRF to provide media stream control and/or mixing. At the time of handoff from GSM to IMS, the MMCF appears as the target MSC to the originating MSC and it appears as the link between the MGCF and the dual mode handset in the IMS domain.

In addition, it provides location update information when the dual mode handset "roams" between the two domains. When the E.164 number is owned by the CS domain, the MMCF acts as a VLR for the GSM phone when it roams into IMS coverage, thus allowing a telephony call to the E.164 number to route to the IMS location. When the E.164 number is owned by the IMS domain, the MMCF acts as an HLR for the IMS phone when it roams into GSM coverage, thus allowing a telephony call to the E.164 number to route to the GSM location.

6.4.1.2 Mobility Management Control Function

The Mobility Management Control Function (MMCF) Function manages the overall handoff process. It interfaces to the S-CSCF via SIP and has a role similar to an IMS Application Server. It interfaces to the MGCF and MRF acting as a CSCF. It interfaces to the 2G Network via the MAP protocol and acts as a VLR and HLR.

Editor's Note: the role of MMCF acts as VLR and HLR needs further elaboration.

The MMCF sub functions are:

MMCF Registrar Function:

- receives registration requests from the S-CSCF when the UE registers in the IMS domain

MMCF HO Procedure Function:

- SIP Event driven mobile interface
- may subscribe to a UE for event notifications or publish information to a UE after registration
- Target system determination function

MMCF HO Function:

- Pool of temporary E.164 numbers and tel URIs

MMCF HO Connection Function:

- Inserted in the call when the HO begins and juggles the legs of the call to switch the bearer path from source to target
- Remains in the call after the HO, acting as a B2BUA
- Call parking function
- Call rendezvous function
- Call transfer function
- SIP interface to BGCF and MGCF

MMCF MSC/VLR:

- Used to setup handover legs with MAP procedures
- Does update location / cancel location when the UE does a SIP register/deregister
- Assigns MSRN when HLR does a request
- Responds to MGCF request (via HO Connection Function) to resolve MSRN on receipt of incoming PSTN calls

- MAP interface to GSM MSC

Editor Note: The requirement of the location update and cancel location need further elaboration.

Editor Note: How to discover the 3G Cell ID or RNC ID information for IMS to CS handover need to be investigated.

Editor Note: Further elaboration on the SIP message encapsulation for IMS to CS handover.

6.4.2 Routing Selection Decision

6.4.3 Registration

There are 4 different registration scenarios to consider based on where the user's telephone number is provisioned. The provision of the user telephone number is based on one user subscription.

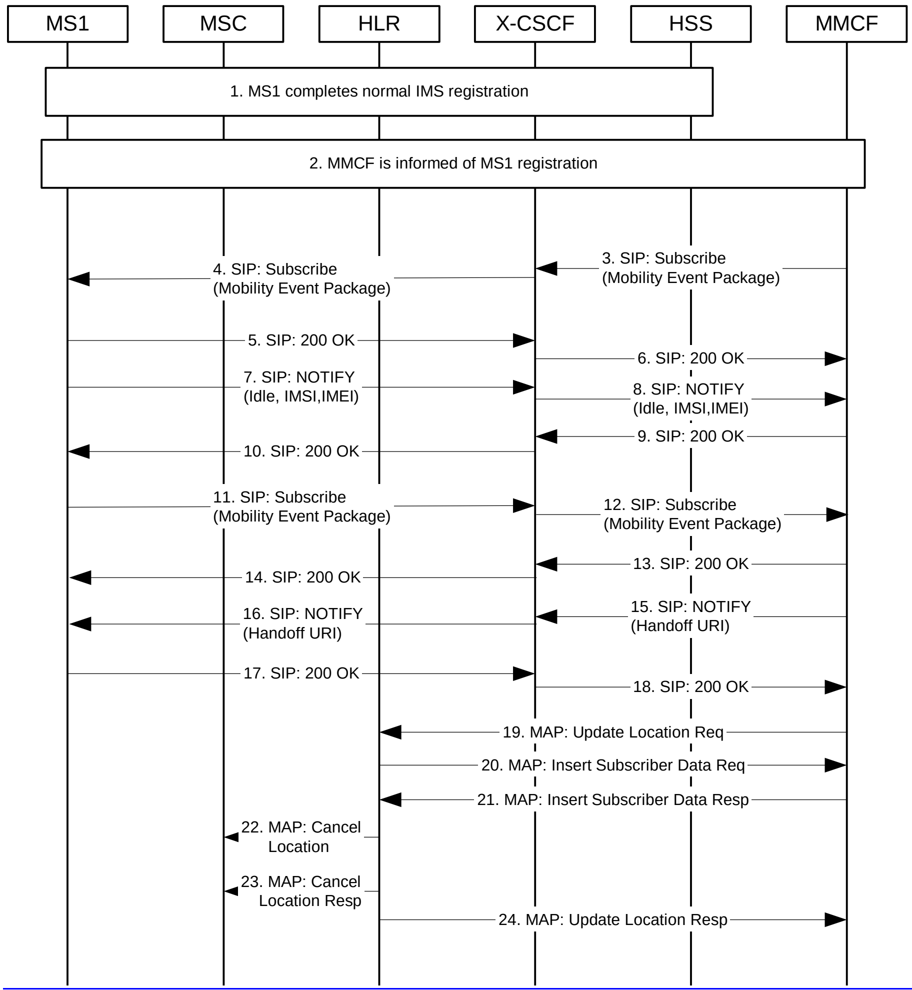
1. The telephone number is provisioned in the GSM HLR and the user is registering in the GSM domain. This is a normal GSM registration. The only change from the normal registration scenario is that the MMCF may have been acting as a VLR (see registration case 2) and thus the GSM registration will cause the HLR to send a cancel location to the MMCF to remove the user from the VLR.

2. The telephone number is provisioned in the GSM HLR and the user is registering in the IMS domain for voice calls(i.e., it wants to do voice via the IMS domain and not the CS domain). This is a normal IMS registration (flow line 1) with the following additional steps:

    a. The MMCF is informed of MS1 registration. This could be via proxy registration, MS1 publishing its registration status to the MMCF or some similar mechanism (flow line 2).

    b. The MMCF and dual mode handset subscribe to each other's mobility event package. As part of the subscription, the MMCF sends a handoff URI to the dual mode handset that will be used during the IMS to CS handover procedure (flow lines 3 – 18).

    c. If the dual mode handset's mobility event package indicates that the user is not in a call (i.e., this is an idle state registration), the MMCF creates a VLR and does an update location procedure with the GSM HLR (flow lines 19 – 24).



3. The telephone number is provisioned in the IMS HSS and the user is registering in the IMS domain for voice calls(i.e., it wants to do voice via the IMS domain and not the CS domain). This is a normal IMS registration with the following additional steps:

   a. The dual mode handset includes a trigger function that causes the S-CSCF to do a proxy registration with the MMCF.

   b. The MMCF and dual mode handset subscribe to each other's mobility event package. As part of the subscription, the MMCF sends a handoff URI to the dual mode handset that will be used during the IMS to CS handover procedure.

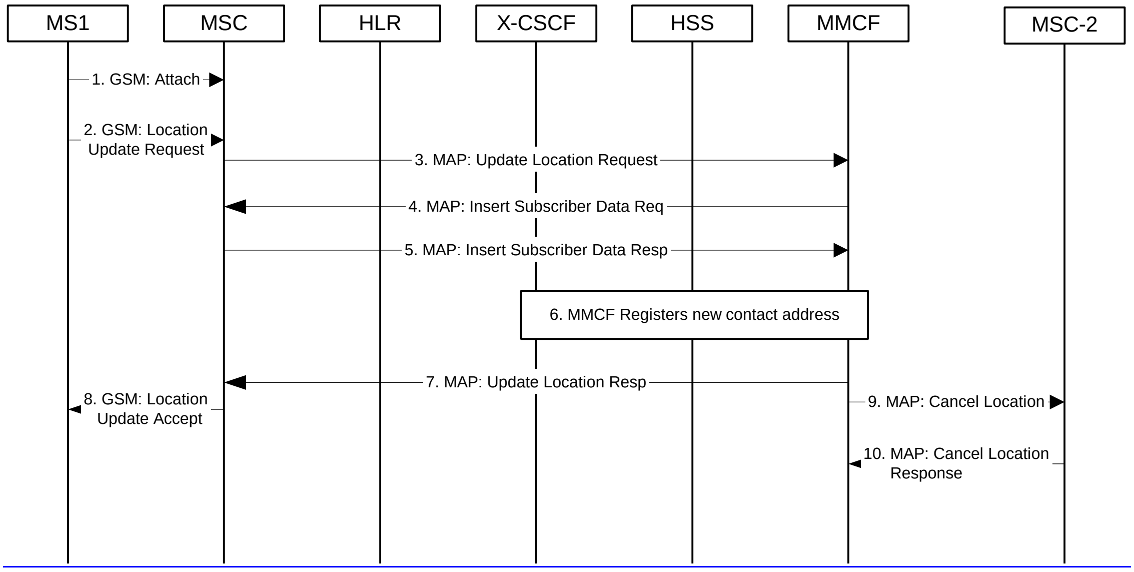
   c. If the dual mode handset's mobility event package indicates that the user is not in a call (i.e., this is an idle state registration), the MMCF will send a cancel location to the previous MSC if applicable.

4. The telephone number is provisioned in the IMS HSS and the user is registering in the GSM domain. This is a normal roaming registration in the GSM domain, with the following differences described below and shown in the attached flow diagram:

   a. When the visited MSC creates the VLR, it communicates with the MMCF, which appears like an HLR to the MSC (flow lines 3 – 5, 7).

   b. The MMCF will use the HSS to validate the GSM credentials and then will create a registration in the HSS that causes the S-CSCF to treat the MMCF as one of the registered contact addresses for the public userid of the dual mode handset. The contact address provided by the MMCF should be sufficient for the MMCF to be able to do a provide roaming number request when requested by the S-CSCF (flow line 6).

c. The MMCF will send a cancel location to a previous MSC if applicable (flow lines 9 – 10).



Editor's Note: for the above figure step 3, consider to terminate at HSS instead of MMCF as shown.

Editor's Note: How to address VLR neighboring configuration by MMCF.

Editor's Note: Need to elaborate the inter VLR communication by MMCF.

Editor's Note: whether study is required into e.g. terminating SMS the behavior, when the subscriber has CAMEL trigger. Consider new location update procedure?

6.4.4 Origination

6.4.4.1 IMS origination

When a user who is registered in IMS initiates a voice session via a SIP INVITE to another telephone number or IMS public user id, normal IMS domain call processing occurs. There is no special processing because this is a dual mode handset.

6.4.4.2 GSM/UMTS CS origination

When a user who is registered in GSM/UMTS initiates a voice session to another telephone number normal CS domain call processing occurs. There is no special processing because this is a dual mode handset.

6.4.5 Termination

6.4.5.1 IMS termination

The anchored call control model does not assume that the GSM operator has an IMS domain or that the IMS operator has a GSM domain. Thus there is no assumption that there is a shared HSS/HLR between the 2 domains.

There are two cases depending on where the dual mode handset is registered:

1. dual mode handset is registered in IMS domain; IMS owned telephone number

    a. Prior to the call, the mobile powers up in the WLAN or moves back into the WLAN while idle, then registers with IMS. This registration includes the subscription for the Mobility Event Package. The MMCF acts as an HLR and cancels any prior GSM registration (Cancel Location).

    b. Normal IMS call termination procedures are followed for calls from the PSTN to the dual mode handset.

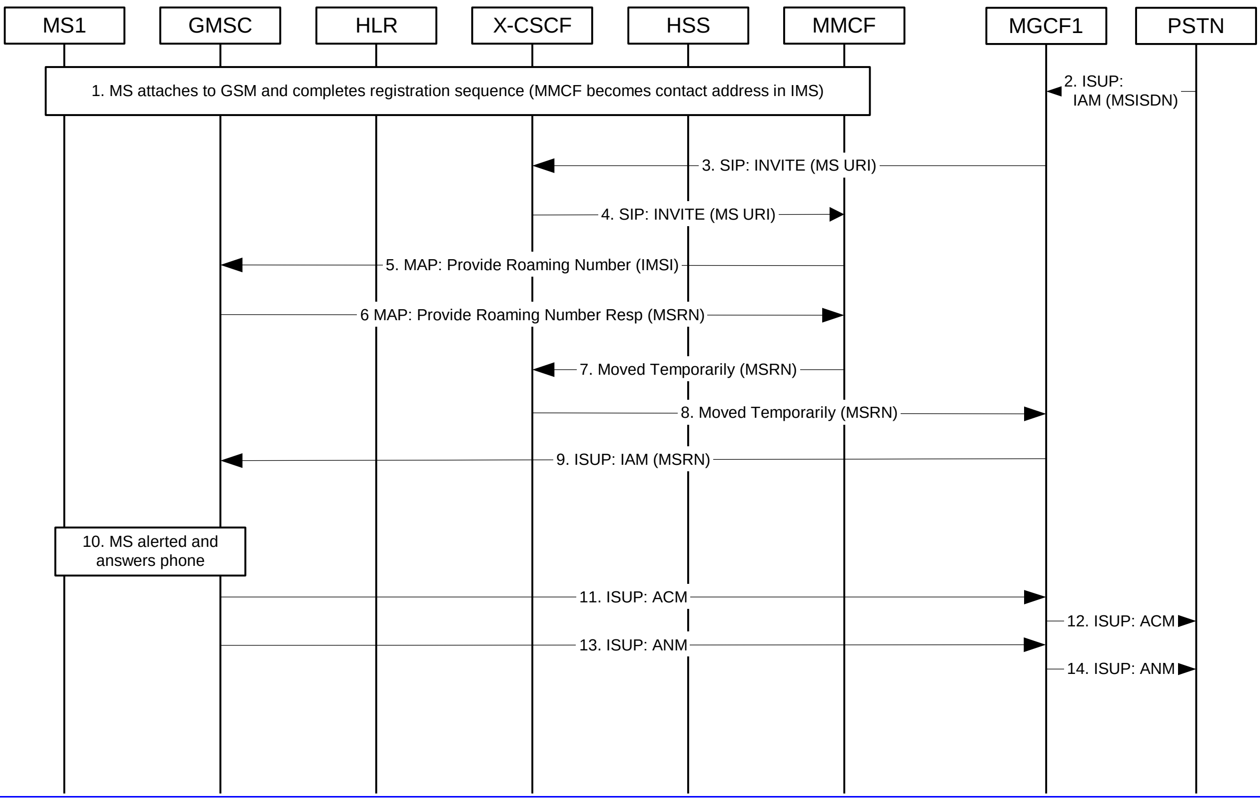
2. Dual mode handset is registered in GSM domain; IMS owned telephone number

   a. Prior to the call, the mobile powers up in the GSM domain or moves back into the GSM domain while idle, then it accesses the control channels and registers in GSM. Since the MMCF is acting as the HLR for this telephone number, the MSC sends an Update Location Request to it and the MMCF registers itself as the contact address for the dual mode handset (flow line 1).

   b. Normal call termination procedures are now followed for calls from the PSTN to the dual mode handset (flow lines 2 – 3)

   c. The S-CSCF determines that the MMCF is the current contact address for dual mode handset URI and sends the INVITE to the MMCF, who then gets the MSRN and redirects the INVITE to the MSRN (flow lines 4 – 8).

   d. The MGCF sets up a call to the GMSC (flow lines 9 - 14).



6.4.5.2 GSM/UMTS CS termination

The anchored call control model does not assume that the GSM operator has an IMS domain or that the IMS operator has a GSM domain. Thus there is no assumption that there is a shared HSS/HLR between the 2 domains.

There are two cases depending on where the dual mode handset is registered:

1. Dual mode handset is registered in GSM domain; GSM owned telephone number

   a. Prior to the call, the mobile powers up in the GSM domain or moves back into the GSM domain while idle, then it accesses the control channels and registers. If the dual mode handset had previously been registered in the IMS domain, the HLR will send a Cancel location to MMCF. The MMCF will send a deregistration to the S-CSCF to cancel the IMS registration for that contact.

   b. A normal mobile termination from the PSTN happens. This is the normal GSM terminating call flow. There are no special steps because this is a dual mode dual mode handset.

2. Dual mode handset is registered in IMS domain; GSM owned telephone number

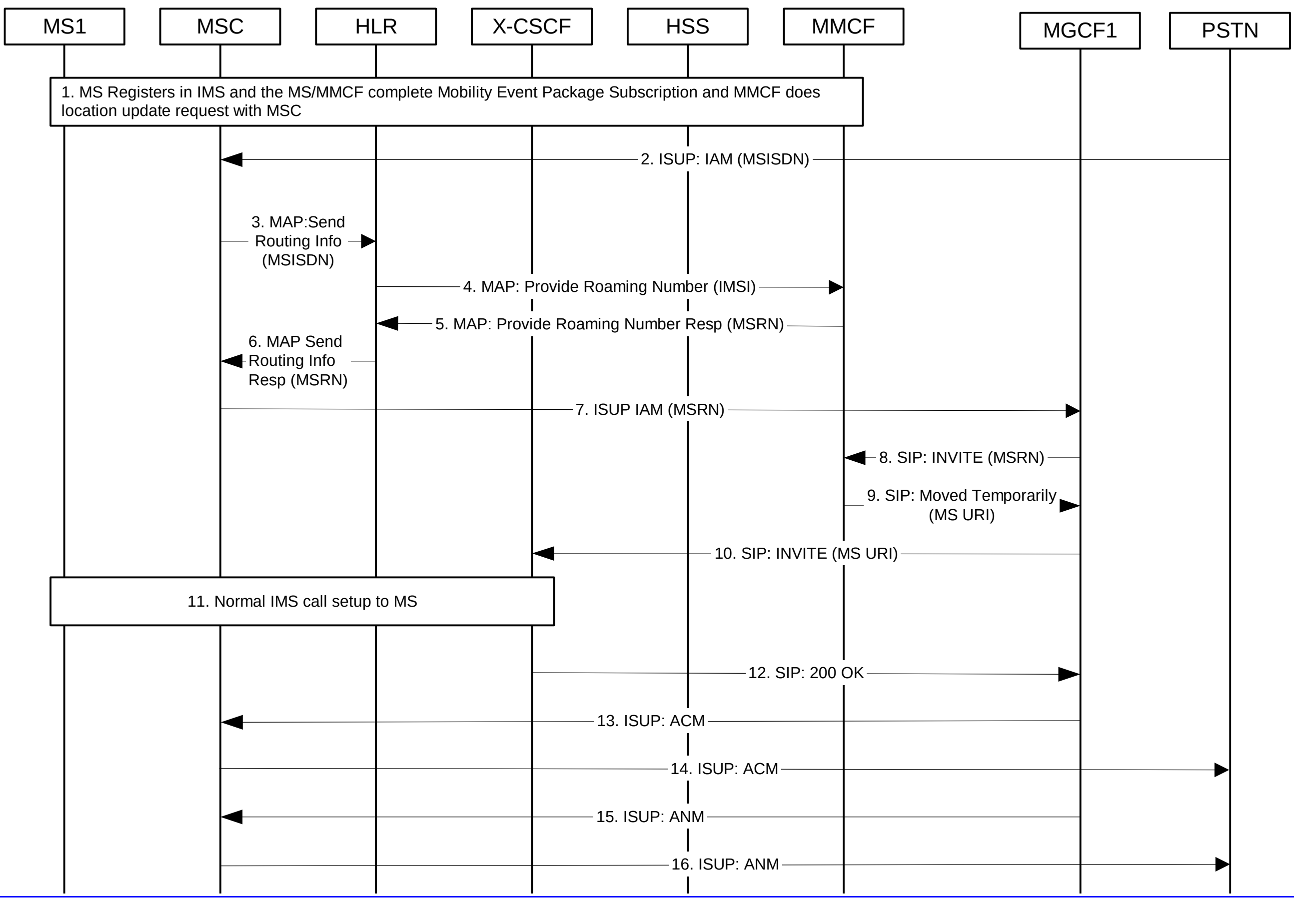
a. Prior to the call, the mobile powers up in the WLAN or roams into the WLAN while idle, then registers with IMS. This registration includes the subscription for the Mobility Event Package. Because the mobile is Idle, a VLR record is established by the MMCF and it sends a location update to the HLR. This procedure may start immediately after the initial NOTIFY from the mobile. The HLR will inform the last known serving MSC to delete the mobile's VLR record (flow line 1).

b. A normal mobile termination from the PSTN happens. The GMSC connects to the HLR to get routing information and the HLR requests a roaming number from the MMCF (flow lines 2 – 4).

c. The MMCF issues a temporary dual mode handset roaming number. This number will cause a call to be routed to an MGCF (flow lines 5 – 7).

d. The MGCF does an ENUM query or some other means to determine that the roaming number belongs to MMCF and then forwards the call to MMCF via an INVITE. MMCF converts the roaming number to the dual mode handset URI and sends a moved temporarily response back to the MGCF. The MGCF routes to the correct I-SCSCF (flow lines 8 – 10).

e. Normal call termination procedures are now followed for calls from the PSTN to the dual mode handset (flow lines 11 – 16).



Editor note: investigate how to satisfy operator policy for domain selection.

Editor note: identify the difference between this solution and Siemens' anchor solution.

Editor note: Elaborate the role of MMCF acting as HLR function.

Editor note: replace GSM with CS domain.

6.4.6 Handover Scenarios

6.4.6.1 CS UE to CS UE call

This Use Case illustrates the architecture used for handing off a dual mode dual mode handset on a circuit voice call on a GSM/UMTS system to a VoIP call on a WLAN/IMS system.

In Figure 6.4.6.1-1, the dual mode handset is operating in the GSM domain and has two active GSM calls. One session is to a PSTN telephone (which could be a wireline or wireless phone) . The other session is to a GSM dual mode handset connected to the same MSC. Since these are GSM calls, the MSC manages both call legs for services such as call waiting or 3-way calling. Any mixing of the media streams is managed internal to the MSC.

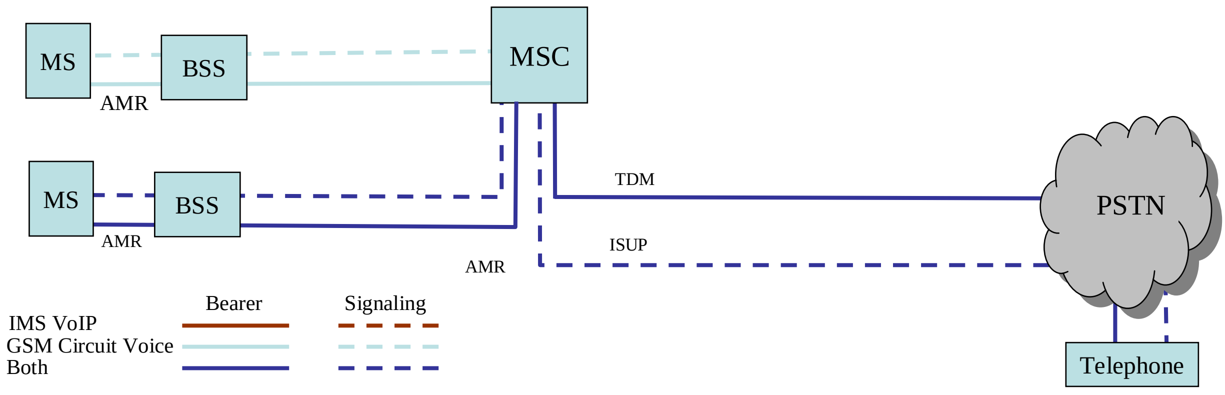


Figure 6.4.6.1-1: Initial State GSM - dual mode handset has 1 GSM call and 1 PSTN Call active (either 3-way or call waiting)

Figure 6.4.6.1-2 shows the state of the calls after the dual mode handset had handed over to the IMS domain. Since the GSM MSC remains in the call as the anchor MSC, only 1 call leg is handed over. The GSM MSC, as anchor MSC still manages the call and the various call legs. The dual mode handset sends and receives call state change to the MMCF as DTAP messages embedded in SIP NOTIFY messages (SIP INFO messages could be used as an alternative). The MMCF acts as a pass thru point to the GSM MSC for these DTAP messages, as is the normal procedure with GSM handovers.

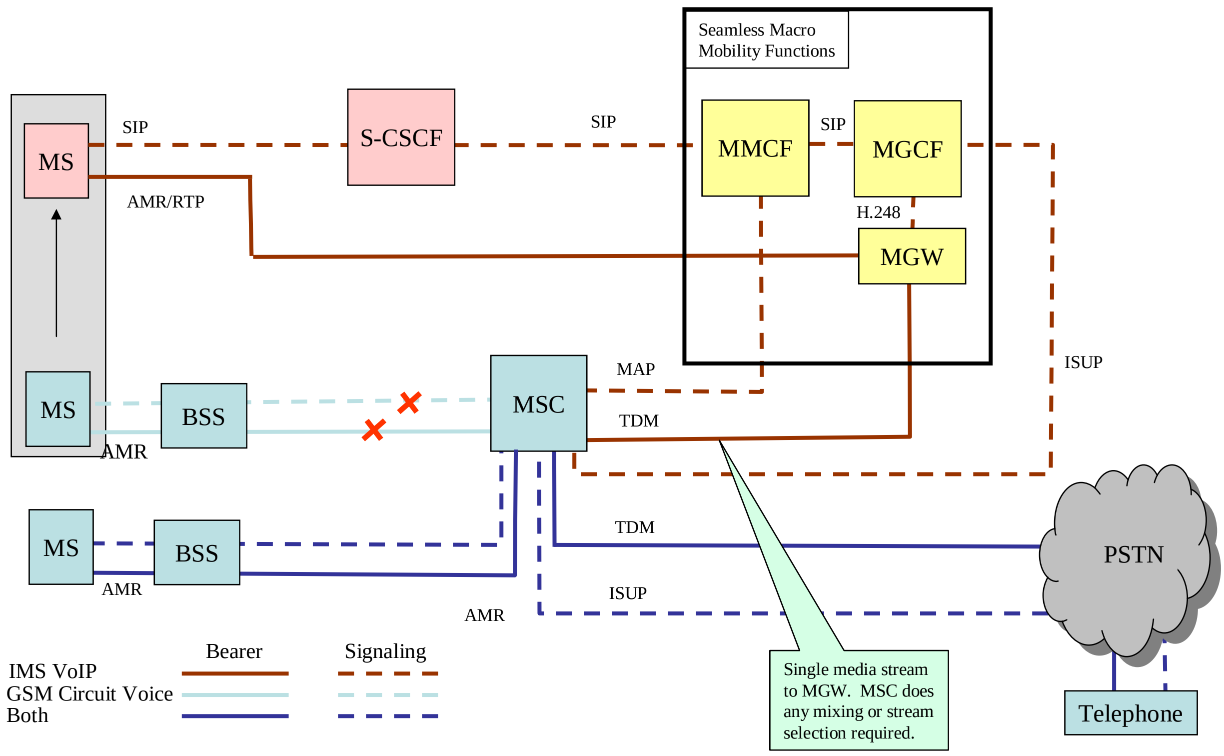


Figure 6.4.6.1-2: Handover - GSM Circuit Voice to IMS VoIP

The steps in the procedure for handoff of a GSM circuit voice call to a WLAN VoIP to are as follows.

The dual mode handset is in a call with two parties, one is another GSM MS served by the MSC and the other is a PSTN party which could be an IMS based user. This could be a 3-way call or it could be a call waiting call.

1. The dual mode WLAN/GSM mobile, upon entry into a WLAN coverage area, must register with the IMS (IP Multimedia Subsystem) network and a Macro-Mobility Control Function (MMCF). The S-CSCF does a proxy register with the MMCF. The MMCF will use a SIP SUBSCRIBE/NOTIFY method to subscribe to a newly defined Mobility Event Package with the mobile, and the mobile will likewise subscribe to the same event package with the MMCF. The dual mode handset notifies the MMCF that it is in a GSM call and provides its IMSI. In this case, the MMCF sends a null notification to the dual mode handset (flow line 1).

2. On the GSM circuit voice network, the mobile initiates the GSM handover procedure via the MSC and the mobile is instructed to tune its WLAN radio to the WLAN-IMS (flow lines 2 – 8).

   - dual mode handset now initiates a normal GSM handoff, with the target MSC pointing to the MMCF. Even though there are multiple connections, only 1 connection is setup from the MSC to the IMS network. The MSC anchors the call and will continue to manage both the call leg to MS2 and the PSTN.
   - The MMCF provides a handover number to the MSC and the MSC initiates a call to the handover number.
   - When the MGCF receives the call setup message, it uses ENUM or some other means to translate the handover number to a handover number URI that routes to MMCF. It then initiates a call to MMCF.

- On the GSM side, the dual mode handset receives a handover command with the reference number and the fake traffic channel info.

3. On the WLAN-IMS network, the mobile uses the SIP NOTIFY to inform the MMCF that it is handing off and includes the reference number that it received (flow lines 9 – 12).

4. The MMCF then establishes a SIP session with the mobile and the mobile now switches to its WLAN radio (flow lines 13 – 17).

   - MMCF sends an invite to dual mode handset with MGW1 as the endpoint. It is able to do this since the reference number allows it to correlate the dual mode handset with the incoming call to the handover number URI from MGCF1.

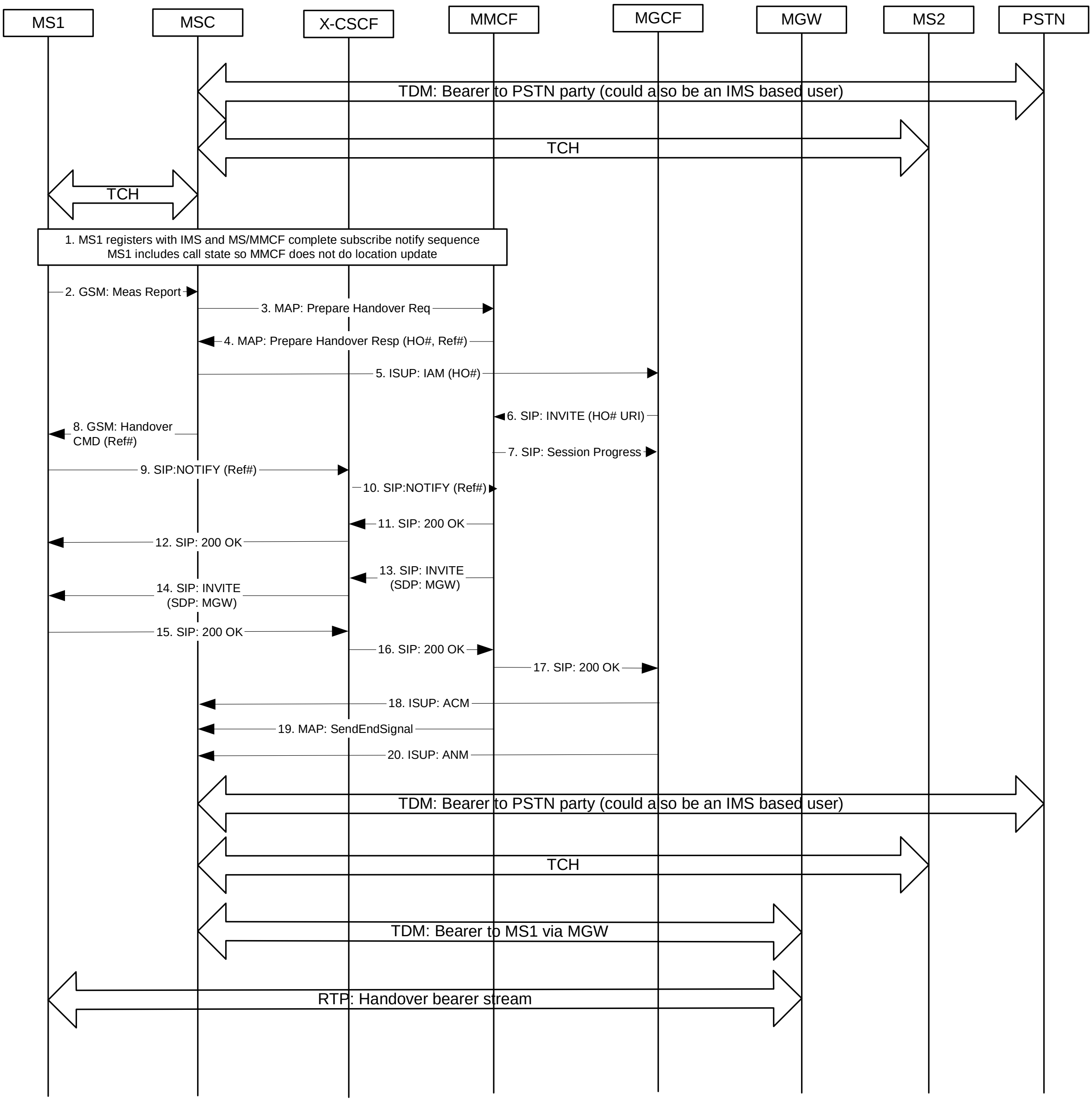
   - After dual mode handset accepts the call, MMCF can respond to the call from the MGCF.

   - The MGCF uses the dual mode handset endpoint information and connects the MGW to the dual mode handset.

5. The GSM network is informed the handoff is complete and the GSM core network and radio access network resources are released (flow lines 18 – 20).

   - The MGCF provides an answer message to the MSC.

   - The MSC now clears the connection to the dual mode handset.



6.4.6.2 CS UE to IMS UE call

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

Editor note: Elaborate on the simultaneous registration for CS and PS domain,

6.4.6.3 IMS UE to IMS UE call

This Use Case illustrates the architecture used for handing off a dual mode dual mode handset on a VoIP call from a WLAN/IMS system to a circuit voice call on a GSM/UMTS MSC.

In Figure 6.4.6.3-1, the dual mode handset is operating in the WLAN domain and has two active IMS call sessions. One session is to a PSTN telephone (which could be a wireline or wireless phone) . The other session is to an IMS dual mode handset. Since these are IMS sessions, the dual mode handset manages both sessions for services such as call waiting or 3-way calling. Any mixing of the media streams is managed internal to the dual mode handset.

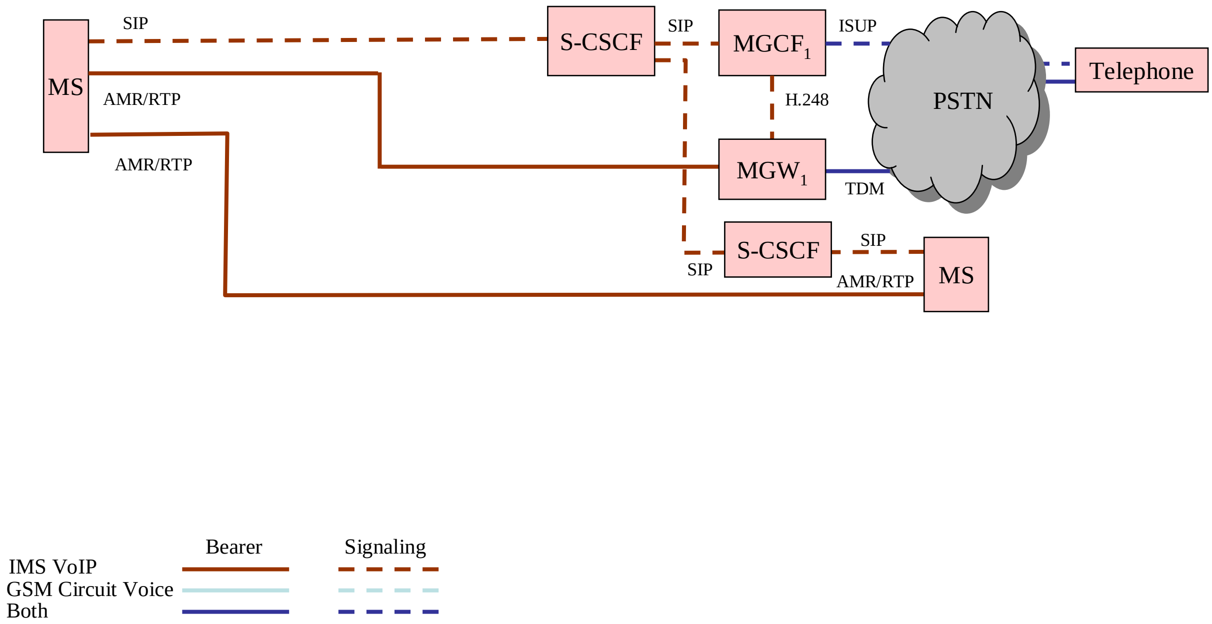


Figure 6.4.6.3-1: Initial State IMS - dual mode handset has 1 IMS session and 1 PSTN Call Active (either 3-way or call waiting)

Figure 6.4.6.3-2 shows the state of the sessions after the dual mode handset has handed over to the GSM domain. Since the GSM domain only supports 1 media stream to the dual mode handset, the MMCF has introduced an MRF in the IMS domain to terminate the IMS sessions that have been transferred to the MMCF. The MMCF appears as the anchor MSC to the GSM MSC after handover and still manages the call and the various call legs. The dual mode handset sends and receives session state change messages to and from the MMCF via DTAP messages, as is the normal procedure with GSM handovers.

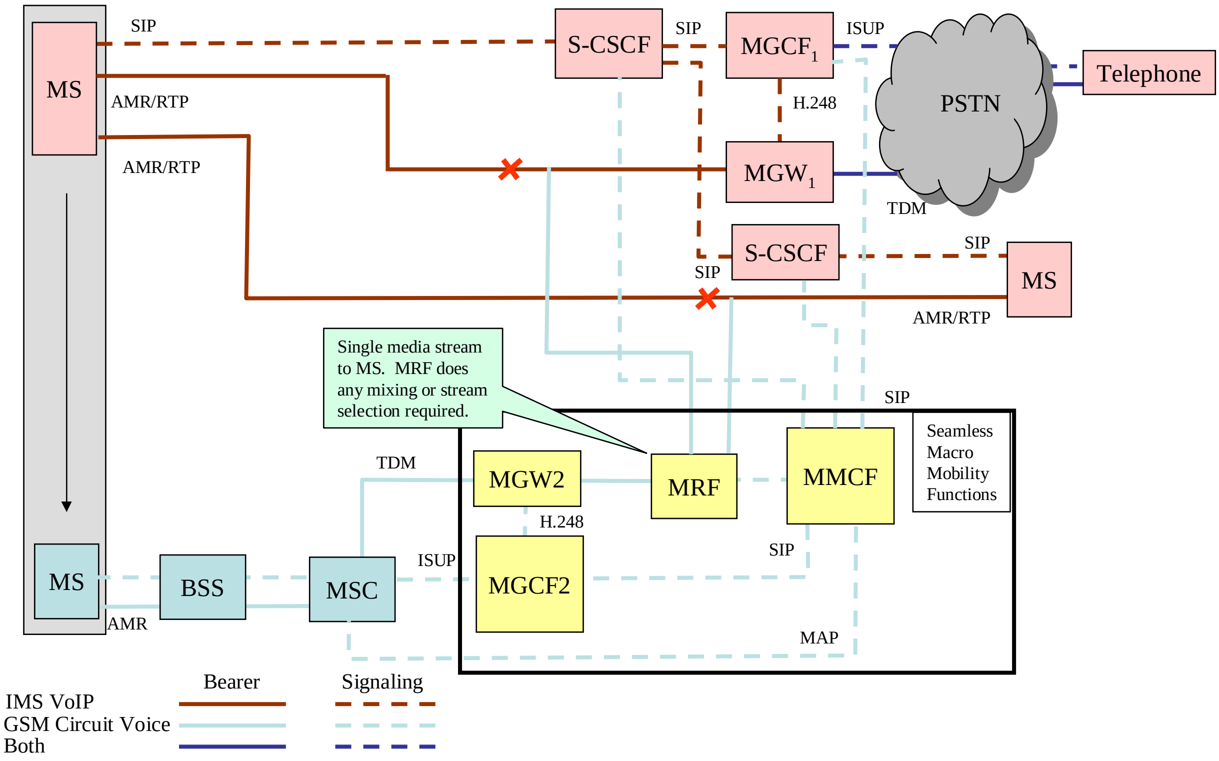


Figure 6.4.6.3-2: Handover - MMCF and MRF replace dual mode handset for IMS session and PSTN call

The steps in the procedure for handoff of a WLAN VoIP to a GSM circuit voice call and a subsequent hand back to WLAN are as follows.

1. STEP 1: The dual mode WLAN/GSM mobile, upon entry into a WLAN coverage area, must register with the IMS (IP Multimedia Subsystem) network and a Macro-Mobility Control Function (MMCF). The MMCF will use a SIP SUBSCRIBE/NOTIFY method to subscribe to a newly defined Mobility Event Package with the mobile, and the mobile will likewise subscribe to the same event package with the MMCF (flow line 1).

   - Standard IMS registration occurs, with the dual mode handset providing an indications that this is a dual mode dual mode handset.
   - The S-CSCF does a proxy register with the MMCF
   - The MMCF and dual mode handset subscribe to each others mobility events. The dual mode handset notifies the MMCF that it is idle and provides its IMSI. The MMCF notifies the dual mode handset of a handoff URI to use when a handoff is required. The handoff URI will uniquely identify the session as one being handed off for this dual mode handset.

2. The Mobile sets up the PSTN call and IMS dual mode handset VoIP session. In IMS, call waiting and 3-way call may be client features and not network features.

   - In the example flow, the dual mode handset has 2 sessions active
   - The first session is a call to the PSTN
   - The second session is an IMS VoIP call to another IMS dual mode handset.
   - In this example, both media streams are connected to the dual mode handset and the dual mode handset will do any media stream selection (e.g. for call waiting) or media stream mixing (e.g. for 3-way calling)

3. When the mobile is engaged in a call with another party on the PSTN (or with another mobile) and it detects that it is leaving WLAN coverage and entering the coverage area of the GSM network, the mobile makes the decision to hand over the call. It initiates 3 different steps in parallel:

   a. It notifies the MMCF that it wants to handoff .

   b. It initiates IMS session transfer procedures with the active PSTN session to transfer it to the MMCF.

   c. It initiates IMS session transfer procedures with the active MS2 session to transfer it to the MMCF.

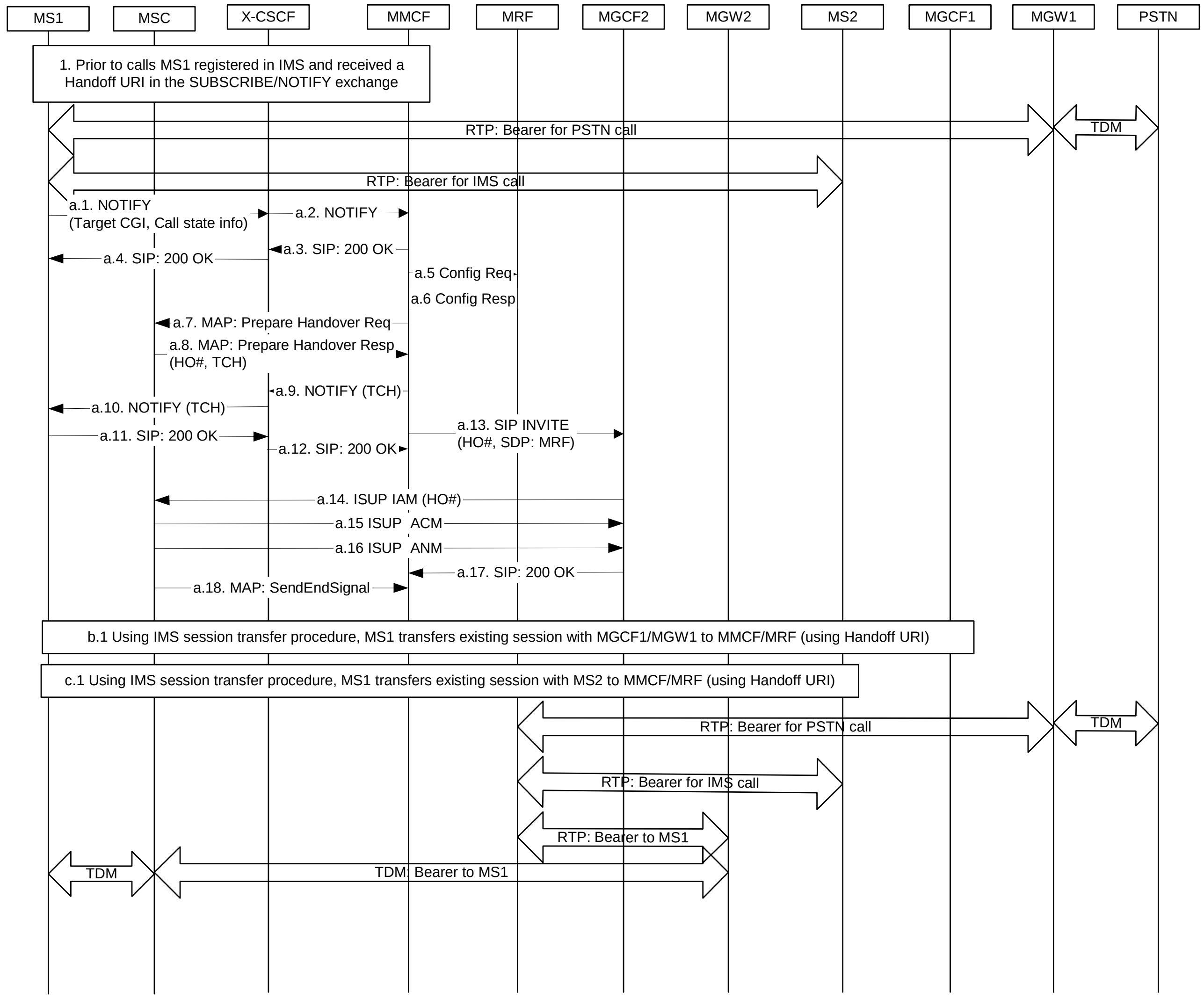
4. STEP a: The dual mode handset will notify the MMCF and pass it parameters that identify the target GSM system and cell ID. This marks the start of the handover sequence (a.1 – a.18).

   - The dual mode handset sends a notify to the MMCF with target information as well as information on the state of its existing IMS call sessions (e.g. there are 2 sessions, actively talking to MS2, but the PSTN session is on-hold in call waiting)
   - The MMCF will use GSM Handoff MAP signaling procedures to communicate with the GSM network and a circuit switched MSC.
     - Standard inter-MSC handoff MAP messages are sent and the MMCF receives back the target traffic channel
     - The MMCF notifies the dual mode handset which traffic channel to switch to.
     - When the dual mode handset switches to the traffic channel, the MSC completes the signaling with the MMCF to bring up the path from the dual mode handset to the MMCF via the GSM network. This results in connecting the GSM media stream to the MRF via MGW2.

5. STEP b: Using IMS session transfer procedures, MS1 transfers the existing session with the PSTN via MGCF1/MGW1 to the MMCF/MRF. MS1 uses the Handoff URI to inform MGCF1 of the destination. The MMCF had supplied this Handoff URI to MS1 in a NOTIFY at registration time (flow line b.1).

6. STEP c: Using IMS session transfer procedures, MS1 transfers the existing session with MS2 to the MMCF/MRF. MS1 uses the same Handoff URI to inform MS2 of the destination (flow line b.2).

With the completion of these steps, the handover has been completed. The bearer stream now flows to the MRF from MS2 and MGW1 on the IMS side. Media is combined or selected by the MRF and sent to MGW2 and then to the MSC and out to the dual mode handset.



6.4.6.4 IMS UE to CS UE call

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

Editor note: investigate the starting of the DTAP state machine in response for the need of handover.

Editor note: elaborate the encapsulation of the SIP message in DTAP.

Editor note: elaborate the Iur communication is required.

Editor note: consider point to point call scenario (e.g., IMS to IMS).

Editor note: fixed the call flow for the bearer to terminate at MS2 instead of MGCF1.

6.4.7 Impact on Supplementary Services

6.4.8 Evaluation of the model

This clause presents the evaluation of the service continuity solution against the set of criteria

6.5 Service Continuity Model: CS to IMS Voice Call Continuity – MS assisted

6.5.1 General Description

In this approach the MS initiates the appropriate sequence towards the IMS domain once it detects that a handoff/transition is required.

After the handoff/transition (in this document we use the terms handoff, handover and transition synonymously) is completed, Leg C would have replaced leg B as shown in the figure below.

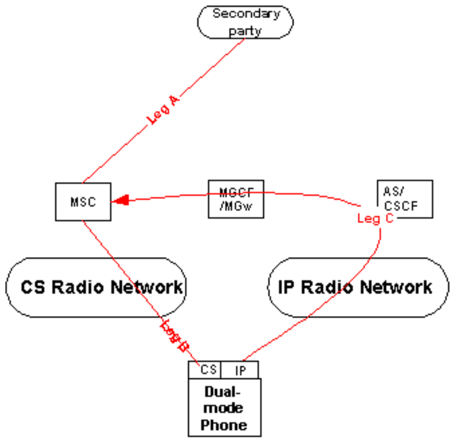


Figure 2. CS to IMS Voice Call Continuity – MS assisted (high level view)

This approach does not use an anchor point for the handoff until one is actually needed. As such, it employs an anchor point dynamically once a handoff is to occur.

6.5.2 Routing Selection Decision

6.5.3 Registration

6.5.4 Origination

6.5.4.1 IMS origination

6.5.4.2 GSM/UMTS CS origination

6.5.5 Termination

6.5.5.1 IMS termination

6.5.5.2 GSM/UMTS CS termination

6.5.6 Handover Scenarios

6.5.6.1 CS UE to CS UE call

The call flow to accomplish the above is illustrated below.

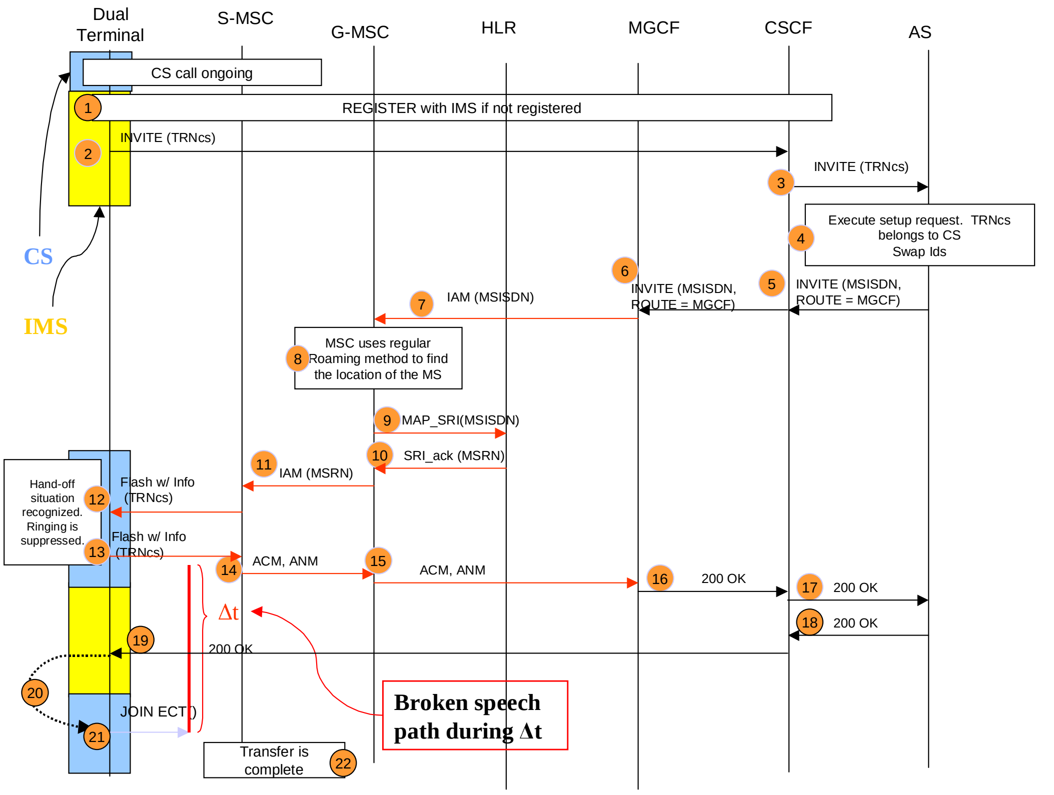


Figure 3. Call flow for CS to IMS Voice Call Continuity – MS assisted

1. The MS recognizes that a new access is available and makes the decision to handoff to this access. The MS will register the contact information and associated network access.

2. To start the setup of the IMS leg (Leg C), the terminal initiates a new call over IMS by sending an INVITE to the CSCF. A special transfer number (TRNcs) is used as an address uniquely identifying the call as a handoff call. The UE own MSISDN is included as calling party number (CgPN)

3. The INVITE is sent to the Application Server (AS) associated with the user, as result of an activated trigger for the originating INVITEs.

4. The AS analyses the call and identify the request as a handoff of voice service (based on the TRNcs) and that the call shall be connected to the CS domain. The AS swaps the MSISDN with the TRNcs to start a new call leg.

5. The AS sends an INVITE to the CSCF with destination MS address in the CS domain (Called Party Number - CdPN = MSISDN, CgPN = TRNcs, etc.) and ROUTE = MGCF.

6. The CSCF forwards the INVITE to MGCF.

7. The MGCF send an IAM with CdPN = MSISDN, CgPN = TRNcs to the GMSC

8. The GMSC uses regular methods to request routing information from the HLR.

9. The GMSC sends a MAP_SRI to the HLR

10. The HLR returns a routing number to the G-MSC

11.The call is forwarded to the Serving MSC.

12. The MSC finds that the MS is busy in a call, and sends a Call Waiting indication to the MS.

13. The ringing in the MS is suppressed. Note: The MS can recognize that the CgPN is this specific TRNcs.

14. The MS automatically accepts the incoming. The ongoing call (Leg A + Leg B) is put on hold.

15. The Serving MSC sends ACM/ANM to the MGCF to indicate that the call has been accepted.

16-18. The MGCF sends a SIP 200 OK to indicate that the call shall be setup. The message is routed through the CSCF, AS and to the MS (IMS Part). The call media path is setup between the IP and the CS part of the MS.

19. The MS receives the SIP 200 OK and triggers the Explicit Call Transfer.

20-21. The MS send an ECT JOIN to the serving MSC over leg B. This connects the A party with the IMS side of the MS (Leg A + Leg C). The Leg over the CS radio is disconnected (Leg B).

22. Handoff is complete

6.5.6.2 CS UE to IMS UE call

6.5.6.3 IMS UE to IMS UE call

6.5.6.4 IMS UE to CS UE call

6.5.7 Impact on Supplementary Services

6.5.8 Evaluation of the model

The call will have a short voice interruption time, as indicated in the figure 2.

This approach has the advantage that no changes are required to the CS domain nodes.

The AS and the UE implement all the needed functionality to achieve the desired behaviour.

Given the fact that CS coverage is quite adequate, the risk of losing CS coverage before the handoff/transition is completed is very slim, there are no impacts on nodes in CS domain, it is recommended to adopt that alternative for CS-IMS handoff/transition.

It is also important to note that there is a need in this approach to ensure that service interaction is properly handled to ensure that the handoff call is successfully delivered. To that extent, unconditional transfer services and all transfer services in general must become deactivated prior to handoff. In addition call waiting and CLIP must be activated. Following the handoff all of the above services must be restored to their original condition.

Finally in this approach, the handoff cannot succeed if the UE is already engaged in a 2-way call prior to the handoff or the call is initiated as an emergency call.

The figure below presents the user media plane topologies before and after the handover procedures.

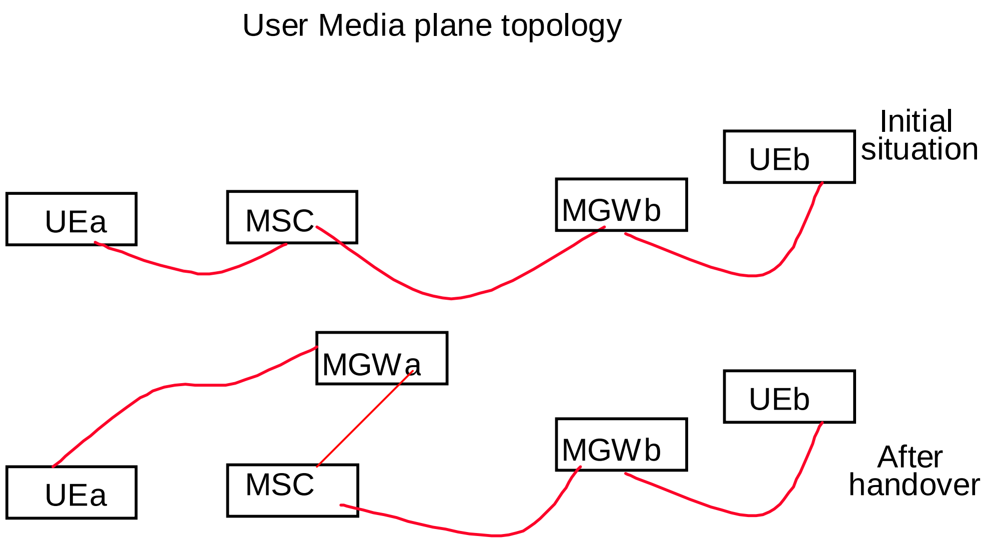


Figure 4. User media path from CS to IMS, UEb in IMS domain

6.6 Service Continuity Model: IMS to CS Voice Call Continuity – MS assisted

6.6.1 General Description

In this approach, there is a voice anchor point within the IMS domain to handle the call leg toward the MS via the CS domain.

After the handoff/transition is completed, Leg C would have replaced leg B as shown in the figure below.

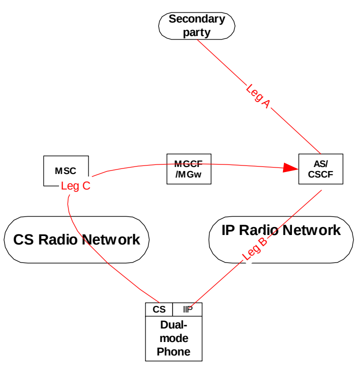


Figure 5. IMS to CS Voice Call Continuity – Anchoring model (high level view)

There are two approaches described here, one as static anchoring and the other as dynamic anchoring. Static anchoring means an AS will always be involved in every IMS call, acting as a B2BUA. Dynamic anchoring means the AS will only be involved when voice continuity to CS domain is needed.

6.6.2 Routing Selection Decision

6.6.3 Registration

6.6.4 Origination

6.6.4.1 IMS origination

6.6.4.2 GSM/UMTS CS origination

6.6.5 Termination

6.6.5.1 IMS termination

6.6.5.2 GSM/UMTS CS termination

6.6.6 Handover Scenarios

6.6.6.1 CS UE to CS UE call

6.6.6.2 CS UE to IMS UE call

6.6.6.3 IMS UE to IMS UE call

6.6.6.3.1 Static Anchoring Model

The call flow to accomplish the Static Anchoring Model is illustrated below.

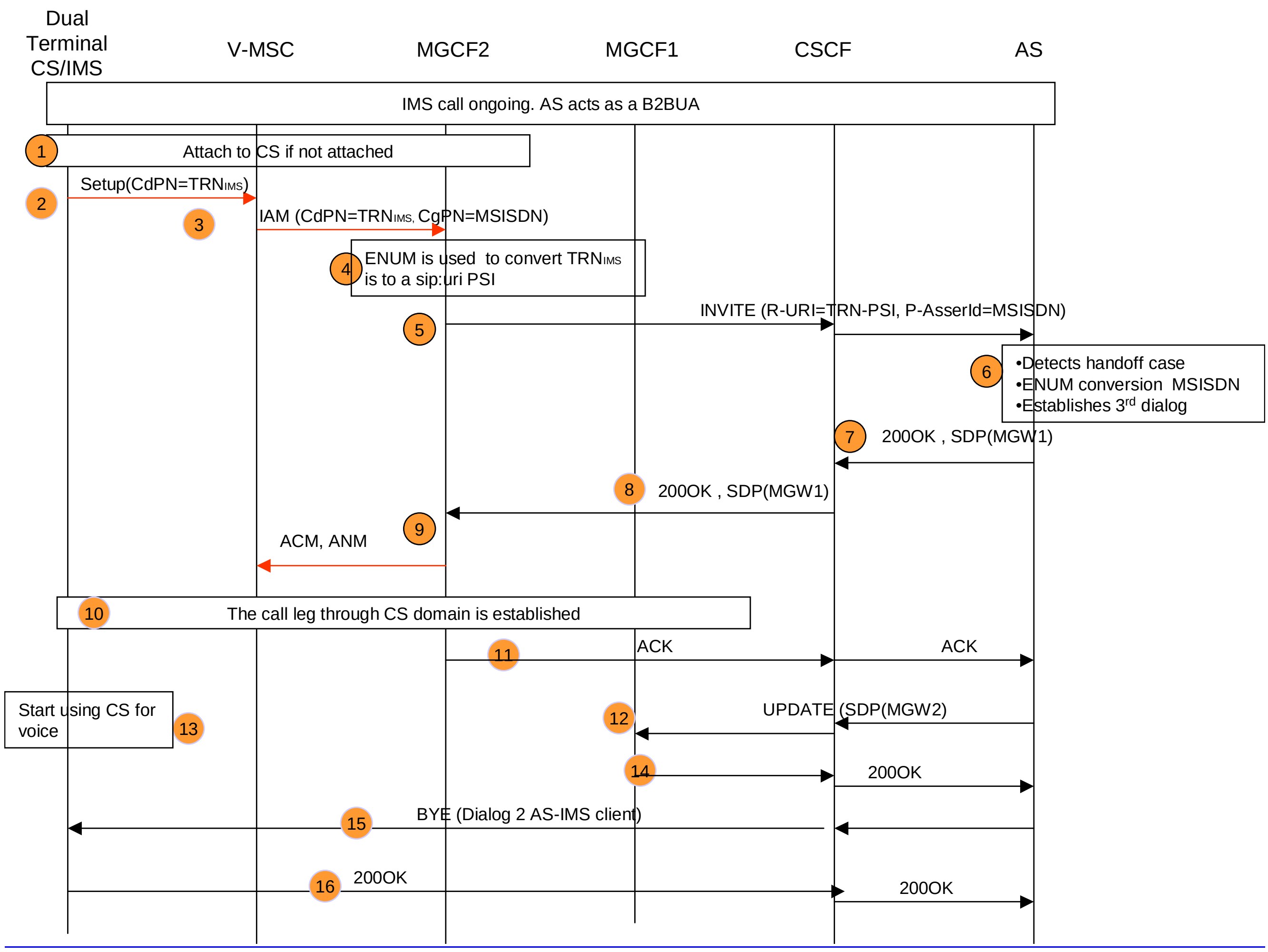


Figure 6. Call flow for IMS to CS Voice Call Continuity – Static Anchoring Model

1. The MS registers and attaches to the CS domain

2. To start the new leg (Leg C) the MS initiates a call in the CS domain. The address used is the TRNims uniquely identifying the call as a handoff call.

3. The Serving MSC adds the CgPN = MSISDN to the IAM and forwards the call to MGCF2 (depending on the actual network configuration, the IAM might transit the G-MSC).

4. MGCF2 (or an I-CSCF/S-CSCF which are not shown, but which might have been contacted by MGCF) interrogates ENUM database and converts the TRNims to a sip:uri Public Service Identifier (PSI). The information from the CgPN is copied in the P-Asserted ID SIP header

5. By performing normal SIP routing procedures the INVITE is send to an AS hosting the hand-off service, identified by the PSI.

6. The AS detects the handoff case, performs ENUM conversion on the MSISDN, then establishes a third dialog.

7. The AS sends a 200 OK establishing the third dialog.

8. The S-CSCF forwards the 200 OK to MGCF2

9. MGCF2 sends an ANM to the V_MSC.

10. Voice path is through connected, but no voice is yet transmitted from the CS part of the client. The IMS part of the client is the one receiving/transmitting voice at this moment.

11 MGCF2 sends an ACK confirming the Dialog 3 establishment

12. On reception of ACK, AS changes Dialog 1, by sending an UPDATE that informs MGCF1 to start sending/receiving media from MGW2 (associated with MGCF2)

13. The client which is aware of the hand-off situation, starts using the CS bearers for sending/receiving the voice

14. MGCF1 confirms the update. At this moment voice is connected between MGW1 and MGW2

15. At the same time as 12, AS releases the Dialog 2, towards the IMS client, by sending a BYE.

16. On the reception of BYE, the client responds with a 200 OK and releasing Dialog 2

Note that step 16 may not be fully completed if the IMS coverage is lost, but that should not impact the robustness of the procedure. The client design should be built with that assumption in mind.

6.6.6.3.2 Dynamic Anchoring Model

The call flow to accomplish the Dynamic Anchoring Model is illustrated below.

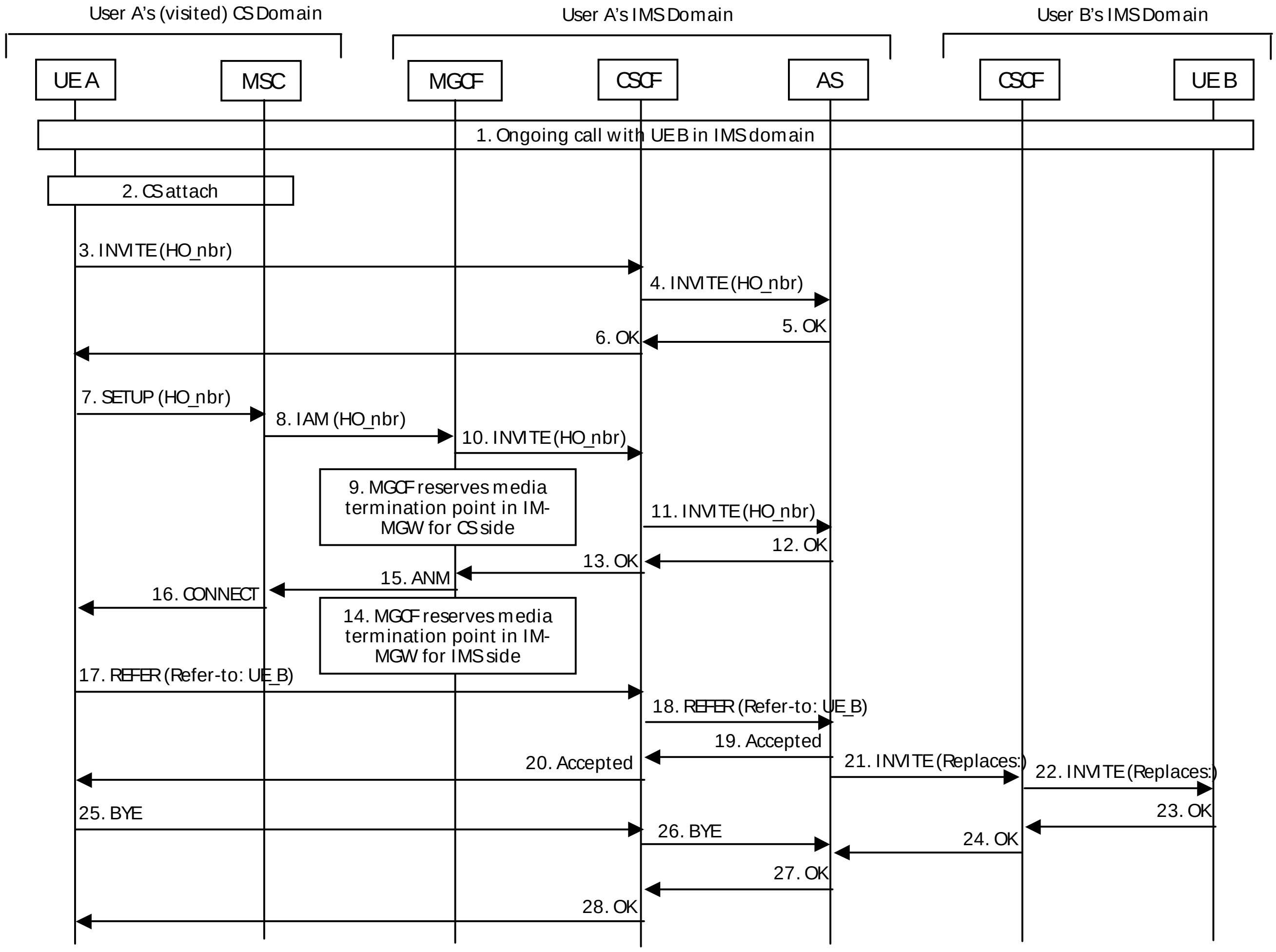


Figure 7. Call flow for IMS to CS Voice Call Continuity – Dynamic Anchoring Model

1. UE A in IMS domain has an ongoing voice session with UE B in IMS domain.

2. UE A compares the cellular and WLAN radio coverages, and decides to perform a handover. UE A performs a CS Attach procedure.

3. UE A sends an INVITE request towards a provisioned E.164 Handover number (HO_nbr) that points to the Application Server. The Handover number may be pre-provisioned during registration or it may be learned during IMS registration procedures.

4. S-CSCF of UE A forwards the request towards the Application Server based on the Request-URI.

5. AS acknowledges the session establishment.

6. Acknowledgement is forwarded to UE A

7. UE A initiates a normal CS call to the Handover number.

8. The Visited MSC routes the call towards the IMS domain via MGCF using normal CS-IMS interworking procedures.

9. MGCF reserves a media termination point for the CS side.

10. MGCF creates a INVITE request and sends it towards the AS.

11. S-CSCF forwards the INVITE towards the AS based in the Request-URI.

12. OK response to the INVITE is forwarded towards the S-CSCF.

13. OK response to the INVITE is forwarded towards the MGFC.

14. MGCF reserves a media termination point in the IM-MGW for the IMS side of the call.

15. MGCF responds with ANM to the Visited MSC.

16. CONNECT response is forwarded to the UE

17. UE A creates a REFER request that instructs the AS to invite UE B to the handover call.

18. S-CSCF forwards the request towards the AS.

19. AS acknowledges the REFER request.

20. Acknowledgement is forwarded to UE A.

21. AS invites UE B to the handover call. The INVITE request has a Replaces header that instructs the UE B to replace the ongoing call with UE A with the new call.

22. S-CSCF of UE B forwards the request to UE B.

23. UE B sends an OK reply to the AS. UE B modifies the ongoing session with UE A so that the media is now routed towards the MGW.

24. S-CSCF of UE B forwards the OK to the AS.

25. UE A releases the ongoing call with UE B.

Editor note: How supplementary services are handled with Dynamic model is FFS

6.6.6.4 IMS UE to CS UE call

6.6.7 Impact on Supplementary Services

6.6.8 Evaluation of the model

For Static Anchoring Method, the MS initiates the setup of the leg C in order to minimize the interruption in the voice call, due to paging or traffic channel setup times, while the IMS domain is in control of the handoff process.

Hence, an AS will be involved in every IMS call, acting as a B2BUA. Before the hand-off occurs the AS controls two SIP dialogs:

1. Dialog1 - MGCF1 to AS

2. Dialog2 - AS to the terminal (IMS client)

The entire handoff procedure should take around 1.3 seconds including the speech switch over.

It is to be noted that even if the IMS coverage is lost during the time it takes to establish the circuit switched leg of the call (Leg C), only one of the IMS legs will be lost for a very brief period < 1.3 seconds.

That brief period is insignificant to the extent that the switch over to the CS will be completed before the end user (at the remote end) perceives any interruption.

For Dynamic Anchoring Model, this alternative is similar to Static Anchoring model with the exception that there is no AS linked in the call at the beginning of the call. The AS is linked in dynamically only when a handoff is required, and is achieved by sending an INVITE to the AS.

Once the AS takes over the handoff process, and if the IMS coverage is lost thereafter during the time it takes to establish the circuit switched leg of the call (Leg C), only one of the IMS legs will be lost for a very brief period <1.3 seconds.

That brief period is insignificant to the extent that the switch over to the CS will be completed before the end user (at the remote end) perceives any interruption.

The switch over and the usage of the CS leg of the call (Leg C) is independent from the clearing of the IMS leg of the call (Leg B) which would not be graceful. The client design should be as such that the switch over to the CS will occur regardless if the IMS leg is gracefully terminated or not due to coverage problems)

Note that if the IMS coverage is lost before the AS takes over then the call will be lost, and the handoff procedure will fail.

This will happen if the INVITE, or the ACK never makes it to the AS. This is a key difference from Static Anchoring Model, which of course required an anchor point at all times.

6.7 Service Continuity Model: Mobility Management AS with Anchoring HO Model

6.7.1 General Description

Mobility Management Control Function as described in ref [4] consists of mainly Registrar, HO, and VMSC/VLR functions, and it is similar to an IMS Application Server. This proposal adopts its Registrar and the VMSC/VLR functions excluding the HO related parts to manage the IMS registration and CS side location update, to synchronise the registration state between the two networks, and to manage the routing of terminating calls between domains. There may be additional functions described in this alternative that are in addition to the ones defined in ref [4].

The registration synchronisation between domains may be affected by both the network and user preferences. Such operator policies may be (pre-) configured to the terminal, or they may be available dynamically. User preferences may also be (pre-) configured to the terminal, or they may be based on user input. This also determines the routing toward the terminating domains for terminating call.

The HO mechanism from CS to IMS is based on ECT (see ref [1,3]) when call is first established on the CS side and it has not been anchored in the IMS domain. Ref [1] indicates how ECT is used to transfer the call to IMS domain and ref [3] is used to indicate that the subsequent transfer due to “ping-pong” HO would not require further ECT usage to avoid further unnecessary call leg establishment between the domains. The HO mechanism from IMS to CS is based on anchoring model (see ref [2]) in which the logics is resided in Application server that has this voice anchoring function.

6.7.2 Routing Selection Decision

The following cases and sub-scenarios are analysed with this architecture for network domain termination.

Case 1: the user can only be reached via WLAN access with IMS

Scenario 1A: user’s E.164 MSISDN is assigned by the IMS and subscription is stored in the HSS.

No impact. In this case, the PSTN routes the called party E.164 number toward the IMS network. Normal IMS routing procedure toward the user takes place and no new procedure is needed in this TR.

Scenario 1B: user's E.164 MSISDN is assigned by the CS Domain and subscription is stored in the HSS/HLR.

During IMS registration, the MMCF acts as a VMSC an update the HSS/HLR with its address. For call termination, the PSTN routes the called party E.164 number toward the CS Domain (GMSC). GMSC queries the HSS/HLR which quires the MMCF for the MSRN for routing toward the IMS network. Procedure is defined in ref [4] to handle this.

Scenario 1C: user has 2 E.164 MSISDNs, one is assigned by IMS and is stored in the HSS. The other MSISDN is assigned by CS domain and is stored in the HSS/HLR.

This case is considered that the user has 2 separate subscriptions. If the dialled number is a CS domain assigned E.164 number then the routing process is based on scenario 1B above. If the dialled number is an IMS assigned E.164 number then the routing process is based on scenario 1A above.

Case 2: user can be reached only via Cellular access with CS Domain

Scenario 2A: user's E.164 MSISDN is assigned by the IMS and subscription is stored in the HSS.

During Location Update procedure in the CS Domain, it contacts the subscriber HPLMN via MAP_UPDATE_LOCATION. The PSTN routes the called party E.164 number toward the IMS network. The Mobile termination call procedure related to unregistered PSI is used and the MMCF acting as an application server forwards the call toward the VPLMN using the MSRN received from the VMSC. Procedure is defined in ref [4] to handle this.

Scenario 2B: user's E.164 MSISDN is assigned by the CS Domain and subscription is stored in the HSS/HLR.

No impact. In this case, the PSTN routes the called party E.164 number toward the CS Domain (GMSC). Normal GSM terminating call routing is used in this case. No new procedure is needed in this TR.

Scenario 2C: user has 2 E.164 MSISDNs, one is assigned by IMS and is stored in the HSS. The other MSISDN is assigned by CS domain and is stored in the HSS/HLR.

This case is considered that the user has 2 separate subscriptions. The USIM from the CS domain is used for registering to the CS domain. After that, this user can only be reached via the CS domain assigned E.164 number. It should be possible that the IMS domain deploys a type of call forwarding service so that all terminated call toward the IMS assigned E.164 could also be forwarded to the CS assigned E.164 number during unregistered state.

Case 3: user can be reached by both the Cellular or WLAN access and is registered to both domains.

Scenario 3A: user's E.164 MSISDN is assigned by the IMS and subscription is stored in the HSS.

During IMS registration, the UE needs to indicate to the MMCF that it has also registered to the CS domain. MMCF receive IMS registration and MAP_UPDATE_LOCATION from the VMSC. The PSTN routes the called party E.164 number toward the IMS network. The S-CSCF invokes the MMCF AS to determine whether the call should be continue routed using the IMS domain or divert the routing toward the VPLMN using the CS access. The decision could be based on e.g., static rule such as use the IMS if an ongoing VoIP session is in place, or based on user's preference or operator's preference.

Scenario 3B: user's E.164 MSISDN is assigned by the CS Domain and subscription is stored in the HSS/HLR.

During IMS registration, the UE needs to indicate to the MMCF that it has also registered to the CS domain. The IMSI indicates to the MMCF that the CS Domain registration is with the CS assigned E.164 number. The logic resided in the MMCF could invoke a MAP_Location_update toward the HSS/HLR to force all the call to be handled via the IMS side. If this is done, it also has to indicate to the UE that it should not perform Update_location procedure on the CS side. Otherwise, it will cancel the MMCF's location update. The decision for whether the MMCF's location update is invoked or not may depended on operator's or the user's preference. If MMCF location update is not performed then the terminating call will be handled via the CS domain as they are today. The user may have a choice to originate the call via CS or IMS, subject to the UE capabilities (e.g., UE may only not allow multiple voice call/sessions simultaneously on both the CS and IMS side).

Scenario 3C: user has 2 E.164 MSISDNs, one is assigned by IMS and is stored in the HSS. The other MSISDN is assigned by CS domain and is stored in the HSS/HLR.

This case is considered that the user has 2 separate subscriptions. During IMS registration, the UE needs to indicate to the MMCF that it has also registered to the CS domain. The IMSI indicates to the MMCF that the CS Domain registration is with the CS assigned E.164 number. In order to terminate CS assigned E.164 number in IMS domain, the logic resided in the MMCF would have to invoke a MAP_Location_update toward the HSS/HLR to force all the CS assigned E.164 call to be handled via the IMS side. In order to terminate IMS assigned E.164 number in the CS side, the logic resided in the MMCF would have to invoke call forwarding service to the CS assigned E.164 number. These logics must be coordinated and could be based on operator's or the user's preference.

Editor note: CS domain subscriber can contain multiple MSISDNs for different services in the subscription; the impact toward the services beside voice (i.e., fax, data, UDI) with MAP_UPDATE_LOCATION from IMS side. This is FFS!

Editor note: In IMS, multiple devices in addition to the dual mode UE can be registered with the same public user identity, how to ensure that the routing of terminating call to the dual mode UE which is now being served in the CS domain will not affect the other IMS devices that are registered using the same public user identity? This is FFS!

Editor note: Assuming the IMS and CS subscriptions are coming from separate domain, the assumption is also that both of these subscriptions are used within the same device. The MMCF has the role of registration synchronisation between domains. If the subscription can be separated, i.e., one device uses IMS, other physically separated device uses CS subscription, will there be any interaction issue to the service provided to each device respectively by each domain. This is FFS!

6.7.3 Registration

6.7.4 Origination

6.7.4.1 IMS origination

6.7.4.2 GSM/UMTS CS origination

6.7.5 Termination

6.7.5.1 IMS termination

6.7.5.2 GSM/UMTS CS termination

6.7.6 Handover Scenarios

6.7.6.1 CS UE to CS UE call

6.7.6.2 CS UE to IMS UE call

6.7.6.3 IMS UE to IMS UE call

6.7.6.4 IMS UE to CS UE call

6.7.7 Impact on Supplementary Services

6.7.8 Evaluation of the model

This clause presents the evaluation of the service continuity solution against the set of criteria

6.8 Service Continuity Model: Mobility Management Application Server

6.8.1 General Description

The Mobility Management Application Server (MM-AS) defines a logical entity that enables roaming and handover functionality between the CS domain and the IMS.

The MM-AS acts as a “virtual” MSC/VLR to the GSM/UMTS domain, when a UE is registered in IMS.

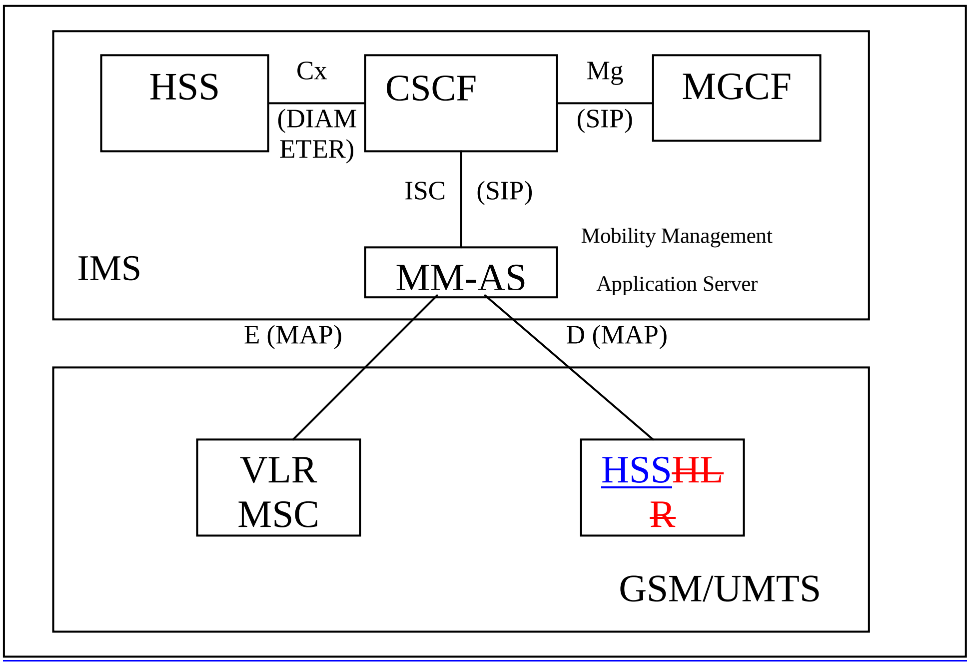


Figure 1: IMS-GSM/UMTS Inter-working Network Architecture

The proposed alternative concept does not require any changes to deployed CS domain infrastructure (HLR, MSC, etc.).

Editor's Note: Alignment with the reference architecture model required.

6.8.2 Routing Selection Decision

6.8.3 Registration

6.8.3.1 General

When a dual mode terminal is switched on, it attempts to register with the GSM/UMTS network (including authentication). After GSM/UMTS registration is successful or if certain network selection settings within the terminal indicate to search for I-WLAN access first, the terminal searches for I-WLAN and attempts for IMS registration.

6.8.3.2 Roaming between GSM/UMTS and IMS

A dual mode terminal periodically polls for I-WLAN signal and attempts to register with I-WLAN, if all selection criteria are fulfilled. When I-WLAN access registration is successful, the dual mode terminal attempts IMS registration.

The MM-AS is configured in the HSS user profile to receive REGISTER requests through the ISC interface.

The Figure 2 shows the roaming scenario for a dual mode handset moving from GSM/UMTS access to I-WLAN access.

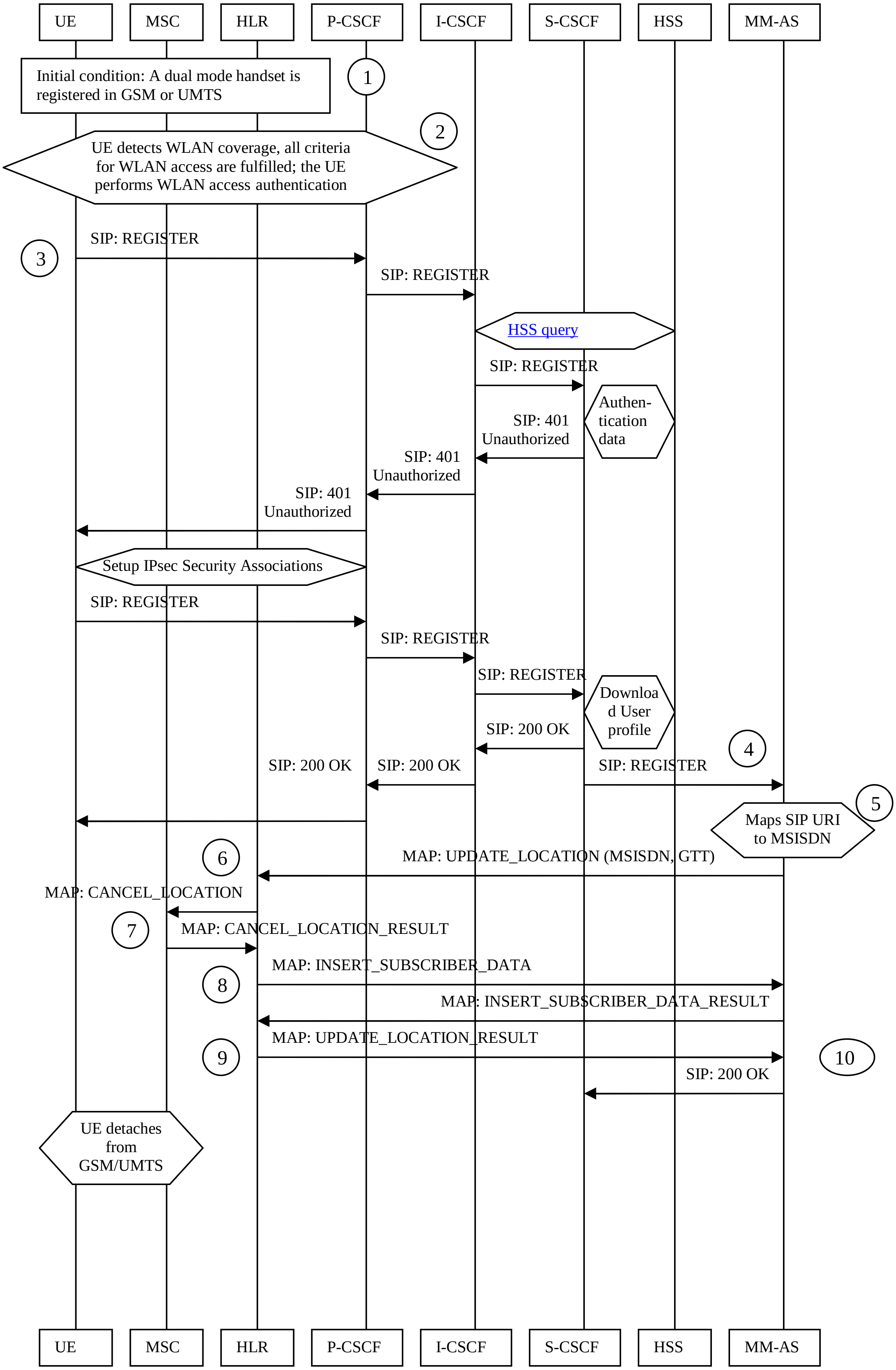

UE
MSC
HLR
P-CSCF
I-CSCF
S-CSCF
HSS
MM-AS
Initial condition: A dual mode handset is registered in GSM or UMTS
1
2
UE detects WLAN coverage, all criteria for WLAN access are fulfilled; the UE performs WLAN access authentication
3
SIP: REGISTER
SIP: REGISTER
HSS query
SIP: REGISTER
Authen-tication data
SIP: 401 Unauthorized
SIP: 401 Unauthorized
SIP: 401 Unauthorized
Setup IPsec Security Associations
SIP: REGISTER
SIP: REGISTER
SIP: REGISTER
Download User profile
SIP: 200 OK
4
SIP: 200 OK
SIP: 200 OK
SIP: REGISTER
5
Maps SIP URI to MSISDN
6
MAP: UPDATE_LOCATION (MSISDN, GTT)
MAP: CANCEL_LOCATION
7
MAP: CANCEL_LOCATION_RESULT
8
MAP: INSERT_SUBSCRIBER_DATA
MAP: INSERT_SUBSCRIBER_DATA_RESULT
9
MAP: UPDATE_LOCATION_RESULT
10
SIP: 200 OK
UE detaches from GSM/UMTS
UE
MSC
HLR
P-CSCF
I-CSCF
S-CSCF
HSS
MM-AS

Figure 2. Roaming from GSM/UMTS to IMS

1. Initial condition: A dual mode UE is attached to the CS domain in GSM/UMTS

2. A dual mode UE enters a WLAN coverage area. The selection criteria for WLAN access are fulfilled. The UE performs WLAN access authentication and authorization and establishes a connection to the serving network (e.g. tunnel to the PDG in the HPLMN).

3. The UE sends a REGISTER request to the user's home IMS network to perform SIP registration (SIP: +49-89-636-12345@siemens.com, user = phone).

4. The MM-AS receives the REGISTER request.

5. The MM-AS maps the user's SIP URI (received in the REGISTER request) to MSISDN from its User Profile Database.

6. The MM-AS performs Global Title Translation on the MSISDN to determine the HLR and sends a MAP: UPDATE_LOCATION to the HLR.

7. The HLR sends MAP: CANCEL_LOCATION to the previous attached VLR/MSC.

Editor's note: It is for further study if the UE can be CS attached too.

8. The HLR sends the user's profile data through MAP: INSERT_SUBSCRIBER_DATA request(s).

9. The HLR acknowledges the MAP: UPDATE_LOCATION to the MM-AS.

10. Finally, the MM-AS accepts the registration with a 200 OK.

Now the user is registered successfully in the IMS and detaches from GSM/UMTS.

6.8.3.3 Roaming from IMS to GSM/UMTS

When a dual mode terminal is registered in IMS, it periodically checks the WLAN signal strength. When the UE senses drop of WLAN signal or because of certain network selection settings, it attempts to register to GSM/UMTS.

The figure 3 shows the roaming scenario for a dual mode handset moving from WLAN (IMS) access to GSM/UMTS access to support voice call continuity.

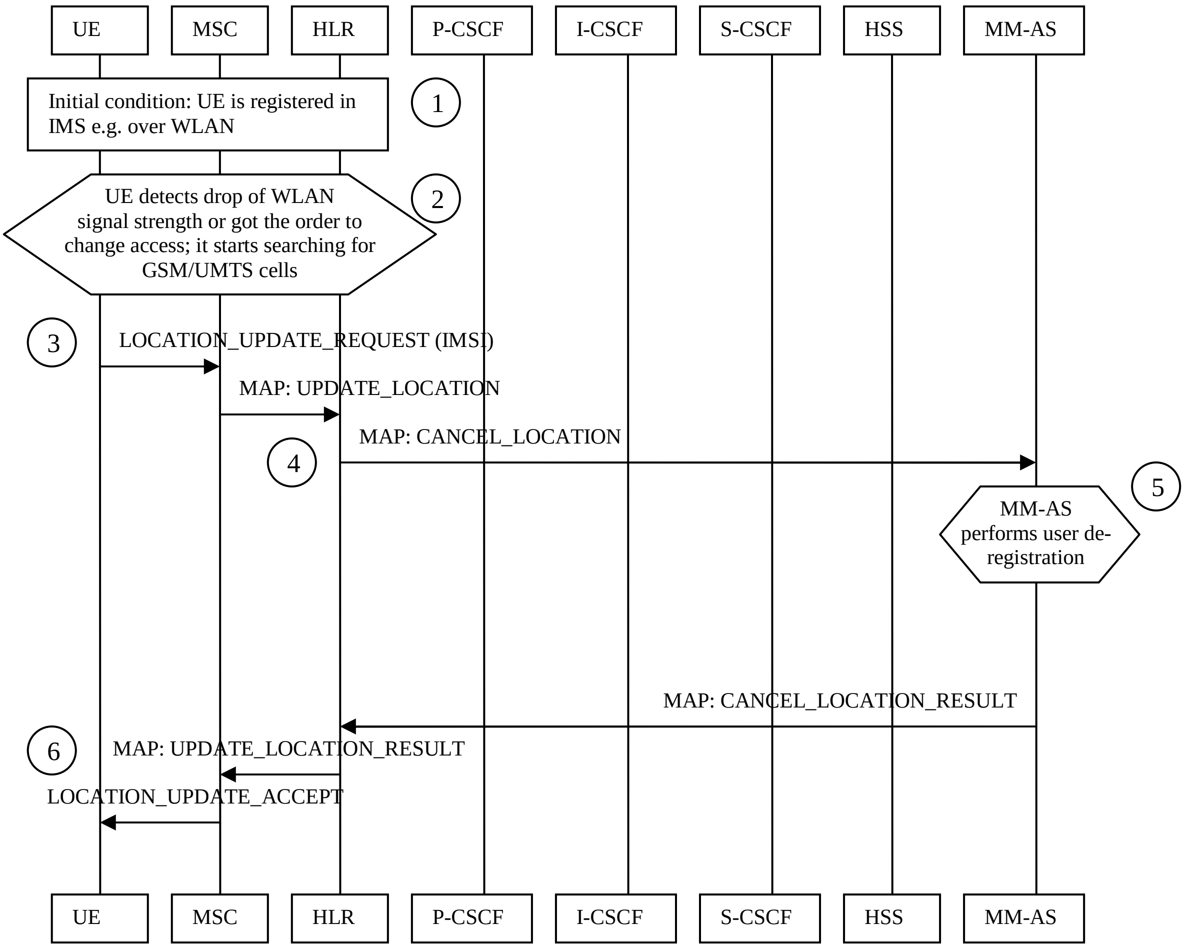


Figure 3. Roaming from IMS to GSM/UMTS

1. Initial condition: A dual mode UE is registered in IMS e.g. over WLAN.

2. UE detects drop of WLAN signal strength or got the order to change access; it starts searching for GSM/UMTS cells.

3. If the UE finds a suitable GSM/UMTS cell, it performs IMSI attach by sending a LOCATION_UPDATE_REQUEST.

4. The HLR informs the MM-AS about the IMSI attach by sending a MAP: UPDATE_LOCATION towards the MM-AS.

5. The MM-AS initiates IMS de-registration for this user. During this “Network Initiated De-registration by Service Platform” procedure, the UE is informed by S-CSCF about the De-registration.

6. The MM-AS acknowledges the HLR by a MAP: CANCEL_LOCATION_RESULT. The user receives the LOCATION_UPDATE_ACCEPT.

7. UE performs IMS de-registration if WLAN signal is still available.

6.8.4 Origination

6.8.4.1 IMS origination

6.8.4.1.1 Mobile Originating Call from IMS to IMS

The MM-AS acts as a "virtual" MSC/VLR to the GSM/UMTS domain, when a UE is registered in IMS.

1. Initial condition: A dual mode terminal is registered in IMS
2. UE1 initiates a VoIP call to UE2, which is also IMS registered.
3. The MM-AS is contacted on the basis of initial filter criteria, which were downloaded by the S-CSCF from HSS during UE1 registration.
4. The MM-AS is contacted on the basis of initial filter criteria, which were downloaded by the S-CSCF from HSS during UE2 registration, i.e. UE2 is also a dual mode terminal.
5. UE2 receives the INVITE and responds with a 200 OK.
6. UE1 receives 200 OK from UE2 and responds with the final ACK.
7. UE2 receives the final ACK, which establishes the call.

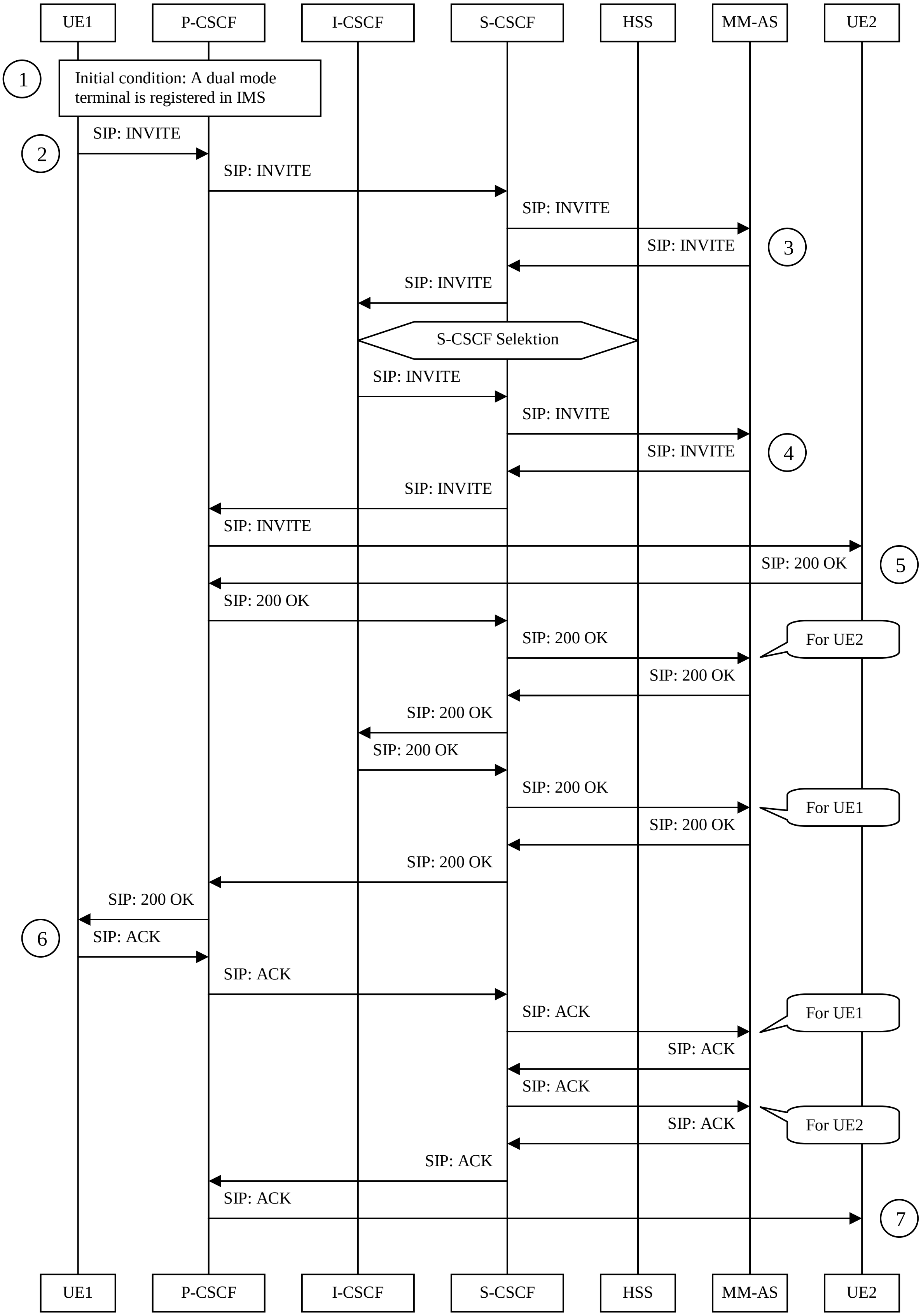


Figure 1. Mobile Originating Call from IMS to IMS

6.8.4.1.2 Mobile Originating Call from IMS to GSM/PSTN

The MM-AS acts as a "virtual" MSC/VLR to the GSM/UMTS domain, when a UE is registered in IMS.

1. Initial condition: A dual mode terminal is registered in IMS
2. The UE initiates a call to a user in GSM or PSTN.
3. The MM-AS is contacted on the basis of initial filter criteria, which were downloaded by the S-CSCF from HSS during UE registration.
4. The MGCF receives the INVITE and sends an ISUP: INITIAL_ADDRESS_MESSAGE (IAM) to the GMSC and PSTN.
5. The GSMC/PSTN sends an ISUP: ADDRESS_COMPLETE_MESSAGE to the MGCF, which sends a SIP: 180 Ringing towards the UE.
6. The ISUP: ANSWER_MESSAGE indicates that the call is confirmed.
7. The UE receives a 200 OK and responds with the ACK, the call is now established.
8. The S-CSCF forwards the ACK to the MM-AS.

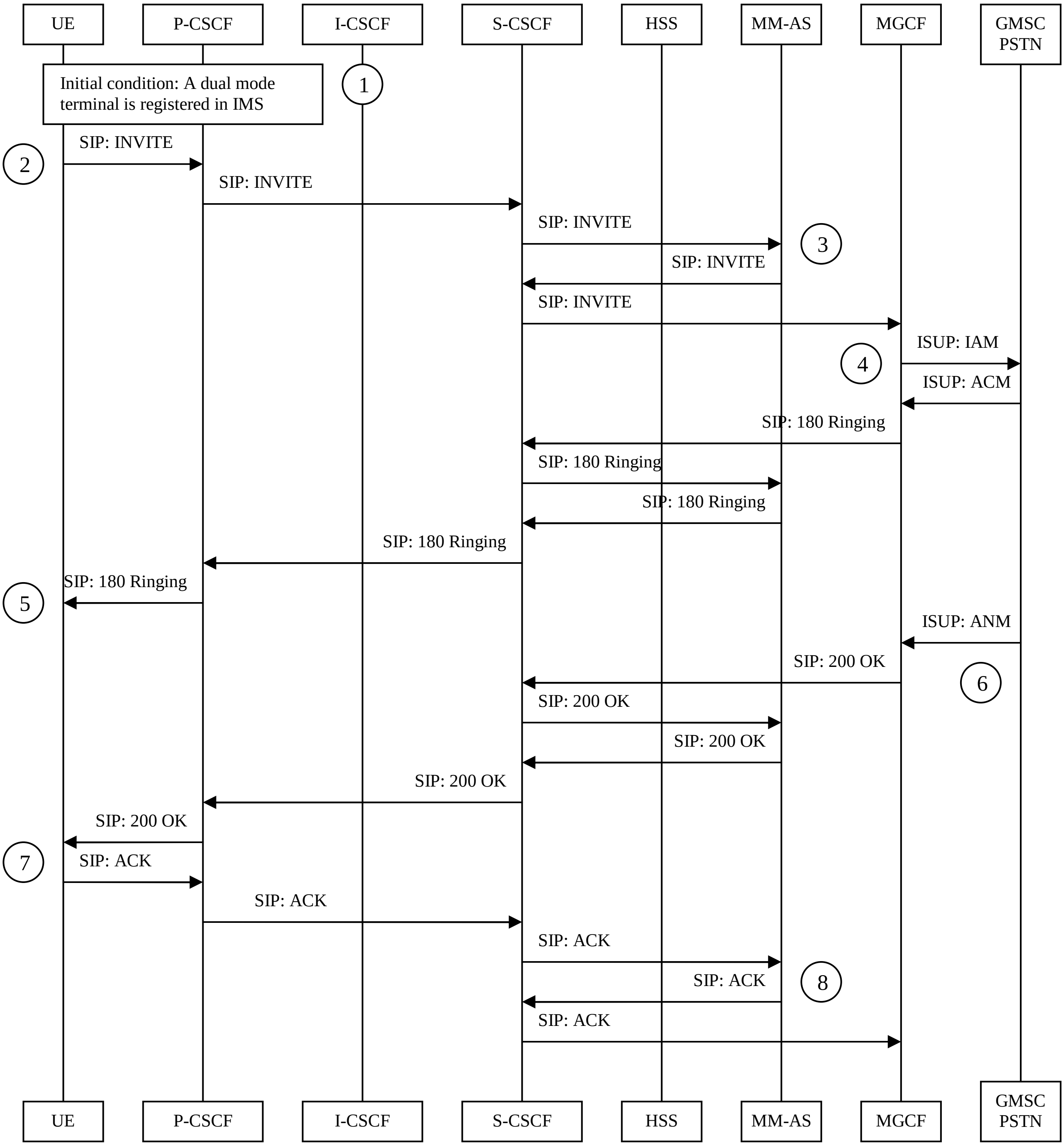


Figure 2. Mobile Originating Call from IMS to GSM/PSTN

6.8.4.2 GSM/UMTS CS origination

6.8.5 Termination

6.8.5.1 IMS termination

The MM-AS acts as a “virtual” MSC/VLR to the GSM/UMTS domain, when a UE is registered in IMS.

1. When the MTC is received, the call is routed to the GMSC (since the number is owned by the GSM domain).

2. GMSC queries the HLR to retrieve a roaming number from the serving MSC/VLR through MAP: SEND_ROUTING_INFO.
3. HLR requests the MM-AS to generate a Roaming Number (MSRN) for the UE through MAP: PROVIDE_ROAMING_NUMBER procedure.
4. If the user is still registered in IMS, the MM-AS generates a MSRN, which allows the GMSC to route to a MGCF in the roaming network. Furthermore, the MSRN should be coded in such a way that the MSISDN can be derived from it.
5. The MM-AS sends the MSRN within the MAP: ROVIDE_ROAMING_NUMBER_ACK to the HLR, the HLR to the GMSC. The GMSC sends an ISUP IAM, which terminates at the MGCF.
6. The MGCF uses the MSRN to generate a SIP: INVITE containing the MSISDN of the user as a tel URL towards the I-CSCF.

Editor's Note: It is FFS, if number translation or ENUM query to derive the tel URL.

7. The MM-AS is contacted on the basis of initial filter criteria, which were downloaded by the S-CSCF from HSS during registration.
8. The UE receives the terminating call, starts ringing (not shown in figure 1) and sends the 200 OK.
9. (and 9') The MM-AS is involved in the whole SIP message flow.
10. The MGCF maps the 200 OK to an ISUP: ANM and sends it towards the GMSC.

~~11.~~ The UE receives the final ACK, which finally establishes the terminating call.

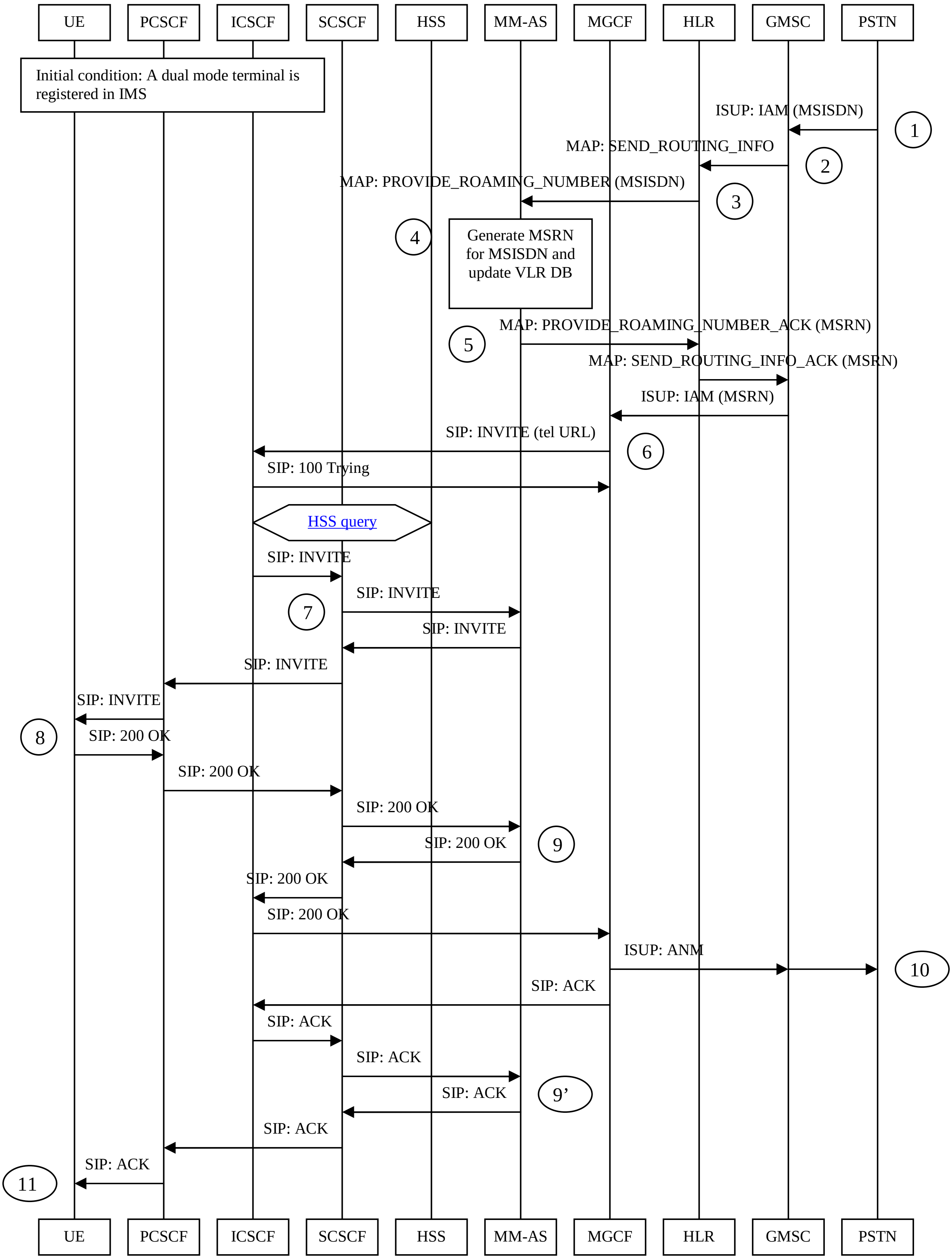


Figure 1. Terminating Call when UE is registered in IMS

6.8.5.2 GSM/UMTS CS termination

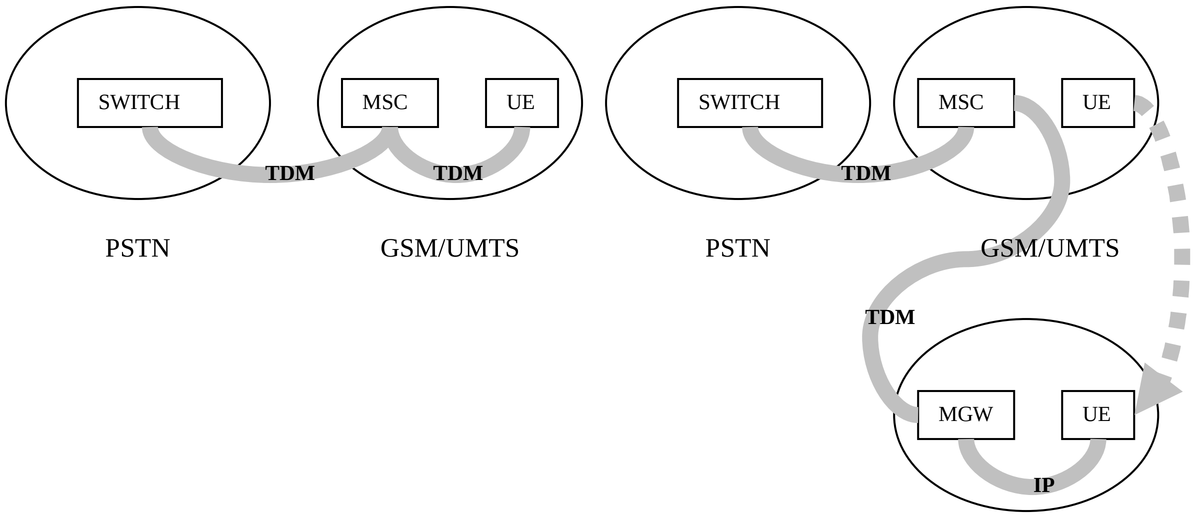
6.8.6 Handover Scenarios

6.8.6.1 CS UE to CS UE call

6.8.6.2 CS UE to IMS UE call

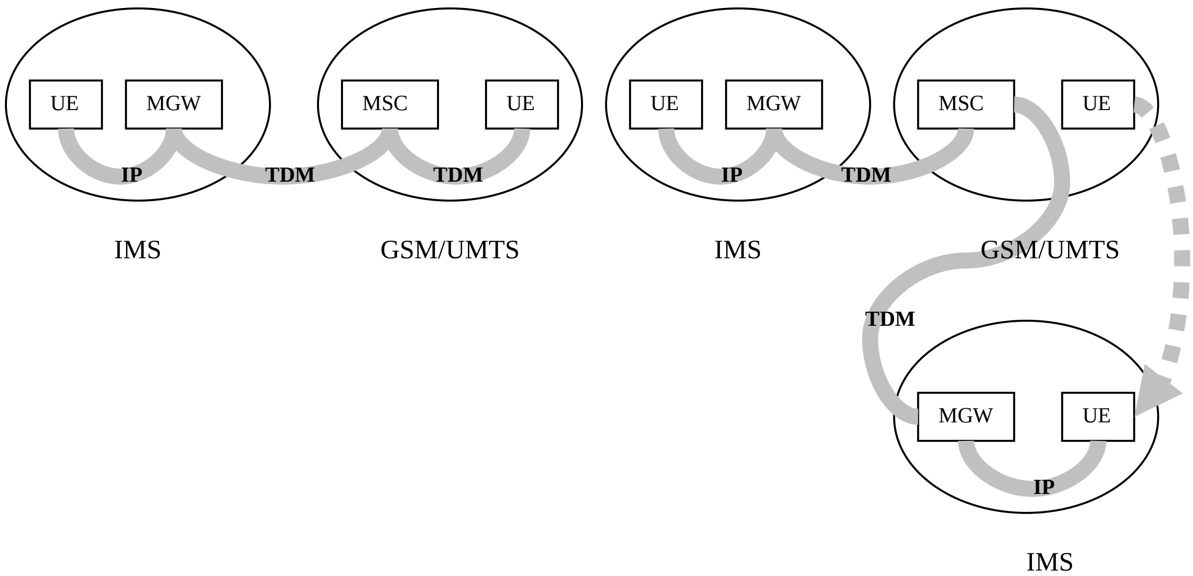
6.8.6.2.1 Bearer Path: Established call with a PSTN subscriber

Before handover: After handover:



6.8.6.2.2 Bearer Path: Established call with GSM or UMTS subscriber

Before handover: After handover:



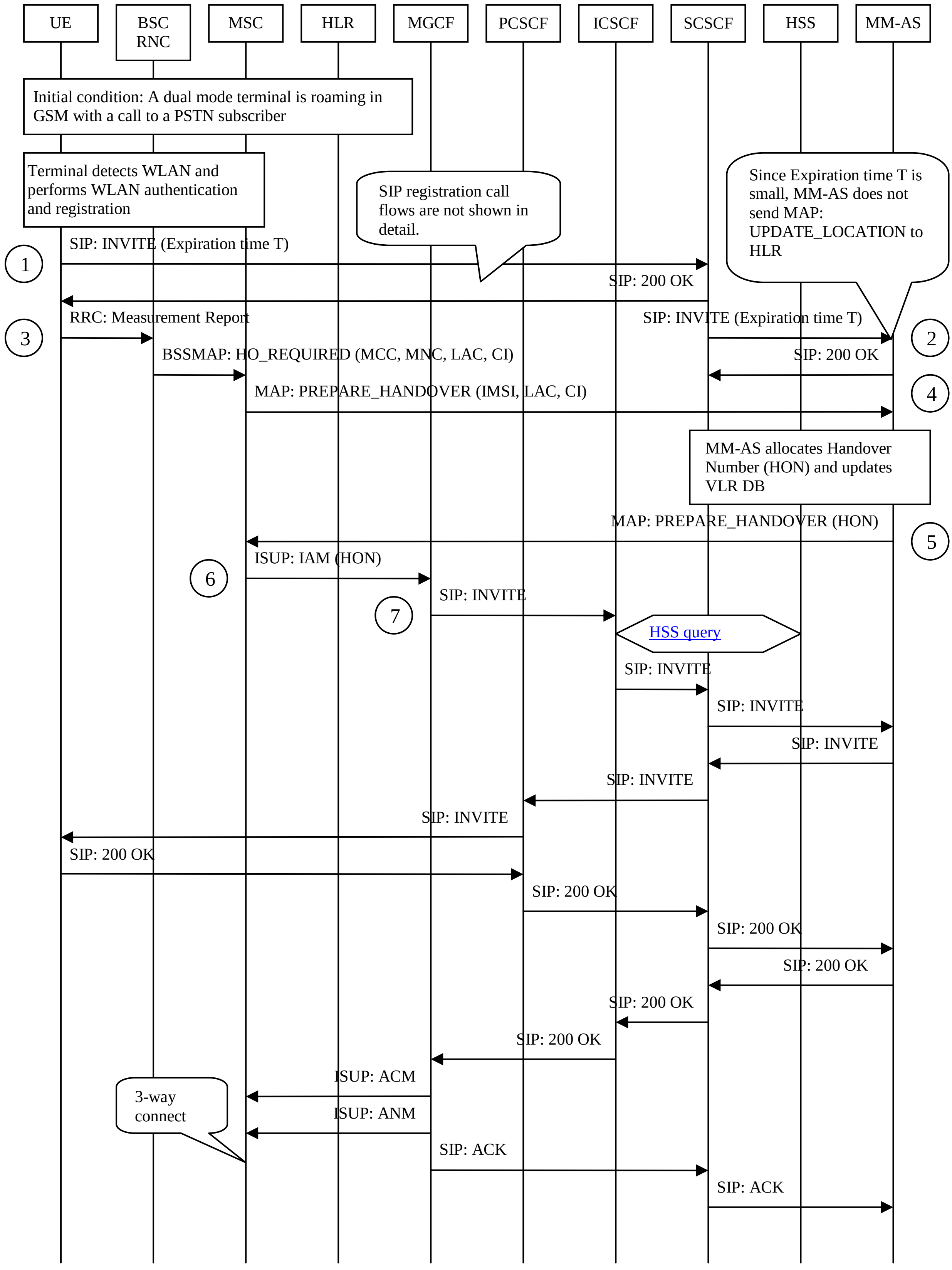
6.3.1.3 Message Flow: From GSM/UMTS to IMS

In this scenario, the dual mode UE has an active call to a PSTN subscriber. Periodically, the UE scans the WLAN signal and when it detects WLAN with sufficient signal strength, it may attempt to attach WLAN.

1. After the UE has performed WLAN access and service authentication, it performs IMS registration. The REGISTER request may contain already an indication to prepare handover. Alternatively, a subsequent SIP message exchange could indicate the handover.
2. The MM-AS is notified about the IMS registration and is informed about the used IP-CAN-CGI. This CGI enables later the MSC to route MAP HO requests and ISUP IAM to the MM-AS.
3. The UE may force handover for the GSM/UMTS call with RRC Measurement Reports indicating the IP-CAN-CGI with the highest signal level. This should trigger the BSC/RNC to request handover to the IP-CAN-CGI by sending an HO Request message to the MSC.

Editor's Note: The neighbour cell list of the BSC/RNC has to be configured in such a way to support a proper handling of the IP-CAN-CGI. enable the correct routing to the MSC.

4. The MSC sends a MAP Prepare Handover Request message to the MM-AS by using the IP-CAN-CGI.
5. The MM-AS returns a MAP Prepare Handover Response containing an E.164 Handover number.
6. The serving MSC establishes an ISUP call to the handover number generated in the previous step. The call is terminated in the MGCF, which routes the SIP INVITE to the I-CSCF based on the SIP URI or Tel URI.
7. Call setup continues similar to the Mobile Terminating Call procedure.
8. After the call is successfully established, the MM-AS sends the MAP: SEND_END_SIGNAL procedure to the MSC that causes the GSM/UMTS resources to be released. Now the call has been handed over to IMS (WLAN).
9. After call is terminated, the UE performs an IMS registration. This would cause the MM-AS to update the location data in the HLR.


UE
BSC RNC
MSC
HLR
MGCF
PCSCF
ICSCF
SCSCF
HSS
MM-AS
Initial condition: A dual mode terminal is roaming in GSM with a call to a PSTN subscriber
Terminal detects WLAN and performs WLAN authentication and registration
SIP registration call flows are not shown in detail.
Since Expiration time T is small, MM-AS does not send MAP: UPDATE_LOCATION to HLR
1
SIP: INVITE (Expiration time T)
SIP: 200 OK
2
SIP: INVITE (Expiration time T)
3
RRC: Measurement Report
SIP: 200 OK
BSSMAP: HO_REQUIRED (MCC, MNC, LAC, CI)
4
MAP: PREPARE_HANDOVER (IMSI, LAC, CI)
MM-AS allocates Handover Number (HON) and updates VLR DB
5
MAP: PREPARE_HANDOVER (HON)
6
ISUP: IAM (HON)
7
SIP: INVITE
HSS query
SIP: INVITE
SIP: INVITE
SIP: INVITE
SIP: INVITE
SIP: INVITE
SIP: 200 OK
SIP: 200 OK
SIP: 200 OK
SIP: 200 OK
SIP: 200 OK
SIP: 200 OK
ISUP: ACM
3-way connect
ISUP: ANM
SIP: ACK
SIP: ACK

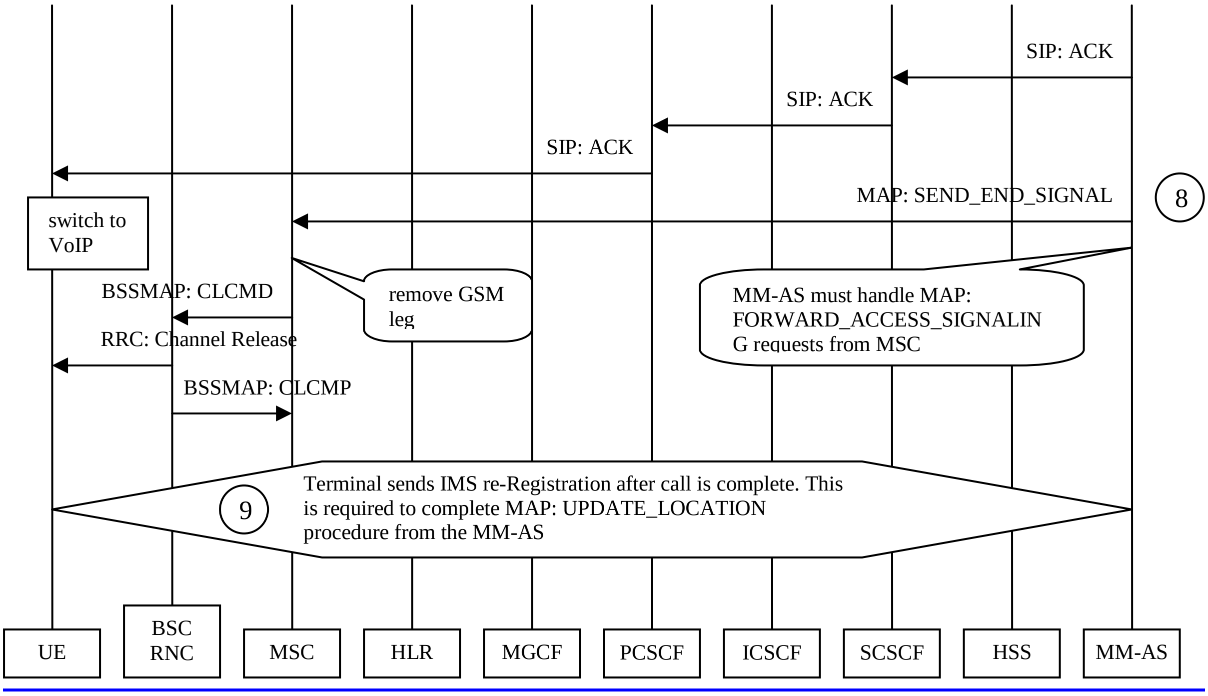


Figure 1: Handover from GSM/UMTS to IMS

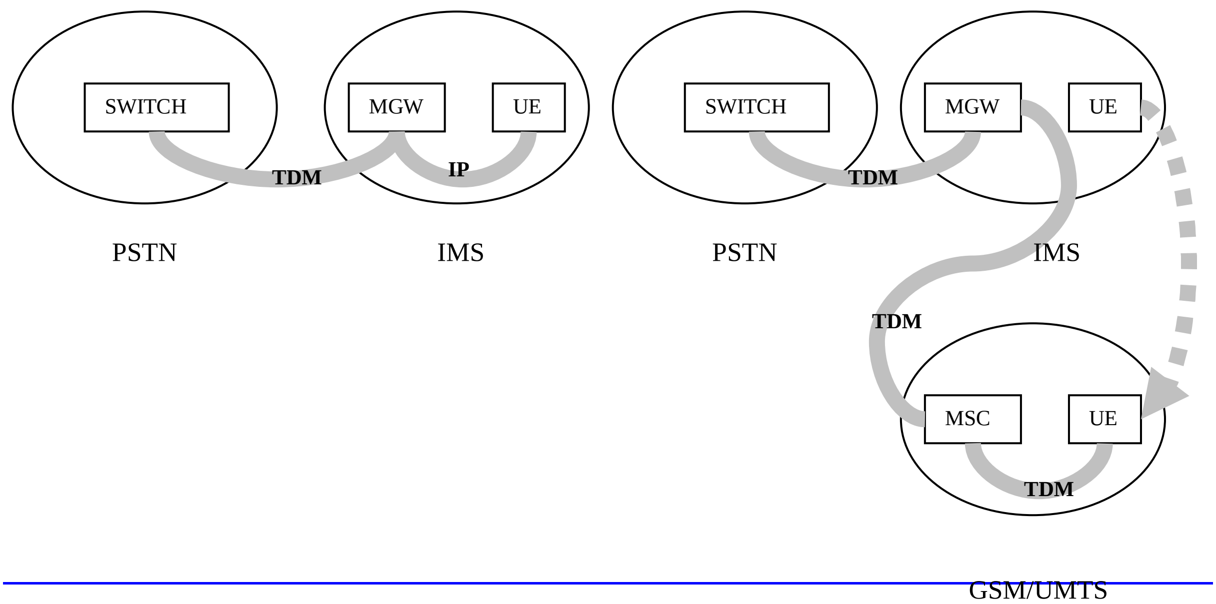
Editor's Note: The message flow does not consider the support of supplementary services.

6.8.6.3 IMS UE to IMS UE call

6.8.6.4 IMS UE to CS UE call

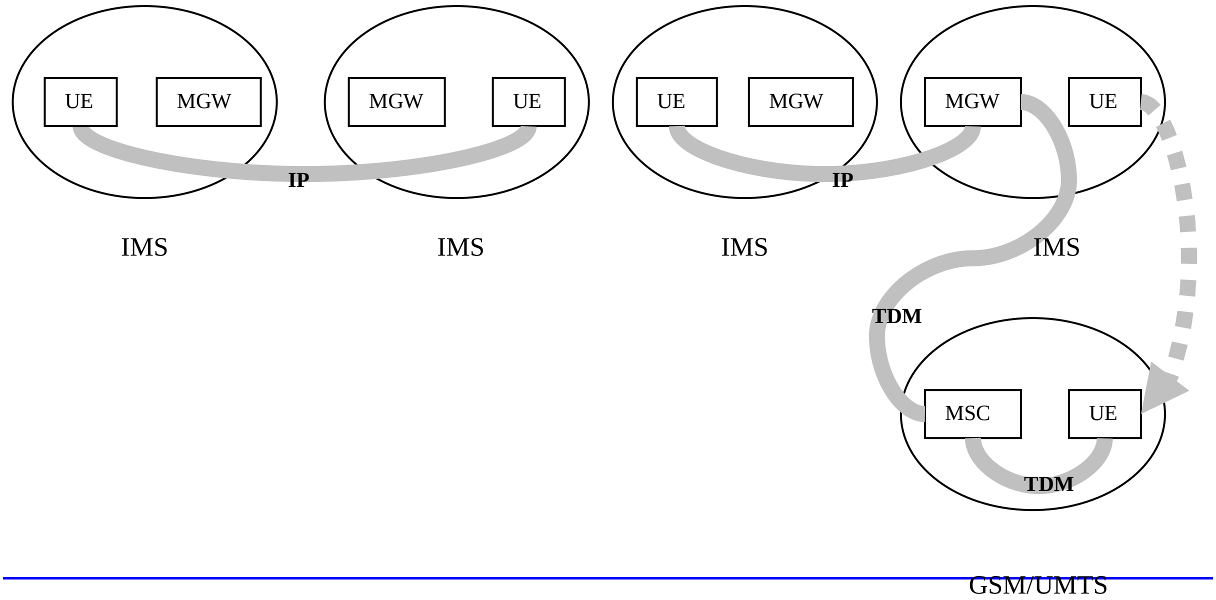
6.8.6.4.1 Bearer Path: Established call with a PSTN subscriber

Before handover: After handover:



6.8.6.4.2 Bearer Path: Established call with a IMS subscriber

Before handover: After handover:



6.8.6.4.3 Message Flow: From IMS to GSM/UMTS

The initial condition is an active call with a PSTN or IMS subscriber, the UE has an active IMS session over I-WLAN. The S-CSCF forwards SIP messages to the MM-AS based on filter criteria:

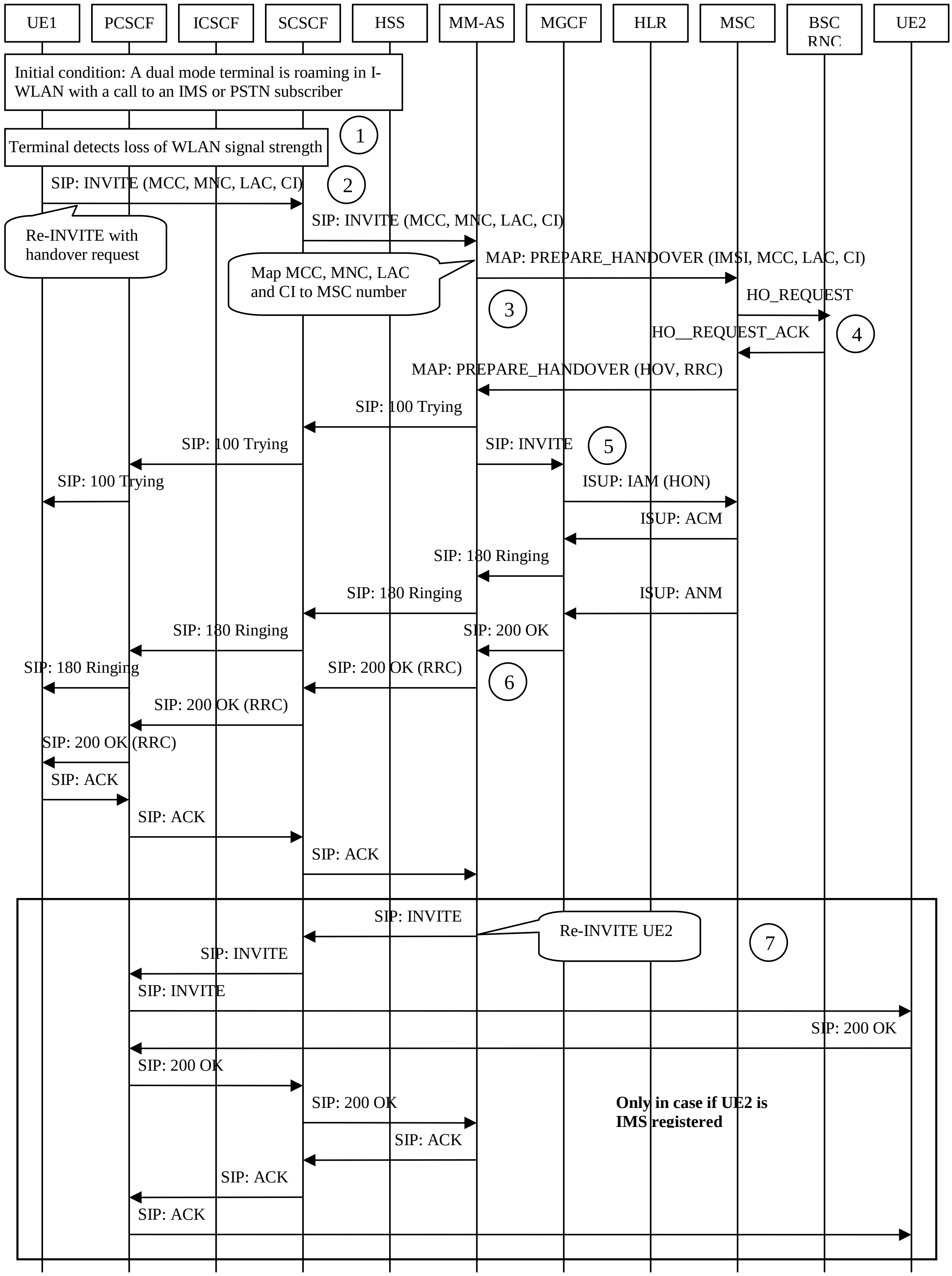
1. The UE periodically polls the WLAN signal strength to decide, if a handover to GSM/UMTS is required.
2. UE requests handover to GSM/UMTS through a re-INVITE in the context of the existing SIP dialog. The re-INVITE contains the desired GSM/UMTS parameters like MCC, MNC, LAC, and CI.

Editor's Note: In UMTS it is FFS how the MM-AS gets the RNC-id.

3. MM-AS sends a MAP: PREPARE_HANDOVER request to the MSC based on the CGI.
4. The MSC sends a HANDOVER_REQUEST to the serving BSC/RNC. After the BSC/RNC acknowledges the request, the MSC responds back to the MM-AS with a Handover number (HON) and the necessary RRC information for the target cell.
5. MM-AS request the MGCF to establish an outgoing ISUP call to the MSC with the Handover Number (HON) by sending a SIP: INVITE. This establishes the bearer towards the target MSC.

Editor's Note: Clarify if the proposed handover scenario is different from existing handover procedures.

6. MM-AS transfers the RRC information to the UE within the SIP: 200 OK.
7. Only in the case if UE2 is registered in IMS: The MM-AS re-INVITEs UE2 to establish a direct UE connection.
8. After ISUP call establishment is completed, the UE tries to access the target GSM/UMTS cell.
9. When handover to GSM/UMTS is complete, the BSC/RNC sends HO-Complete to the MSC. The MSC sends a MAP: Send End Signal to the MM-AS.
10. Only in the case if UE2 is in the PSTN: The MM-AS issues a SIP: REFER (Call Transfer) command to the MGCF to join the two ISUP call legs (with MSC and PSTN). The voice path in the UE will be interrupted for the time needed by the MGCF to perform this operation.
11. The MM-AS releases the IMS connection towards UE1.


UE1
PCSCF
ICSCF
SCSCF
HSS
MM-AS
MGCF
HLR
MSC
BSC RNC
UE2
Initial condition: A dual mode terminal is roaming in I-WLAN with a call to an IMS or PSTN subscriber
Terminal detects loss of WLAN signal strength
1
SIP: INVITE (MCC, MNC, LAC, CI)
2
Re-INVITE with handover request
SIP: INVITE (MCC, MNC, LAC, CI)
Map MCC, MNC, LAC and CI to MSC number
MAP: PREPARE_HANDOVER (IMSI, MCC, LAC, CI)
3
HO_REQUEST
HO__REQUEST_ACK
4
MAP: PREPARE_HANDOVER (HOV, RRC)
SIP: 100 Trying
SIP: 100 Trying
SIP: INVITE
5
SIP: 100 Trying
ISUP: IAM (HON)
ISUP: ACM
SIP: 180 Ringing
SIP: 180 Ringing
ISUP: ANM
SIP: 180 Ringing
SIP: 200 OK
SIP: 180 Ringing
SIP: 200 OK (RRC)
6
SIP: 200 OK (RRC)
SIP: 200 OK (RRC)
SIP: ACK
SIP: ACK
SIP: ACK
SIP: INVITE
Re-INVITE UE2
7
SIP: INVITE
SIP: INVITE
SIP: 200 OK
SIP: 200 OK
SIP: 200 OK
Only in case if UE2 is IMS registered
SIP: ACK
SIP: ACK
SIP: ACK

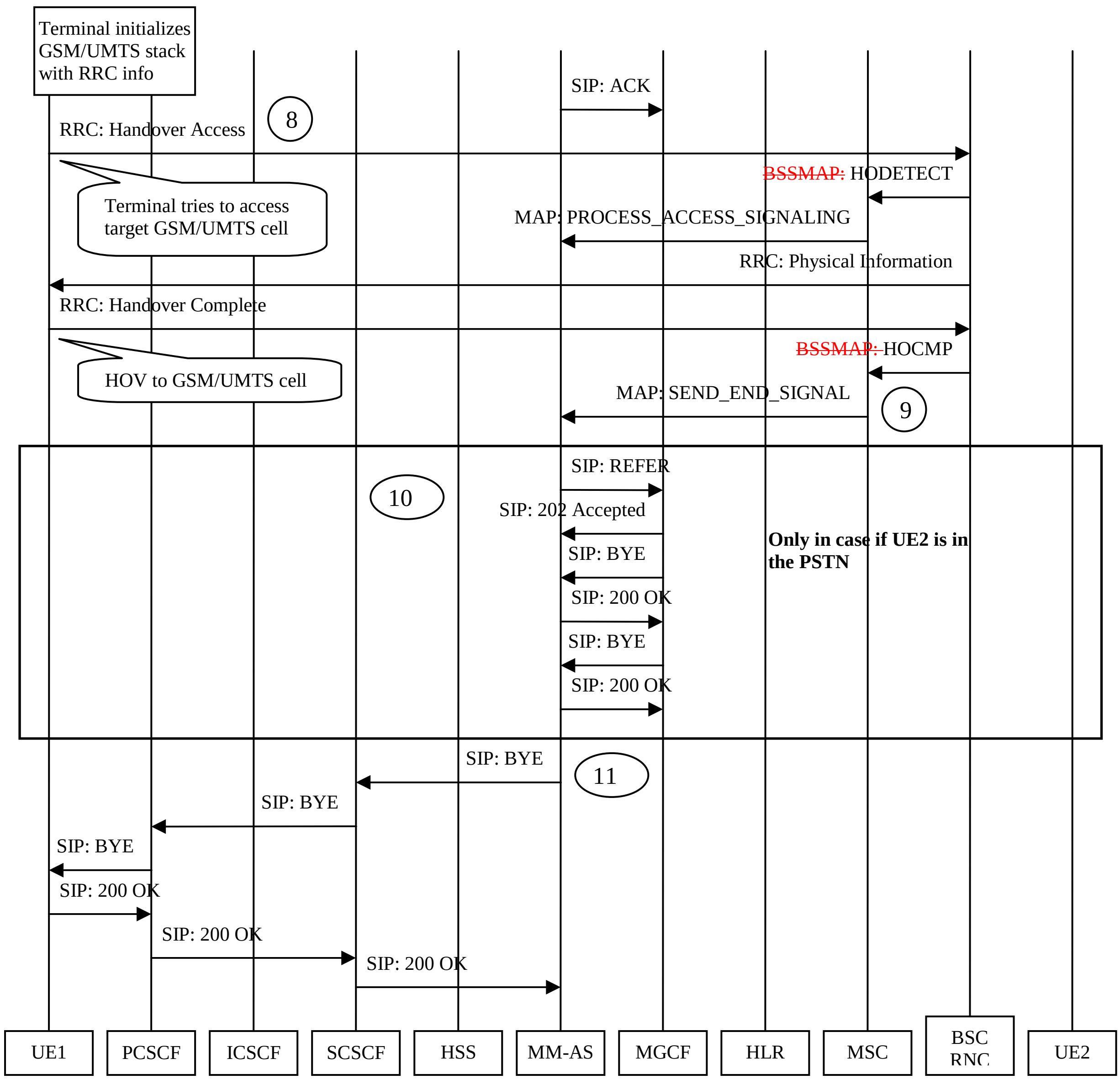


Figure 1: Handover from IMS to GSM/UMTS

6.8.7 Impact on Supplementary Services

6.8.8 Evaluation of the model

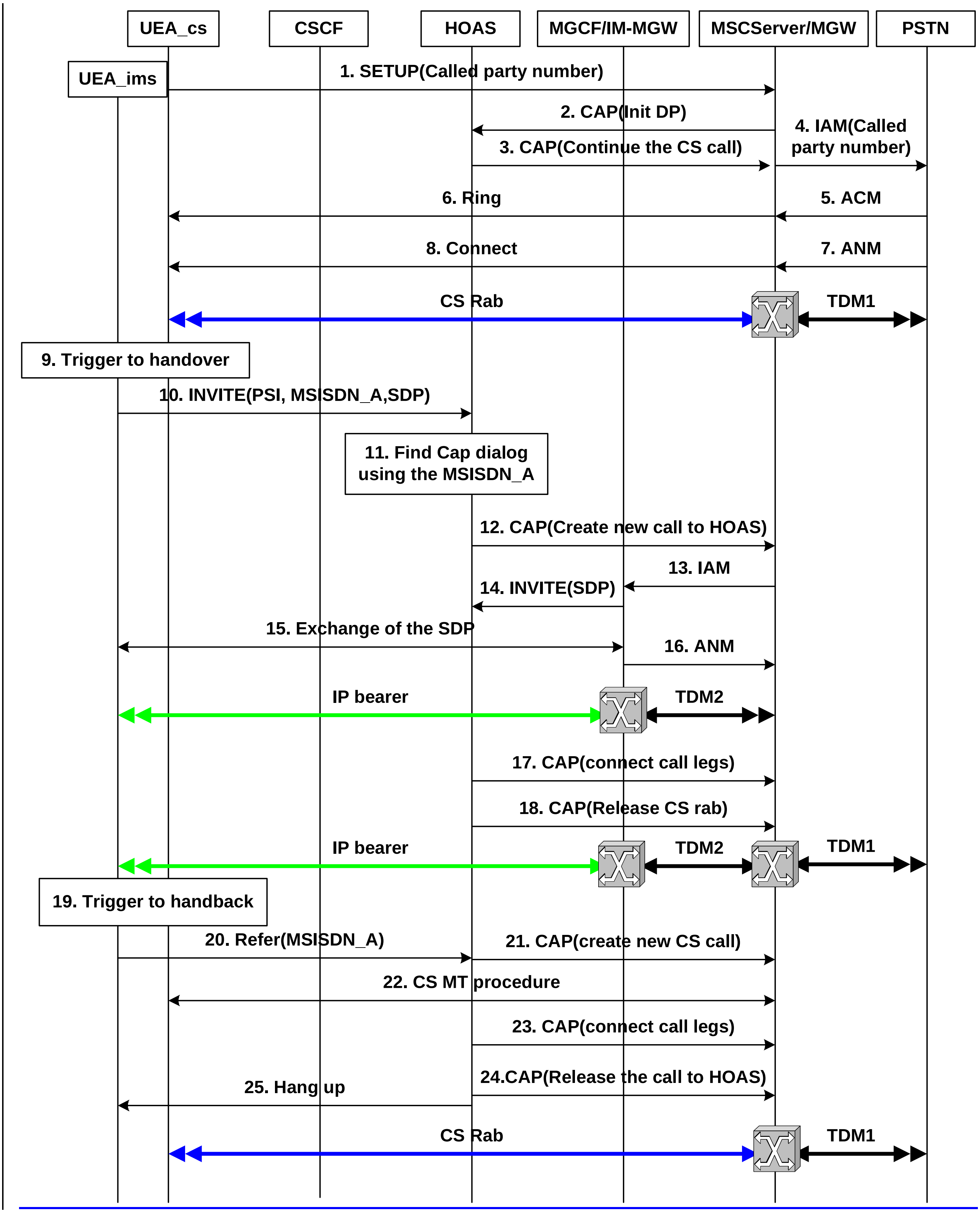
This clause presents the evaluation of the service continuity solution against the set of criteria

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

6.9.1 General Description

This section describes CS-IMS subscriber handover from the CS domain to the IMS. This section focuses on the scenario that a CS-IMS subscriber has an active call in the CS domain to a PSTN subscriber and the CS-IMS subscriber decide to handover from the CS domain to the IMS. The scenario which the called party is a IMS subscriber is very similar with this scenario.

The following steps show the handover procedure from CS domain to IMS.


UEA_cs
CSCF
HOAS
MGCF/IM-MGW
MSCServer/MGW
PSTN
UEA_ims
1. SETUP(Called party number)
2. CAP(Init DP)
3. CAP(Continue the CS call)
4. IAM(Called party number)
5. ACM
6. Ring
7. ANM
8. Connect
CS Rab
TDM1
9. Trigger to handover
10. INVITE(PSI, MSISDN_A,SDP)
11. Find Cap dialog using the MSISDN_A
12. CAP(Create new call to HOAS)
13. IAM
14. INVITE(SDP)
15. Exchange of the SDP
16. ANM
IP bearer
TDM2
17. CAP(connect call legs)
18. CAP(Release CS rab)
IP bearer
TDM2
TDM1
19. Trigger to handback
20. Refer(MSISDN_A)
21. CAP(create new CS call)
22. CS MT procedure
23. CAP(connect call legs)
24.CAP(Release the call to HOAS)
25. Hang up
CS Rab
TDM1

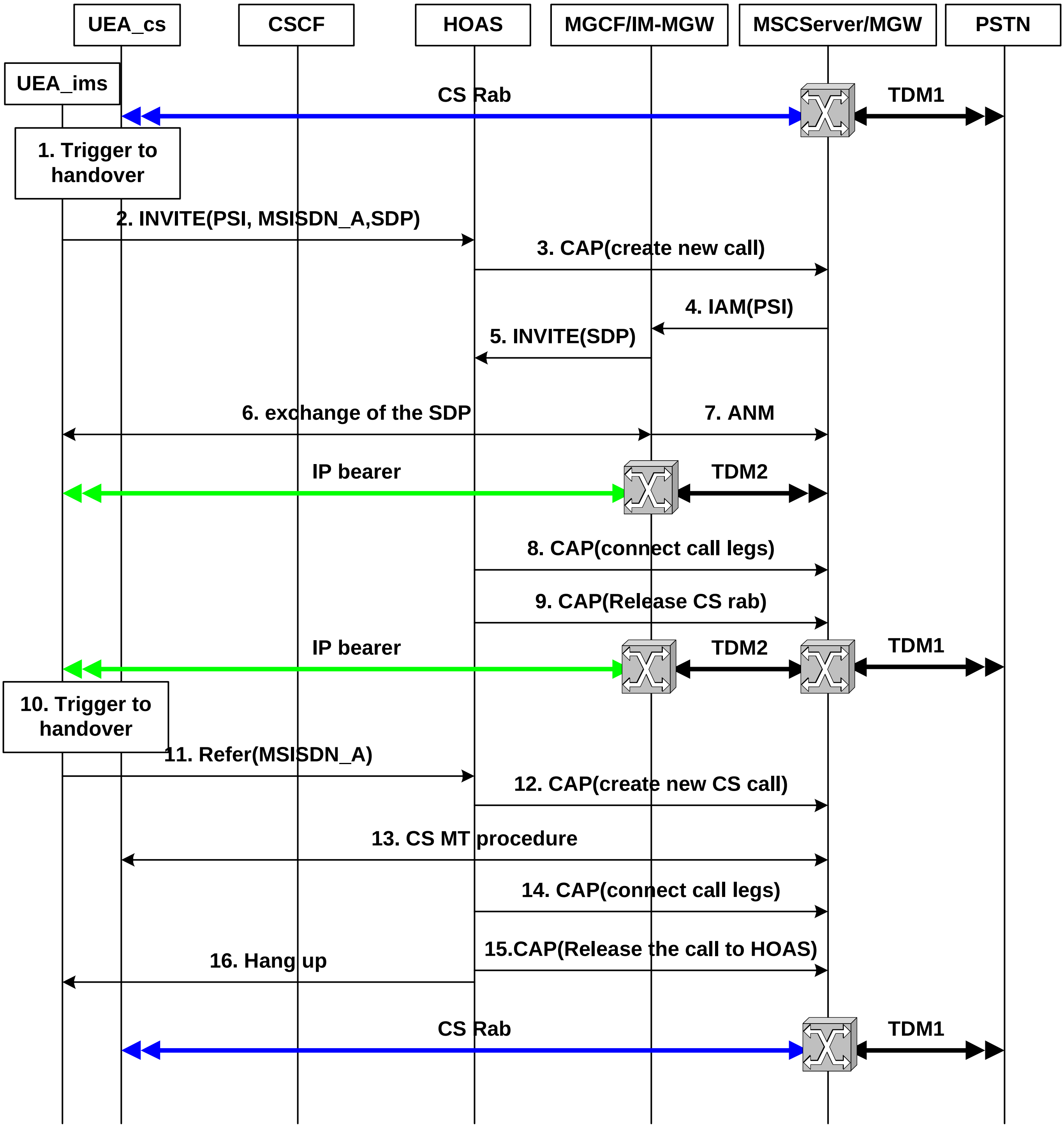


Figure 1

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 dialog to HOAS according to the camel criteria in the MSC Server.

3. The HOAS indicated the MSC Server to continue the CS call to the PSTN user.

4. The MSC Server send a IAM to the PSTN user.

5. The PSTN user send a ACM to the MSC Server to indicate that the PSTN user is ring.

6. After receiving the ACM then the MSC Server send a Ring to the user A.

7. When the PSTN user answer the CS call it shall send ANM to the MSC Server.

8. The MSC Server send a Connect to the user A and connect the bearer paths

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 DTM between the MSC Server/MGW and PSTN, as described in Figure 1. During the CS call procedure the MSC Server establishes a CAP dialog to HOAS.

9. According to the radio condition and/or user preference, the user A decides to handover from CS domain to IMS. User A register in IMS firstly.

   Note: The registration in IM domain may increase the duration of handover. It is FFS how to handle this issue.

10. After the registration, user A send a INVITE request to the IMS core network(PSI of the HOAS, 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 HOAS. The PSI of the HOAS can be automatically transferred in the registration or statically configured in the user A.

11. The HOAS store the initial SDP included in the INVITE request. Using the MSISDN of user A, the HOAS find the CAP dialog established in the CS domain.

12. After find the CAP dialog the HOAS shall indicate the MSC Server through this CAP dialog to create a CS call leg to the HOAS This message include the Tel number of HOAS.

13. Once receive the indication, the MSC select a proper MGCF and send an IAM to this MGCF, the called number is the PSItel number of HOAS, as indicated by the HOAS.

14. After receive the IAM the MGCF shall translate the tel number of HOAS to a tel: URI,which is the PSI of the HOAS. Then the HOAS shall initiate a INVITE request(PSI of the HOAS, MSISDN of user A, SDP). The INVITE request is then routed to the HOAS.

15. When receive the INVITE request, the HOAS shall find the session originated by the user A in step 1 according to the MSISDN of user A. The HOAS shall exchange the SDP between the user A and the MGCF. After the exchange the bearer path between the user A and the IM-MGW controlled by the MGCF is established.

16. After establishment of the bearer path(IP bearer) between the user A and the IM-MGW, the MGCF send a ANM to the MSC Server. So the bearer path(DTM2) between the MSC Server/MGW and IM-MGW is established.

17. The HOAS shall indicate the MSC through the CAP dialog 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

18. The HOAS shall indicate the MSC through the CAP dialog to release the CS connection between the MSC Server/MGW and user A in CS domain. When the user A receive the release message it shall change it's voice channel to IM domain.

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

19. According the radio condition and user preference, user A decide to handback to CS domain.

20. User A will send a REFER to HOAS using the existing session in IMS, containing "Refer-To" the MSISDN of user A.

21. When the HOAS receive the REFER message, the HOAS shall look for whether the CAP dialog of the user A have already existed. If it find the CAP dialog then the HOAS shall indicate through the dialog the MSC to originate a CS call leg to user A in CS domain. If it can not find the CAP dialog it shall reject the handover request.

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

   Note: This step which including the paging procedure may increase the duration of handover. 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 HOAS shall indicate the MSC 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 HOAS shall indicate the MSC Server to release the call leg from MSC Server to HOAS.

25. the HOAS shall release the IMS session from user A to HOAS. 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.

6.9.8 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 handover 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 handover. After the handover from CS domain to IM domain, the IM service in IM domain can also provide to the user. The Camel dialog exist during the whole call so it is possible that the camel logic in HOAS can also control this voice call even after the handover.

The drawbacks of this solution are:

1. The MSC Server must support Camel 4 service.

2. The user which want to handover between the CS domain and IMS must have camel 4 subscription.

7 Security

8 Charging

9 Comparison of the Architecture Model

10 Conclusion

Annex A (informative)

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 handover procedure was described and discussed based on it.

A.1.1 Logical Model Introduction

For the flexibilities from IMS network, handover 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 handover is proposed shown in the figure 1.

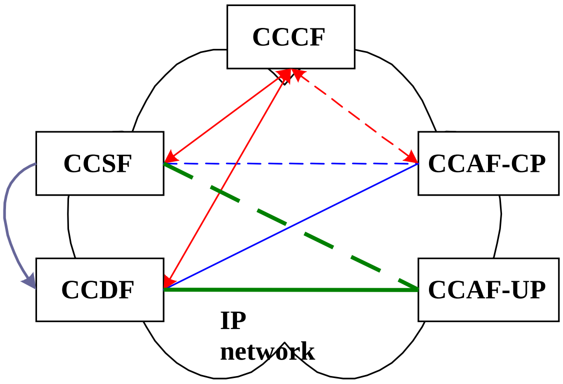


Figure 1: system model

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 handover, 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 handover user establishes session to the remote UE side after handover, 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 handover.

CCAF (Call Continuity Anchor Function): The CCAF acts as an anchor in handover procedure. It establishes the connection with CCSF before handover and with CCDF after handover. 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 handover procedure, establishes control connection at control plane with CCSF and CCDF separately before/after handover. During the course of handover, 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 handover procedure, establishes media connection at user plane with CCSF and CCDF separately before/after handover. During the course of handover, 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 handover request
- Allocating and Controlling CCAF-CP

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

The figure 2 and 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 figure 2 and 3.

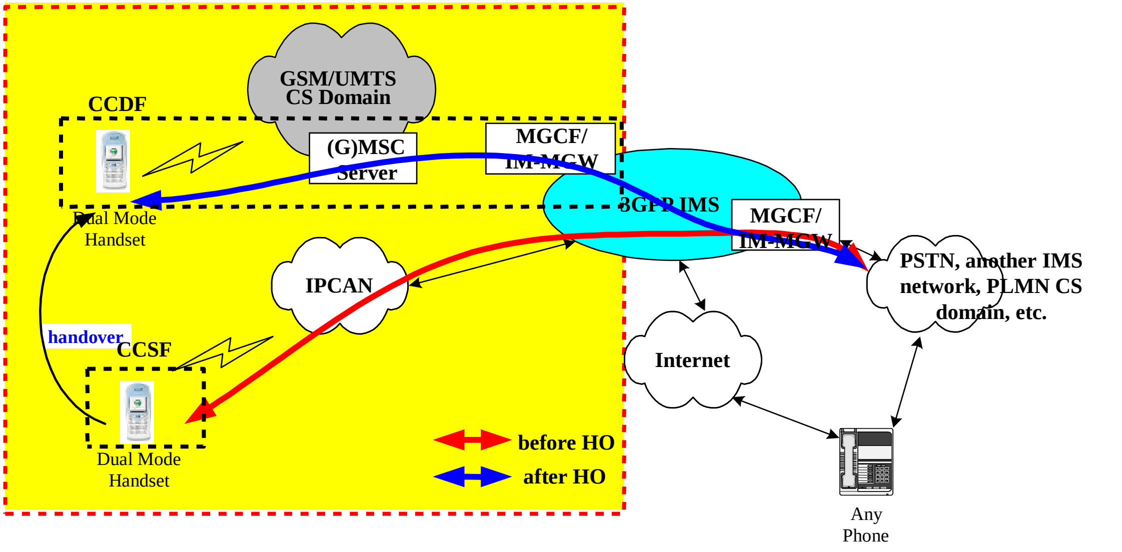


Figure 2: handover from IMS-controlled VoIP call and CS call

(Notes: if the remote user is an IMS user, the MGCF/IM-MGW is not needed in the figure)

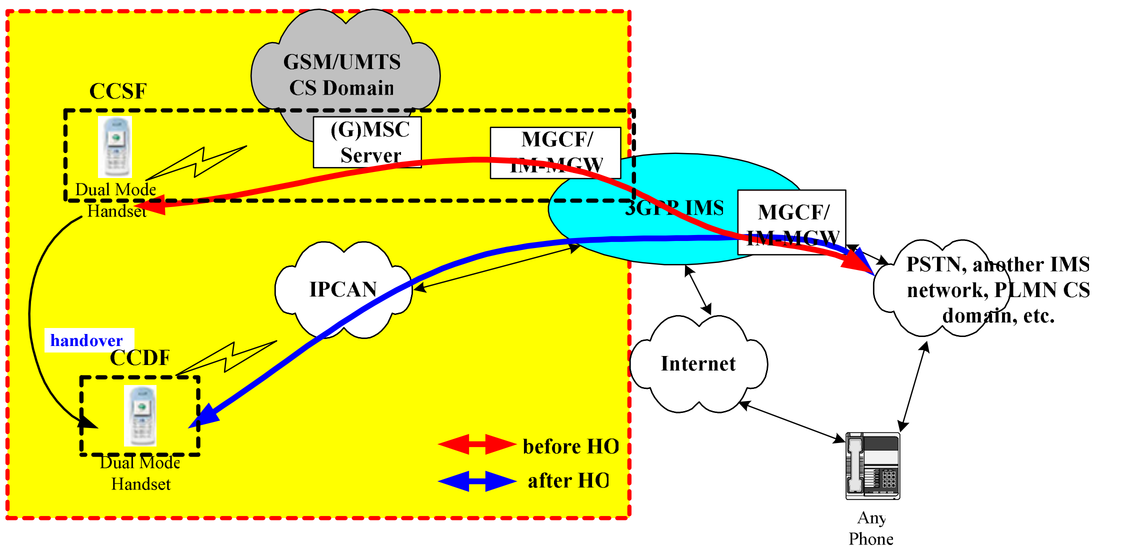


Figure 3: handover from CS call to IMS-controlled VoIP call

(Notes: if the remote user is an IMS user, the MGCF/IM-MGW is not needed in the figure)

Notes: 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 handover.

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 CP&UP-segmented (control-plane & user-plane segmented) mode

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

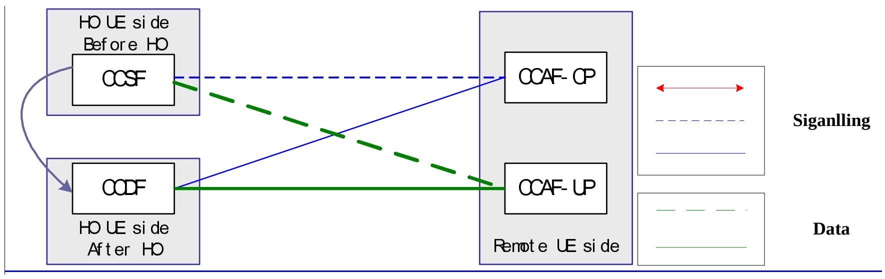


Figure 4: Terminal-controlled Mode

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.
- Handover procedure, including handover detection, handover initiation and handover execution.
  - ✓ Handover detection is executed by CCSF
  - ✓ Handover initiation: CCSF indicates to CCDF or CCAF to perform handover
  - ✓ Handover execution: either CCDF or CCAF may be the master controller of handover, 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 handover 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

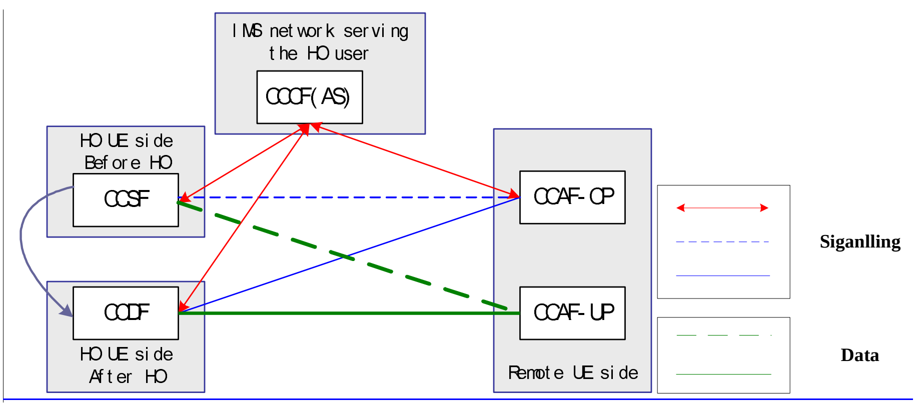


Figure 5: Network-controlled Mode

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.
- Handover procedure, including handover detection, handover authentication, handover initiation and handover execution.
  - ✓ Handover detection is executed by CCSF
  - ✓ Handover authentication: CCSF sends handover request to CCCF and CCCF authenticates the request.
  - ✓ Handover initiation: After completion of handover authentication, CCCF may either indicate directly to CCAF/CCDF to perform handover, or return authentication acknowledgement to CCSF and then CCSF sends handover indication.
  - ✓ Handover 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 handover 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

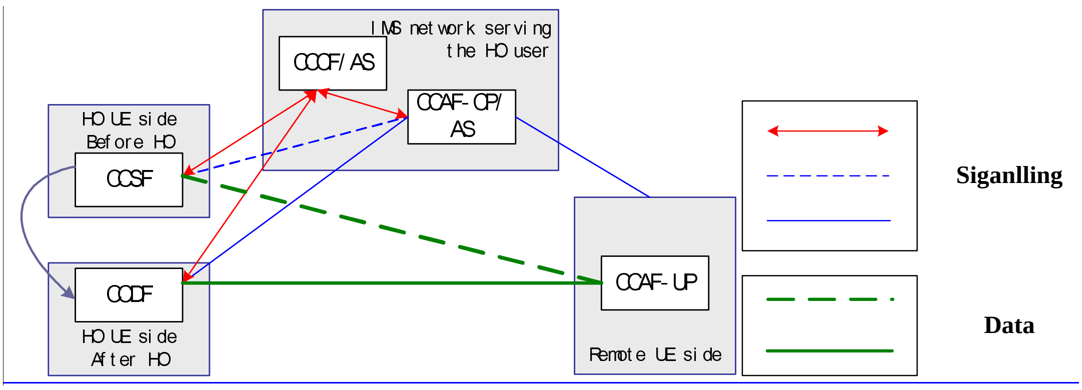


Figure 6: CP-segmented Mode

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.
- Handover procedure, including handover detection, handover authentication, handover initiation and handover execution.
  - ✓ Handover detection, handover authentication, handover initiation and handover execution procedure is the same as that in End-to-End/Network-controlled mode.
  - ✓ During the handover 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 handover

Pros.

i During handover 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 CP&UP-Segmented Mode

Cons.

i. More complicated and 3PCC capability need to be supported by AS comparing with End-to-end Mode

4. CP&UP-segmented Mode

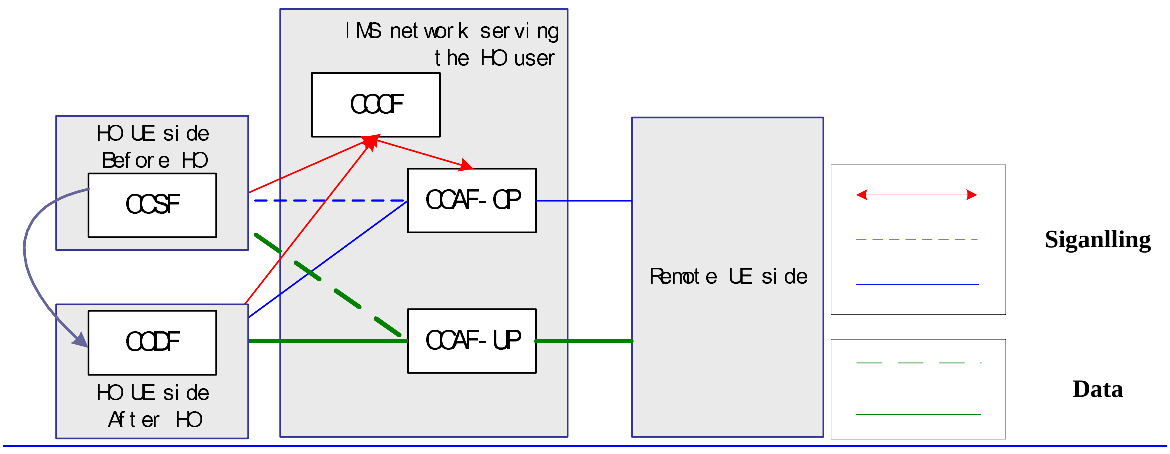


Figure 7: CP&UP-segmented Mode

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.
- Handover procedure, including handover detection, handover authentication, handover initiation and handover execution.
  - ✓ Handover detection, handover authentication, handover initiation and handover 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 handover 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 handover..

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 handover 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 handover 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 handover from IMS to CS, or directly from IMS in case of handover 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 handover:

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 handover 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 handover:

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 handover 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 handover 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.

7 Conclusion and Recommendations

Annex <X>: Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2005-04					*First version created*	*0.0.0*	*0.0.1*
2005-04	*SA2#45*				*Agreed documents from SA2#45 incorporated*	*0.0.1*	*0.1.0*
2005-05	*SA2#46*				*Agreed documents from SA2#46 incorporated*	*0.1.0*	*0.2.0*
2005-05					*Editorial updates based on email comments after SA2#46*	*0.2.0*	*0.2.1*