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GLOBAL SYSTEM FOR
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Keywords

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3GPP

Postal address

3GPP support office address

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

Internet

http://www.3gpp.org

Contents

Foreword

This Technical Report has been produced by the 3rd Generation Partnership Project (3GPP).

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version x.y.z

where:

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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

Development of the architecture for Voice Call Continuity has identified that supporting domain transfer of active mid-call services by implementing these services in both the CS domain and IMS is not a viable solution in the 3GPP Rel-7 timeframe. Therefore it has been proposed that an architecture is necessary that allows implementation of such services in IMS while also allowing control of the services when the serving access network is the CS domain. In addition to the VCC scenario, the increased deployment of VoIP capable access technologies will encourage further service development on IMS also increasing the importance of being able to access these services via CS domain access independently of the support of VCC.

1 Scope

1.1 General

This document contains the results of the feasibility study into the architectural requirements and alternatives for the delivery of consistent services to the user mainly via IMS centralized services regardless of the attached access type; e.g. CS domain access or IP-CAN. Considerations include overall requirements, architectural requirements, alternative architectures and evaluation of potential architectural solutions.

The study shall consider how to access the IMS-based multimedia telephony services while still allowing innovative services. Specifically, it shall provide consideration for the handling of the multiple media types that are enabled by the IMS multimedia telephony communication service (MMTel) utilizing CS services of TS11 and CS video telephony reliant upon BS30. Emergency calls that utilise TS12 are outside the scope of this work item in this release.

It shall include an investigation into call/session establishment via CS domain access and IP-CAN and for calls/sessions transferred across CS domain access and IP-CAN, including the interactions with domain selection. The solution should be applicable for terminals with or without VCC capabilities. Impact on legacy terminals with the same subscription (e.g. SIM swapping) should be studied.

The study shall also investigate the means to support and the need of the evolution of a network towards the IMS centralized services architecture. The assumption for this evolution is that some networks may not immediately migrate all services to the IMS centralized services architecture. In addition, given that some calls may not be rerouted to IMS during the migratory period, the study shall investigate how to ensure that equivalent services are implemented in IMS and the CS domain.

Overall, Centralized IMS Services Control supports the introduction of MMTel by enabling support of IMS bi-directional speech media services when no bi-directional speech media capable IP-CAN is useable. Combined with the domain transfer capability of VCC, service continuity for bi-directional speech media is enabled for MMTel between a bi-directional speech media capable IP-CAN and CS access. Furthermore it supports the introduction of MMTel by enabling terminals without IMS to support receiving IMS-based speech services.

1.2 Motivation and background

Communication networks are evolving towards packet-based infrastructures. A single common consolidated core network offers service providers the possibility of reduced core network complexity and maintenance. As service providers shift their core network infrastructure from the CS domain to a consolidated common IMS infrastructure the need will exist to enable the consistent provision of services to subscribers over a variety of accesses, including CS domain and PS domain accesses.

Initially it can be expected that the coverage of IP-CANs capable of transporting bi-directional speech media will be limited compared to CS domain access networks at least during the introduction period of IP-CANs capable of transporting bi-directional speech media. Therefore a need exists to specify an architecture that supports the provision of IMS based services across a variety of PS domain or CS domain access networks (see figure 1.2-1). This thereby enables a consistent user-experience with bi-directional speech services of IMS subscribers irrespective when being inside or outside the coverage of an IP-CAN capable of transporting bi-directional speech media.

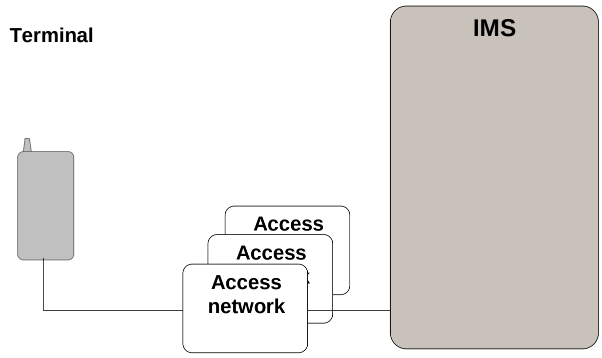


Figure 1.2-1: IMS based services across a variety of PS and/or CS domain access networks

In order to take advantage of all the new capabilities IMS has to offer, it is assumed that new terminals are required. However, there is also an interest to analyze whether IMS bi-directional speech services can be supported to terminals without any IMS capabilities in order to enable centralization of service management in IMS.

Already TR 23.806 [8], introduced the concept of IMS controlled service continuity in clause 6.3 "Service Continuity Model: IMS Controlled Alternative". The Voice Call Continuity (VCC) Application is described in TS 23.206 [3],which implements a distributed service model where both CS and IMS services can be used by a VCC subscriber as follows:

- When using a PS domain access, only IMS services are offered.

- When using a CS domain access, call control and supplementary services are active in IMS with the exception of mid call and presentation services, which are active in the CS domain.

Moreover, the development of the architecture for Voice Call Continuity has identified that supporting domain transfer of active mid-call services by implementing such services in both the CS domain and IMS is not a viable solution in the 3GPP Rel-7 timeframe. When mid-call services are active, the domain transfer cannot be executed unless the mid-call service is finalized first.

Therefore it is proposed to develop an architecture that allows implementation of such services in IMS while also allowing IMS control when the serving access network is in the CS domain. In addition to the VCC scenario, the increased deployment of bi-directional speech media capable access technologies will encourage further service development on IMS. This therefore increases the importance of being able to access these services via CS domain access independently of the support of VCC and also independently of having an IMS enabled terminal. The solution for centralized IMS service control should also support call independent IMS supplementary services management.

Different access network scenarios for centralized IMS service control can be envisioned as depicted in Figure 1.2-2:

- Scenario A: The serving access network is an IP-CAN capable of transporting bi-directional speech media. Here both media transport and session control signalling is carried over the IP CAN.

- Scenario B: The serving access network is a CS access: Here both media transport and session control signalling is carried over CS domain access.

- Scenario C: Both CS domain access and an IP-CAN are serving access networks (possibly, it has been determined that the IP-CAN is not bi-directional speech media capable or operator policy/user preference prevent usage of IP-CAN or usage of IP-CAN for bi-directional speech media communications): Here, media transport is carried over CS access and session and media control signalling is carried either over the CS domain access or over the IP-CAN not capable of transporting bi-directional speech media.

NOTE: If CS access is used for media transport, CS signalling according to TS 24.008 [7] is always used to setup CS bi-directional speech media e.g. TS11 call.

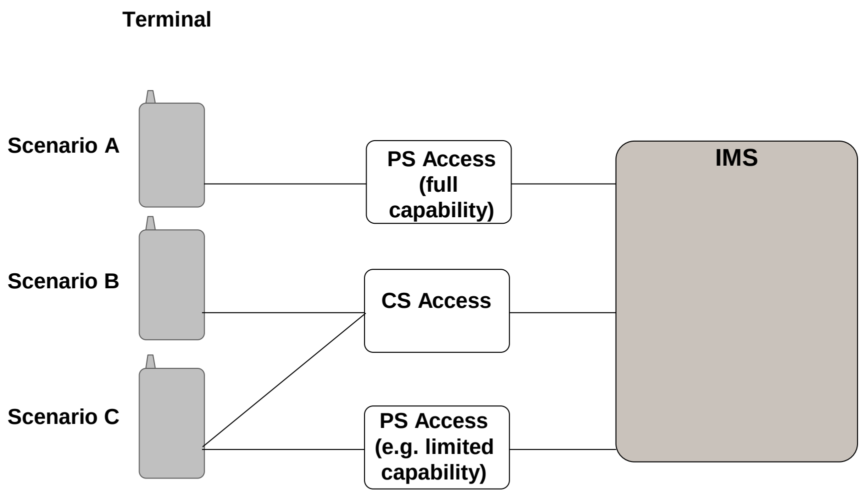


Figure 1.2-2: Scenarios for Centralized IMS Service Control

Within the study of Centralized IMS Service Control both Scenario B and Scenario C need to be investigated for bi-directional speech services. Scenario A shall not be impacted by the study of Centralized IMS Service Control. In Scenario C, depending on operator policy/user preference, access and terminal capabilities, additional IMS controlled media (e.g. video, text) can be carried over an IP-CAN that is not capable of transporting bi-directional speech media.

For the above-introduced scenarios, the following main cases for session continuity for bi-directional speech can be distinguished (see also figure 1.2-3 and figure 1.2-4), both requiring domain transfer for bi-directional speech between bi-directional speech media capable IP-CAN and CS access (Voice Call Continuity as specified in TS 23.206 [3]).

Service Continuity Scenario 1: Access Scenarios A & B:

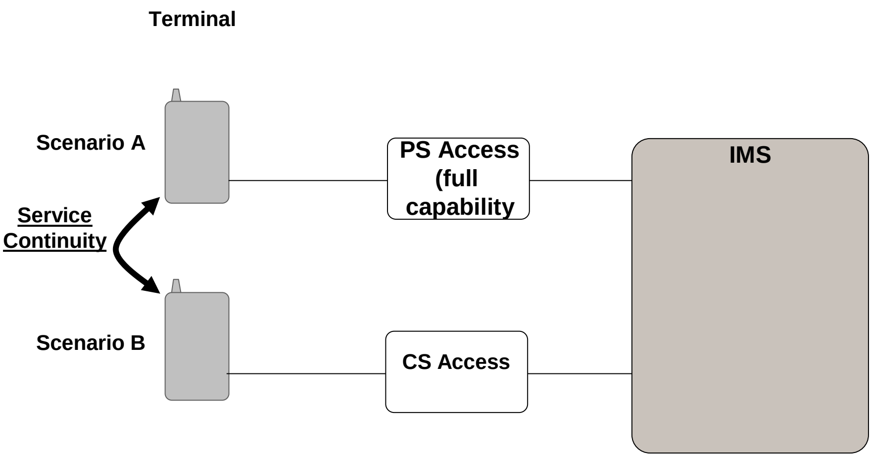


Figure 1.2-3: Service continuity between scenario A and B

Service Continuity Scenario 2: Access Scenarios A & C

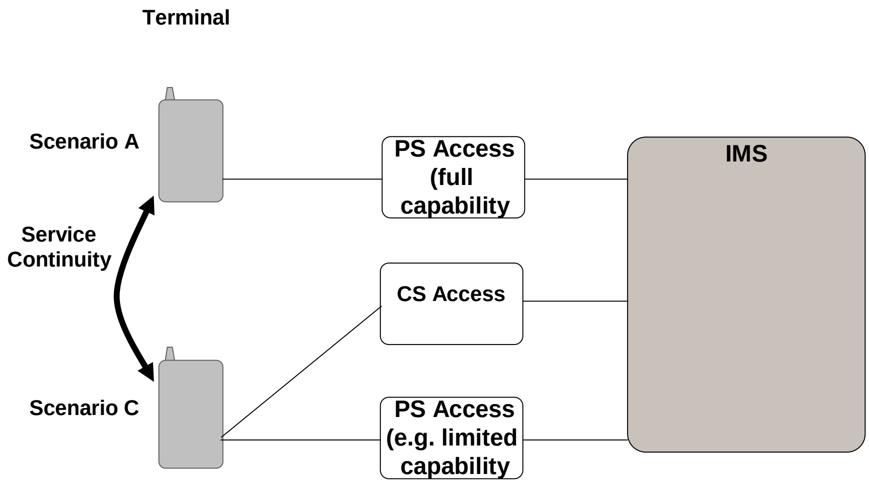


Figure 1.2-4: Service continuity between scenario A and C

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 TS 22.101: "Service principles".

[2] 3GPP TS 22.173: "IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services".

[3] 3GPP TS 23.206: "Voice Call Continuity between CS and IMS".

[4] 3GPP TS 23.228: "IP Multimedia Subsystem (IMS)".

[5] 3GPP TS 24.173: "IMS Multimedia telephony service and supplementary services".

[6] 3GPP TS 24.229: "Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP)".

[7] 3GPP TS 24.008: "Mobile radio interface Layer 3 specification; Core network protocols".

[8] 3GPP TR 23.806: "Voice call continuity between Circuit Switched (CS) and IP Multimedia Subsystem (IMS) Study".

[9] 3GPP TR 23.826: "Feasibility study on Voice Call Continuity (VCC) support for emergency calls".

[10] 3GPP TS 25.413: "UTRAN Iu interface Radio Access Network Application Part (RANAP) signalling".

[11] 3GPP TS 23.009: "Handover procedures".

[12] 3GPP TS 43.055: "Dual Transfer Mode (DTM)".

[13] ITU-T Recommendation H.248.18 (11/2002): "Gateway control protocol: Package for support of multiple profiles".

[14] 3GPP TS 29.232: "Media Gateway Controller (MGC) - Media Gateway (MGW) interface; Stage 3".

[15] 3GPP TS 29.332: "Media Gateway Control Function (MGCF) - IM Media Gateway (IM-MGW); Mn interface".

[16] IETF RFC 3680 (March 2004): "A Session Initiation Protocol (SIP) Event Package for Registrations".

[17] 3GPP TS 23.204: "Support of Short Message Service (SMS) over generic 3GPP Internet Protocol (IP) access".

[18] 3GPP TS 29.163: "Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks".

[19] 3GPP TS 23.893: "Feasibility study on multimedia session continuity".

[20] 3GPP TS 33.210: "3G Security; Network Domain Security; IP network layer security".

[21] 3GPP TS 23.292: "IP Multimedia Subsystem (IMS) Centralized Services; Stage 2".

3 Definitions and abbreviations

3.1 Definitions

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

Bearer Control Signalling Path: Standard CS signalling path used to control the call established to setup CS voice bearer between the UE and IMS.

CS Access Signalling: Standard CS signalling used between the UE and the CS network.

ICS UE: The ICS UE is a User Equipment that contains the new UE capabilities defined in this document and that is capable of receiving telephony services and other services offered by IMS while the voice bearer is established via CS. An ICS UE can also be a UE which can access IMS via an IP-CAN that supports the full duplex speech component of the IMS multimedia telephony service, and follows the procedure defined in TS 22.101 [1], TS 22.173 [2], TS 23.228 [4], TS 24.229 [6] and TS 24.173 [5]. An ICS UE is not necessarily capable of VCC.

ICS Enhanced MSC Server: An MSC Server which has been enhanced with ICS capability.

Non-ICS UE: The non-ICS UE is a User Equipment that does not contain the new capabilities defined in this document. A non-ICS UE is not necessarily capable of VCC.

RUA Leg: The call leg between the ICCF and the remote end. It is formed at the ICCF for presentation of the SIP UA behaviour to IMS on behalf of the UE. The TAS, VCC AS and other Application Servers are invoked on the RUA Leg.

Session Control Signalling Path: Signalling path established between the UE and the ICCF, either directly via PS through an IP-CAN or via CS network elements such as the VMSC and the HSS for enablement of IMS control of user sessions at the ICCF when using CS voice bearers.

Session Transfer Identifier: Session Transfer Identifier is an identifier used by Domain Transfer Function to identify the anchored sessions destined to or originated from a specific UE. It is used by the UE or the network to identify the specific anchored session to perform Domain Transfer.

UE Leg: The call leg between the ICCF and the UE. It is formed at the ICCF by combining of the CS call established between the UE and the ICCF to set up the voice bearer, with the control information communicated between the UE and ICCF to enable the completion of the call towards the remote end.

3.2 Abbreviations

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

CAAF	CS Access Adaptation Function
ICCC	IMS CS Control Channel
I1-cs	IMS CS Control Channel implemented over the CS domain
I1-ps	IMS CS Control Channel implemented over the PS domain
ICS	IMS Centralized Services
ICCP	IMS CS Control Protocol
ICCF	IMS CS Control Function
IMSC	ICS Enhanced MSC Server
L-CAAF	Local CS Access Adaptation Function
L-CAAF-n	Local CS Access Adaptation Function-network equivalent
RUA	Remote User Agent
R-CAAF	Remote CS Access Adaptation Function
VCC	Voice Call Continuity

4 Overall requirements

5 Architectural requirements and considerations

5.1 Basic assumptions

5.1.1 Domain Transfer principles

Domain Transfer procedures are based on the following principles:

- For IMS sessions established over PS access, standard IMS procedures are used. Multiple sessions exist for multiple calls; e.g. two sessions with two different media exists. All of the user sessions are anchored at the DTF in the VCC Application, with one Access Leg and one Remote Leg per session established for all user sessions at the DTF.

- For IMS sessions with bi-directional speech media established over CS access the RUA in the ICCF presents user sessions to IMS as standard IMS sessions after combining with CS media. Multiple sessions exist for multiple calls, but only one CS bearer is used for these sessions; i.e. only one session is active over the CS bearer. All of the user sessions are anchored at the DTF in the VCC Application, with one Access Leg and one Remote Leg per session established for all user sessions at the DTF.

- The Access Legs associated with all the sessions present in the transferring-out domain at the time of Domain Transfer are updated at the DTF in the VCC Application with information from the transferring-in domain. This is required for proper control of current sessions after Domain Transfer and to ensure that the control of these sessions is consistent with any new sessions established after Domain Transfer.

- At most one active session and zero or more inactive sessions with bi-directional speech media is transferred as part of Domain Transfer.

- Domain Transfers of UE-based conferencing is not supported.

5.2 Architectural requirements

5.2.1 Service consistency

The following requirements are defined to ensure service consistency:

- It shall be possible to provide the services offered by the Telephony Application Server to the users who are accessing the network via the CS domain or via an IP-CAN. Support of MMTel full duplex speech service set shall be provided as defined in TS 22.173 [2] and TS 24.173 [5].
  - Current definition of the multimedia telephony service offered by the TAS shall not be changed due to the centralization of services in IMS.
  - Home IMS network services shall be provided when using CS access for speech media transport in home and roaming networks; however a reduced service offering may be provided subject to the constraints of the access network.
  - Home IMS network services shall not be impacted by the solution for Centralized IMS Service Control when using an IP-CAN for speech media transport in home and roaming networks.
  - The solution shall support call independent IMS supplementary services management for users who are accessing the network via the CS domain or via an IP-CAN.
  - Solution needs to work also over international transit networks.

Editors Note 1: The capabilities of the international transit networks need to be identified.

- It shall be possible to support Emergency Call and Priority Services (ETS) for IMS Centralized Services users.

Emergency calls specifically using TS12, SMS, CBS, Fax, Data, Voice Group Call Service and eMLPP/priority services are not centralized in Release 8 in IMS. Provision of SMS in IMS is specified in 23.204 [17].

5.2.2 Core network requirements

5.2.2.1 IMS Core Network

- Impacts on IMS entities should be minimized.

5.2.2.2 CS Core Network

- Impacts on CS network elements should be minimized.

5.2.3 Use of VCC Application Domain Transfer with ICS

Domain Transfer functions of TS 23.206 [3] shall be used to enable service continuity of IMS sessions between PS and CS access for sessions established by ICS UEs capable of VCC.

The Rel-7 VCC Domain Transfer procedures may require enhancements for support of transfers of single sessions between CS and PS access. Further enhancements to these procedures are required for support of Domain Transfer of multiple sessions.

5.2.4 Access domain selection

Access domain selection shall be performed for ICS.

5.2.5 Redirection of CS calls to IMS

Redirection of CS calls to IMS should be possible.

5.3 UE requirements

5.3.1 Support of ICS UEs

Subject to operator policy, the ICS UE shall perform the IMS SIP registration procedure as defined in TS 23.228 [4].

Editor's note: There are no agreed assumptions on how ICS will be rolled out and deployed in networks and whether the feature will be subject to a specific subscription. The operation of an ICS UE in a network where some users have an ICS subscription and others do not is FFS.

5.3.2 Support of Pre-Rel-08 non ICS Enhanced UEs

Pre-Release 8 UEs (without IMS Centralized Services enhancements) should be supported. When services are provided by the IMS, a reduced supplementary service set may be allowed.

Editor's Note: The impact of providing IMS Centralized Services for a non-ICS enhanced UE e.g. in terms of service data synchronization requirements, needs to be studied.

5.4 Service continuity requirements

- The IMS Centralized Services solution is required to enable subscribers to have consistent service behaviour upon Domain Transfers between access networks, subject to the constraints of the device and access network.

Editor's Note: multimedia and non-voice services require further consideration for seamless transparency.

5.4.1 Priority Service (ETS)

The IMS centralized services solution needs to support Domain Transfers of Priority Service.

Editor's Note: It is for further study whether Priority Service can be provided from IMS domain when using CS access.

5.4.2 Supplementary services

The IMS centralized services solution needs to provide subscribers with a consistent supplementary service behaviour upon Domain Transfers (e.g. Call Forwarding, Communication Waiting, Call Hold, etc.).

5.5 Session scenarios

5.5.1 Session scenarios when a PS network is used that supports the full duplex speech component of the IMS multimedia telephony service

When the UE has access to a PS network that supports the full duplex speech component of the IMS multimedia telephony service, then standard IMS session scenarios are supported according to the procedures specified in TS 22.101 [1], TS 22.173 [2], TS 23.228 [4], TS 24.229 [6] and TS 24.173 [5], no further amendments required.

5.5.2 Session scenarios when a PS network is used that does not support the full duplex speech component of the IMS multimedia telephony service

When the UE has access to a PS network that does not support support the full duplex speech component of the IMS multimedia telephony service or where the full duplex speech component of the IMS multimedia telephony service is not allowed but where the UE can use a CS domain to access IMS services, then the solution shall support the IMS session scenarios specified in clause 5.5.3.

NOTE: It is not clear how the UE understands if a PS network is capable of supporting the full duplex speech component of the IMS multimedia telephony service, but this issue is considered out of the scope of this document.

5.5.3 Session scenarios when a CS network is used to access IMS services

5.5.3.1 Session scenarios for an ICS UE

When an ICS UE accesses IMS services over a CS network, then the following IMS session scenarios shall be supported according to the procedures specified in TS 22.101 [1], TS 22.173 [2], TS 23.228 [4], TS 24.229 [6] and TS 24.173 [5], along with the solution specified in this document.

- Basic voice service origination and terminating sessions.
- Voice origination and termination service sessions with Line ID services (e.g. OIP, OIR, TIP, TIR) controlled in IMS.
- Voice origination and termination service sessions with Communication Barring services controlled in IMS.
- Voice termination service sessions with Communication Diversion services controlled in IMS.
- Voice origination and termination service sessions with mid-call services (e.g. Hold/Resume, Conferencing, CW, ECT) controlled in IMS.
- Communication services setting modifications (e.g. changing forwarding info or activating barring services, etc).

Editor's note: Procedure for the communication services setting modifications where PS domain is not available is FFS.

The solution shall provide also generic capabilities to enable introduction of new bi-directional speech related IMS services via CS bearer without further standardisation.

5.5.3.2 Additional session scenarios for ICS UE capable of VCC

In addition to session scenarios specified in clause 5.5.3.1 for ICS UEs, the solution shall support the following VCC session scenarios according to network procedures specified in TS 23.206 [3] when a ICS UE is capable of VCC:

- Domain Transfers of basic voice service sessions in both directions (between networks that support the full duplex speech component of the IMS multimedia telephony service over IP-CAN and CS).
- Domain Transfers of voice sessions with non mid call services in both directions (between networks that support the full duplex speech component of the IMS multimedia telephony service over IP-CAN and CS).
- Domain Transfers of voice sessions with mid call services in both directions (between networks that support the full duplex speech component of the IMS multimedia telephony service over IP-CAN and CS).
- Domain Transfers of Emergency sessions in both directions using the work being done in TR 23.826 [9], VCC support for Emergency Calls as a basis.

5.5.3.3 Session scenarios for non-ICS UEs

5.5.3.3.1 When using CS networks upgraded with ICS capability

The solution shall support the following IMS session scenarios together with the procedures specified in TS 22.101 [1], TS 22.173 [2], TS 23.228 [4], TS 24.229 [6] and TS 24.173 [5] when a non-ICS UE accesses IMS services via GSM/UMTS CS network which has been upgraded with the ICS capability:

- Basic voice service origination and terminating sessions.
- Voice origination and termination service sessions with Line ID services (e.g. OIP, OIR, TIP, TIR) controlled in IMS.
- Voice origination and termination service sessions with Communication Barring services controlled in IMS.
- Voice termination service sessions with Communication Diversion services controlled in IMS.
- Voice origination and termination service sessions with mid-call services (Hold/Resume, Conferencing, CW, ECT) controlled in IMS.

5.5.3.3.2 When using CS networks not upgraded with ICS capability

The solution shall support the following IMS session scenarios together with the procedures specified in TS 22.101 [1], TS 22.173 [2], TS 23.228 [4], TS 24.229 [6] and TS 24.173 [5] when a non-ICS UE accesses IMS services via GSM/UMTS CS network which has not been upgraded with the ICS capability:

- Basic voice service origination and terminating sessions.
- Voice origination and termination service sessions with Communication Barring services controlled in IMS.
- Voice termination service sessions with Communication Diversion services controlled in IMS.

Support of the following session scenarios may not be possible when a non-ICS UE accesses IMS services via GSM/UMTS CS network not upgraded with the ICS capability; these session scenarios may be supported with control of respective Supplementary Services in CS domain if necessary.

- Voice origination and termination service sessions with Line ID services (OIP, OIR, TIP, TIR) controlled in IMS.
- Voice origination and termination service sessions with mid-call services (Hold/Resume, Conferencing, CW, ECT) controlled in IMS.

5.5.3.3.3 Additional session scenarios for non-ICS UE capable of VCC

In addition to session scenarios specified in the clause 5.5.3.3.1 and clause 5.5.3.3.2 for a non-ICS UE, the solution shall support the following VCC session scenarios according to procedures specified in TS 23.206 [3] when a non-ICS UE capable of VCC accesses IMS services via GSM/UMTS CS network:

- Domain Transfers of basic voice service sessions in both directions.
- Domain Transfers of voice sessions with non mid call services in both directions.

5.5.4 Session scenarios for co-existence of ICS UE and Non ICS UEs solutions

Co-existence of ICS UE and non ICS UE in the same network shall be supported. A network with a VMSC enhanced for support of non-ICS UE shall be able to support call originations, terminations and Domain Transfer scenarios for ICS UE as specified in clause 5.5.3.1 Session scenarios for an ICS UE.

Roaming of ICS users to a network with a VMSC enhanced for non ICS UE shall be supported. A visited network with a VMSC enhanced for support of non ICS UE shall be able to support call originations, terminations and Domain Transfer scenarios for ICS UE as specified in clause 5.5.3.1 Session scenarios for an ICS UE when an ICS user with a ICS UE roams into this network.

ICS UE that connects to the ICCF via VMSC that has been enhanced for ICS will use the ICCC to access the ICS services unless ICCC connectivity is not available in HPLMN or VPLMN. However, when the ICS UE is not able to use ICCC (e.g. due to lack of UTRAN in case of I1-ps), the ICS UE falls back to non-ICS UE behaviour.

Editor's Note: It is FFS whether the IMSC needs to know whether the UE is an ICS UE or non ICS UE for the support if I1-cs which terminates on the IMSC.

When it serves the ICS user, the VMSC enhanced for ICS initiates SIP session to route the UE initiated calls to IMS, regardless whether the user uses ICS UE or non ICS UE, hence the VMSC enhanced for ICS does not perform CAMEL service execution to reroute the UE initiated calls.

ICCF does not need to be aware whether the session has been initiated via ICS enhanced VMSC or regular VMSC.

6 Architecture alternatives

Editor's Note: This clause will describe and evaluate detailed reference architectures, including network elements, interfaces and reference points, suitable to provide Centralized IMS Services

6.1 Introduction

The architectural alternatives described here may broadly be divided into those that have UE impacts and those that do not. Descriptions of the alternatives that impact the UE follow the ICS Reference Architecture and Reference Point clauses. The alternatives that do not impact the UE are described in a clause following the ICS Reference Architecture.

In the architecture alternatives described here a fundamental part of the architecture is the concept of the IMS CS Control Channel (ICCC). This is a logical control channel, established between the UE and an IMS network element for establishment and/or service control of IMS sessions using CS voice bearers as described below.

6.1.1 IMS CS Control Channel (ICCC)

The IMS CS Control Channel (ICCC) is a logical control channel used to transport signalling between the ICS UE or L-CAAF-n and the IMS when accessing IMS services via the CS domain. ICCC is used when needed, e.g. for IMS registration, on session establishment and/or service control of IMS sessions using CS voice bearers. The ICCC between the UE and the ICCF can in principle be established over the CS domain network, in which case it is referred to as I1-cs, or over the PS domain, in which case it is referred to as I1-ps. When I1-cs is established it can also be used for effecting communication service setting modifications or ICS UE could establish a circuit switch data connection and run Ut over it.

Editor's Note 1: Selection of circuit switch data or I1-cs for effecting communication service setting modifications is for further study.

Editor's Note 2: Fitting communication service setting modification messages in a minimal number of USSD messages is for further study.

- Transport mechanism for I1-cs is USSD and the transport mechanism for I1-ps is a PS bearer.

- IP connectivity is needed between the L-CAAF-n and ICCF for the support of ICCC.

The USSD transport mechanism does not offer as much bandwidth as the PS bearer so when using the USSD transport mechanism the limited bandwidth has to be taken into account and a suitable IMS CS Control Protocol (ICCP) is required. The ICCP may be implemented as a functional or stimulus protocol as driven by the architectural requirements. Details of ICCP implementation are for further study.

IMS SIP signalling is carried over the I1-ps.

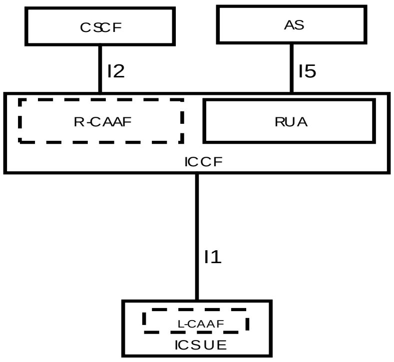
Editor's Note 3: Procedures for use of ICCC for IMS registration when using CS access, and session establishment and/or service control of IMS sessions established using CS voice bearers are FFS.

Editor's Note 4: ICCC only applies to MMTEL services. Potential use of ICCC for non-MMTEL services is FFS.

6.2 Reference architecture for ICS UEs

6.2.1 Reference architecture diagram

The figure below provides a Reference Architecture for IMS voice sessions established by an ICS UE using CS voice bearers and for voice sessions transferred between CS and PS access.



NOTE: Only relevant functions are shown.

Figure 6.2.1-1: ICS – Reference Architecture

The architecture introduces two new logical functions, the IMS CS Control Function (ICCF) and the ICS UE, as described below.

6.2.2 IMS CS Control Function (ICCF)

The IMS CS Control Function (ICCF) provides functions necessary for provision of IMS services for calls originated or terminated over CS access networks, for calls transferred between CS and PS access networks. and for communication services setting modifications over I1-cs. Note: alternatively, ICS UE could establish a circuit switch data connection and run Ut over it. Note: Communication setting modifications via Ut are not affected. The ICCF belongs to the IMS home network and is includes the following functions: the CS Access Adaptation Function (CAAF) and the Remote User Agent (RUA).

Based on information received and decoded by CAAF, the ICCF will determine the one or more commands to be executed, causing information to be encoded and exchanged by functions internal to the ICCF (e.g. RUA, RPC).

6.2.2.1 Remote User Agent (RUA)

The Remote User Agent (RUA) performs SIP User Agent functions on behalf of the ICS UE for IMS voice sessions established using CS voice bearers.

Editor's Note: Interaction details between the redirection functionality in the ICCF versus the one used in the VCC application is FFS

The RUA combines the CS call established between the UE and the RUA to set up a voice bearer, and the ICCC established between the RUA and the UE either directly or via the CAAF. It enables the completion of the call leg towards the UE, referred to hereafter as the "UE Leg"; and presents the session through the S-CSCF toward the other party, on a call leg referred to hereafter as the "RUA Leg". The UE Leg and the RUA Leg form a B2BUA at the RUA. The TAS and other Application Servers are executed on the RUA Leg as part of standard service execution logic at the S-CSCF. The session processing complies with the current IMS procedures (e.g. MMTel as in TS 24.173[5] for standardized supplementary services). In other words TAS and other IMS Application Servers do not see a difference regarding the current IMS/MMtel procedures, whether it serves a UE roaming in CS or in IP-CAN. This does not exclude access specific information passing via SIP and its use when necessary by the services.

6.2.2.2 Realization of the ICCF

The ICCF is realized with an ISC interface to S-CSCF for both I1-cs and I1-ps. It is invoked as the very first SIP AS in the originating call and the last one in terminating calls i.e. closest to the access. It needs to be ensured that the AS implementing DTF of VCC Application is either second for originating calls or second last for terminating calls.

Further details for the realization of ICCF vary based on the various architecture alternatives as described below.

6.2.2.2.1 I1-ps approach

The Gm reference point is used to realize the I1 reference point with use of 3GPP SIP signalling for registration and session control. The ISC interface and the Ma reference point are used for PSI routing to the ICCF to establish voice bearer via the CS domain. The Ut reference point is used for communication services setting modifications.

The ISC interface is used for realization of the I2 reference point.

Figure 6.2.2.2.1-1 illustrates the realization of ICCF for the I1-ps approach.

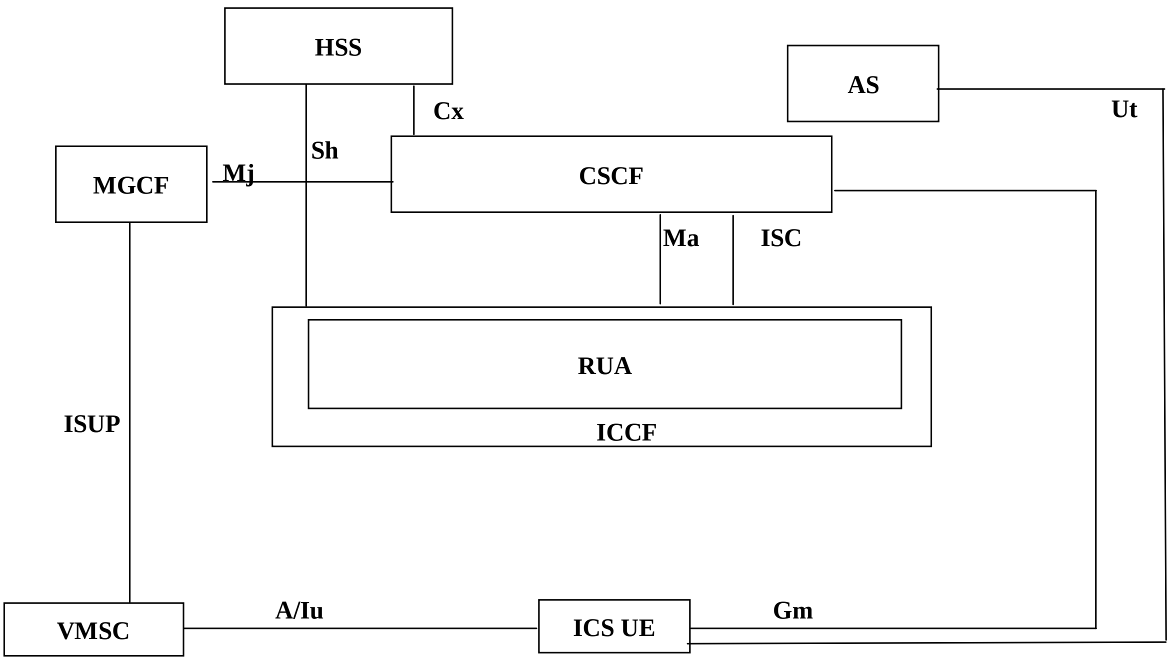


Figure 6.2.2.2.1-1: Realization of ICCF for I1-ps approach

6.2.2.2.2 I1-cs approach

The A/Iu interface as well as the gsmSCF-HLR interface is used to realize the I1 reference point with use of USSD transport to carry information required to generate IMS signalling at the ICCF. The ISC interface and the Ma reference point are used for PSI routing to the ICCF to establish voice bearer via the CS domain.

6.2.2.2.2.1 I1-cs: unregistered user solution

For realization of I2, the ISC interface is used for originations and terminations. IMS Registration via CS access is not supported.

Figure 6.2.2.2.2.1-1 illustrates the realization of ICCF for the I1-cs: unregistered user solution.

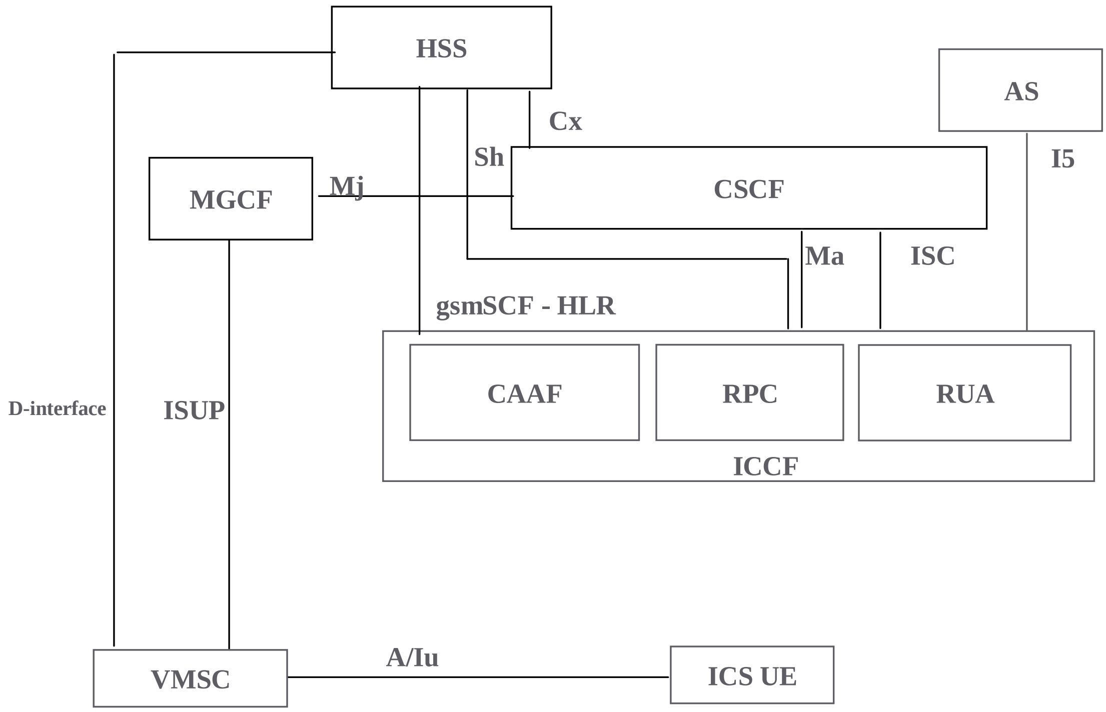


Figure 6.2.2.2.2.1-1: Realization of ICCF for I1-cs: unregistered and registered user solution

The CAAF of the ICCF presents gsmSCF-HLR interface towards the HSS for USSD signalling. The ICCF functionality is similar to the ICCF in the I1-ps approach except that it does not the include the CAAF for USSD handling.

6.2.2.2.2.2 I1-cs approach: registered user solution

6.2.2.2.2.2.1 I1-cs approach: registered user solution-ISC model

The ISC is used for realization of I2, i.e., origination and termination procedures are executed over the ISC interface.

Editor's note: Two solution alternatives have been suggested for execution of the IMS Registration via CS access; one using the ISC and the other using the Cx interface. Selection of a particular alternative is expected at conclusion of this study.

Figure 6.2.2.2.2.1-1 illustrates the realization of ICCF for the I1-cs: registered user solution-ISC model.

The CAAF of the ICCF presents gsmSCF-HLR interface towards the HSS for USSD signalling. The ICCF functionality is identical to the I1-ps approach except the inclusion of CAAF for USSD handling.

6.2.2.2.2.2.2 I1-cs approach: registered user solution-Mw model

The Mw is used for realization of I2, i.e. the IMS Registration via CS access, origination and termination procedures are executed over the Mw reference point.

Figure 6.2.2.2.2.2-1 illustrates the realization of ICCF for the I1-cs: registered user solution.

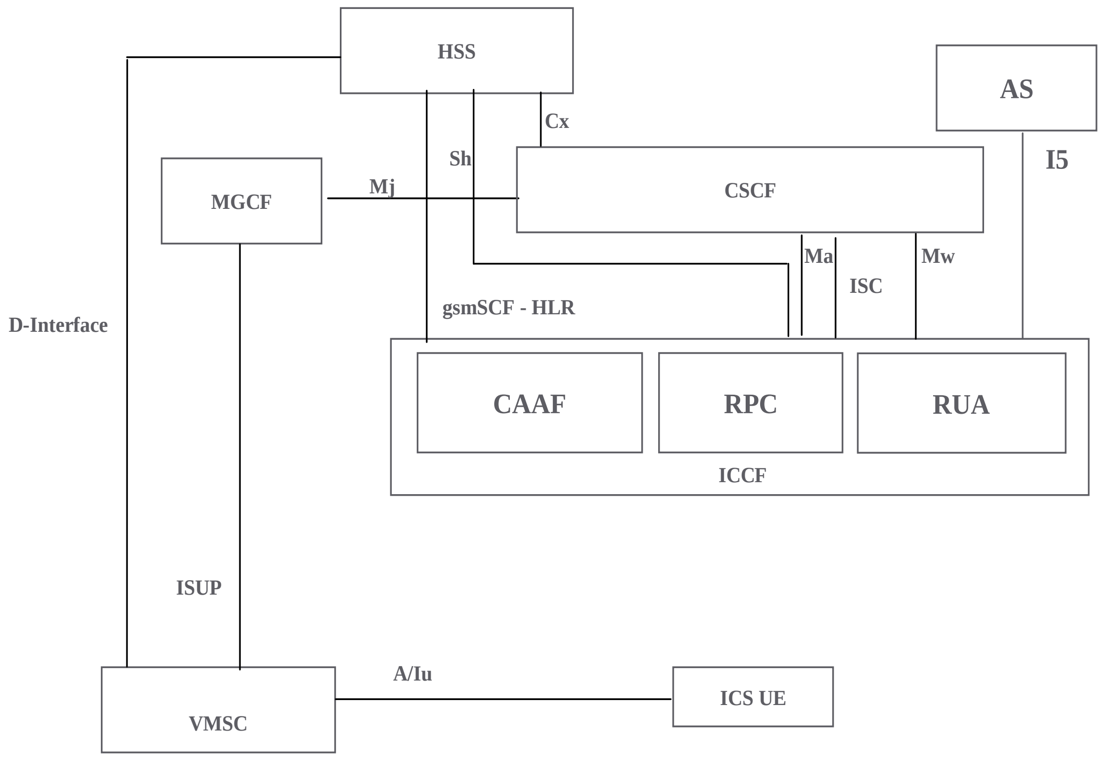


Figure 6.2.2.2.2.2.2-1: Realization of ICCF for I1-cs only: registered user solution-Mw model

The CAAF of the ICCF presents gsmSCF-HLR interface towards the HSS for USSD signalling.

Editor's Note 1: Use of Access Selection Function for ICCF when using I1-ps and I1-cs is FFS.

Editor's Note 2: Whether to use ISC or Mw for I2 reference point for registered user solution of I1-cs is FFS.

Editor's Note 3: It is for FFS whether to support only I1-cs, only I1-ps or both.

Editor's Note 4: A decision would be required for selection of one physical implementation of RUA.

Editor's Note 5: The interaction between VCC and the ICCF is for further study.

Editor's Note 6: The handling of registration via the I2 reference point is for further study.

Editor's Note 7: It is FFS whether IMS Registration is required over CS access.

6.2.2.3 CS Access Adaptation Function (CAAF)

The CS Access Adaptation Function (CAAF) is an ICCF internal adaptation function for the service control signalling between CS domain and IMS. The CAAF conveys the information received from the ICS UE over CS access signalling to ICCF functions such as the RUA and RPC, and vice versa. The RUA uses the information received from the CAAF for initiation and control of SIP sessions.

The Remote CS Access Adaptation Function (R-CAAF) resides in the ICCF with a Local CS Access Adaptation Function (L-CAAF) provided in the ICS UE.

The CAAF can also be used as an adaptation function for requests for communication services setting modifications. In this instance the CAAF conveys the information received from the ICS UE over CS access signalling to a RPC.

Editor's Note: Use of USSD for transport of communication services setting related request is FFS.

The CAAF is not employed in the ICCF and the UE when the I1-ps is used.

6.2.2.4 Remote Provisioning Client (RPC)

The Remote Provisioning Client (RPC) is an ICCF internal adaptation function for the communication setting modifications signalling between CS domain and IMS. The RPC conveys communication setting modifications information received from the ICS UE over CS access signalling to the TAS. The settings to be modified are defined by as part of the definition of current IMS procedures (e.g. MMTel as in TS 24.173[5] for standardized supplementary services). The ASes involved do not see the difference, whether they serve a UE roaming in CS or in IP-CAN.

6.2.3 ICS UE

The ICS UE is a UE capable of ICS. An ICS UE communicates via I1 with the ICCF.

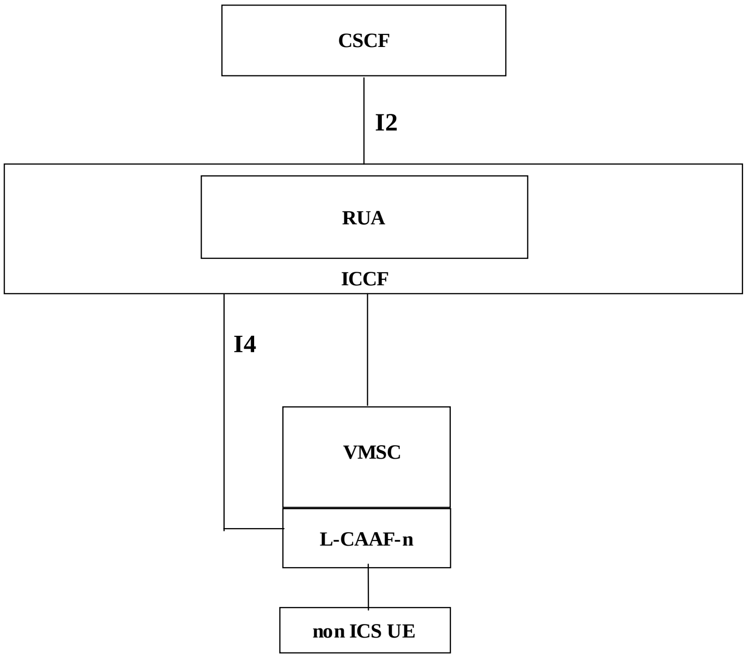
The UE contains the L-CAAF for enablement of I1-cs.

6.3 Architecture for support of non-ICS UE

6.3.1 L-CAAF-n approach

6.3.1.1 Reference architecture

The ICS Ref Architecture in figure 6.3.1-1 may be used to provide ICS support but requires CS core network upgrades. The limitations of the legacy UEs need to be taken into consideration.



NOTE: Only relevant functions are shown.

Figure 6.3.1-1: Architecture for support of non ICS UE, L-CAAF-n approach

6.3.1.2 Extensions to the CAAF in the ICCF for support of non ICS UEs

The CAAF provides necessary adaptation needed for enablement of the RUA in the ICCF.

The CAAF is provided as a functional element; i.e. for non ICS UEs, the L-CAAF logic which is otherwise associated with ICS UEs is provided by a CS Core Network function, the Local CS Access Adaptation Function-network (L-

CAAF-n) on the access edge of the VMSC on behalf of the UE. The L-CAAF-n also provides interworking with TS 24.008 [7] as necessary.

The CAAF is exclusively provided by the L-CAAF-n associated with the VMSC and therefore the R-CAAF component in the ICCF is not required.

Editor's Note: Call independent aspects such as user configuration of supplementary services are FFS.

6.3.1.3 Service Support for non ICS UE using L-CAAF-n

When using the L-CAAF-n approach for non ICS UE, the RUA of the ICCF provides SIP UA behaviour on behalf of the non ICS UE, with the L-CAAF-n providing the interworking between CS Access Signalling and ICCC; this enables exclusive control of services in IMS.

Note that the service behaviour is subject to the limitations of the UE and CS access signalling, and is comparable to that of normal CS users.

6.3.1.3.1 Line ID Services (OIP, OIR, TIP, TIR)

With the L-CAAF-n providing the interworking between CS Access Signalling and ICCC and the RUA of the ICCF providing SIP UA behaviour on behalf of the non ICS UE, these services are provided in IMS.

6.3.1.3.2 Communication Diversion Services

6.3.1.3.2.1 Communication Diversion services exclusively controlled in home IMS (CFU, CFNL)

With the L-CAAF-n providing the interworking between CS Access Signalling and ICCC and RUA of the ICCF providing the SIP UA behaviour on behalf of the non ICS UE, these services are provided in IMS.

6.3.1.3.2.2 Communication Diversion services requiring participation of serving network (CFNR, CFB)

With the L-CAAF-n providing the interworking between CS Access Signalling and ICCC and the RUA of the ICCF providing SIP UA behaviour on behalf of the non ICS UE, these services are provided in IMS.

6.3.1.3.3 Communication Deflection

With the L-CAAF-n providing the interworking between CS Access Signalling and ICCC and the RUA of the ICCF providing SIP UA behaviour on behalf of the non ICS UE, these services are provided in IMS.

6.3.1.3.4 Communication Barring

With the L-CAAF-n providing the interworking between CS Access Signalling and ICCC and the RUA of the ICCF providing SIP UA behaviour on behalf of the non ICS UE, these services are provided in IMS.

6.3.1.3.5 Mid call services (Communication Hold, CW, Conf, Communication Transfer)

With the L-CAAF-n providing the interworking between CS Access Signalling and ICCC and the RUA of the ICCF providing SIP UA behaviour on behalf of the non ICS UE, these services are provided in IMS.

6.3.1.3.6 Voice Call Continuity

The Voice Call Continuity for non ICS UEs capable of VCC is based on TS 23.206.

6.3.1.3.7 User configuration of Supplementary Services

Service data management and control of user configuration of services data is provided by IMS.

6.3.1.3.8 Serving Domain Considerations

Table 6.1.3.8-1 below provides recommendation for serving domain for the services discussed in this clause:

Table 6.1.3.8-1: Service set for non ICS UE solution; L-CAAF-n approach

Service Capability	Serving Domain
Supplementary Services	
Originating Identification Presentation	IMS
Originating Identification Restriction	IMS
Terminating Identification Presentation	IMS
Terminating Identification Restriction	IMS
Communication Diversion (CFU, CFNL)	IMS
Communication Diversion (CFNR, CFB)	IMS
Communication Deflection	IMS
Call Wait	IMS
Communication Hold	IMS
Communication Barring	IMS
Conference	IMS
Explicit Communication Transfer	IMS
Service Continuity	
Basic Service Continuity	IMS[1]
Service Continuity on non mid call services	IMS[1]
Service Continuity with mid call services	N/A
Other capabilities	
Call Independent Supplementary Services Operations	IMS
[1] Applicable to non ICS UE capable of VCC.	

6.3.1.4 Physical realization of ICCF with L-CAAF-n approach

The following figure illustrates the physical realization of the ICCF with the L-CAAF-n approach for non ICS UE.

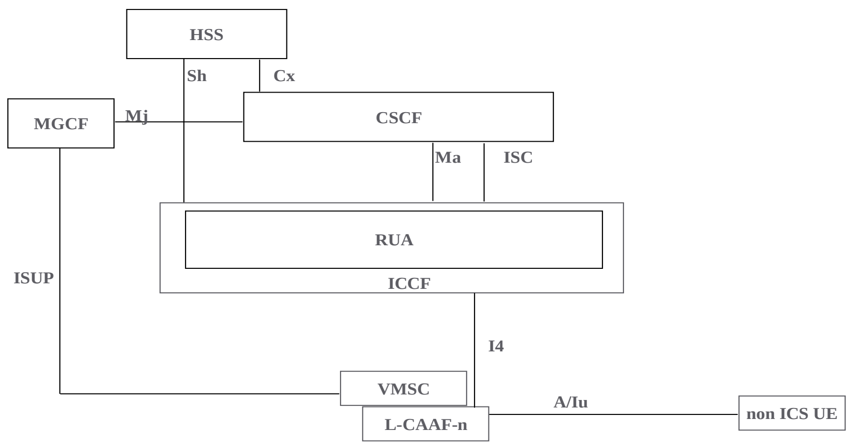


Figure 6.3.1.3-1: Realization of ICCF; L-CAAF-n approach for non ICS UE

Editor's note: Details of session control procedures on I4 are FFS.

The ISC interface is used for realization of the I2 reference point.

6.3.2 CAMEL approach

The ICS reference architecture in figure 6.3.2-1 may be used to provide reduced ICS support without requiring VMSC enhancements. When considering deployment of this option the limitations of the legacy UEs need to be taken into consideration.

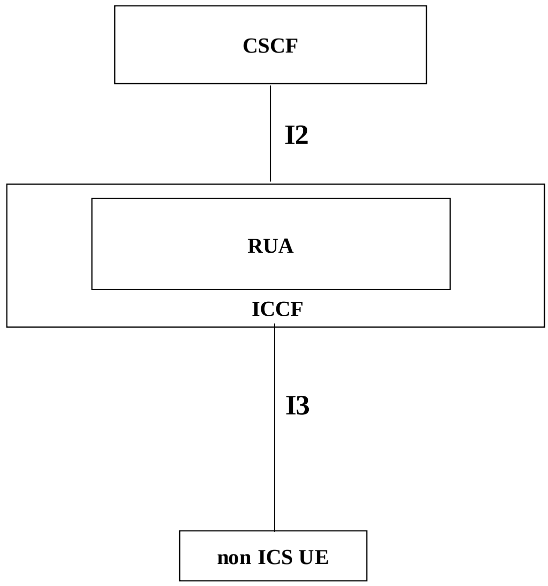


Figure 6.3.2-1: Architecture for non ICS UE support, CAMEL approach

NOTE: Only relevant functions are shown.

Standard CS call control procedures are used for setup of CS sessions with CAMEL used for redirection of the CS calls to IMS, as specified in TS 23.206 [3], in conjunction with the standard MGCF providing the interworking between the CS call control and SIP. SIP is used for call control with the ICCF providing SIP UA behaviour on behalf of the UE for control of user sessions.

Control in IMS of some serving network services e.g. line id services and mid call services is not possible with this option. These services may alternatively be provided in the CS domain for sessions established for non ICS UEs.

Editor's note 1: interaction with VCC to support VCC and ICS users for non ICS UEs is FFS.

Editor's note 2: Solution for Call independent aspects such as user configuration and interworking/alignment of supplementary services data across CS domain and IMS is to be provided.

6.3.2.1 Service Support for non ICS UE, using the Architecture with CAMEL approach

In the CS network agnostic solution for the non ICS UEs, the ICCF provides a SIP UA behavior on behalf of the non ICS UE for CS sessions established using standard CS procedures, and redirected to IMS using techniques similar to the procedures defined for VCC in TS 23.206. Since the solution uses standard CS procedures for sessions established over CS access, it is limited in its service offering as discussed below:

6.3.2.1.1 Line ID Services (OIP, OIR, TIP, TIR)

Since standard session setup procedures are used to establish the non ICS UE sessions, with the VMSC controlling session origination and session delivery from/to the non ICS UEs, Line ID services configuration in the HLR component of the HSS for appropriate service execution at the VMSC is required. Some control of these services may also be provided in IMS with an AS manipulating the information being presented from/to the ICS subscriber using a non ICS UE.

6.3.2.1.2 Communication Diversion Services

6.3.2.1.2.1 Communication Diversion services exclusively controlled in home IMS (CFU, CFNL)

With the ICCF providing the SIP UA behaviour on behalf of the non ICS UE, these services are provided in IMS.

6.3.2.1.2.2 Communication Diversion services requiring participation of serving network (CFNR, CFB)

Some control of these services may be provided in IMS. Since standard session setup procedures are used to establish the non ICS UE sessions, with the VMSC controlling session origination and session delivery from/to the non ICS UEs, some handling of these services is also required at the VMSC.

6.3.2.1.2.3 Communication Deflection

This service can only be provided by the VMSC.

6.3.2.1.3 Communication Barring Services (CB)

With the ICCF providing the SIP UA behaviour on behalf of the non ICS UE, these services are provided in IMS.

6.3.2.1.4 Mid call services (Communication Hold, CW, Conf, Communication Transfer)

Since standard session setup procedures are used to establish the non ICS UE sessions, with the standard CS signalling for invocation of these services, these services can only be provided in CS domain.

6.3.2.1.5 Session continuity

The Voice Call Continuity for non ICS UEs capable of VCC is based on TS 23.206. The session continuity of ICS video call would require DTF to be enhanced with capability of anchoring multimedia call which is under study in MMSC WI .TS 23.206 modification are required.

6.3.2.1.6 User configuration of Supplementary Services

Service data management and control of user configuration of services data is provided by the IMS for services controlled in IMS and by the CS domain for services controlled in CS domain

6.3.2.1.7 Serving domain considerations

Table 6.3.2.1.7-1 below provides recommendation for serving domain for the services discussed in this clause:

Table 6.3.2.1.7-1: Service set for non ICS UE with CS network agnostic solution

Service Capability	Serving Domain
Supplementary Services	
Originating Identification Presentation	IMS[2]
Originating Identification Restriction	IMS[2]
Terminating Identification Presentation	IMS[2]
Terminating Identification Restriction	IMS[2]
Communication Diversion (CFU, CFNL)	IMS
Communication Diversion (CFNR, CFB)	IMS [2]
Communication Deflection	CS or n.a.[1]
Call Wait	CS or n.a.[1]
Communication Hold	CS or n.a.[1]
Communication Barring	IMS
Conference	CS or n.a.[1]
Explicit Communication Transfer	CS or n.a.[1]
Service Continuity	
Basic Service Continuity	IMS[3]
Service Continuity on non mid call services	IMS[3]
Service Continuity with mid call services	n.a.
Other capabilities	
Call Independent Supplementary Services Operations	IMS/CS[4]

[1] Control in IMS of this service is not possible. As an alternative, this service may be provided by CS domain: In this case a configuration in the HLR component of the HSS for appropriate service execution at the VMSC is required.

[2] Exclusive control in IMS of this service is not possible, some support from the CS domain may be required. Configuration in the HLR component of the HSS for appropriate service execution at the VMSC is required.

[3] Applicable to non ICS UEs capable of VCC.

[4] Service data management provided by the IMS for services controlled in IMS and by CS domain (if services are provided) for services controlled in CS domain.

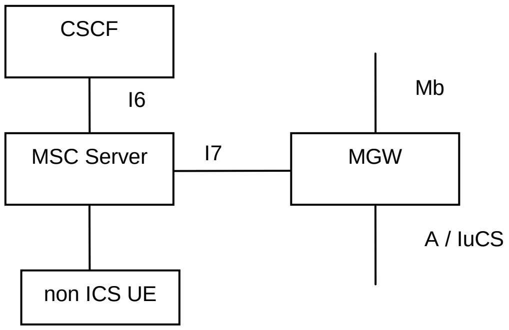
The HLR configuration that enables the execution of CLIR, CLIP, COLP, COLR, Call Deflection, Call Wait, Call Hold, Multiparty, Explicit Call Transfer in the CS domain should be used when the ICS subscriber is using a non-ICS UE and according to operator policy, in order to avoid conflicts with the corresponding services provided by IMS.

Editor's Note: How to trigger the use of the appropriate HLR configuration is to be investigated.

6.3.3 Architecture with Enhanced MSC Server

6.3.3.1 Reference architecture

The ICS reference architecture in figure 6.3.3-1 may be used to provide ICS support but requires MSC Server upgrades in both the home and visited networks. The limitations of the legacy UEs need to be taken into consideration.



NOTE: Only relevant functions are shown.

Figure 6.3.3-1: Architecture for ICS support without terminal impact, with MSC Server enhancements

The MSC Server provides the necessary interworking between TS 24.008 [7] and SIP. The MSC Server also controls a MGW to provide the user plane interworking between CS access and RTP bearers.

Editor's note: Physical realization of this MGW is for further study.

6.3a ICS Architecture for ICS UE and non-ICS UE

This architecture alternative combines the architectures in clause 6.2 and 6.3 for a unified ICS solution that provides the following capabilities for ICS UE and non ICS UE:

- Consistency and continuity of operator services with an ICS UE that uses both I1-ps and I1-cs.
- ICCC transport change between I1-ps and I1-cs based on access network capabilities for ICS UE.
- IMS service control and R7 VCC based Domain Transfers with a non ICS UE.
- Facilitate Session Continuity according to TR 23.893 [19].

6.3a.1 IMS CS Control Channel (ICCC)

Same as defined in clause 6.1.1 with the following difference:

I1-cs runs transparently to the VMSC when the ICS UE is attached to a standard VMSC. I1-cs need not be transparent to the VMSC when the ICS UE is attached to an ICS Enhanced MSC Server; i.e. the ICS capabilities of the ICS Enhanced MSC Server may be used to interwork I1-cs with SIP at the ICS Enhanced MSC Server for signalling efficiencies; e.g. to avoid unnecessary HSS hop when processing USSD messages.

6.3a.2 Reference architecture diagram

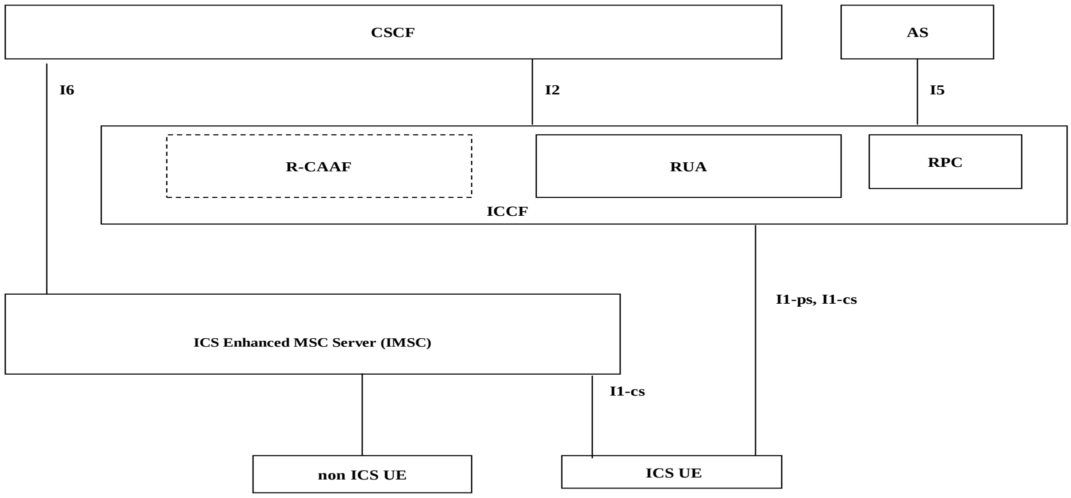


Figure 6.3a.2-1 ICS Architecture for ICS and non ICS UE

This architecture introduces new logical functions as described below.

6.3a.2.1 IMS CS Control Function (ICCF)

Same as defined in clause 6.2.

The R-CAAF is employed only when using I1-cs which is established via a standard VMSC, or when using I1-cs which is established via IMSC Server which does not implement the option for non transparent I1-cs. To be able to employ the R-CAAF, the ICCF must know whether there is an IMSC that implements the option for non transparent I1-cs or not. This capability is negotiated during the registration procedure between the IMSC and ICCF.

For ICS User using ICS UE using I1-ps and I1-cs established via an IMSC and for non ICS UE, and for session established via IP-CAN, the RUA of the ICCF is inserted in the ICS User session path as part of the originating and terminating iFC execution. For I1-cs established via a standard VMSC, the RUA is invoked via USSD or through PSI routing.

The RUA provides UA behaviour for all ICS User sessions; the ICS UE or an IMSC enables the RUA in the ICCF, with a SIP UA also running on the ICS UE or the IMSC.

The RUA maintains the access network information required to enable the T-ADS; it also maintains the end-to-end SIP session state for all ICS User sessions to enable Service Continuity and change of ICCC transport while maintaining an active session.

The ICCF is not required when using non ICS UEs which are capable of receiving voice telephony services only over GSM/UMTS CS access.

6.3a.2.2 ICS Enhanced MSC Server (IMSC)

Selected the approach as documented in the Conclusion clause of this TR.

6.3a.2.3 ICS UE

Same as defined in clause 6.2.3

ICS UE receives ICS when using an ICS Enhanced VMSC Server or a standard VMSC.

6.3a.2.4 Remote Provisioning Client (RPC)

Same as defined in clause 6.2.2.4.

6.3a.2.5 Reference Points

6.3a.2.5.1 I1 Reference Points

Same as defined in clause 6.4.1 with the following difference:

I1-ps is established using the Gm reference point as in clause 6.4.

When I1-cs is established via standard VMSC, it is transparent to the serving CS network.

When I1-cs is established via an IMSC, it may not be transparent to the IMSC if the R-CAAF resides in the IMSC as an internal adaptation function.

The I1 reference points are used only for ICS UE.

6.3a.2.5.2 I2 Reference Point

Same as defined in clause 6.4.2.

The I2 reference point is used for ICS UE and non ICS UE.

6.3a.2.5.3 I5 Reference Point

Same as defined in clause 6.4.5.

6.3a.2.5.4 I6 Reference Point

Same as defined in clause 6.4.6.

The I6 reference point is used for routing of session control signalling to IMS, with the ICCF inserted in the session path via iFC execution.

The I6 reference point is used for ICS UE when using I1-cs via an IMSC and for non ICS UE.

Editor's note: Use of the I4 reference point for ICS capability exchange between the ICCF and the ICS Enhanced MSC Server is FFS.

6.3a.3 Signalling and Bearer Architecture for ICS UE: I1-ps with Bearer Control Signalling Path established via standard VMSC

Same as in clause 6.5.1 Signalling and bearer architecture for full duplex speech over CS access.

6.3a.4 Signalling and Bearer Architecture for ICS UE: I1-ps with Bearer Control Signalling Path established via IMSC

The signalling and bearer setup for I1-ps when using an IMSC for CS bearer setup is the same as when using a standard VMSC with a difference that the IMSC sets up the Bearer Control Signalling Path as a standard SIP session towards home IMS, and that IMSC registers to IMS on behalf of the ICS UE.

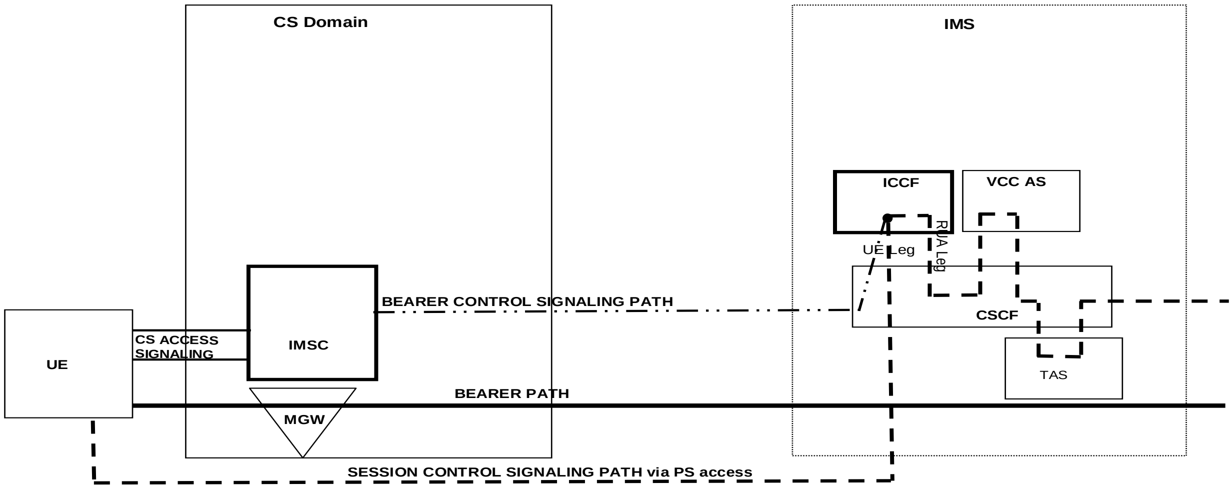


Figure 6.3a.4-1: Signalling/Bearer Paths for PS transport of ICCC; CS bearer setup via IMSC

6.3a.5 Signalling and Bearer Architecture for ICS UE: I1-cs using a standard VMSC

Same as in clause 6.6.1 Signalling and bearer architecture for full duplex speech over CS access.

6.3a.6 Signalling and Bearer Architecture for ICS UE with I1-cs using an IMSC and non ICS UE

When attached to an IMSC, the ICS UE or a non ICS UE may receive IMS services as enabled by interworking of CS Access Signalling for the I6 reference point by the IMSC.

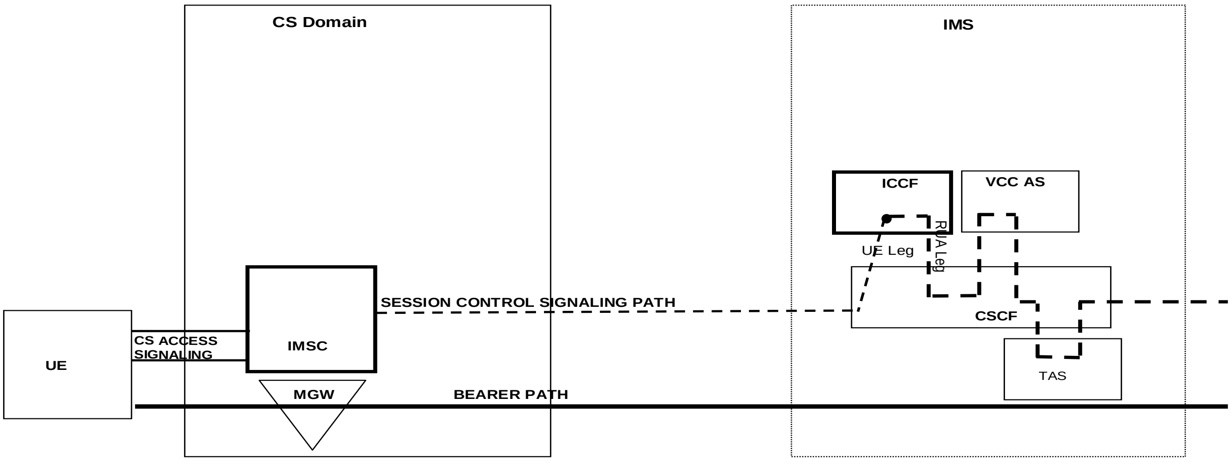


Figure 6.3a.6-1: Signalling/Bearer Paths for I1-cs of ICS UE and non ICS UE attached to a IMSC

In this model ICS is enabled by the ICS Enhanced MSC Server interworking the CS Access Signalling with 3GPP IMS SIP. New procedures over the I6 reference point are used to set up the Session Control Signalling Path between the IMSC and remote end with the ICCF inserted in the session path using originating and terminating iFCs. The media is established between the UE and the remote end through a MGW controlled by the IMSC. The SDP information related to the media is carried in the Session Control Signalling Path. The ICCF maintains the end-to-end session state and access specific information to enable Service Continuity and T-ADS. When IMSC serves the ICS UE using I1-cs, the IMSC registers to IMS on behalf of the UE, regardless whether the IMSC implements the transparent or non transparent option of I1-cs.

Interworking of CS Access Signalling with 3GPP SIP for session setup for this model enables the capability to provide all services exclusively by IMS.

NOTE: When IMSC is serving ICS UE using I1-cs, the I1-cs may or may not be transparent to IMSC depending on the implementation option. The figure 6.3a.6-1 shows only the non-transparent option.

6.3a.7 Signalling and Bearer Architecture for an ICS User who only owns a non ICS UE which is capable of receiving voice telephony services only over GSM/UMTS CS access

Same as in clause 6.8.a.1.1.

6.4 ICS Reference Points

6.4.1 I1 Reference Point

For ICS UEs, the I1 reference point is used between the ICS UE and the ICCF. The I1 reference point implements either the I1-cs or the I1-ps. The I1-cs may be established through the CS domain network using CS Access Signalling and the I1-ps may be established through PS domain using PS transport.

6.4.2 I2 Reference Point

The I2 reference point is used between the ICCF and the CSCF for presentation of the SIP UA behaviour toward IMS for control of user sessions.

6.4.3 I3 Reference Point

The I3 reference point is established between non ICS UE and the ICCF through the non-enhanced CS network using standard CS network procedures for CS session establishment and using CAMEL for redirection of CS originating sessions to IMS.

The I3 reference point is only applicable to the architecture for ICS support without terminal impact or VMSC enhancements.

6.4.4 I4 Reference Point

For non ICS UEs, the I4 reference point is used between the L-CAAF-n associated with the VMSC and the ICCF for establishment and control of ICCC.

6.4.5 I5 Reference Point

The I5 reference point is used between the ICCF and the AS for securely managing and configuring data for ICS when I1-cs is used or when a circuit switch data connection with Ut over it is established. This reference point uses capabilities defined for the Ut reference point as defined in TS 23.002 [18].

6.4.6 I6 Reference Point

The I6 reference point is established between the MSC Server and the home IMS of the ICS subscriber using a non ICS UE for presentation of the SIP UA behaviour toward IMS for control of user sessions. The Mw reference point is used as the baseline for I6.

6.4.7 I7 Reference Point

The I7 reference point is established between the MSC Server and the MGW for interworking the user plane on CS access interfaces (A and IuCS reference points) with the user plane in IMS (Mb reference point).

The I7 reference point is realized by adding the Mn reference point to the MSC Server. This allows the MSC Server to control both a CS-MGW and an IM-MGW, which is sufficient to perform the necessary user plane interworking. This is depicted in the following figure.

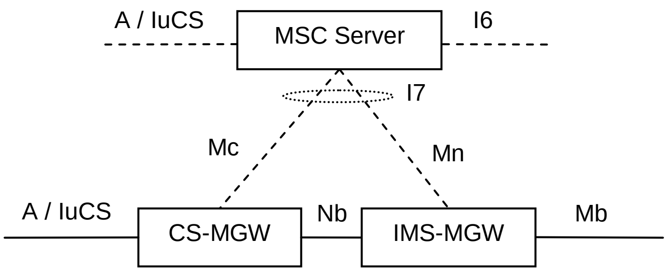


Figure 6.4.7-1

For resource efficiency, H.248.18 [13] can be used to allow the enhanced MSC Server and its MGW to negotiate the use of both the Mc and Mn profiles, defined in TS 29.232 [14] and TS 29.332 [15] respectively.

The TS shall not define a new reference point, but instead shall add the Mn reference point to the MSC Server.

6.4a Applicability of the Domain Selection Function to ICS

6.4a.1 Service Domain Selection (SDS) for ICS

Service control for all ICS user sessions is provided by IMS, hence the service domain selection, if applied, results in selection of IMS as the serving domain for all ICS user sessions.

SDS may be required in the CS domain core network for originating sessions established using CS call control and for terminating sessions directed to the CS domain so that the ICS user sessions are redirected to IMS for service control.

6.4a.2 Access Domain Selection (ADS) for ICS

6.4a.2.1 ADS for originating sessions

ADS for originating sessions is executed in the UE, as specified in TS 23.221, clause 7.2a Domain selection for UE originating sessions / calls.

6.4a.2.2 ADS for terminating sessions

The ADS for terminating sessions is executed by the last Application Server as part of the terminating iFC.

For a network which implements the ICCF, the ICCF is the last AS for the ICS user.

NOTE 1: The ADS need not be physically collocated with the ICCF.

NOTE 2: ADS selection criteria is in TS 23.221 clause 7.2b, Access Domain Selection for terminating sessions

Alternatively for the I1-ps approach and the I1-cs approach, the initial ADS for terminating sessions could be executed in the UE in conjunction with the network based on operator policy and taking into account its own capabilities and those of the access network. Figure 6.4a.2.2-1 shows an example handling logic for UE based domain selection in ICS.

In both cases, the access domain used by all ICS UE and non ICS UE active sessions is taken into account during ADS for any new incoming call.

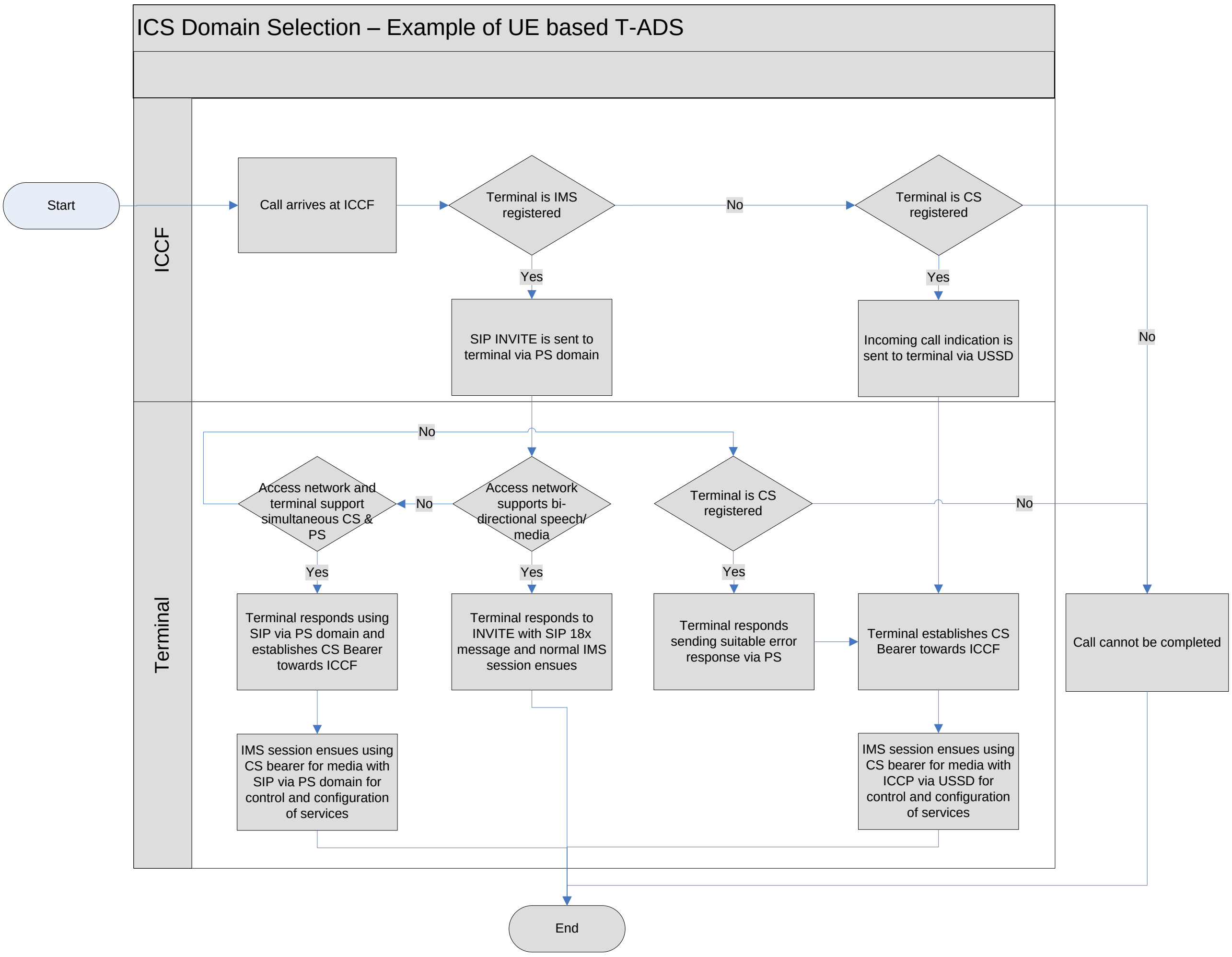


Figure 6.4a.2.2-1: Example handling logic for UE based Terminating Access Domain Selection for ICS

The ICCF will be included in the call flow for all MT calls.

1 If the terminal is IMS registered the ICCF sends an INVITE to the terminating UE

   a. If the terminal determines it is in a bi-directional speech/media capable IP-CAN it will respond to the INVITE with a standard 18x response and the session will be established via IMS.

   b. If the terminal determines it is in a IP-CAN where bi-directional speech/media cannot be supported, but where the access network can support simultaneous PS and CS, it will respond with a 18x response, that indicates that the media should be carried via a CS bearer. The UE will then establish a CS bearer towards the ICCF, and the ICCF will correlate the CS bearer with the IMS session.

   c. If the terminal determines it is in a IP-CAN where bi-directional speech/media cannot be supported and where there is no simultaneous CS and PS in the access network, it responds by sending a suitable error response towards the ICCF to indicate that the session will be established via the CS domain using I1-cs procedures. Further control and configuration of services will be via USSD.

2 If the terminal is not IMS registered via an IP-CAN but is CS registered, the ICCF will use USSD to send the incoming call notification towards the terminal. The UE will establish a CS bearer towards the ICCF using I1-cs call establishment procedures that the ICCF will correlate with the IMS session. Further control and configuration of services will be via USSD.

6.4a.2.2.1 Interactions with VCC for ADS of terminating sessions

In a network that implements the ICCF the ADS for terminating sessions established for ICS UE capable of VCC is executed by the ICCF which is invoked after the DTF of the VCC AS, as part of the terminating iFC, or by network in conjunction with the UE based on operator policy and taking into account the capabilities of the access network. TS 23.206 modifications are required.

NOTE: How the UE determines the bi-directional speech/media capabilities of the access network is not within the scope of this study.

Editor's note: Details of modifications to TS 23.206 are FFS.

ADS for terminating sessions established for a VCC subscriber who is not an ICS user is provided as specified in TS 23.206.

It is not specified which AS performs T-ADS for a network that does not implement ICCF or VCC.

ADS is performed only once per session.

6.5 Architectural alternative: I1-ps approach

6.5.1 Signalling and bearer architecture for full duplex speech over CS access

When in CS coverage with simultaneous PS access available, e.g. UTRAN, the ICS UE may use IMS SIP signalling over PS bearers for enablement of I1-ps when support of the full duplex speech component of the IMS multimedia telephony service is not available over PS bearer.

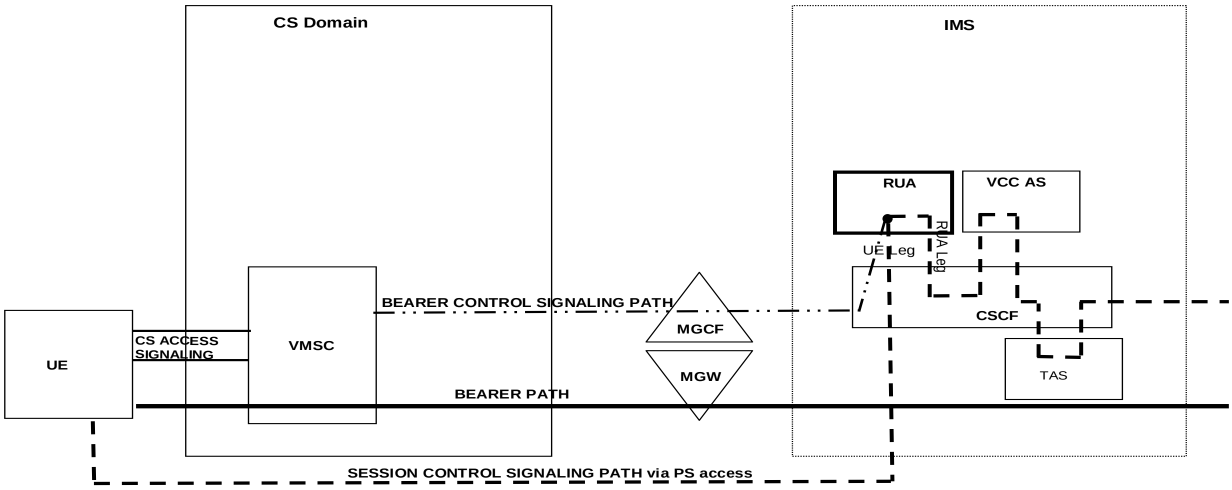


Figure 6.5.1-1: Signalling/Bearer Paths for PS transport of ICCC

In this model the I1-ps is enabled by using IMS SIP signalling over PS bearers, and is used for session setup when establishing IMS voice sessions using CS voice bearers. IMS SIP signalling is used in the UE for control of all calls with the I1-ps established through the Session Control Signalling Path over PS access. Standard IMS call control procedures are used to set up the Session Control Signalling Path between the UE and remote end with the RUA inserted in the session path. No bi-directional speech media is transmitted over the IP-CAN. In parallel, the UE establishes a Bearer Control Signalling path with the RUA by establishing a CS call toward the RUA. The Bearer Control Signalling and Session Control Signalling stimuli are combined at the RUA for presentation of the IMS session toward the CSCF on behalf of the UE.

The UE maintains the SIP/SDP state machine with RUA also maintaining a copy of the state data when present in the session path.

Use of the I1-ps for session setup for this model enables the capability to provide all services exclusively by IMS.

The CAAF is not required when using I1-ps with RUA providing control of IMS sessions using CS voice bearers.

6.5.2 Information flows

6.5.2.1 Registration

Provides an example flow for Registration made by an ICS UE where by a dynamic RUA DN is allocated to be used for session originations where CS domain is used for bearer plane.

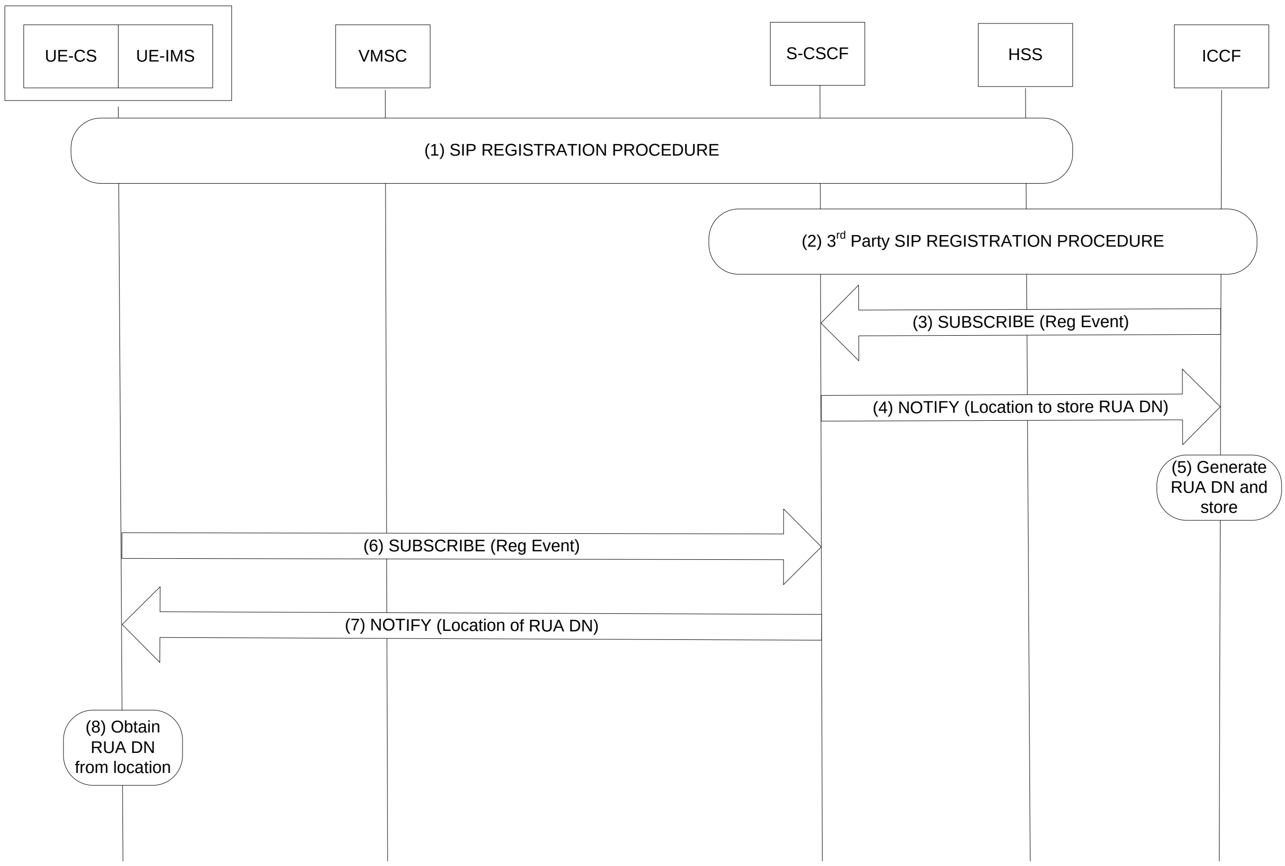


Figure 6.5.2.1-1

1. IMS registration procedure per TS 23.228 [4] is performed. As part of the IMS registration procedure subscriber data shall be loaded into S-CSCF. This data shall include a URL that identifies the location of where the configuration data (RUA DN) for the subscriber shall be stored.

2. 3rd party registration shall be performed to the ICCF as per TS 23.228 [4]

3. ICCF shall subscriber to the REG Event package per TS 23.228 [4]

4. ICCF shall receive a SIP NOTIFY per TS 23.228 [4] in response to the REG EVENT subscription. The SIP NOTIFY shall include a URL that identifies the location of where the configuration data (RUA DN) for the subscriber shall be stored per RFC 4483.

5. The ICCF allocates a unique RUA DN that has the properties that when received in a SIP INVITE from a MGCF in a future session set-up is ,the ICCF can retrieve the necessary call information from the initial SIP INVITE that indicated that the voice media is not conveyed via the IP-CAN. The RUA DN shall be stored at the location as identified by the URL received in the SIP NOTIFY.

6. The ICS UE shall subscribe to the Reg Event Package per TS 23.228 [4]

7. The ICS UE shall receive a SIP NOTIFY that includes a URL that identifies the location of where the configuration data (RUA DN) for the subscriber shall be stored per RFC 4483.

8. The ICS UE shall retrieve the configuration data from the URL as identified in step 7.

6.5.2.2 Origination

6.5.2.2.1 Calls established using CS bearers with use of I1-ps

Figure 6.5.2.2.1-1 provides an example flow for a call made by an ICS UE-A to the other end B with I1-ps.

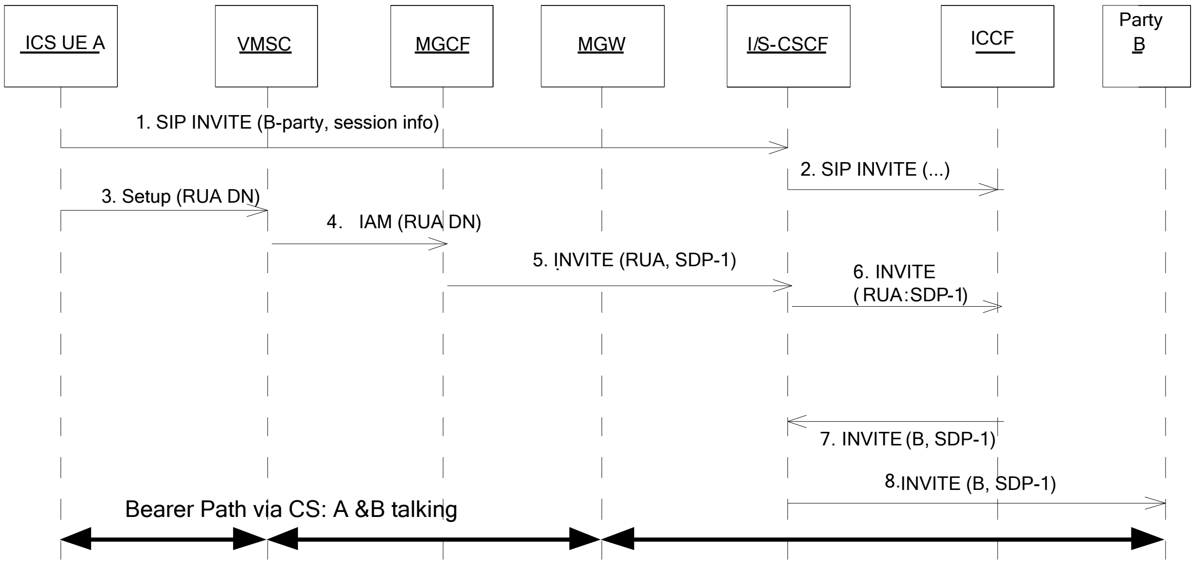


Figure 6.5.2.2.1-1: ICS UE origination: PS transport

1,2. ICS UE-A generates a SIP INVITE initiating a session toward the other end B, and sends it toward the home IMS. The SIP INVITE message has sufficient information needed at the RUA of the ICCF to correlate the session request with the Bearer Control Session established with the RUA of the ICCF in steps 3 through 6. It also contains information that the voice media is not converyed via the IP-CAN. The PCRF or P-CSCF needs to recognise that the PS media is not used for the voice component in order to avoid the PCC authorizing or provisioning PS bearer resources for them.

NOTE 1: The SIP Session Control Signalling is used to carry information that would be otherwise (i.e. in regular IMS procedures) carried in the same SIP dialog than voice media (examples: voice media is inactivated, location of the UE is carried in PIDF-LO element and SIP Call-Info header carries e.g. business card). In other words, the SIP Session Control Signalling is not used for SIP messages that are sent out of the SIP dialog of voice media (examples: REFER, SUBCSCRIBE, NOTIFY, PUBLISH, MESSAGE as these messages are typically sent out of the existing voice SIP dialog).

NOTE 2: How the UE selects when to use the voice SIP dialog also for other media will follow the requirements defined in the communication service framework in TS 23.228 [4], clause 4.13.

3. The ICS UE initiates standard CS procedures for establishing a CS originated call with the RUA of ICCF to establish the Bearer Control Signalling session with the RUA of ICCF. The UE-A sends a CS SETUP message to DN associated with the RUA of ICCF. The DN used by the UE to establish the Bearer Control Signalling session is either statically configured on the UE or assigned to the UE by the ICCF upon IMS Registration. The Bearer Control Signalling session is identified at the ICCF by the caller's identity for correlation with the Session Control Signalling session as there can be only one Bearer Control Signalling session for an ICS UE at any given time. If the ICS UE has multiple IMS sessions for voice, a single circuit bearer is alternated between the multiple sessions.

NOTE 3: The ISUP link might (especially in roaming situations) not be able to transfer enough information to correlate the SIP INVITE. Under such circumstances the ICCF may need to interact with a CAMEL Service.

NOTE 4: Steps 3-5 may occur in parallel to steps 1-2.

4. Standard VMSC procedure for CS origination.

5. Standard MGCF procedure for PSTN origination. The I-CSCF routes the INVITE based on the standard procedures "PSI based Application Server Termination (direct or –indirect) procedures in TS 23.228 [4]. S-CSCF selected may not be the same one as in step 1. The PSI routing points to the same RUA of the ICCF as in step 1 (e.g. with the use of dedicated PSI per user).

6. Standard IMS procedure to route the SIP INVITE message to the RUA of the ICCF is invoked as part of standard iFC processing at the S-CSCF.

7. The RUA of the ICCF the invokes a B2BUA, terminating the UE Leg and originating the RUA Leg for presentation of an IMS session toward Other end B on behalf of ICS UE-A. The RUA of the ICCF combines the SDP offers received for the Session Control Signalling and Bearer Control Signalling as one offer towards the other end B. The SDP offer used to establish the voice media send towards other end B follows the regular IMS procedures for a VoIP offer for voice media.

8. Standard IMS originated session processing at the CSCF.

NOTE 5: VCC and Originating Supplementary services (e.g. OCB, OIR), if any, are executed in the Application Servers in home IMS. These are not shown in the figure.

NOTE 6: VCC Application may anchor the SIP session before it leaves the originating IMS (as per regular Rel-07 VCC procedures), these are not shown in the figure.

NOTE 7: Once the B party answers to the SDP offer, the RUA of the ICCF needs to modify also the leg towards the MGCF. The RUA of the ICCF sends a Re-INVITE to MGCF, MGCF modifies the remote MGW resource reservation accordingly. The Re-INVITE modifies the remote IP address and codec of MGW in order to send and receive RTP stream from/to B party. This is not shown in the figure.

NOTE 8: E.164 resources are considered scarce. Assigning multiple E.164 numbers to each ICS UE may not be possible for some operators.

6.5.2.2.2 Calls established using CS bearers with use of I1-ps (Dynamic RUA DN allocation)

When an ICS UE establishes a call using CS bearers with use of I1-ps, it will setup a call using an RUA DN that has the properties that when received, the ICCF can retrieve the necessary call information stored upon receipt of a first INVITE. This RUA DN is either provisioned on demand (i.e. during a call setup) or at another time (e.g. during registration). In order to conserve E.164 numbers, such an RUA DN is likely to be released after a (configurable) period. In such a case, another RUA DN would have to be used.

Figure 6.5.2.2.2-1 provides an example flow for a call made by an ICS UE-A to the other end B with I1-ps, where the ICS UE-A uses a dynamic RUA DN).

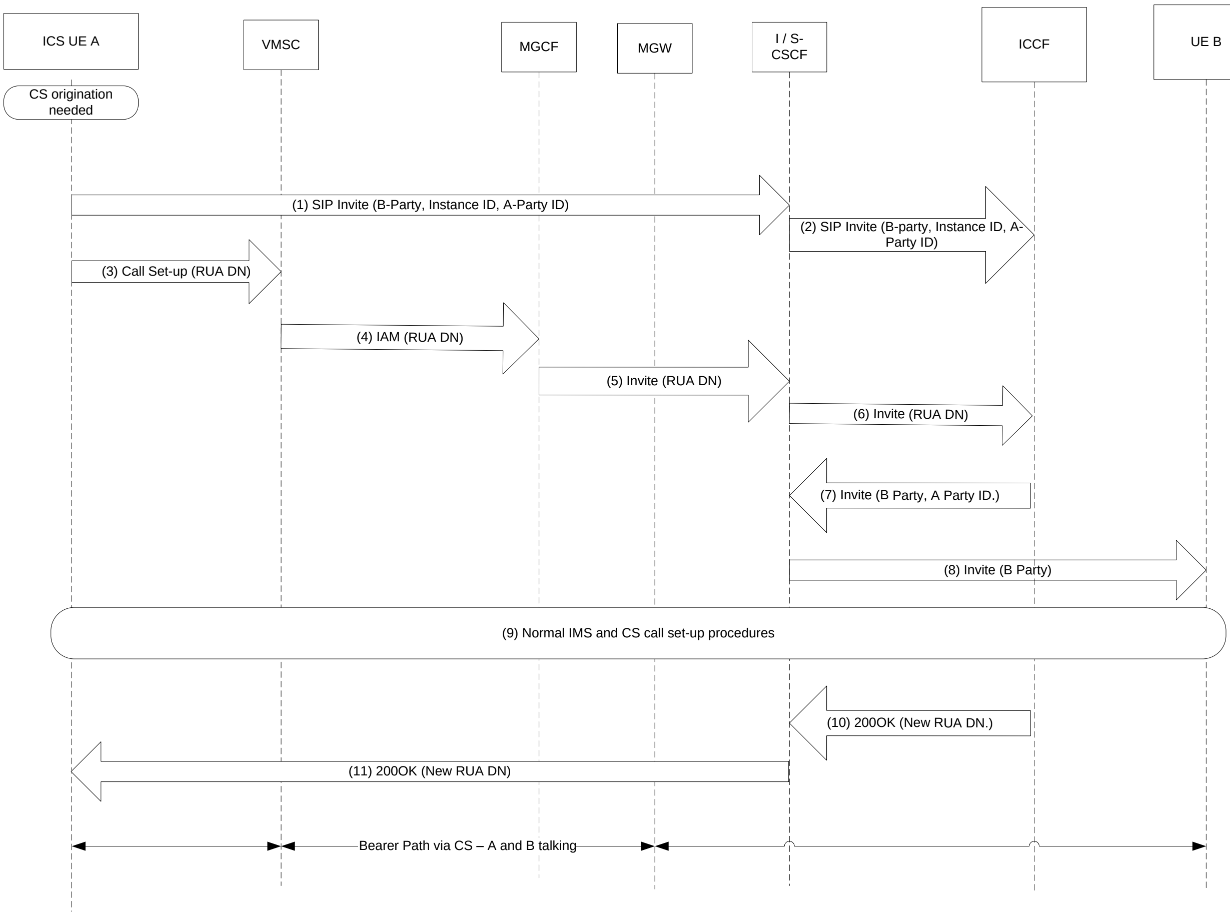


Figure 6.5.2.2.2-1: ICS UE origination: PS transport

1,2. ICS UE-A generates a SIP INVITE initiating a session toward the other end B, and sends it toward the home IMS. The SIP INVITE message has sufficient information needed at the RUA to set up the outgoing SIP INVITE in step 6. The SIP INVITE also contains information that the voice media is not conveyed via the IP-CAN. The PCRF or P-CSCF needs to recognise that the PS media is not used for the voice component in order to avoid the PCC authorizing or provisioning PS bearer resources for them.

Editor's Note 1: The details on correlating the session request with the Bearer Control Session established with the RUA are FFS.

NOTE 1: The SIP Session Control Signalling is used to carry information that would be otherwise (i.e. in regular IMS procedures) carried in the same SIP dialog than voice media (examples: voice media is inactivated, location of the UE is carried in PIDF-LO element and SIP Call-Info header carries e.g. business card). In other words, the SIP Session Control Signalling is not used for SIP messages that are sent out of the SIP dialog of voice media (examples: REFER, SUBCSCRIBE, NOTIFY, PUBLISH, MESSAGE as these messages are typically sent out of the existing voice SIP dialog).

NOTE 2: How the UE selects when to use the voice SIP dialog also for other media will follow the requirements defined in the communication service framework in TS 23.228 [4], clause 4.13.

3. The ICS UE initiates standard CS procedures for establishing a CS originated call with the RUA of ICCF to establish the Bearer Control Signalling session with the RUA of ICCF. The UE-A sends a CS SETUP message to the RUA DN that was received in a previous 200Ok (see step 10) for session establishment or as a result of registration

4. Standard VMSC procedure for CS origination.

5. Standard MGCF procedure for PSTN origination. The I-CSCF routes the INVITE based on the standard procedures "PSI based Application Server Termination (direct or –indirect) procedures in TS 23.228 [4]. S-CSCF selected may not be the same one as in step 1. The PSI routing points to the same RUA of the ICCF as in step 1.

6. Standard IMS procedure to route the SIP INVITE message to the RUA of the ICCF is invoked as part of standard iFC processing at the S-CSCF.

7. Upon receipt of the SIP INVITE, the RUA will use the received RUA DN to correlate the original SIP INVITE in step 1 with the received SIP INVITE. B2BUA will be invoked, terminating the UE Leg and originating the RUA Leg for presentation of an IMS session toward Other end B on behalf of ICS UE-A. The RUA of the ICCF combines the SDP offers received for the Session Control Signalling and Bearer Control Signalling as one offer towards the other end B. The SDP offer used to establish the voice media send towards other end B follows regular IMS procedures.

Editor's Note 3: the behaviour of the ICCF upon receipt of a released RUA DN is FFS.

8-9. Standard IMS originated session processing at the CSCF.

10-11. Upon receipt of the SIP 200Ok from UE-B, the ICCF allocates a unique RUA DN that has the properties that when received in a 2nd SIP INVITE in a future session set-up (step 6) the ICCF can retrieve the necessary call information from the initial SIP INVITE received to initiate a session (step 2). The RUA DN is sent back to ICS UE-A in SIP 200OK per RFC 4483.

NOTE 4: VCC and Originating Supplementary services (e.g. OCB, OIR), if any, are executed in the Application Servers in home IMS. These are not shown in the figure.

NOTE 5: VCC Application may anchor the SIP session before it leaves the originating IMS (as per TS 23.206 [x]), these are not shown in the figure.

NOTE 6: Once the B party answers to the SDP offer, the RUA of the ICCF needs to modify also the leg towards the MGCF. The RUA of the ICCF sends a Re-INVITE to MGCF, MGCF modifies the remote MGW resource reservation accordingly. The Re-INVITE modifies the remote IP address and codec of MGW in order to send and receive RTP stream from/to B party. This is not shown in the figure.

6.5.2.3 Termination

6.5.2.3.1 Calls established using CS bearers with use of I1-ps

The figure 6.5.2.3-1 provides an example flow for a call destined to an ICS UE when the PS transport alternative for ICCC is used to support the setup of terminating sessions.

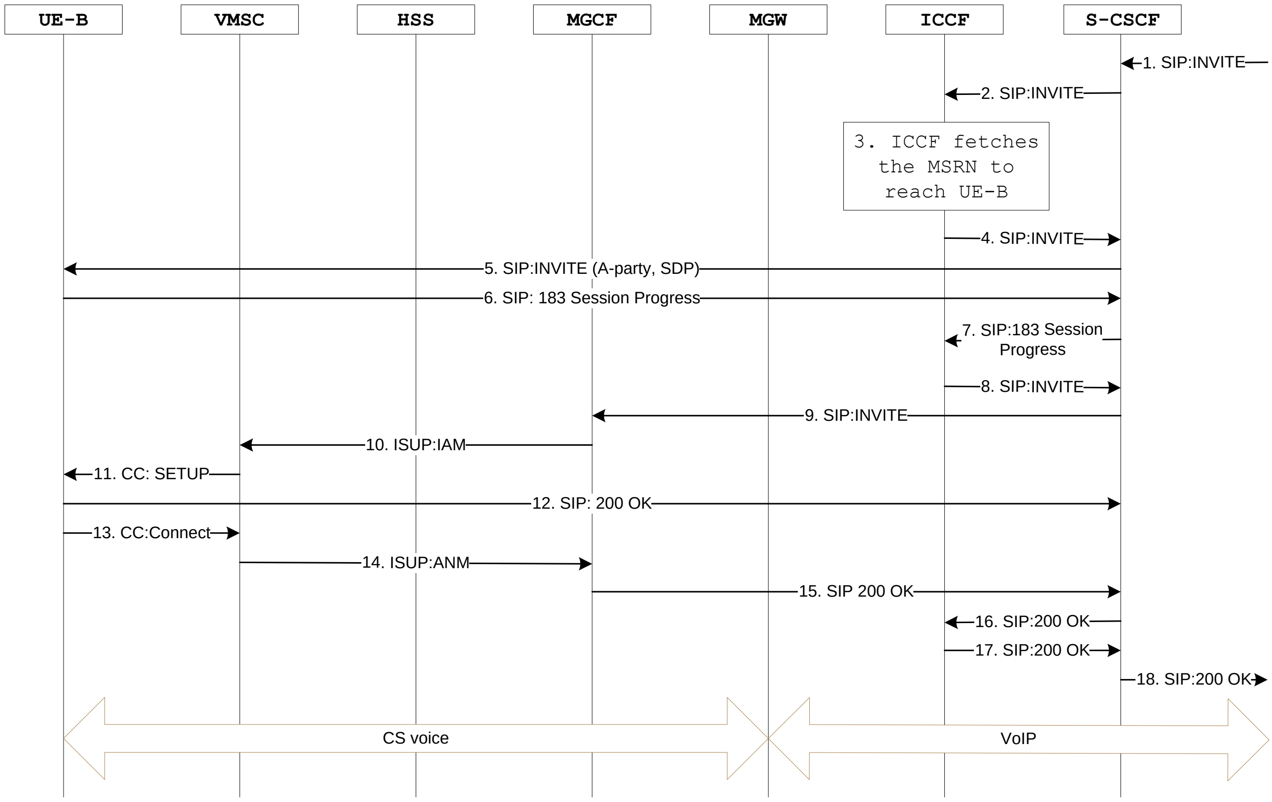


Figure 6.5.2.3.1-1: ICS UE termination using CS bearers with use of I1-ps

1. An incoming SIP INVITE is received at the S-CSCF of the B party.
2. The S-CSCF forwards the INVITE to the ICCF.
3. The ICCF fetches a number such as MSRN if a media connection is not already established to UE-B.

Editor's Note 1: The type of number to be used and how the number is allocated is FFS. The Relationship and interworking with DSFVCC is FFS.

4. The ICCF generates an INVITE to the other end B indicating the use of a CS bearer.
5. The INVITE is sent to the UE-B via I1-ps. It also contains information that the voice media is not conveyed via the IP-CAN. The PCRF or P-CSCF needs to recognise that the PS media is not used for the voice component in order to avoid the PCC authorizing or provisioning PS bearer resources for them.
6. UE-B responds with a 183 Session Progress. The user is not alerted.
7. The ICCF receives the 183 Session Progress from the UE.
8. The ICCF sends an INVITE (tel URI = MSRN) to the S-CSCF.
9. The Request URI has been modified, the S-CSCF skips further service execution and routes the call towards the CS domain.
10. The MGCF sends an IAM to the VMSC.
11. VMSC sends SETUP to UE-B.
12. The user accepted the call, the UE sends the 200 OK.
13. UE-B responds with a CONNECT message.
14. VMSC responds with ANM towards the IMS.
15. MGCF sends the 200 OK to the S-CSCF.

16. S-CSCF forwards the 200 OK to the ICCF.

17. The ICCF sends a 200 OK as response to the INVITE in step 2 towards the S-CSCF.

18. S-CSCF forwards the 200 OK towards the A party.

6.5.2.3.2 Using CS Origination procedures to set up the Bearer Control Signalling session

The figure 6.5.2.3.2-1 provides an example flow for a call destined to an ICS UE when the PS transport alternative for ICCC is used to support the setup of terminating sessions using standard CS Origination procedures for setting up Bearer Control Signalling session.

NOTE: The un-bolded steps are related to the Session Control Signalling set-up procedures. The bolded steps are related to the Bearer Control Signalling session set-up procedures.

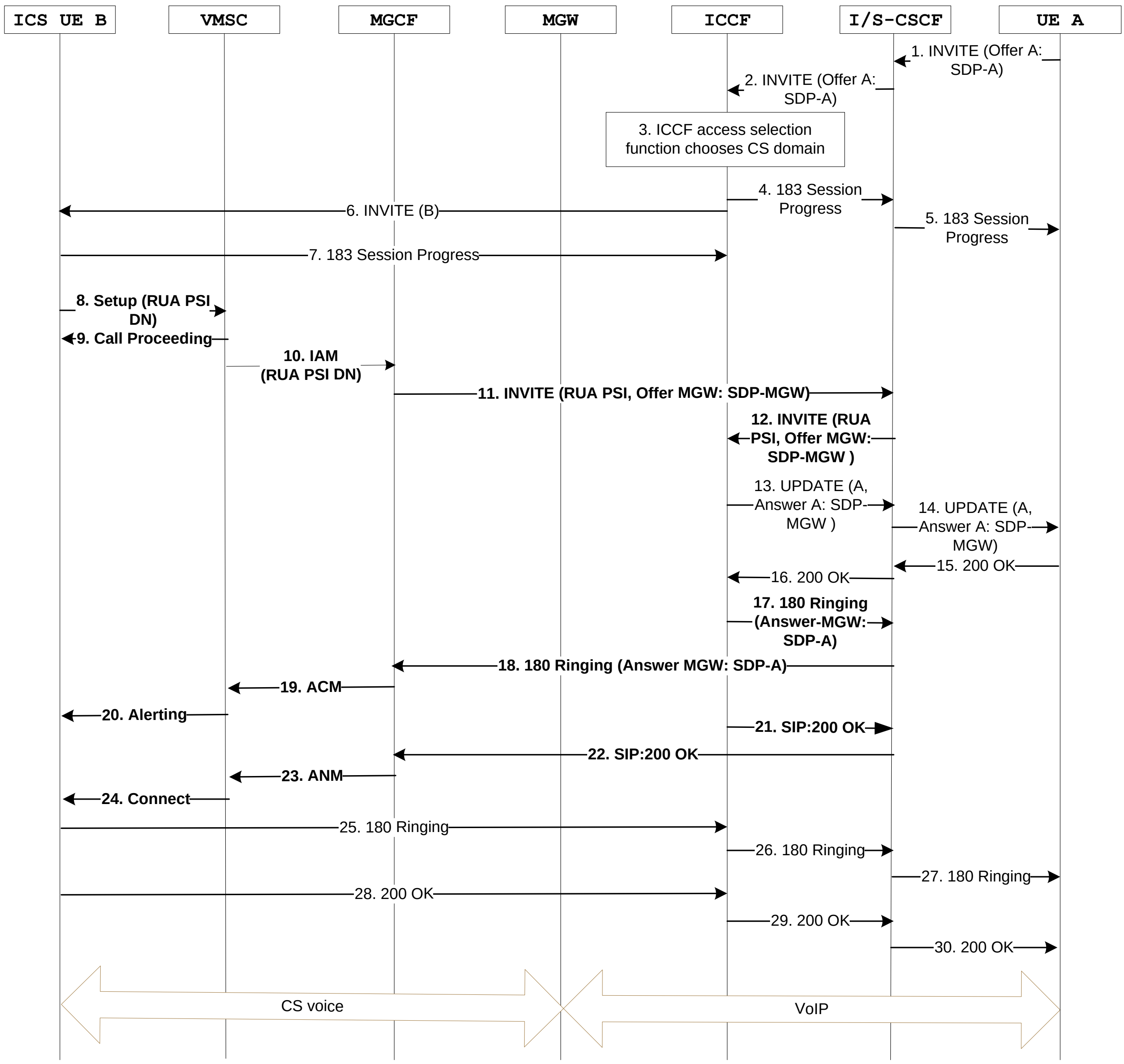


Figure 6.5.2.3.2-1: ICS UE termination using CS bearers using CS Origination procedures with use of I1-ps

1. An incoming SIP INVITE is received at the S-CSCF of the B party with an Offer containing SDP from the A-party.

2. The S-CSCF forwards the INVITE to the ICCF.

3. The ICCF performs access selection and chooses the CS domain.

4. The ICCF sends a 183 Session Progress back to the S-CSCF.

5. The S-CSCF sends the 183 Session Progress to the A-party.

6. The ICCF (acting as a B2BUA) establishes a session over I1-ps by sending an INVITE to the B-party. This INVITE contains an indication to inform the UE to initiate the CS bearer establishment procedure. The INVITE also contains a dynamic RUA PSI to enable the ICCF to later on correlate the outgoing session control signalling with the incoming bearer control signalling.

7. UE-B responds with a 183 Session Progress.

8. UE-B sends a SETUP message to the VMSC to establish the Bearer Control Signalling session. The SETUP message includes the RUA PSI DN as the called party number. This will establish the circuit voice bearer between the UE and IMS.

9. The VMSC responds with a call proceeding message and begins to set up the Bearer Control Signalling session.

10. The VMSC processes the SETUP message and sends an IAM to the MGCF. The SETUP message contains the RUA PSI DN. [NOTE: Standard VMSC procedures for CS origination]

11. The MGCF performs a setup of the MGW and creates an INVITE with an Offer containing the SDP from the MGW. The INVITE is sent to the ICCF (via the I/S-CSCF). [NOTE: Standard MGCF procedure for PSTN origination]

12. The I/S-CSCF forwards the INVITE to RUA of ICCF.

13. The ICCF sends an UPDATE to the S-CSCF with an Answer to the Offer from the A-party, containing the SDP from the MGW.

14. The S-CSCF forwards the UPDATE to the A-party.

15. The S-CSCF receives a 200 OK to the UPDATE.

16. The S-CSCF forwards the 200 OK to the ICCF.

17. The ICCF sends a 183 Session Progress to S-CSCF with an Answer to the Offer from the MGW, containing the SDP from the A-party.

18. The S-CSCF sends the 183 Session Progress to the MGCF.

19. The MGCF creates an ACM and sends it to the VMSC.

20. Alerting is sent from the VMSC to UE-B.

21. The ICCF sends a 200 OK to S-CSCF in response to the INVITE in Step 12.

22. The S-CSCF forwards the 200 OK to the MGCF.

23. The MGCF creates an ANM and sends it to the VMSC.

24. The VMSC sends a Connect to UE-B. The set up of the bearer is complete.

25. User alerting at UE-B; UE-B sends 180 Ringing to the ICCF.

26. The ICCF forwards the 180 Ringing to the S-CSCF for communicating toward UE-A.

27. The S-CSCF forwards the 180 Ringing to UE-A.

28. User answer at UE-B; UE-B sends a 200 OK to the ICCF for the INVITE in Step 6.

29. The ICCF forwards the 200 OK to the S-CSCF for communicating toward UE.

30. The S-CSCF forwards the 200 OK towards to the A-party.

NOTE: Steps 21-24 above related to the Bearer Control Session set-up procedures may alternatively be sent after the 200 OK has been sent from the ICCF towards UE-A.

6.5.2.4 Mid-call services

6.5.2.4.1 Calls established using CS bearers with use of I1-ps

Figure 6.5.2.4.1-1 provides an example flow for a call made by an ICS UE-A to the other end C after holding the other end B.

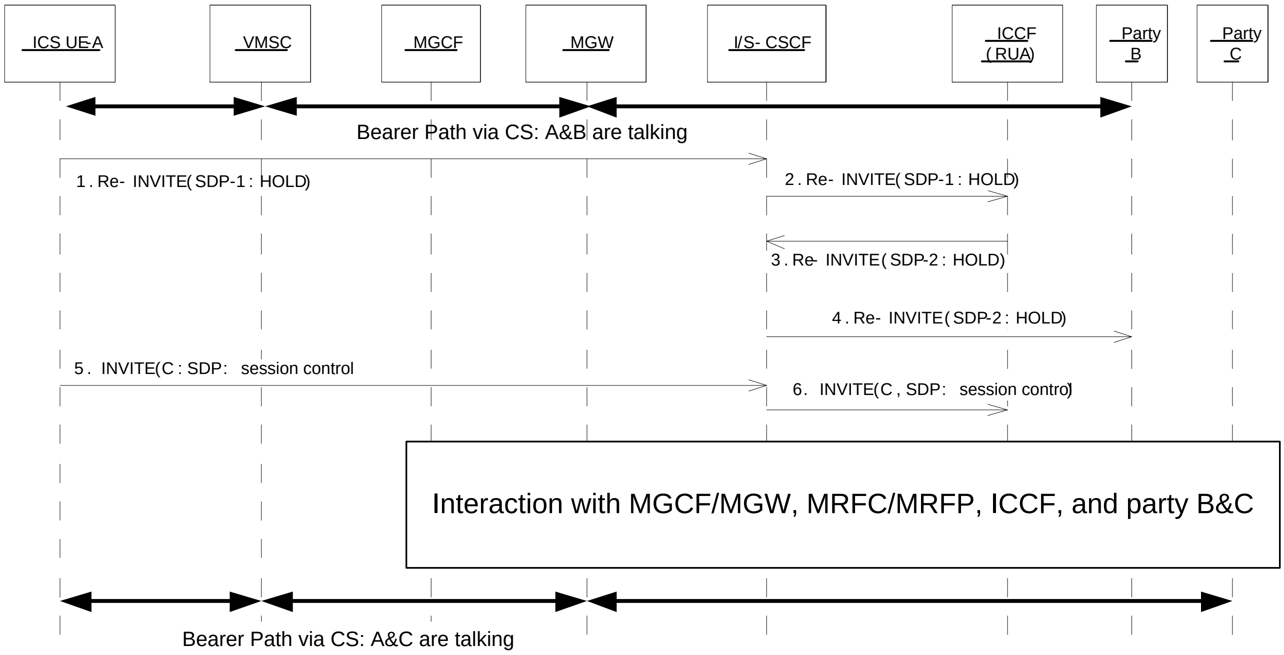


Figure 6.5.2.4.1-1: ICS UE mid-call service-A holds B, calls C: PS transport

1. ICS UE-A Holds Call to the other end B by sending a "HOLD indication" via SIP Re-INVITE message to RUA.
2. Standard IMS procedure to communicate the SIP message to the RUA.
3. RUA composes an SDP offer towards the B party to put the voice media on hold. SDP offer/answer for hold follows the standard MMTel procedures.
4. Standard IMS procedure to communicate the SIP message to the B party.
5. UE-A generates a SIP INVITE initiating a session toward the other end C, and sends it toward the P-CSCF and I/S-CSCF. The SIP INVITE message has sufficient information needed at the RUA to correlate the session request with the Bearer Control Session established previously with the RUA. The SDP offer in INVITE indicates to the RUA that the UE-A is willing to establish a voice call via CS bearer, and the actual details for voice media will be offered by MGCF.
6. Standard IMS procedure to communicate the SIP message to the RUA.

NOTE: Annex B describes two ways to handle media/RTCP report of the held party (party-B) and the media connection toward ICS UE-A and party C. As the result of the procedure in Annex B, ICS UE-A & party C are talking.

6.5.2.5 Domain Transfer to CS

The following figure provides an example flow for Domain Transfer to CS when the user is engaged in a held voice session with the other end B and an active voice session with the other end C, when using ICS UE with I1-ps and the AS approach for the ICCF.

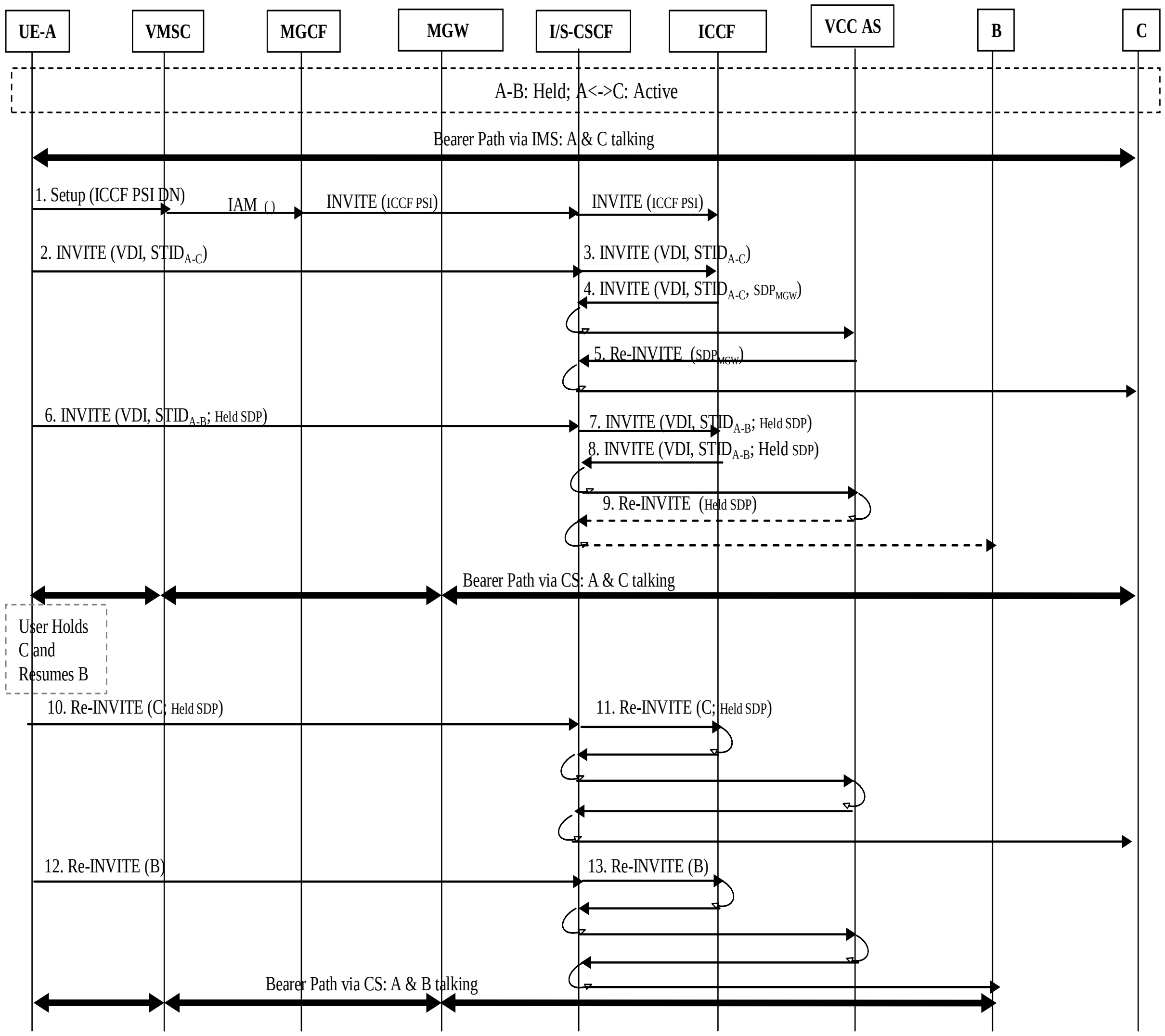


Figure 6.5.2.5-1: Domain Transfer to CS

As a starting point, user A has two ongoing voice calls, one in Active state with C, and another on Hold with B. This means the UE-A has two ongoing SIP dialogs which both include Session and Bearer Control Signalling, one dialog towards UE-C and another UE-B. Neither of them traverses via RUA. Procedures to establish such dialogs follow the procedures in TS 24.173 [5].

1. Upon detection of conditions requiring Domain Transfer, UE-A initiates the Domain Transfer to CS by establishing a Bearer Control Path with RUA of the ICCF through the VMSC.

2. UE-A sends an INVITE over GPRS transport requesting the Domain Transfer of the active session with the other end C. If the UE received the STI from the DTF during session establishment then the domain transfer request contains an STI, otherwise, only the VDI, as defined in TS 23.206 [3], is used. The transfer session is created in the UE with the current state information of the session being transferred.

NOTE 1: Due to change of the IP address, the UE needs to re-register to IMS over the transferred-in access before it is able to send INVITE to VDI for Domain Transfer.

3. Standard originating iFC execution processing at the CSCF results in execution of the RUA.

   The Bearer Control Signalling session established in the INVITE sent by S-CSCF in step 1 is identified at the ICCF by the caller's identity for correlation with the Session Control Signalling session transferred via the INVITE sent by the S-CSCF in step 3 as there can be only one Bearer Control Signalling session for an ICS UE at any given time. If the ICS UE has multiple IMS sessions for voice, a single circuit bearer is alternated between the multiple sessions.

NOTE 2: The ISUP link might (especially in roaming situations) not be able to transfer enough information to correlate the SIP INVITE. Under such circumstances the ICCF may need to interact with a CAMEL Service.

4. The INVITE is extended toward the CSCF by the RUA after combining with the media components established via the MGCF/MGW in Step 1. The INVITE is processed at the CSCF using standard IMS procedures and the DTF of the VCC AS is invoked as part of standard iFC processing at the CSCF, as in regular Rel07 VCC procedures.

5. The DTF uses Access Leg Update procedure to update the Access Leg and the Remote Leg associated with the other end C's session. The DTF extends a Re-INVITE toward the other end C for update of SDP. Standard Domain Transfer procedure at CSCF for extending the Re-INVITE toward the other end C. A CS bearer path is re-established between the UE-A and the other end C via MGW as a result of the Re-INVITE processing at the other end C. The DTF and UE subsequently release the other end C's Access Leg previously established via IMS.

6. UE-A sends an INVITE over GPRS with held SDP requesting the Domain Transfer of the held session with the other end B. If the UE received the STI from the DTF during session establishment then the domain transfer request contains an STI, otherwise. only the VDI, as defined in TS 23.206 [3], is used.

Step 6 may execute in parallel with procedures initiated by Step 2.

7. Standard processing at the CSCF for directing toward RUA.

8. The INVITE is extended back to the CSCF by the RUA.

9. The INVITE is processed at the CSCF using standard procedures and the DTF is invoked as part of standard iFC processing at the CSCF.

   The DTF identifies this session as a held session and updates the Access Leg with the information received in the INVITE. This is needed for completion of the call control signalling path between the UE and the other end B via the Access Leg established with DTF via CS domain. An Access Leg update of the held session toward the other end B, in other words, the updating of the Remote Leg of the B's session may not be necessary at this point since active media is not needed for this session until B is "Resumed". However, this procedure may be carried out in parallel to completion of Domain Transfer of the active session toward the other end C as a UE implementation option.

NOTE 3: The DTF must send re-INVITE also for the held session for continuation of the RTCP reports if previously being sent to the UE port.

   The DTF and UE subsequently release the other end B's Access Leg previously established via IMS, as in regular Release 7 VCC procedures. This completes the Domain Transfer procedures.

10. After the Domain Transfer the User A may want to swap the calls between B and C. User A subsequently Holds the other end C and Resumes the other end B. A Re-INVITE is sent with held SDP for C's session is sent by UE-A.

11. The Re-INVITE with held SDP is communicated to the other end C and the session is Held.

12. A Re-INVITE for Resuming B's session is also sent by the UE-A.

13. The Re-INVITE is processed at the CSCF and the DTF for Access Leg Update of B's session resulting in establishing and/or updating of an IP bearer between the other end B and the UE-A (bearer is established if it was not established in the Held state as part of procedure initiated in Step 9; bearer is updated if it was established in the Held state as part of procedure initiated in Step 9).

6.5.2.5.3 Assignment of Session Transfer Identifier

See clause 6.6.2.

6.5.2.6 I1-ps to I1-cs fallback

The Figure 6.5.2.6-1 provides an example flow for Session Control Signalling Path handover from I1-ps to I1-cs.

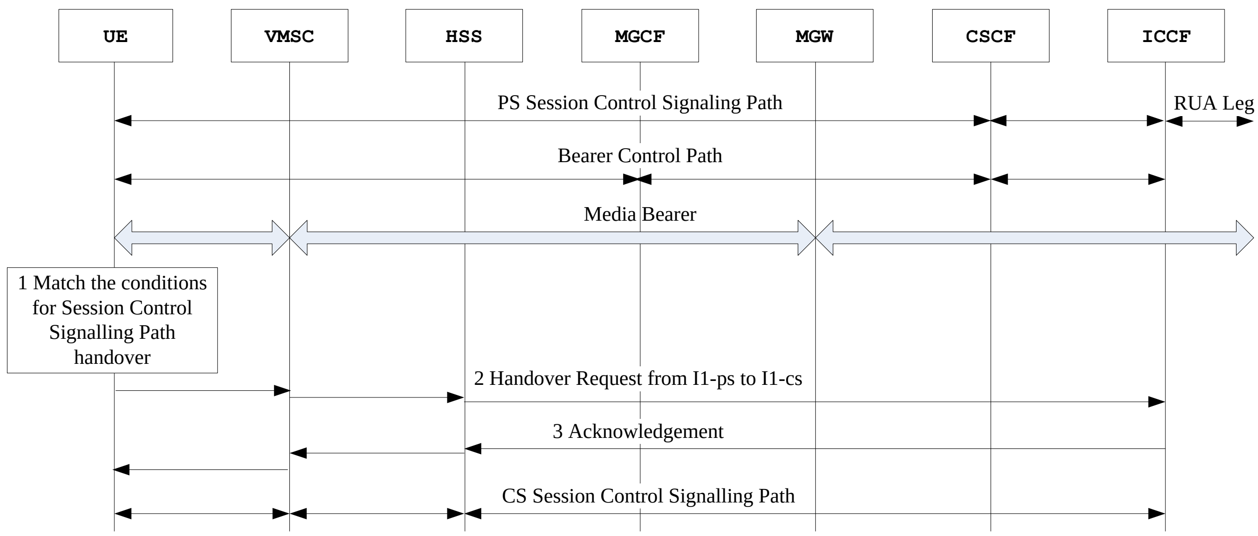


Figure 6.5.2.6-1: Session Control Signalling Path handover from I1-ps to I1-cs

1. Upon detecting that PS access can not satisfy the signalling communication requirement for PS session control signalling path or becomes unavailable, ICS UE triggers the session control signalling path handover.

2. ICS UE sends the session control signalling path handover request with PS session control signalling path information to ICCF over I1-cs.

3. ICCF identifies the PS session control signalling path to be handed over, and decides to accept the handover request, then returns the acknowledgement to ICS UE.

After the above steps, the session control signalling path is handed over from the PS access to the CS access, and the subsequent session control signalling will be over I1-cs.

If the ICCF detects that the ICS UE is not reachable by PS and the UE has not already made a fallback to I1-cs, the ICCF shall clear all session related to the user currently being controlled over I1-ps.

6.5.3 I1-ps handover scenarios

6.5.3.1 Introduction

When in CS coverage with simultaneous PS access available, e.g. UTRAN, the ICS UE may use IMS SIP signalling over PS bearers for enablement of I1-ps when support of the full duplex speech component of the IMS multimedia telephony service is not available over PS bearer. However, it is possible that session and media control signalling transport over the PS access in an ongoing session becomes unavailable while the CS access in the same session is still available. Under such circumstance, to keep the control signalling continuity, the signalling transport is supposed to be handed over from PS access to CS access.

6.5.3.2 Handovers within 3G access for IMS sessions using CS voice bearer established with I1-ps

Figure 6.5.3.2-1 below provides signalling and bearer paths for an IMS session using CS voice bearer established in 3G access with I1-ps before and after handover to another 3G access.

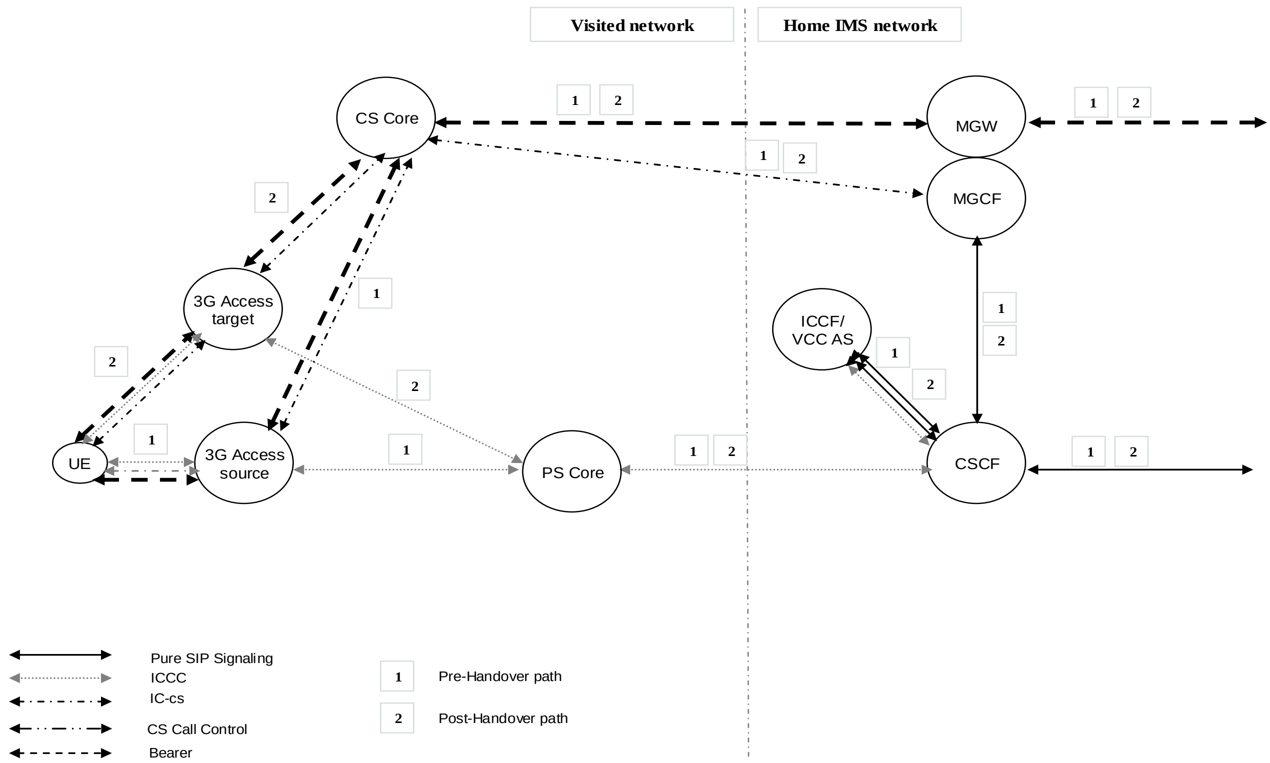


Figure 6.5.3.2-1: Handovers from 3G to 3G with I1-ps

Standard UMTS Relocation procedures are used for handover of the Session Control Signalling Path and the CS Access Signalling along with the associated circuit voice bearer to the target access network. The coordination between the two Iu signalling connections, one with the PS core network for Session Control Signalling Path and the other with the CS core network for CS Access Signalling is performed by the target access network as described in TS 25.413 [10].

The I1-ps is maintained upon handover to the target 3G access network.

6.5.3.3 Handovers of IMS sessions using CS voice bearer established in 3G access with I1-ps to 2G access

6.5.3.3.1 PS Domain available after HO to 2G access

Figure 6.5.3.3.1-1 below provides signalling and bearer paths for an IMS session using CS voice bearer established in 3G access with I1-ps before and after handover to 2G access where PS domain is available (e.g. DTM).

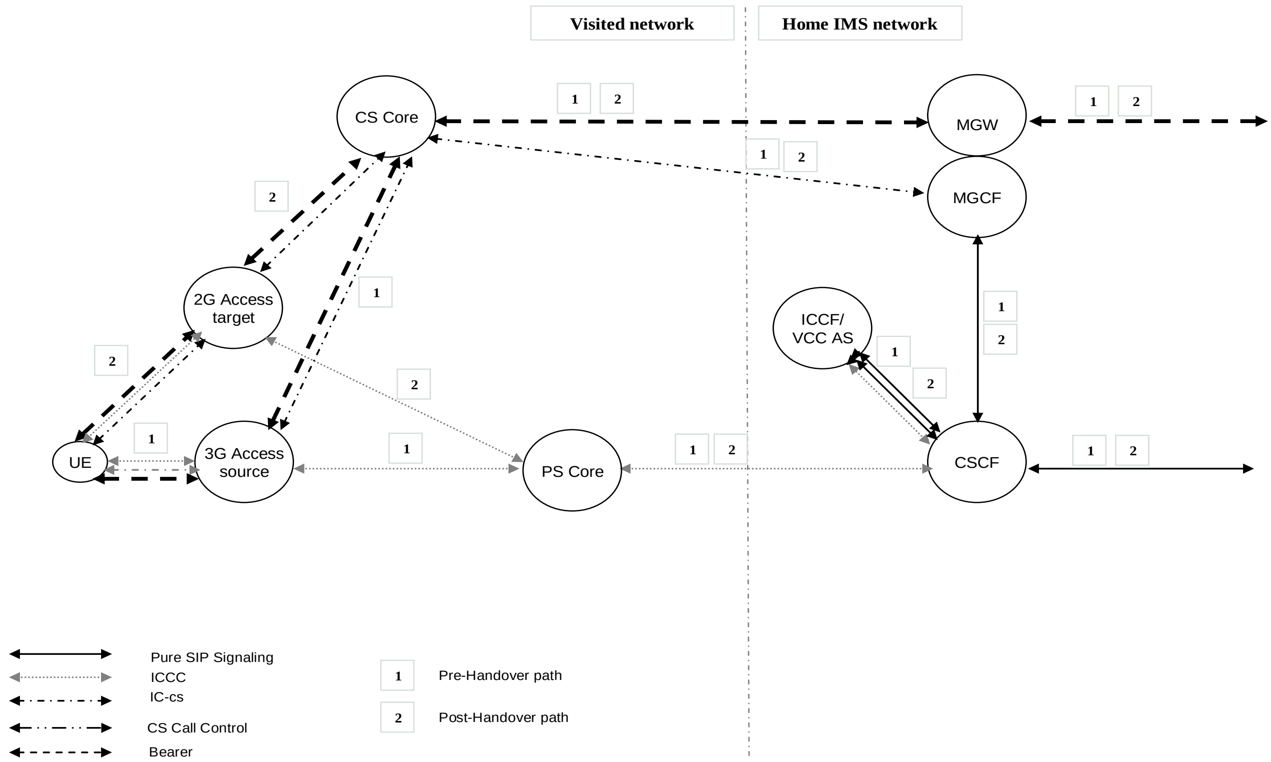


Figure 6.5.3.3.1-1: Handovers from 3G to 2G with I1-ps

Standard Rel-07 DTM Handover procedures, TS 43.055 [12], defined for DTM/3G handovers are used for handover of the Session Control Signalling Path and the CS Access Signalling along with the associated circuit voice bearer to the target access network.

The I1-ps is maintained upon handover to the target 2G access network.

6.5.3.3.2 PS Domain is not available after HO to 2G access

Figure 6.5.3.2.2-1 below provides signalling and bearer paths for an IMS session using CS voice bearer established in 3G access with I1-ps before and after handover to any 2G access where PS domain is not possible (e.g. non-DTM cell at target access).

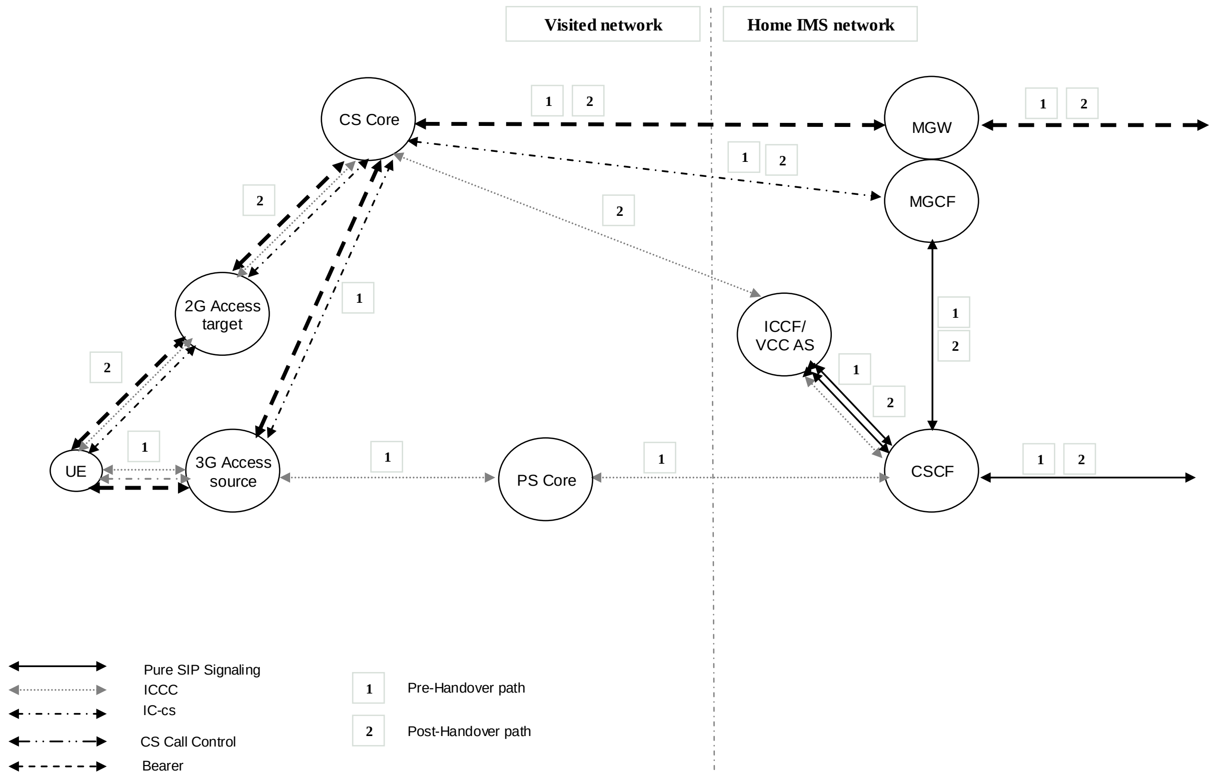


Figure 6.5.3.3.2-1: Handovers from 3G to 2G with I1-ps

Standard CS handover procedure, TS 23.009 [11], is used to relocate the CS Access Signalling and the associated circuit voice bearer from the source 3G access to target 2G access. Upon completion of the CS handover of the CS Access Signalling and the associated circuit voice bearer to the target 2G access, the UE sends a handover notification message using ICCP to establish the I1-cs for the Session Control Signalling Path with the ICCF.

The I1-cs is used for session control signalling post handover to 2G access.

6.5.3.4 Handovers of IMS sessions using CS voice bearer established in 2G access with I1-ps to 2G/3G access

6.5.3.4.1 PS Domain is available after HO to target access

Figure 6.5.3.4.1-1 below provides signalling and bearer paths for an IMS session using CS voice bearer established via GPRS EDGE with DTM using I1-ps before and after handover to any 2G/3G access where PS domain is available (e.g. DTM or Multi-RAB).

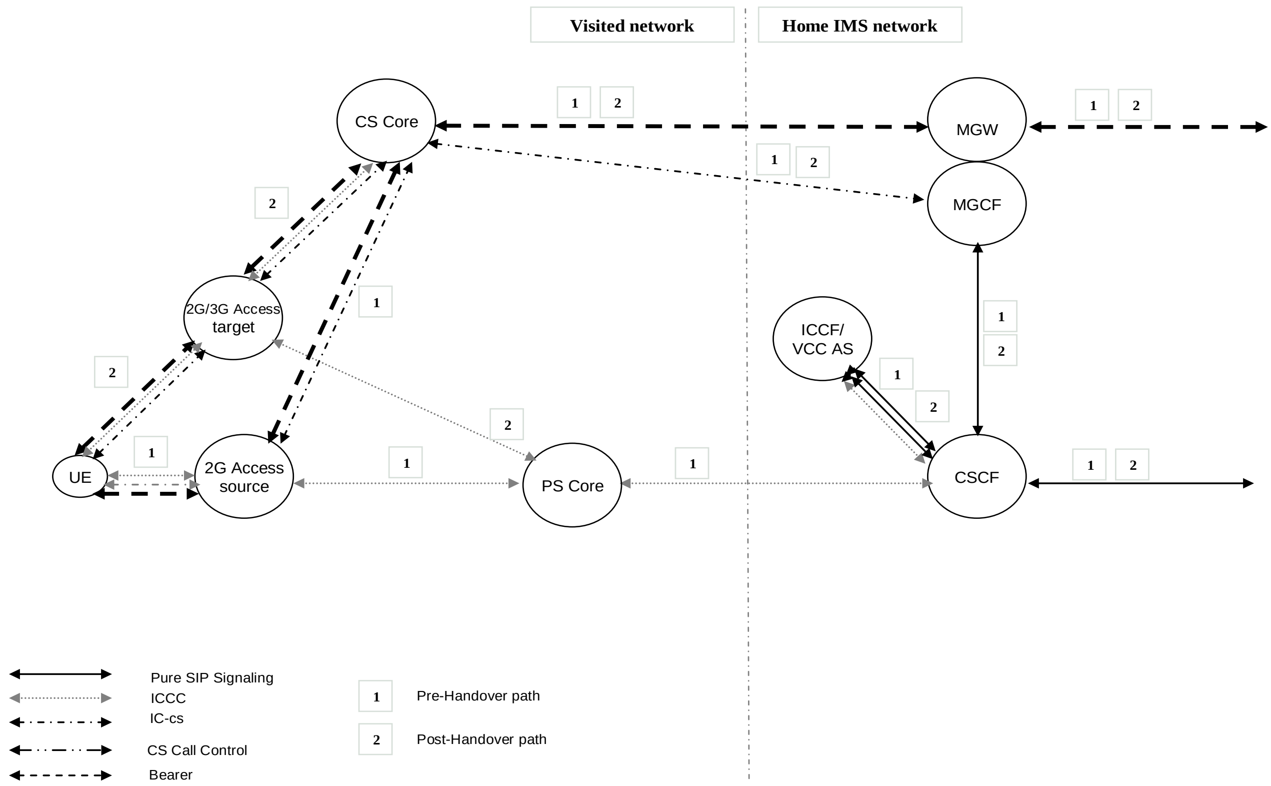


Figure 6.5.3.4-1: Handovers from 2G to 2G/3G with I1-ps

Standard Rel-07 DTM Handover procedures, TS 43.055 [12], defined for DTM/3G handovers are used for handover of the Session Control Signalling Path and the CS Access Signalling along with the associated circuit voice bearer to the target access network.

The I1-ps is maintained upon handover to the target 2G access network.

6.5.3.4.2 PS Domain is not available after HO to target access

Figure 6.5.3.4.2-1 below provides signalling and bearer paths for an IMS session using CS voice bearer established via GPRS EDGE with DTM using I1-ps before and after handover to any 2G/3G access.

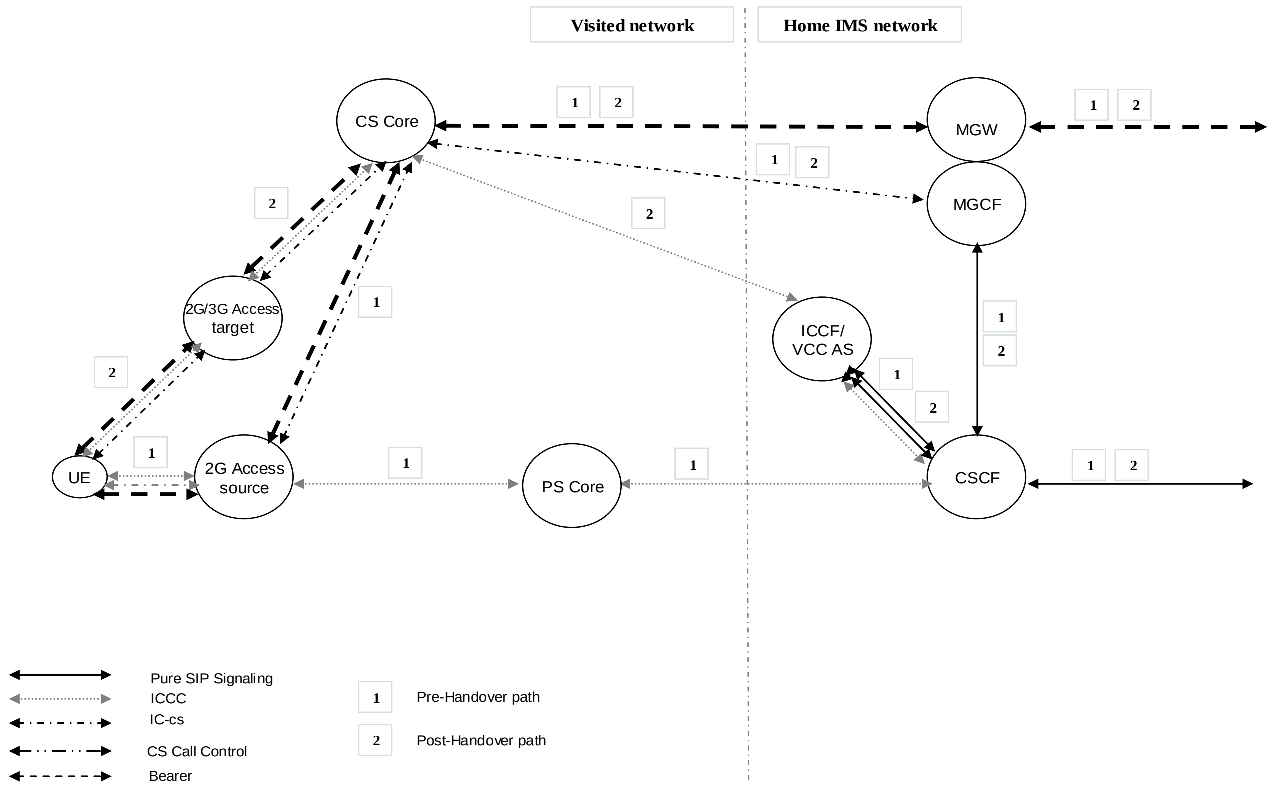


Figure 6.5.3.4.2-1: Handovers from 2G to 2G/3G with I1-ps

Standard CS handover procedure, TS 23.009 [11] is used to relocate the CS Access Signalling and the associated circuit voice bearer from the source GPRS EDGE capable access of DTM to target 2G/3G access. Upon completion of the CS handover of the CS Access Signalling and the associated circuit voice bearer to the target 2G/3G access, the UE sends a handover notification message using ICCP to establish the I1-cs for the Session Control Signalling Path with the ICCF.

The I1-cs is used for session control signalling after handover from the GPRS EDGE access capable of DTM to any 2G/3G access.

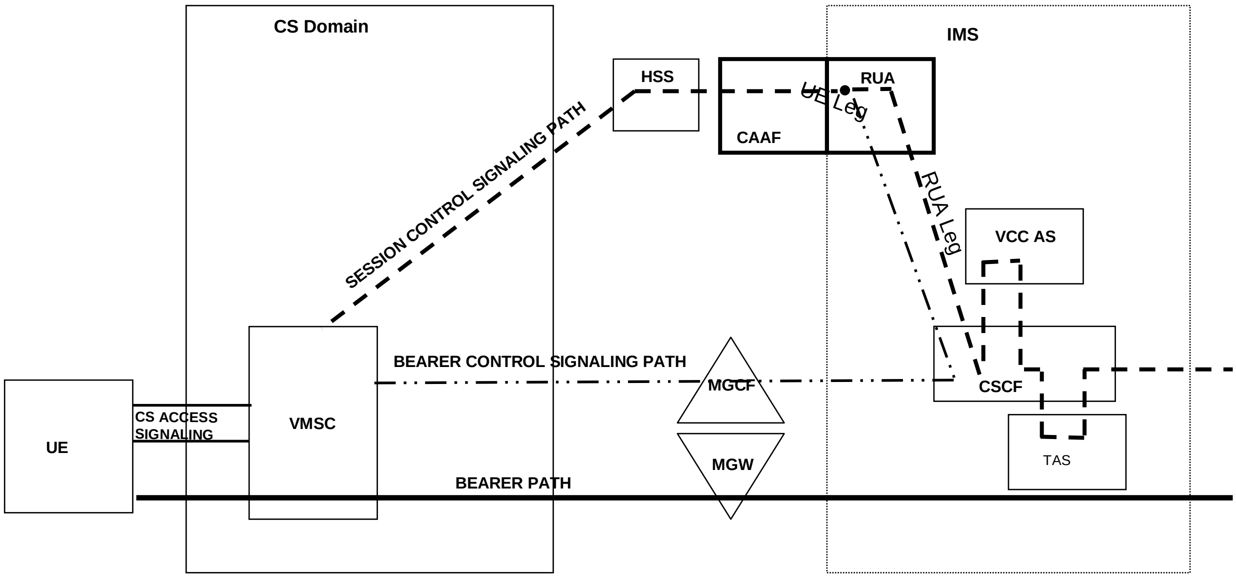
6.6 Architectural alternative: I1-cs Application Server approach

6.6.1 Signalling and bearer architecture for full duplex speech over CS access

6.6.1.1 Calls established using CS bearers

6.6.1.1.1 Introduction

The Signalling/bearer paths for an IMS session established via CS access with USSD transport of the I1-cs are described in figure below.



NOTE: gsmSCF used for redirection of CS calls and USSD processing is not shown for brevity.

Figure 6.6.1.1.1-1: Signalling/Bearer Paths-USSD transport

USSD is used for transport of the I1-cs communicating the Session Control Signalling via the VMSC to the CAAF. The CAAF performs necessary adaptation when relaying the Session Control Signalling to RUA which presents SIP UA behaviour on behalf of the UE toward IMS.

The UE establishes a Bearer Control Signalling path with the RUA by establishing a CS call toward the RUA. In parallel, it establishes the I1-cs through the Session Control Signalling Path with the CAAF and RUA using USSD. The Bearer Control Signalling and Session Control Signalling stimuli are combined at the RUA for presentation of SIP UA behaviour for establishment of an IMS session.

The UE maintains the key elements of the SIP/SDP state machine and the RUA also maintains a copy of the state data when present in the session path.

Use of the I1-cs for session setup for this model enables the capability to provide all services exclusively by IMS.

6.6.2 Assignment of Session Transfer Identifier

The DTF assigns a Session Transfer Identifier on the VCC's Access Leg when it receives an originating session request from the ICS UE or a session request towards ICS UE while there is another ongoing session (or sessions) for that IMPU, and transports it to the ICS UE (possibly through the ICCF when the UE is using CS access). The STI assigned by the DTF shall be different from the VDN/VDI. Similarly, if multiple UEs share the same IMPU, sessions initiated by the second (and subsequent) UEs will be assigned STI.

When the UE initiates a session transfer request, it includes the received Session Transfer Identifier and/or VDN/VDI (possibly through the ICCC to the DTF), the DTF uses the Session Transfer Identifier to identify which session will be transferred.

NOTE: If the UE did not receive the STI from the DTF then the domain session transfer request does not contain an STI. In this case the VDN/VDI, as defined in TS 23.206 [3], is used.

6.6.2a Registration

6.6.2a.1 IMS Registration via CS access

6.6.2a.1.1 Introduction

The following clauses provide alternative approaches for IMS registration, or analogous procedures.

6.6.2a.1.2 IMS Registration via CS access using ISC

6.6.2a.1.2.1 Assumptions

- ICCF is a home IMS node, hence considered a trusted node.
- General assumptions related to roaming relationships apply for SS7 connectivity between the home IMS network and the visited CS network.
- The CS subscription is issued by an operator trusted by the IMS operator.
- The ICS user is authenticated with the home CS network via the visited CS network.
- The user is authenticated in CS prior to when the UE establishes the ICCC-cs with the ICCF for registration in IMS. The user does not need to be re-authenticated as part of IMS Registration because it has been authenticated by the CS network.
- It is assumed that the identities that the UE and network uses for CS domain access using ICS are based on IMSI or derived from IMSI (for IMS). Note that the IMSI is required for CS service.
- It is assumed that the ISC interface is used for the IMS Registration procedure with the ADS executed as part of the ICCF.

6.6.2a.1.2.2 Example Information flow

An example information flow for a user registering in IMS via the CS access is provided in the figure below.

NOTE: This procedure only applies to the Trusted Registration performed by the ICCF; as a pre-condition CS authentication is assumed.

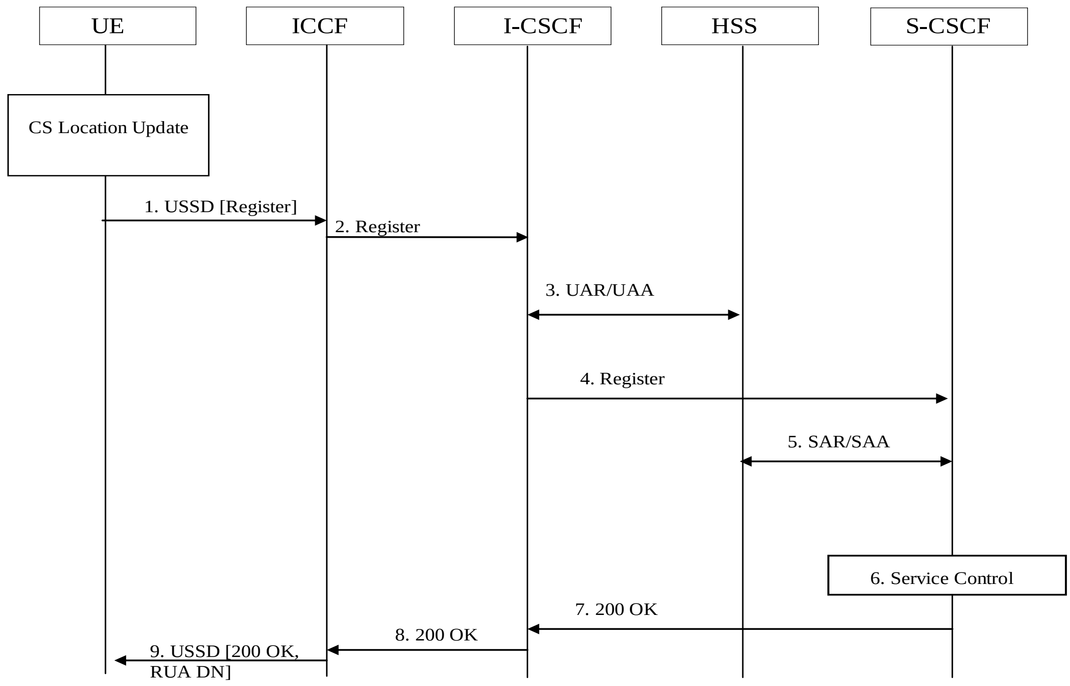


Figure: ICS UE IMS Registration via CS Access

1. After the UE has performed Location Update in the CS network, it may request the RUA of the ICCF to perform the IMS registration as trusted registration for the UE. To do so, the UE sends the content of SIP Register encoded in ICCP, in a USSD message to the ICCF using IMSI as the identity. Alternatively CAMEL mobility triggers may be used to inform the ICCF about the CS attach of the ICS user.

2. Upon receipt of the register information flow, the ICCF initiates a Trusted Registration for the user by sending the Register to the I-CSCF with required information (e.g. Public User Identity derived from IMSI, ICCF IP address as "Contact" address, etc). The Register message from the ICCF does not include an Authorization header field or the header fields or header field values as required by RFC3329. This is same as the registration procedures used with early IMS security (TR 33.978).

3. The I-CSCF performs User Authorization Request /User Authorization Answer with the HSS to authorise the Register request and locate the S-CSCF if already assigned to the user or allocate an S-CSCF otherwise. Note that the private identity is obtained from the IMSI (which is contained in the T-IMPU).

4. The I-CSCF forwards the Register to the S-CSCF.

5. The S-CSCF identifies the Register as a Trusted Registration from the ICCF which is a trusted Application Server (either based on a dedicated network interface for the ICCF or based on a Security Association with the ICCF, such as IPSec SA as in NDS/IP TS 33.210 [20]). The S-CSCF then checks the contact address against a configured list of transport address/range associated with the ICCF Application Servers. If there is a match, then, the S-CSCF skips any further authentication procedures as it is assumed that the user has already been authenticated in the CS domain prior to contacting the ICCF.

   The S-CSCF performs the Server Assignment Request/Server Assignment Answer with the HSS resulting in the user status as "registered" and the S-CSCF name stored in the HSS, if initial registration, and download of the user profile data (e.g. the registered and unregistered iFCs). The S-CSCF also obtains the Implicitly Registered IMPUs from the HSS as part of the SAR/SAA. The S-CSCF stores the user information as downloaded from the HSS. The S-CSCF stores the RUA's address/name, as supplied by the RUA for each of the implicitly registered IMPUs. This represents the address/name that the home network forwards the subsequent terminating session signalling to the ICS UE. If the user is already in "registered" state for the same IMPU, but with a different contact address (e.g. due to IMS registration using IP-CAN), then the S-CSCF creates another record for the IMPU with the contact address with the RUA's address for that IMPU.

If a Register request is received with an Authorization header anytime during/afterwards, then, the S-CSCF behaves according to clause 5.4.1.2 of TS 24.229 [6].

6. Service control execution at the S-CSCF is performed according to the standard procedures.

7. The S-CSCF returns the 200 OK to the I-CSCF.

8. The I-CSCF sends the 200 OK to the ICCF.

9. The ICCF processes this information and relays via the CAAF, the result back to the UE in a USSD message. The ICCF allocates a unique RUA DN that has the properties that when received in a SIP INVITE from a MGCF in a future session set-up is ,the ICCF can retrieve the necessary call information . This RUA DN is included in the USSD back to the UE.

Editor's note: The relation between RUA DN, VDI/VDN and STI is FFS.

6.6.2a.1.3 CS Registration Status Push and Pull over Cx and Sh

6.6.2a.1.3.1 Proposal

A new procedure is proposed for subscriber registration in IMS when attaching via the CS network. Using this procedure, the subscriber may be treated as "Registered" for the purposes of service delivery even when only available over CS access. This new registration procedure uses the knowledge of the user's CS location data in the HSS to mark the subscriber as "CS Registered" in the user's IMS subscription data, so that the CS Registration status is made available to the S-CSCF on IMS Registration and/or ICS user session establishment prior to IMS Registration to enable registered service delivery for ICS user sessions. The CS Registration status is used at the S-CSCF along with the IMS user status to evaluate filter criteria for the invocation of the respective Application Servers.

The HSS stores the location information for a user. When a UE attaches to the CS network, the VLR sends a MAP Update Location message to the HSS to inform the HSS of the registered VLR and MSC.

When there is a change in a user's CS location data at the HSS, i.e. one of the following events:

- A change from NIL-VLR-Number to a registered VLR-Number.

- A change from a registered VLR-Number to a NIL-VLR-Number (i.e. user roams into a VLR area that s/he is not allowed to have access or the operator manually cancels the CS registration for the user).

- The VLR purges the user's record (i.e. the HSS receives a Purge-MS MAP message and sets the MS-Purged flag. From the point of view of the HSS, the user has no CS location).

Then the HSS stores a new registration status called "CS Registration status" in the user's IMS subscription data. The CS Registration status has two possible states: "Registered" or "Not registered".

The CS Registration status is always kept up to date in the HSS and pushed down to the S-CSCF when possible:

- In the case that the HSS has an S-CSCF name already stored (e.g. when the user is IMS Registered) then the CS registration status change signalled from the VMSC to the HSS (stored as CS Registration Status) is simply pushed to the S-CSCF over the Cx interface.

- In the case that the HSS does not have an S-CSCF name already stored, then the CS Registration Status is held at the HSS in the IMS subscription until one of the following events, where upon the CS Registration Status is pulled from the HSS along with the Service Profile:

  1. Standard IMS SIP Registration.

  2. IMS Un-registration (i.e. terminating call to a not registered user that has unregistered services configured).

  3. UE initiates a new session when in CS access when the user is not currently registered in IMS.

The CS Registration Status may be provided in a new AVP or could be contained in the service profile along with the modified filter criteria. The benefit of a new AVP allows the change in CS Registration Status to be signalled without the need to send the service profile again to the S-CSCF.

Application Servers can also be made aware that the ICS user is registered to receive services over CS access. The Sh interface could be utilized to good effect here. Without changes to the Sh interface, a new data reference can be provided to allow the AS to query (Sh-Pull), or subscribe to be notified (Sh-Subs-Notif) and be notified (Sh-Notify) if the user is registered or not registered via CS access to receive IMS Centralized Services.

The CS Registration status may also be used to configure a new set of iFCs, "CS Registered iFC", specifically for the service execution when the ICS user is registered only via the CS access. CS Registered iFC may be utilized based on service provider policy. Alternatively, a new SPT may be configured to filter on the CS Registration status.

If a new condition is added inside SPT for CS registration status, unregistered iFC's can be configured which check if CS registration is registered/not-registered, and thus registered or unregistered state service may be invoked as appropriate.

Editors Note: The possibility to use a new set of iFCs is FFS.

Table 6.6.2a.1.3.1-1 indicates how triggering could be invoked if new type(s) of iFC were specified.

Table 6.6.2a.1.3.1-1: Use of Registration via CS to influence the invocation of iFC in the S-CSCF

IMS Registration Status	CS Registration Status	iFC invoked in the S-CSCF
Not Registered	Not Registered	Unregistered iFC
Registered	Not Registered	Registered iFC
Not Registered	Registered	CS Registered iFC or Registered iFC, or Unregistered iFC (based on service provider policy)
Registered	Registered	Registered iFC

NOTE: If there are no data services in the network, then the Registered-iFC and the CS-Registered -iFC may be the same.

Existing IMS SIP Deregistration procedures are not impacted by the introduction of the CS Registration Status and iFC modifications. When the user is IMS Registered and CS Registered and an IMS Deregistration occurs, the S-CSCF may remove the user-profile from the S-CSCF as a result of the deregistration. In this case, the S-CSCF is furnished with the CS Registration Status on the next originating or terminating session.

6.6.2a.1.3.2 Use cases

Use Case 1: Standard IMS Registration followed by CS Registration

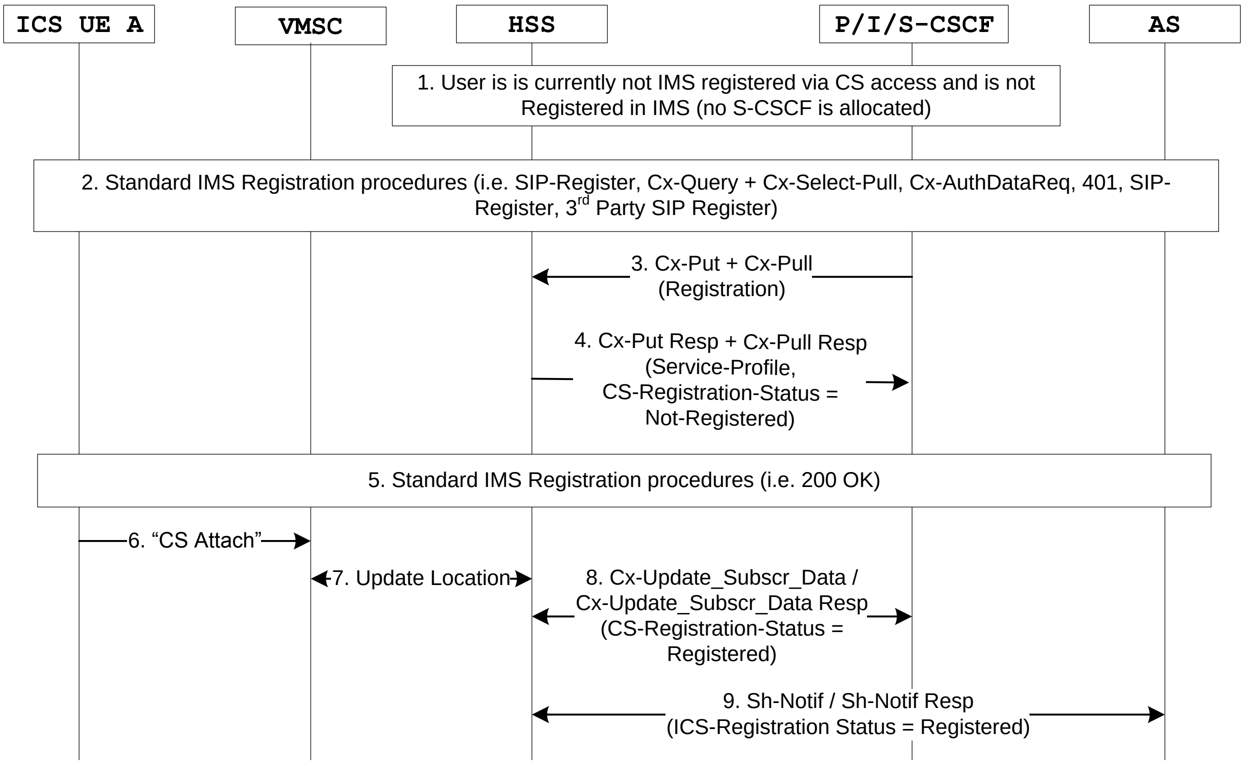


Figure 6.6.2a.1.3.2-1: Use Case 1: Standard IMS Registration followed by CS Registration

1. The user is currently not IMS registered via CS access and is not registered in IMS (no S-CSCF allocated).

2. Standard IMS SIP Registration procedure occurs via an IP-CAN (involving SIP-Register, Cx-Query + Cx-Select-Pull, Cx-AuthDataReq, 401 Unauthorized, second SIP-Register, 3rd Party Register).

3. On successful authentication, the S-CSCF sends a Cx-Put + Cx-Pull to the HSS. This corresponds to the Server Assignment Request (SAR) command with a Server Assignment Type (SAT) set to "Registration".

4. The HSS responds with a Cx-Put Resp + Cx-Pull Resp containing the complete user profile together with the CS Registration Status set to Not-Registered. This corresponds to the Server Assignment Answer (SAA) command.

5. Standard IMS SIP Registration completion via an IP-CAN

NOTE: Steps 2-5 illustrates the standard IMS SIP Registration procedure.

6. ICS User roams from an area where s/he did not have CS access to an area where CS access is allowed and performs a CS Attach.

7. VLR sends an Update Location to the HSS

8. The HSS updates the user's IMS data with the CS registration status of Registered. As the user is already IMS registered (and has an S-CSCF name stored), the CS registration status of Registered is "pushed" to the S-CSCF using a Cx-Update_Subscr_Data request to inform the S-CSCF that the user is available for services over CS access. This corresponds to a Push Profile Request (PPR) command. The S-CSCF responds with a Cx-Update_Subcr_Data Resp. This corresponds to the Push Profile Answer (PPA) command.

9. Assuming that an AS (e.g. TAS, ICCF, VCC AS) has previously subscribed to be notified of any changes in the CS-Registration-Status, the HSS sends a Sh-Notif request to the AS together with the CS-Registration-Status. This corresponds to the Push Notification Request (PNR) command. The AS responds with a Sh-Notif Resp. This corresponds to the Push Notification Answer (PNA) command.

Use Case 2: CS Registration followed by Standard IMS Registration

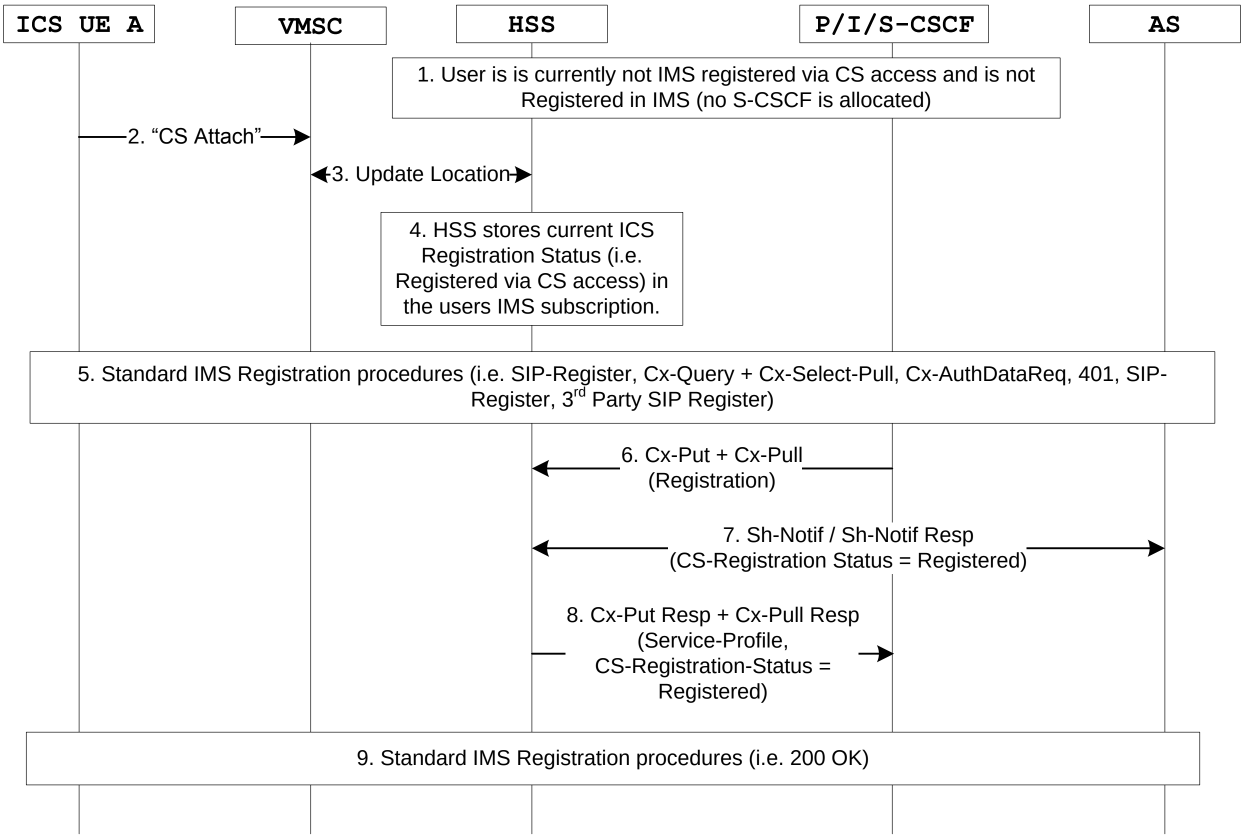


Figure 6.6.2a.1.3.2-2: Use Case 2: CS Registration followed by Standard IMS Registration

1. The user is currently not IMS registered via CS access and is not registered in IMS (no S-CSCF is allocated).

2. ICS User roams from an area where s/he did not have CS access to an area where CS access is allowed and performs a CS Attach.

3. VLR sends an Update Location to the HSS.

4. As the user is not IMS registered (i.e. no S-CSCF name is stored), the HSS stores the current CS Registration Status (i.e. Registered) in the user's IMS data.

5. Standard IMS SIP Registration procedure occurs via an IP-CAN (involving SIP-Register, Cx-Query + Cx-Select-Pull, Cx-AuthDataReq, 401 Unauthorized, second SIP-Register, 3rd Party Register).

6. On successful authentication, the S-CSCF sends a Cx-Put + Cx-Pull to the HSS. This corresponds to the Server Assignment Request (SAR) command with a Server Assignment Type (SAT) set to "Registration".

7. Assuming that an AS (e.g. TAS, ICCF, VCC AS) has previously subscribed to be notified of any changes in the CS-Registration-Status, the HSS sends a Sh-Notif request to the AS together with the CS-Registration-Status. This corresponds to the Push Notification Request (PNR) command. The AS responds with a Sh-Notif Resp. This corresponds to the Push Notification Answer (PNA) command.

8. The HSS responds with a Cx-Put Resp + Cx-Pull Resp containing the complete user profile together with the CS Registration Status set to Registered. This corresponds to the Server Assignment Answer (SAA) command.

9. Standard IMS SIP Registration completion via an IP-CAN.

NOTE: Steps 5-9 illustrate the standard IMS SIP Registration procedure with notification to AS occurring in step 7.

Use Case 3: CS Registration followed by CS-Origination

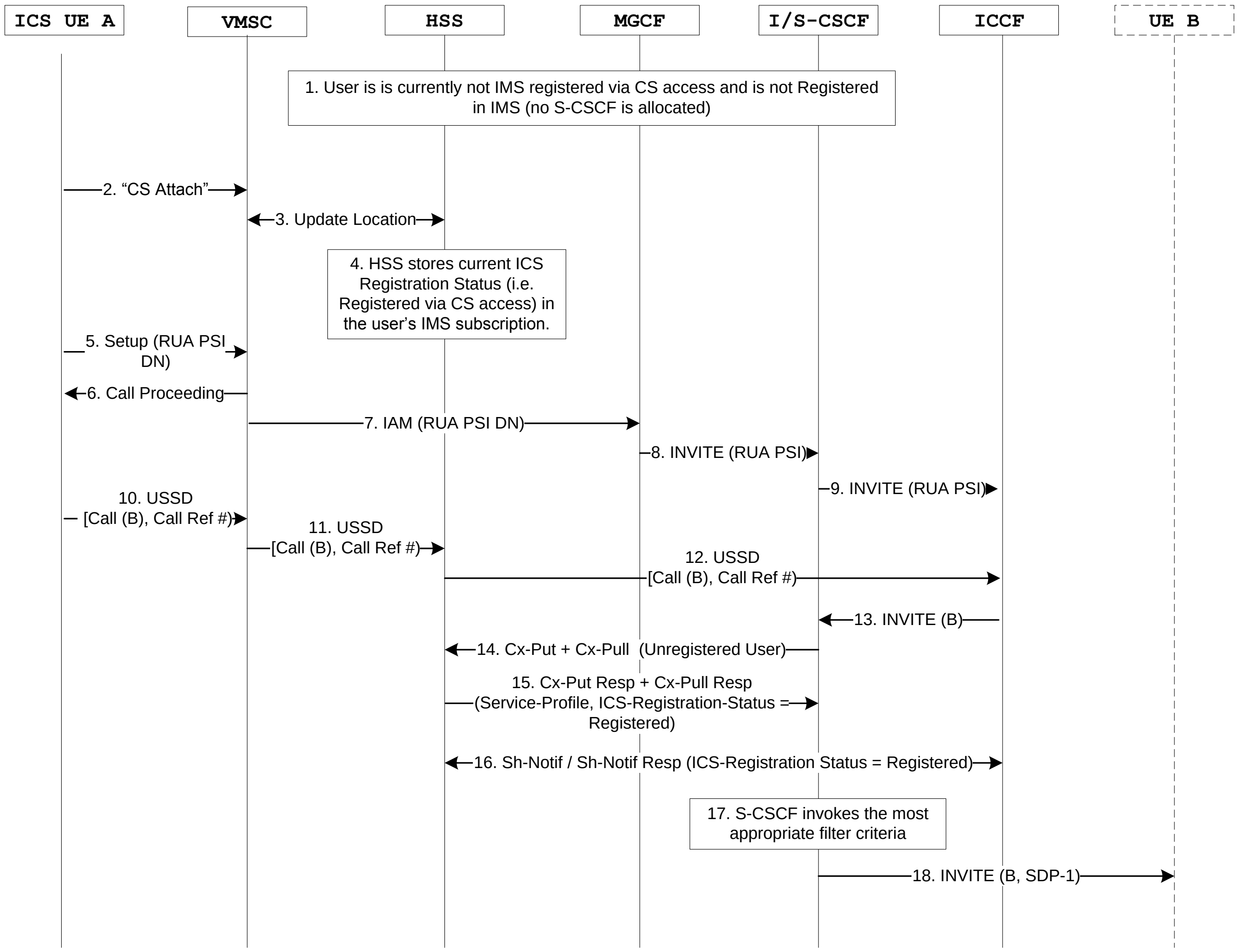


Figure 6.6.2a.1.3.2-3: Use Case 3: CS Registration followed by CS-Origination

Steps 1 to 4: Same as in Use Case 2.

Steps 5 to 13: CS Origination procedures for an ICS UE. This particular example is showing I1-cs as documented in figure 6.6.2.1.1-1, steps 1 to 9.

NOTE: The I-CSCF locates the S-CSCF by sending a Cx-Location-Query to the HSS and the HSS responds with the experimental result code of "Diameter Unregistered Service". For brevity, this is not shown in the diagram above.

14. The I-CSCF contacts the S-CSCF. The S-CSCF sends a Cx-Put + Cx-Pull to the HSS for the unregistered user. This corresponds to the Server Assignment Request (SAR) command with a Server Assignment Type (SAT) set to "Unregistered User".

15. The HSS responds with a Cx-Put Resp + Cx-Pull Resp containing the complete user profile together with the CS Registration Status set to Registered. This corresponds to the Server Assignment Answer (SAA) command.

16. Assuming that an AS (e.g. ICCF, VCC AS, TAS) has previously subscribed to be notified of any changes in the CS-Registration-Status, the HSS sends a Sh-Notif request to the AS together with the CS-Registration-Status. This corresponds to the Push Notification Request (PNR) command. The AS responds with a Sh-Notif Resp. This corresponds to the Push Notification Answer (PNA) command.

17. The S-CSCF invokes the most appropriate iFC as the user is registered via CS access, but not registered in IMS.

18. After triggering and service execution, the S-CSCF sends the INVITE to the remote end.

Use Case 4: CS Registration followed by CS-Termination

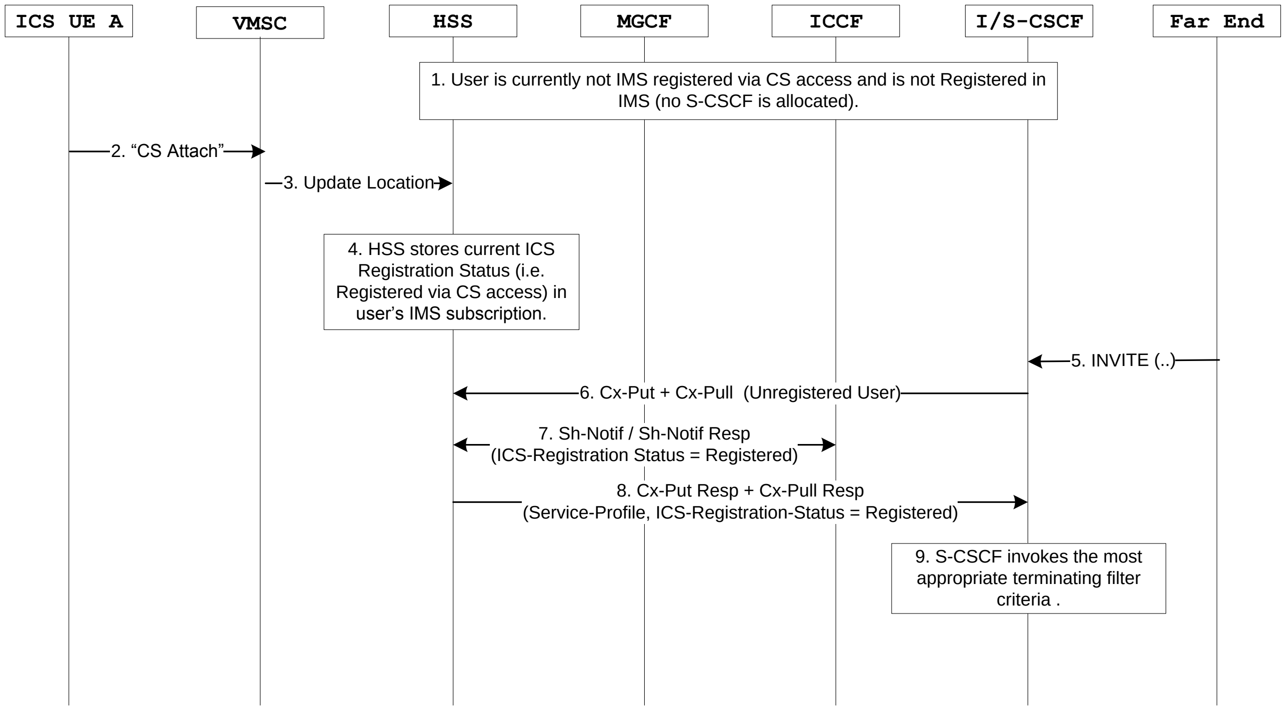


Figure 6.6.2a.1.3.2-4: Use Case 4: CS Registration followed by CS-Termination

Steps 1 to 4: Same as in Use Case 2

5. Invite sent in from the Far End to I-CSCF.

NOTE: The I-CSCF locates the S-CSCF by sending a Cx-Location-Query to the HSS and the HSS responds with the experimental result code of "Diameter Unregistered Service". For brevity, this is not shown in the figure above.

6. I-CSCF contacts the S-CSCF. S-CSCF sends in an SAR to the HSS with SAT set to "Unregistered User".

7. Assuming that an AS (e.g. ICCF, VCC AS, TAS) has previously subscribed to be notified of any changes in the CS-Registration-Status, the HSS sends a Sh-Notif request to the AS together with the CS-Registration-Status. This corresponds to the Push Notification Request (PNR) command. The AS responds with a Sh-Notif Resp. This corresponds to the Push Notification Answer (PNA) command.

8. HSS downloads the complete profile in the SAA together with the CS Registration Status set to "IMS-Registered-via-CS-access".

9. The S-CSCF invokes the most appropriate filter criteria as the user is CS registered but IMS not registered. Termination procedures continue.

Use Case 5: CS Deregistration

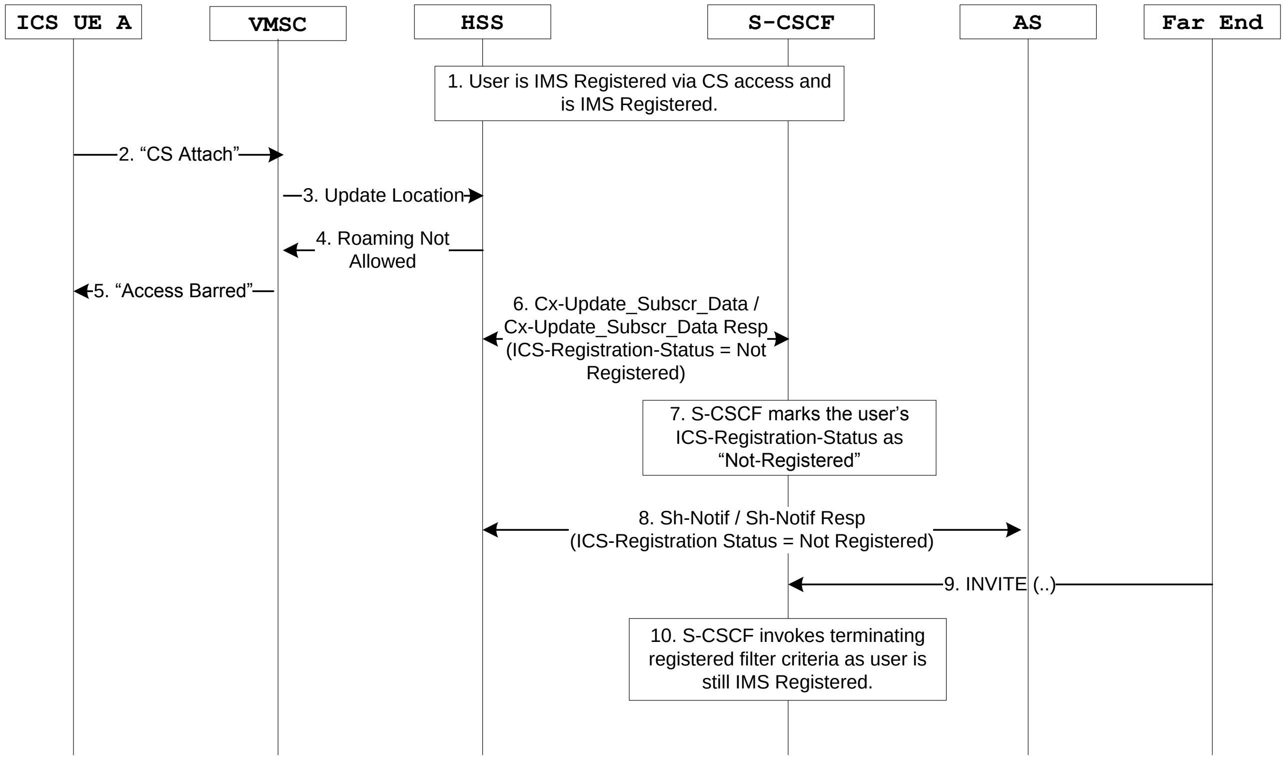


Figure 6.6.2a.1.3.2-5: Use Case 5: CS Deregistration

1. User is currently registered in IMS and is registered in IMS via CS access. The S-CSCF contains these registration statuses and the complete user profile.

2. The ICS User roams into a new VLR area and does a CS attach.

3. The VLR sends an Update Location to the HSS.

4. The user's roaming entitlement in the HSS does not allow the user to roam the current VLR area. The HSS sends back a "Roaming Not Allowed" response.

5. CS access is barred for the user in the current VLR area.

6. The HSS updates the user's IMS data with the CS Registration Status of "Not Registered". As the HSS has an S-CSCF name stored for the user, the HSS sends a Cx-Update_Subscr_Data request to the S-CSCF with the CS Registration Status set to "Not Registered". This corresponds to a Push Profile Request (PPR) command. The S-CSCF responds with a Cx-Update_Subcr_Data Resp. This corresponds to the Push Profile Answer (PPA) command.

7. The S-CSCF marks the user CS Registration Status as "Not Registered".

8. Assuming that an AS (e.g. TAS, ICCF, VCC AS) has previously subscribed to be notified of any changes in the CS-Registration-Status, the HSS sends a Sh-Notif request to the AS together with the CS-Registration-Status. This corresponds to the Push Notification Request (PNR) command. The AS responds with a Sh-Notif Resp. This corresponds to the Push Notification Answer (PNA) command.

9. Invite sent in from the Far End to I-CSCF. I-CSCF sends in a Location Information Retrieval (LIR) request to the HSS and the HSS returns "Diameter Success". I-CSCF contacts S-CSCF.

10. S-CSCF invokes terminating registered filter criteria as the user is still registered in IMS. Terminating procedures continue.

6.6.2a.1.4 RUA Registration in IMS when attached to CS access

See clause 6.7.1a.1.

For I1-cs Application Server model, ICCF is included in terminating iFC and would be invoked via iFC. The ICCF is the last AS and collocates with DSF. It is invoked via the filter criteria and is always involved whether ICS UE is in CS access or in PS access. There are also two cases here:

- If CS access is selected, the INVITE message is routed to CS access (either via DSF fetching CSRN or using ICCP).
- If PS access is selected, DSF indicates sufficient information in the INVITE message only to select PS contact in the S-CSCF.

6.6.2b Video call

ICS video call handling principles:

1 Solutions of multimedia inter-working from TS 29.163 [18], version 7.6.0, for fallback and service change are applicable with ICS video call.

2 In case of the ICS UE originating Non-SCUDIF service change, the mechanism Re-dial with release of the radio connection (TR 23.801, TS 24.008) to setup a new CS bearer supporting the new media type is needed.

3 Sharing one CS bearer between voice session and video session are not supported in this stage.

4 If there are two or more sessions sharing one CS bearer, Whether to support service change (e.g. from video to speech) for one of the sessions is FFS.

6.6.2b.1 Fallback to speech at session establishment

If Fallback occurs on the CS side, normal SDP offer/answer procedure is performed in the IMS side. There is no new requirement to ICS architect and already be supported by current architect.

6.6.2b.2 Service change with SCUDIF

6.6.2b.2.1 IM CN subsystem originated change

If IM CN subsystem originated SCUDIF service change, normal media re-negotiation procedure is performed in the IMS side. There is no new requirement to ICS architect and already be supported by current architect.

6.6.2b.2.2 CS network originated change

If IM CN subsystem originated SCUDIF service change, normal media re-negotiation is performed in the IMS side. There is no new requirement to ICS architect and already be supported by current architect.

6.6.2b.3 Non-SCUDIF case (ISUP or BICC without SCUDIF)

If IM CN subsystem originated Non-SCUDIF service change, normal media re-negotiation is performed in the IMS side. There is no new requirement to ICS architect and already be supported by current architect.

In TS 29.163 [18], when service change is initiated by the CS network or CS UE, Re-dial mechanism with release of the CS connection is recommended solution for switching between Circuit Switched voice and video services when SCUDIF is not supported by the CS network.

Under ICS architecture, a more effective and user friendly method is provided for switching between voice and video services when SCUDIF feature is not supported.

Figure 6.6.2b.3-1 shows an ICS UE originates Non-SCUDIF service change from speech to video during an ongoing session.

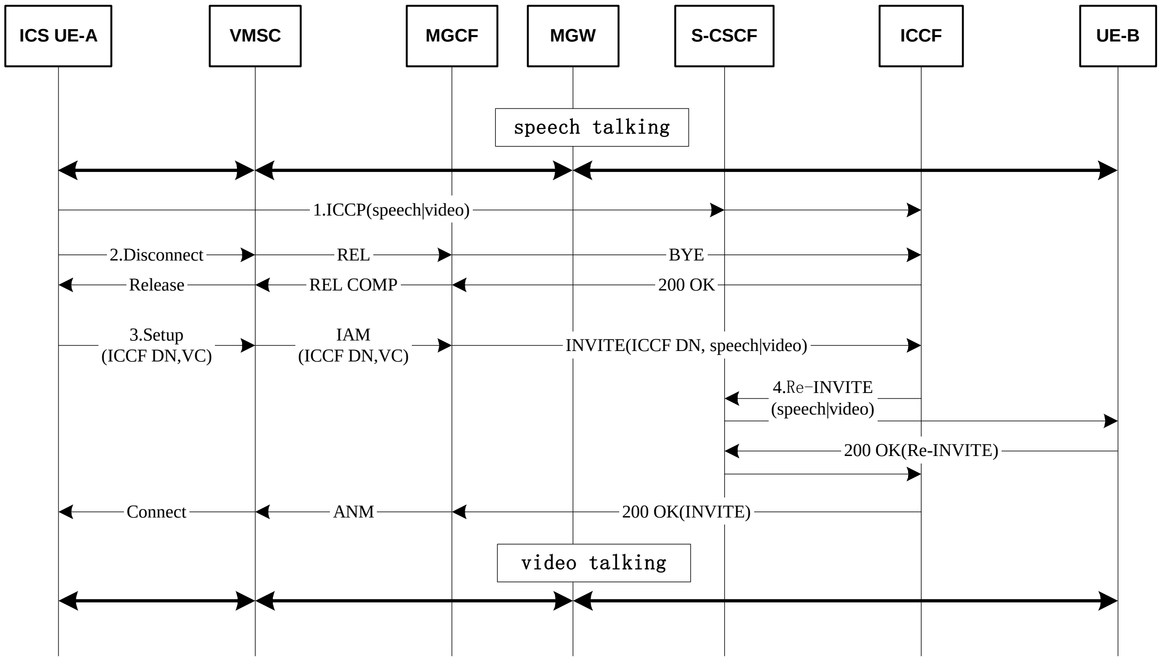


Figure 6.6.2b.3-1: ICS UE originates Non-SCUDIF service change from speech to video during an ongoing session

1. UE-A and UE-B perform speech talking. After verbal agreement, UE-A sends a video call invitation to the ICCF.
2. UE-A disconnects the existing CS call leg with the ICCF. According to clause 6.11 Media Handling, the ICCF can utilize MRFC/MRFP to be sure that the RTCP packets would be sent continuously as "remote end-point aliveness information" to UE-B.
3. UE-A setups a new CS call leg supporting video call with the ICCF and the ICCF receives an INVITE with MGW SDP from the MGCF including video media invitation.
4. The ICCF sends a Re-INVITE/UPDATE with MGW SDP including video media invitation to UE-B. UE-B feeds back 200 OK with video media information to accept the invitation.

Then UE-A and UE-B perform video talking.

6.6.2b.4 Normal video call flow

6.6.2b.4.1 Origination

6.6.2b.4.1.1 Origination Calls established using CS multimedia bearers

Figure 6.6.2b.4.1.1-1 provides an example call flow for a video call made by an ICS UE-A to the other end B with I1-cs for call control, the AS approach for RUA implementation. The H.245 negotiation procedures take place after the CS call being established. Note that other origination approaches (e.g. I1-ps, CAMEL) are applicable.

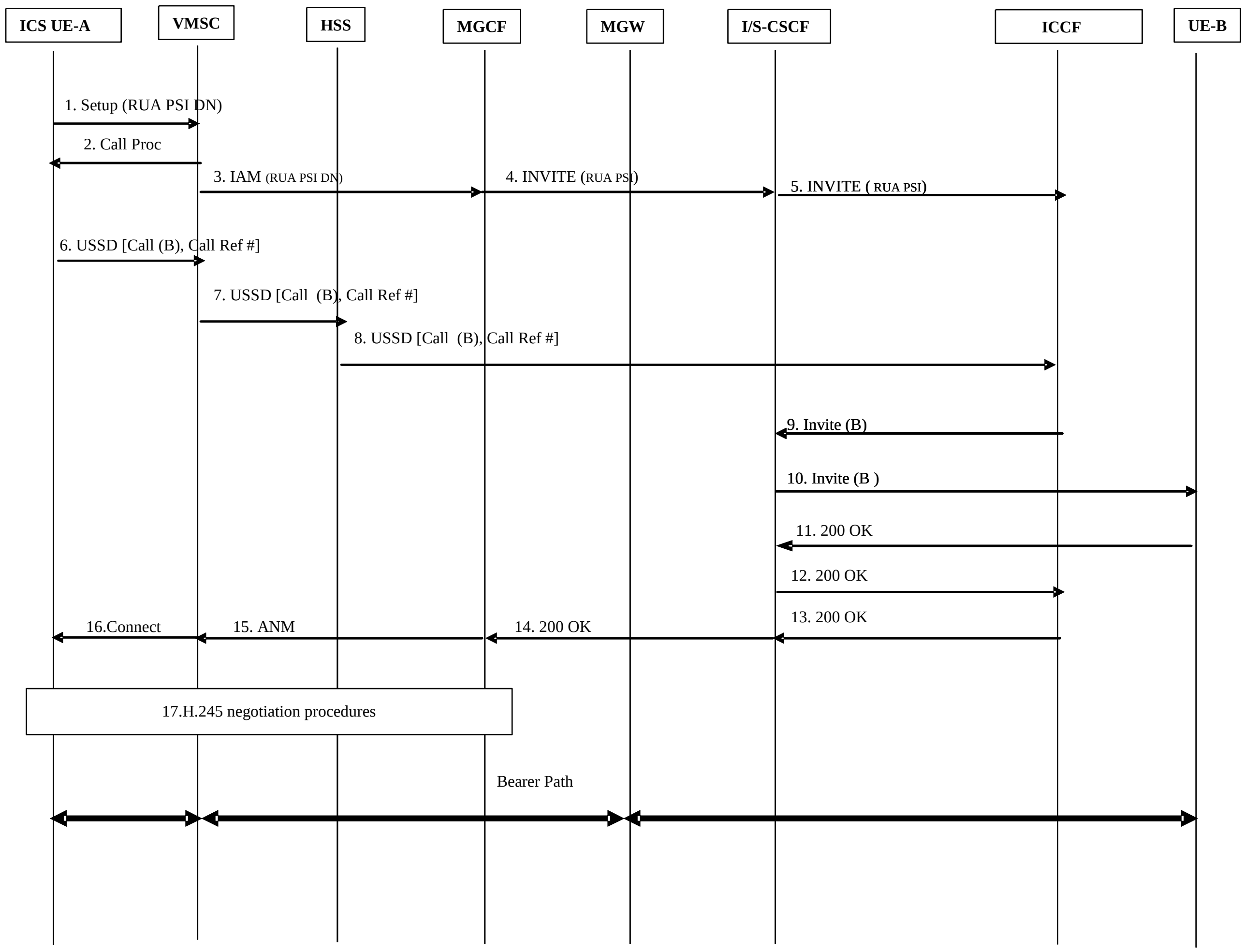


Figure 6.6.2b.4.1.1-1: ICS UE video call origination: USSD transport–AS Approach–I1-cs for call control

1. ICS UE-A sends a SETUP message to VMSC to establish the CS multimedia bearer leg. The SETUP message includes the RUA PSI DN as the called party number. This will establish the CS domain multimedia bearer path with the ICCF.

2. The VMSC responds with a call proceeding message and begins to setup the CS multimedia bearer path.

3. The VMSC processes the SETUP message and sends an IAM to the MGCF. The IAM message contains the RUA PSI DN.

NOTE 1: Standard VMSC procedure for CS origination.

4. The MGCF performs a setup of the MGW and creates an INVITE that is sent to the S-CSCF that serves the ICCF. This may involve sending the INVITE to the I-CSCF first to locate the S-CSCF and to translate the RUA PSI DN to a SIP-URI.

NOTE 2: Standard MGCF procedure for PSTN origination.

5. The S-CSCF forwards the INVITE to RUA of ICCF. When the RUA has received both the USSD message (steps 6-8) and the INVITE, it has sufficient information to setup a session with party B.

6. ICS UE-A encodes in ICCP sufficient information (e.g. containing B party address (SIP or Tel URI) for the RUA of the ICCF to generate a SIP INVITE initiating a session toward the other end B, to be sent in a USSD message, and sends it toward the CAAF of the ICCF via the VMSC. The USSD message has sufficient information needed at the ICCF to correlate the session request with the multimedia bearer path established with the RUA in steps 1-5.

Editor's note: Whether SIP Invite is compressed as such to fit to USSD, or similar information is otherwise carried on top of USSD is for further study.

7. Standard VMSC procedure to communicate the USSD message to the Home network.

8. Standard HSS procedure to communicate the USSD message to the associated service node.

   Step 6 - 8 occurs in parallel to steps 2 - 5. The UE may also invoke some supplementary service from home IMS like temporary OIR.

NOTE 3: Signalling Optimizations may be achieved by combining the two step procedure for setting up of Session Control Signalling Path and the Bearer Control Signalling session in one step or by omitting the Session Control Signalling set-up for basic calls when no service needs to be invoked.

9 CAAF of ICCF performs necessary adaptation and relays information needed for generation of the SIP INVITE at the RUA of ICCF. The RUA of ICCF acting as a B2BUA creates a SIP INVITE with appropriate SDP information and sends it to the S-CSCF (i.e., terminating the UE Leg and originating the RUA Leg for presentation of IMS session toward other end B on behalf of UE-A).

10 The S-CSCF routes the INVITE to the correct termination for the party B based on standard IMS originated session processing at the S-CSCF.

NOTE 4: VCC and Originating Supplementary services (e.g. OCB, OIR), if any, are executed on the RUA Leg in the Application Servers in home IMS. These are not shown in the figure.

11~16. The UE-B answers the call and the multimedia BC (bearer capability) is established.

17. H.245 negotiation procedures are performed between the MGCF and ICS UE-A. Then the CS multimedia bearer is established.

6.6.2b.4.2 Termination

6.6.2b.4.2.1 Termination Calls established using CS multimedia bearers

The figure 6.6.2b.4.2.1-1 provides an example call flow for a video call destined to ICS UE-B with I1-cs for call control, the AS approach for RUA implementation. The H.245 negotiation procedures take place after the CS call being established. Note that other termination approaches (e.g. Using CS Origination procedures) are applicable.

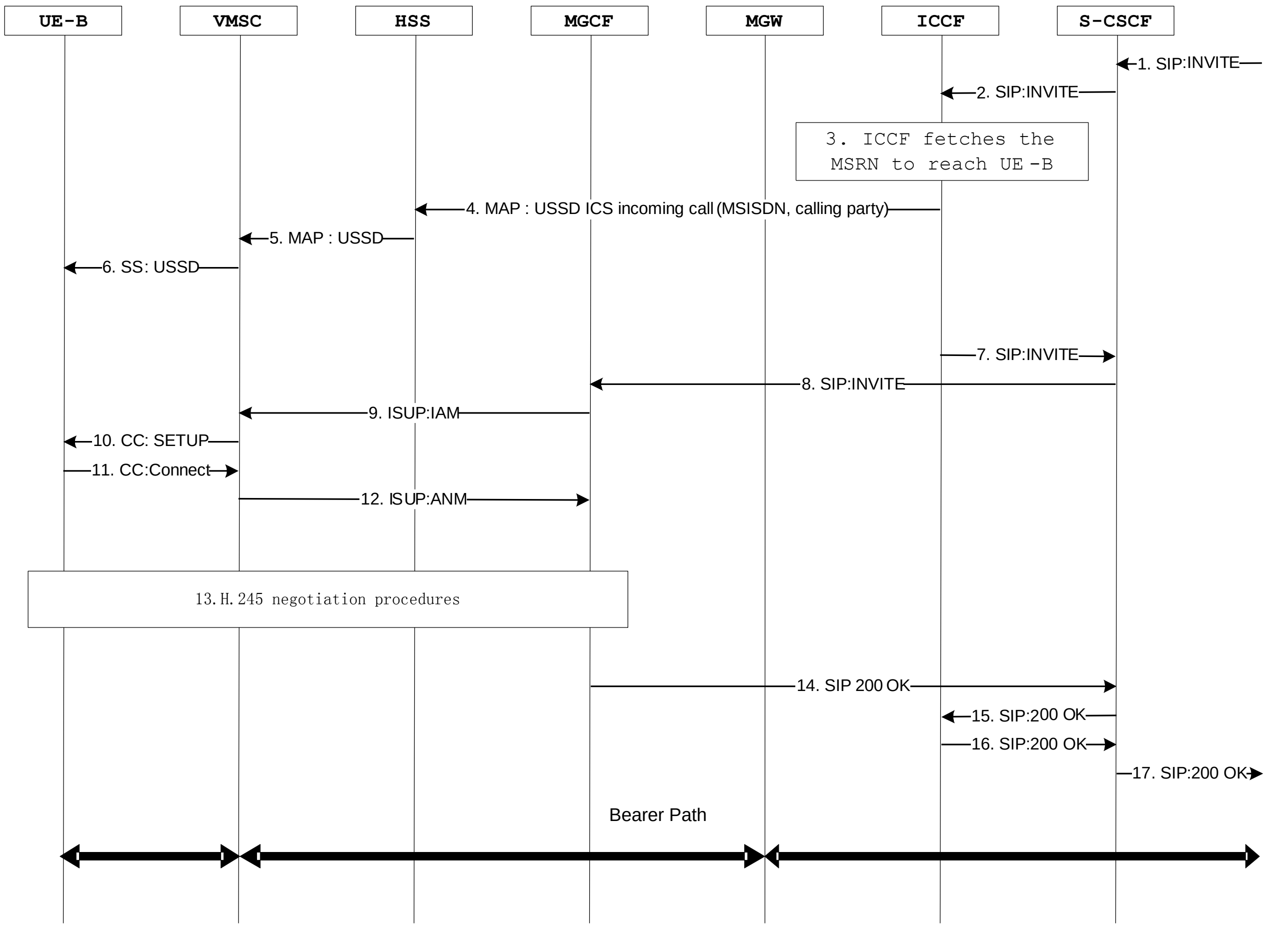


Figure 6.6.2b.4.2.1-1: ICS UE termination using CS multimedia bearers with use of I1-cs

1. An incoming SIP INVITE is received at the S-CSCF of the B party.
2. The S-CSCF forwards the INVITE to the ICCF.
3. The ICCF performs Access Domain Selection, and then fetches a number such as MSRN if a media connection is not already established to UE-B.

Editor's note: The type of number to be used and how the number is allocated is FFS. The Relationship and interworking with DSFVCC is FFS.

4. ICCF initiates an ICS Incoming Call Request to the HSS.

Editor's note: Whether Sh is used instead of MAP is FFS.

5. The HSS sends an USSD message to the VMSC.
6. The UE-B receives the USSD including the calling party ID.
7. The ICCF sends an INVITE (tel URI = MSRN) to the S-CSCF.
8. The Request URI has been modified, the S-CSCF skips further service execution and routes the call towards the CS domain.
9. The MGCF sends an IAM to the VMSC.
10. VMSC sends SETUP to UE-B.
11. UE-B responds with a CONNECT message.
12. VMSC responds with ANM towards the IMS.

13. H.245 negotiation procedures are performed between the MGCF and UE-B. Then the CS multimedia bearer is established.

14. MGCF sends the 200 OK to the S-CSCF.

15. S-CSCF forwards the 200 OK to the ICCF.

16. The ICCF sends a 200 OK as response to the INVITE in step 2 towards the S-CSCF.

17. S-CSCF forwards the 200 OK towards the A party.

6.6.2b.5 Session continuity

6.6.3 Information flows

6.6.3.1 I1-cs unregistered user solution

6.6.3.1.1 Origination

6.6.3.1.1.1 Use of USSD over I1 reference point for Call Origination

Figure 6.6.3.1.1.1-1 provides an example flow for a call made by an ICS UE-A to the other end B with USSD transport of ICCC, the AS approach for RUA implementation when I1 reference point is used for call setup.

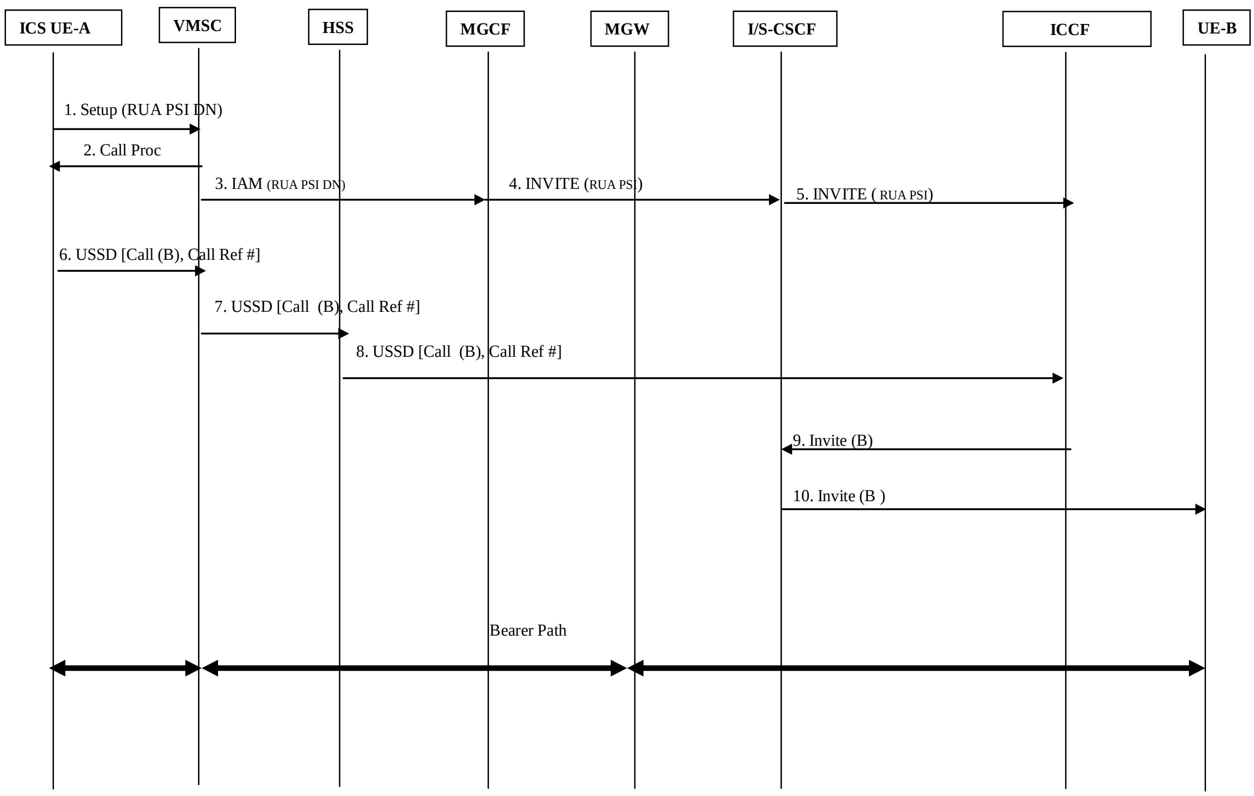


Figure 6.6.3.1.1.1-1: ICS UE origination: USSD transport–AS Approach–I1 reference point for call control

1. ICS UE-A sends a SETUP message to VMSC to establish the bearer leg. The SETUP message includes the RUA PSI DN as the called party number. This will establish the CS domain bearer path with the ICCF.

2. The VMSC responds with a call proceeding message and begins to setup the bearer path.

3. The VMSC processes the SETUP message and sends an IAM to the MGCF. The SETUP message contains the RUA PSI DN.

NOTE 1: Standard VMSC procedure for CS origination.

4. The MGCF performs a setup of the MGW and creates an INVITE that is sent to the S-CSCF that serves the ICCF. This may involve sending the INVITE to the I-CSCF first to located the S-CSCF and to translate the RUA PSI DN to a SIP-URI.

NOTE 2: Standard MGCF procedure for PSTN origination.

5. The S-CSCF forwards the INVITE to RUA of ICCF. When the RUA has received both the USSD message (steps 6-8) and the INVITE, it has sufficient information to setup a session with party B.

6. ICS UE-A encodes in ICCP sufficient information (e.g. containing B party address (SIP or Tel URI) for the RUA of the ICCF to generate a SIP INVITE initiating a session toward the other end B, to be sent in a USSD message, and sends it toward the CAAF of the ICCF via the VMSC. The USSD message has sufficient information needed at the ICCF to correlate the session request with the bearer path established with the RUA in steps 1 - 5.

Editor's note: Whether SIP Invite is compressed as such to fit to USSD, or similar information is otherwise carried on top of USSD is for further study.

7. Standard VMSC procedure to communicate the USSD message to the Home network.

8. Standard HSS procedure to communicate the USSD message to the associated service node.

   Steps 6 - 8 occurs in parallel to Steps 2 - 5. The UE may also invoke some supplementary service from home IMS like temporary OIR.

NOTE 3: Signalling Optimizations may be achieved by combining the two step procedure for setting up of Session Control Signalling Path and the Bearer Control Signalling session in one step or by omitting the Session Control Signalling set-up for basic calls when no service needs to be invoked.

9. CAAF of ICCF performs necessary adaptation and relays information needed for generation of the SIP INVITE at the RUA of ICCF. The RUA of ICCF acting as a B2BUA creates a SIP INVITE with appropriate SDP information and sends it to the S-CSCF (i.e., terminating the UE Leg and originating the RUA Leg for presentation of IMS session toward other end B on behalf of UE-A).

10. The S-CSCF routes the INVITE to the correct termination for the party B based on standard IMS originated session processing at the S-CSCF.

NOTE 4: VCC and Originating Supplementary services (e.g. OCB, OIR), if any, are executed on the RUA Leg in the Application Servers in home IMS. These are not shown in the figure.

6.6.3.1.1.2 Use of USSD over I1 reference point for Call Origination – alternative procedure1

The figure 6.6.3.1.1.2-1 provides an example flow for a call made by an ICS UE-A to the other end B with USSD transport of ICCC, the AS approach for RUA implementation when I1 reference point is used for call setup.

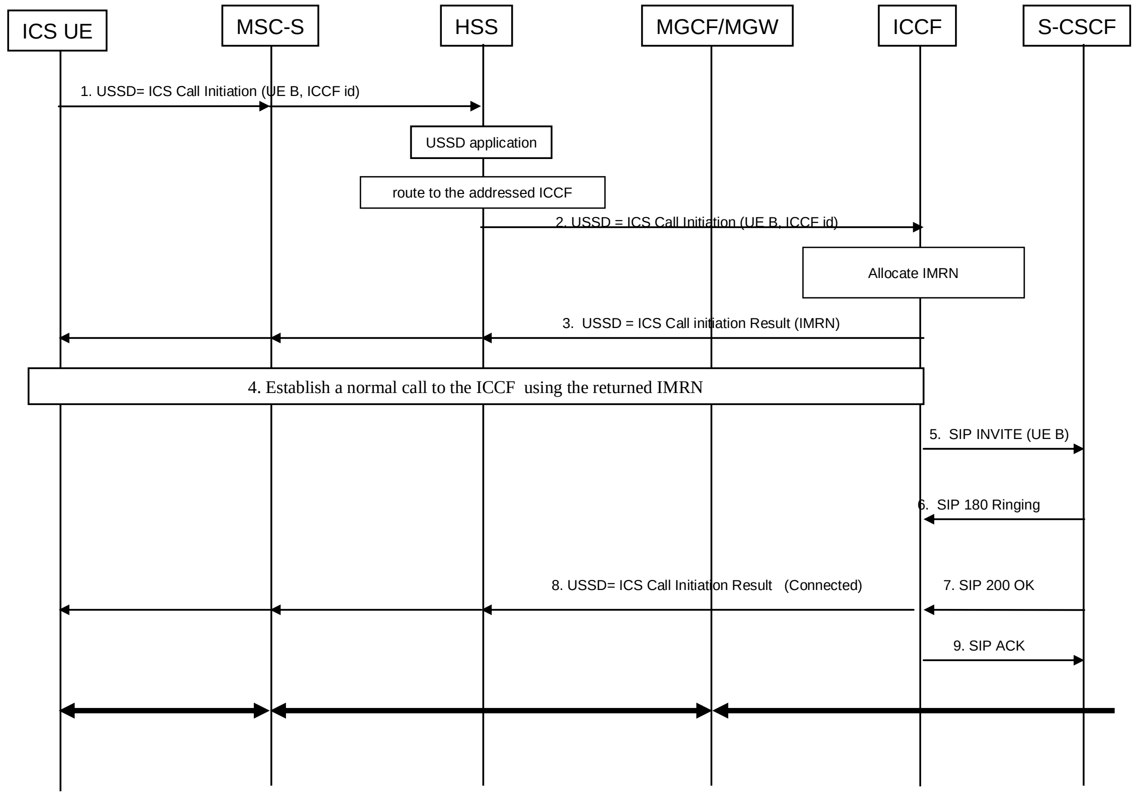


Figure 6.6.3.1.1.2-1: ICS UE origination: USSD transport–AS Approach–I1 reference point for call control alternative procedure

1. The ICS UE initiates a call by sending an ICS Call Initiation Request inside a USSD dialog to the HSS containing B party address (SIP or Tel URI).

2. The USSD application within the HSS forwards the USSD Request to the ICCF instance addressed in the HSS. The ICCF shall store the received B Party Address's against an IMRN.

3. The ICCF allocates a roaming number if a media connection is not already established to the ICS UE. The ICCF returns the IMRN in an ICS Response within the USSD dialog. The USSD response is returned all the way to the ICS UE.

4. The ICS UE uses the IMRN to establish a normal CS call set up to the ICCF via MGCF. PSI routing is used to route the call from MGCF to ICCF.

5. Following that, ICCF initiates the call towards UE-B by sending a SIP INVITE to the S-CSCF.

6. ICCF receives SIP 180 ringing.

7. ICCF receives SIP 200 OK.

8. ICCF send an ICS Call initiation response to the ICS UE that includes the connected status.

9. ICCF sends an ACK to the UE-B and the media is now established end to end.

6.6.3.1.1.3 Use of USSD over I1 reference point for Call Origination – alternative procedure 2

Figure 6.6.3.1.1.3-1 provides an example flow for a call made by an ICS UE-A to the other end B with USSD transport of ICCC, the AS approach for RUA implementation when I1 reference point is used for call setup.

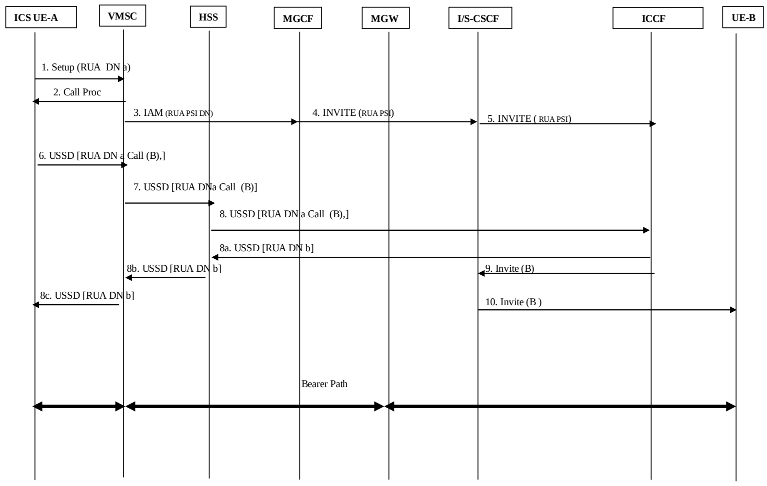


Figure 6.6.3.1.1.3-1: ICS UE origination: USSD transport–AS Approach–I1 reference point for call control – alternative procedure 2

1. ICS UE-A sends a SETUP message to VMSC to establish the bearer leg. The SETUP message includes the RUA DNa that was received in a previous USSD (step 8c – RUA DNb) as the called party number. This will establish the CS domain bearer path with the ICCF.

2. The VMSC responds with a call proceeding message and begins to setup the bearer path.

3. The VMSC processes the SETUP message and sends an IAM to the MGCF. The SETUP message contains the RUA PSI DN.

NOTE 1: Standard VMSC procedure for CS origination.

4. The MGCF performs a setup of the MGW and creates an INVITE that is sent to the S-CSCF that serves the ICCF. This may involve sending the INVITE to the I-CSCF first to located the S-CSCF and to translate the RUA PSI DN to a SIP-URI.

NOTE 2: Standard MGCF procedure for PSTN origination.

5. The S-CSCF forwards the INVITE to RUA of ICCF. When the RUA has received both the USSD message (steps 6-8) and the INVITE, it has sufficient information to setup a session with party B.

6. ICS UE-A encodes in ICCP sufficient information (e.g. containing B party address (SIP or Tel URI), RUA DNa for the RUA of the ICCF to generate a SIP INVITE initiating a session toward the other end B, to be sent in a USSD message, and sends it toward the CAAF of the ICCF via the VMSC. The USSD message has sufficient information needed at the ICCF to correlate the session request with the bearer path established with the RUA in steps 1 - 5.

Editor's note: Whether SIP Invite is compressed as such to fit to USSD, or similar information is otherwise carried on top of USSD is for further study.

7. Standard VMSC procedure to communicate the USSD message to the Home network.

8. Standard HSS procedure to communicate the USSD message to the associated service node.

8a-8c. Upon receipt of the USSD from UE-A, the ICCF allocates a unique RUA DNb that has the properties that when received in a future session set-up (step 5) the ICCF can retrieve the necessary call information from the initial SIP INVITE received to initiate a session (step 2). The RUA DNb is sent back to ICS UE-A in USSD.

Steps 6 - 8 occurs in parallel to Steps 2 - 5. The UE may also invoke some supplementary service from home IMS like temporary OIR.

NOTE 3: Signalling Optimizations may be achieved by combining the two step procedure for setting up of Session Control Signalling Path and the Bearer Control Signalling session in one step or by omitting the Session Control Signalling set-up for basic calls when no service needs to be invoked.

9. CAAF of ICCF performs necessary adaptation and relays information needed for generation of the SIP INVITE at the RUA of ICCF. The RUA of ICCF acting as a B2BUA creates a SIP INVITE with appropriate SDP information and sends it to the S-CSCF (i.e., terminating the UE Leg and originating the RUA Leg for presentation of IMS session toward other end B on behalf of UE-A).

10. The S-CSCF routes the INVITE to the correct termination for the party B based on standard IMS originated session processing at the S-CSCF.

NOTE 4: VCC and Originating Supplementary services (e.g. OCB, OIR), if any, are executed on the RUA Leg in the Application Servers in home IMS. These are not shown in the figure.

6.6.3.1.2 Termination

6.6.3.1.2.1 Introduction

Example information flows for session terminations using different models for the signalling/bearer architecture for full-duplex speech service with IMS centralized services over CS access are discussed in the following.

6.6.3.1.2.2 Calls established using CS bearers with use of I1-cs

The figure 6.6.3.1.2.2-1 provides an example flow for a call destined to an ICS UE when the USSD transport alternative for ICCC is used to support the setup of terminating sessions.

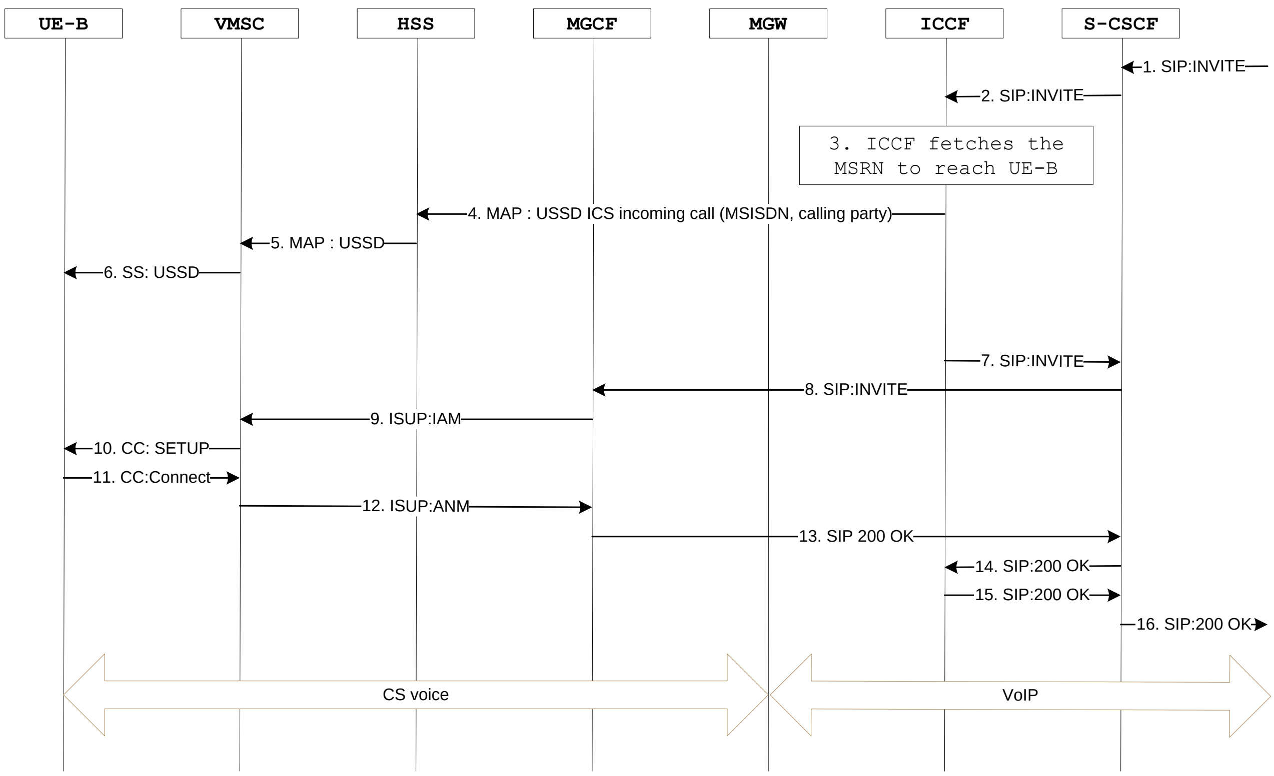


Figure 6.6.2.2.2-1: ICS UE termination using CS bearers with use of I1-cs

1 An incoming SIP INVITE is received at the S-CSCF of the B party.

2 The S-CSCF forwards the INVITE to the ICCF.

3 The ICCF performs Access Domain Selection, and then fetches a number such as MSRN if a media connection is not already established to UE-B.

Editor's note 1: The type of number to be used and how the number is allocated is FFS. The Relationship and interworking with DSFVCC is FFS.

4 ICCF initiates an ICS Incoming Call Request to the HSS.

Editor's note 2: Whether Sh is used instead of MAP is FFS.

5 The HSS sends an USSD message to the VMSC.

6 The UE-B receives the USSD including the calling party ID.

7 The ICCF sends an INVITE (tel URI = MSRN) to the S-CSCF.

8 The Request URI has been modified, the S-CSCF skips further service execution and routes the call towards the CS domain.

9 The MGCF sends an IAM to the VMSC.

10 VMSC sends SETUP to UE-B.

11 UE-B responds with a CONNECT message.

12 VMSC responds with ANM towards the IMS.

13 MGCF sends the 200 OK to the S-CSCF.

14 S-CSCF forwards the 200 OK to the ICCF.

15 The ICCF sends a 200 OK as response to the INVITE in step 2 towards the S-CSCF.

16 S-CSCF forwards the 200 OK towards the A party.

6.6.3.1.2.2.1 Using CS Origination procedures to set up the Bearer Control Signalling session

The figure 6.6.3.1.2.2.1-1 provides an example flow for a call destined to an ICS UE when the USSD transport alternative for ICCC is used to support the setup of terminating sessions using CS Origination procedures for setting up the Bearer Control Signalling session.

NOTE: The un-bolded steps are related to the Session Control Signalling set-up procedures. The bolded steps are related to the Bearer Control Signalling session set-up procedures.

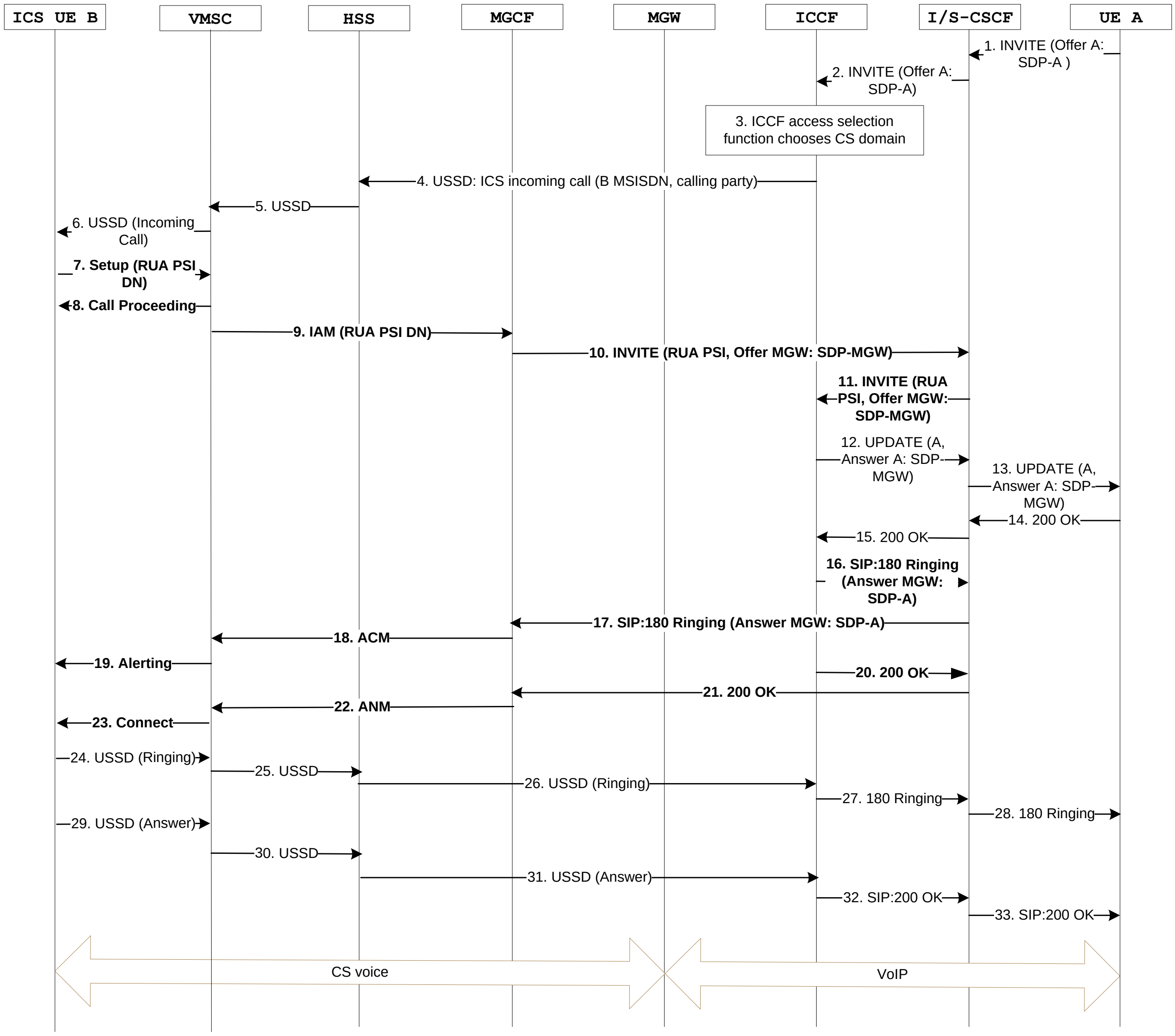


Figure 6.6.3.1.2.2.1-1: ICS UE termination using CS bearers using CS Origination procedures with use of I1-cs

1. An incoming SIP INVITE is received at the S-CSCF of the B party with an Offer containing SDP from the A-party.

2. The S-CSCF forwards the INVITE to the ICCF.

3. The ICCF performs access selection and chooses the CS domain.

4. The ICCF (acting as a B2BUA) establishes a session over I1-cs by sending an Incoming Call Request (in a USSD message) to the HSS. The request contains a SIP or Tel URI of the A and B parties (B subject to privacy), GRUU information and any other necessary information to complete the call. Additionally, an E.164 number is sent in the outbound USSD which is then used by the ICCF later on to correlate the incoming call.

5. The HSS sends an USSD message to the VMSC.

6. UE-B receives the USSD message from the VMSC after paging for USSD as necessary.

7. UE-B sends a SETUP message to the VMSC to establish the Bearer Control Signalling session. The SETUP message includes the RUA PSI DN as the called party number. This will establish the circuit voice bearer between the UE and IMS.

8. The VMSC responds with a call proceeding message and begins to set up the Bearer Control Signalling session.

9. The VMSC processes the SETUP message and sends an IAM to the MGCF. The SETUP message contains the RUA PSI DN. [NOTE: Standard VMSC procedures for CS origination]

10. The MGCF performs a setup of the MGW and creates an INVITE with an Offer containing the SDP from the MGW. The INVITE is sent via the I/S-CSCF to the ICCF. [NOTE: Standard MGCF procedure for PSTN origination]

11. The I/S-CSCF forwards the INVITE to RUA of ICCF.

12. The ICCF sends an UPDATE to the S-CSCF with an Answer to the Offer from the A-party, containing the SDP from the MGW.

13. The S-CSCF forwards the UPDATE to the A-party.

14. The S-CSCF receives a 200 OK to the UPDATE.

15. The S-CSCF forwards the 200 OK to the ICCF.

16. The ICCF sends a 183 Session Progress to S-CSCF with an Answer to the Offer from the MGW, containing the SDP from the A-party.

17. The S-CSCF sends the 183 Session Progress to the MGCF.

18. The MGCF creates an ACM and sends it to the VMSC.

19. Alerting is send from the VMSC to UE-B.

20. The ICCF sends a 200 OK to S-CSCF in response to the INVITE in Step 12.

21. The S-CSCF forwards the 200 OK to the MGCF.

22. The MGCF creates an ANM and sends it to the VMSC.

23. The VMSC sends a Connect to UE-B. The set up of the bearer is complete.

24. User Alerting at UE-B; UE-B sends Ringing in a USSD message to the VMSC.

25. The VMSC forwards the USSD (Ringing) to the HSS.

26. The HSS forwards the USSD (Ringing) to the CAAF of the ICCF.

27. The CAAF of the ICCF performs the necessary adaptation and relays the information needed for generation of the SIP message at the RUA of the ICCF. The RUA of the ICCF acting as a B2BUA creates a 180 Ringing and sends it to the S-CSCF for communicating with UE-A.

28. The S-CSCF forwards the 180 Ringing to UE-A.

29. User Answer at UE-B; UE-B sends Answer in a USSD message to the VMSC.

30. The VMSC forwards the USSD (Answer) to the HSS.

31. The HSS forwards the USSD (Answer) to the CAAF of the ICCF.

32. The CAAF of the ICCF performs the necessary adaptation and relays the information needed for generation of the SIP message at the RUA of the ICCF. The RUA of the ICCF acting as a B2BUA creates a 200 OK and sends it to the S-CSCF for communicating with UE-A.

33. The S-CSCF forwards the 200 OK to UE-A.

NOTE: Steps 20-23 above related to the Bearer Control Session set-up procedures may alternatively be sent after the 200 OK has been sent from the ICCF towards UE-A.

6.6.3.1.2.3 Calls established using CS bearers without use of ICCC

The figure 6.6.3.1.2.3-1 provides an example for setup of terminating sessions towards a legacy UE or if ICCC is not required, e.g. for provisioning the caller ID.

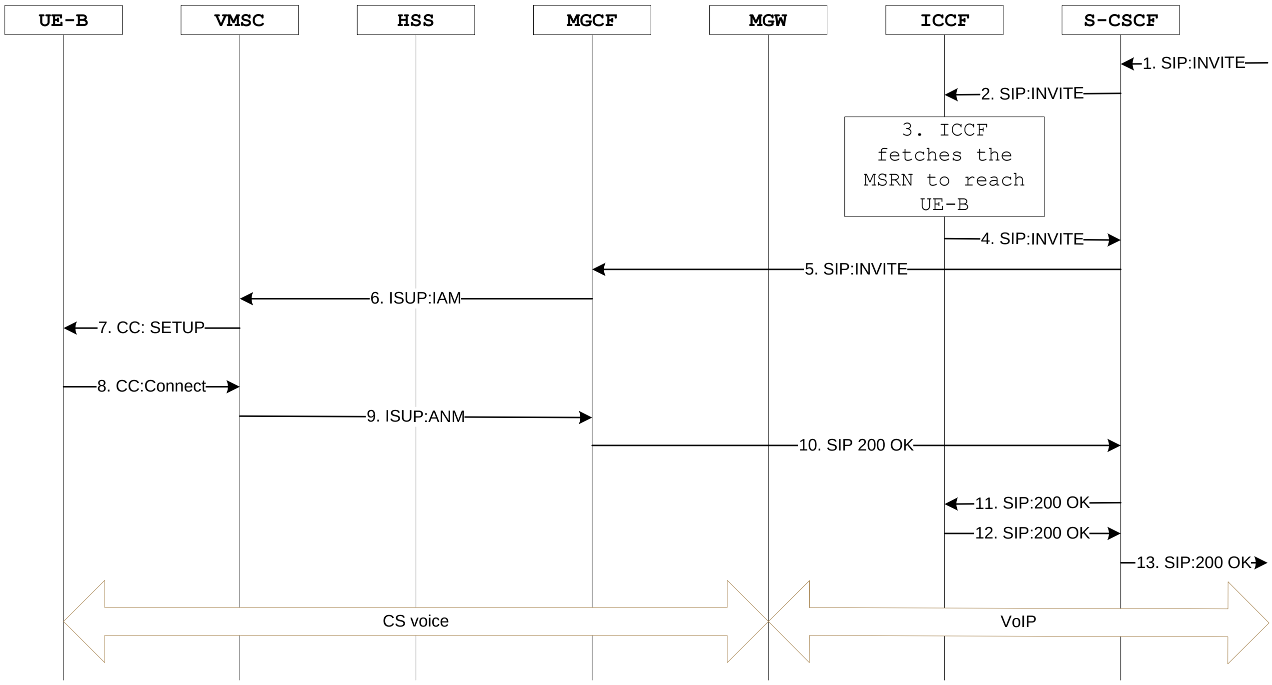


Figure 6.6.3.1.2.3-1: ICS UE termination using CS bearers without use of ICCC

1 An incoming SIP INVITE is received at the S-CSCF of the B party.

2 The S-CSCF forwards the INVITE to the ICCF.

3 The ICCF performs Access Domain Selection, and then fetches a number such as MSRN if a media connection is not already established to UE-B.

Editor's note: The type of number to be used and how the number is allocated is FFS.

4 The ICCF sends an INVITE (tel URI = MSRN) to the S-CSCF.

5 The Request URI has been modified, the S-CSCF skips further service execution and routes the call towards the CS domain.

6 The MGCF sends an IAM to the VMSC.

7 VMSC sends SETUP to UE-B.

8 UE-B responds with a CONNECT message.

9 VMSC responds with ANM towards the IMS.

10 MGCF sends the 200 OK to the S-CSCF.

11 S-CSCF forwards the 200 OK to the ICCF.

12 The ICCF sends a 200 OK as response to the INVITE in step 2 towards the S-CSCF.

13 S-CSCF forwards the 200 OK towards the A party.

6.6.3.1.3 Mid-call services

NOTE: The same call flows apply to the IMS adaptor model (see clause 6.7.2.3).

6.6.3.1.3.1 Calls established using CS bearers with use of USSD transport for I1-cs, AS approach for RUA

Figure 6.6.3.1.3.1-1 provides an example flow for a call made by an ICS UE-A to the other end C after holding the other end B.

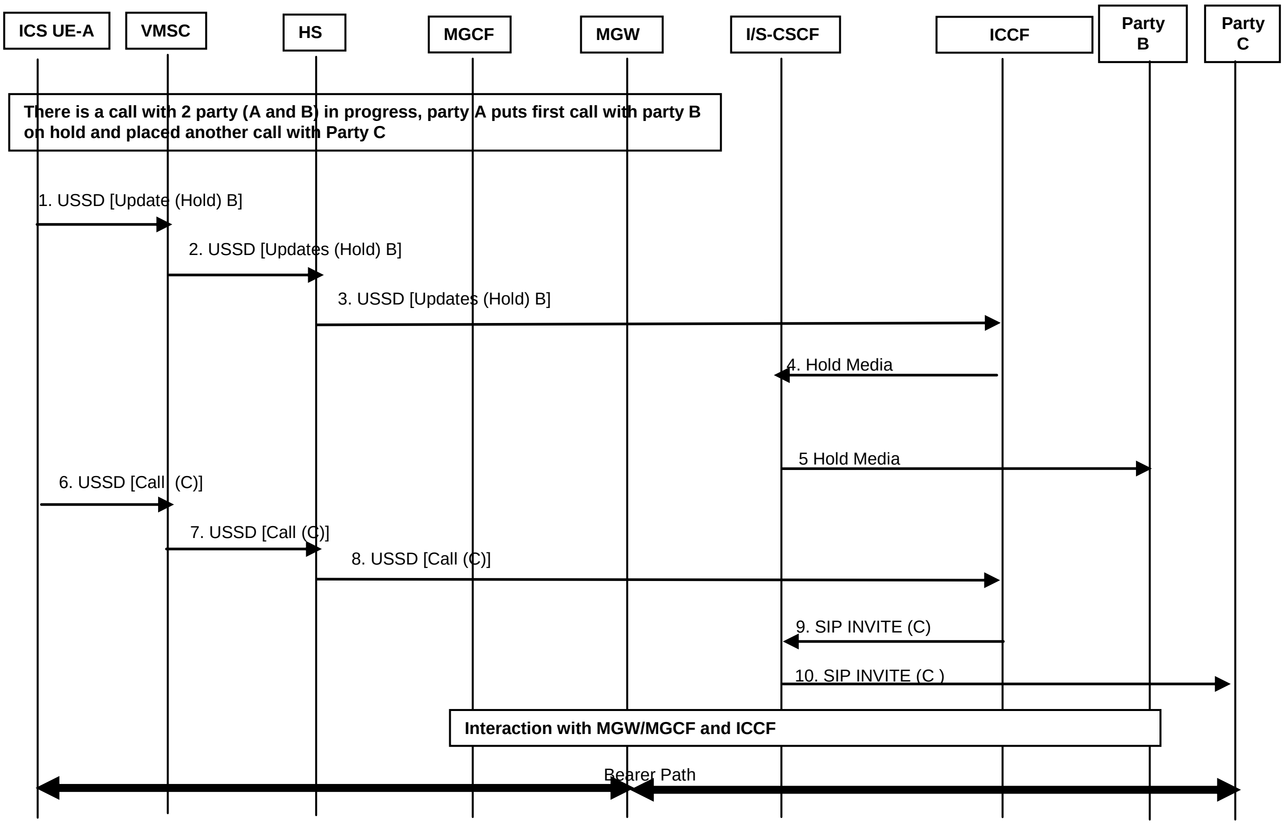


Figure 6.6.3.1.3.1-1: ICS UE mid-call service-A holds B, calls C: USSD transport, AS approach for RUA

1 ICS UE-A to place the call with the other end B on hold. The ICS UE-A encodes in ICCP sufficient information for the RUA of the ICCF to generate a SIP message to hold the media and sends it in a USSD message to the CAAF of the ICCF.

2 Standard VMSC procedure to communicate the USSD message to the Home network.

3 Standard HSS procedure to communicate the USSD message to the associated service node.

4 The CAAF of ICCF terminates the USSD message and communicates it to the RUA of the ICCF. The RUA of the ICCF generates a SIP message to party B to put the media on HOLD.

5 Standard CSCF procedure toward the other end B.

6 ICS UE-A sets up a call toward the other end C by sending a USSD message to the CAAF of the ICCF.

7 Standard VMSC procedure to communicate the USSD message to the Home network.

8 Standard HSS procedure to communicate the USSD message to the associated service node.

9 The CAAF of the ICCF terminates the USSD message; and communicates it to the RUA of the ICCF. The RUA of the ICCF generates a SIP Invite to initiate an IMS session toward the other end C.

10 Standard CSCF procedure toward the other end C.

Editor's Note: The interaction between ICCF/MGCF/MGW in order to connect party A and C is FFS.

6.6.3.1.3.2 Explicit Call Transfer service

6.6.3.1.3.2.1 ICS UE being a transferor

Some messages are omitted for brevity.

Figure 6.6.3.1.3.2.1-1 provides an example flow for ICS UE-A transferring the session to 3rd party.

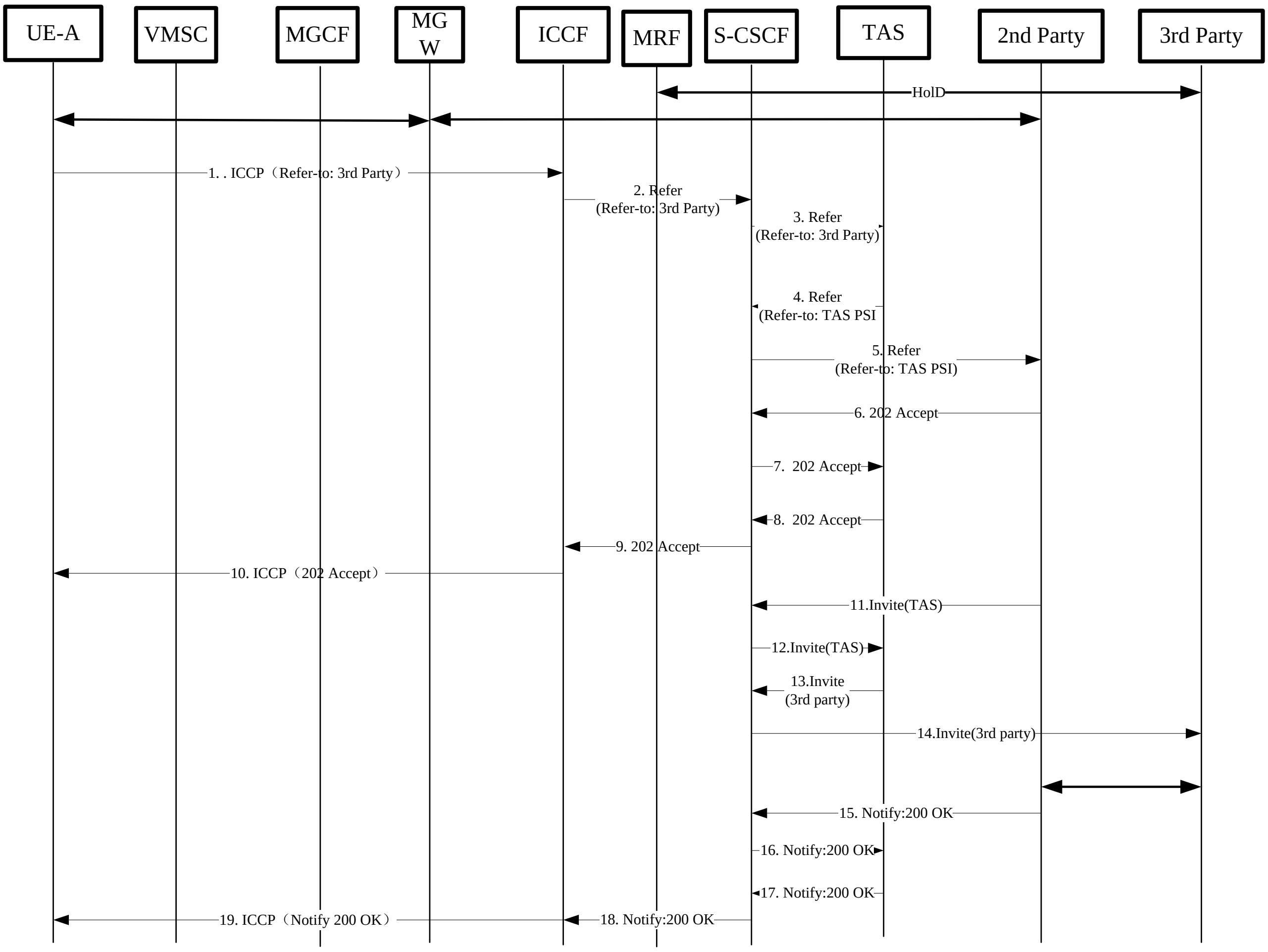


Figure 6.6.3.1.3.2.1-1: ICS UE-A being a transferor transfers the session to 3rd party

1. UE-A have two sessions, the session with 3rd party is on hold， the session with 2nd party is active. UE-A initiates ECT service and sends REFER request with address of 3rd Party to ICCF using ICCP protocol;
2. ICCF sends SIP Refer message to S-CSCF;
3. S-CSCF sends SIP Refer message to TAS.
4. TAS sends SIP Refer message to S-CSCF;
5. S-CSCF sends SIP Refer message to 2nd party.
6. 2nd party sends SIP 202 Accept to S-CSCF;
7. S-CSCF forwards SIP 202 Accept towards the TAS.
8. TAS sends SIP 202 Accept to S-CSCF;
9. S-CSCF sends SIP 202 Accept to ICCF;
10. ICCF sends 202 accept to UE using ICCP protocol.

11~14. Standard ECT procedure to establish a session between 2nd party and 3rd party.

15~18. 2nd party sends notify 200 OK to UE-A.

19. ICCF sends notify 200 OK to UE-A using ICCP protocol.

NOTE: TISPAN is considering that TAS changes the Refer message to Re-invite in order to avoid requirements on transferee (using IP-CAN access).

6.6.3.1.3.2.2 ICS UE being a transferee

Figure 6.6.3.1.3.2.2-1 provides an example flow for ICS UE-A being a transferee, 2nd transfer the session to 3rd party.

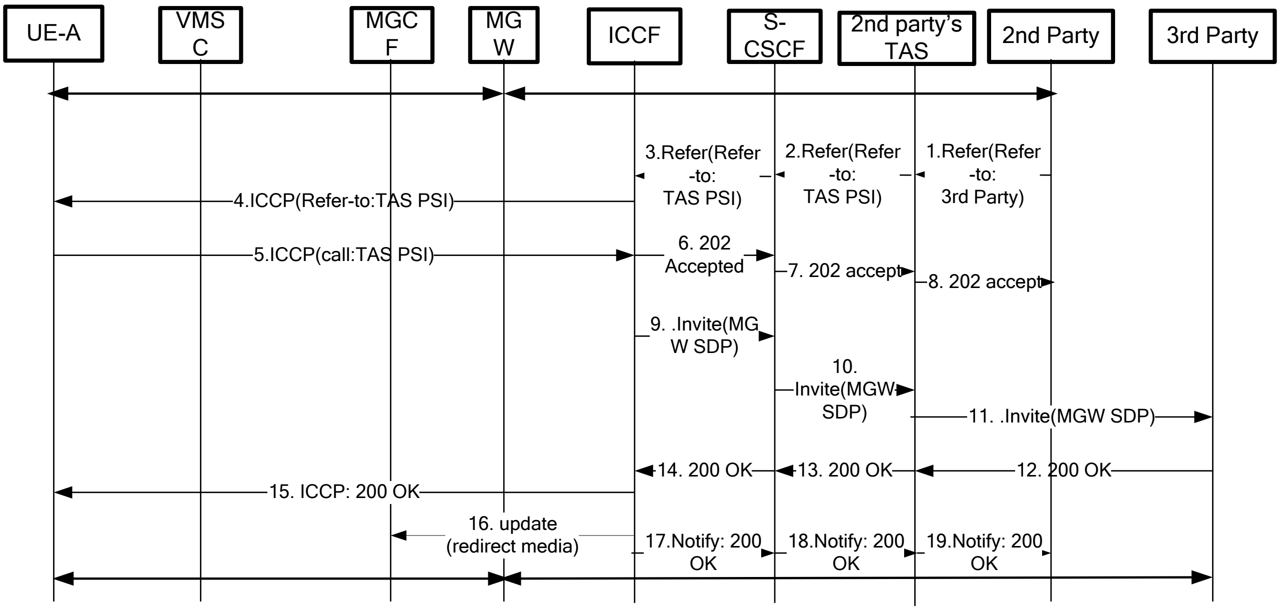


Figure 6.6.3.1.3.2.2-1: ICS UE-A being a transferee, 2nd transfers the session to 3rd party

1. UE-A and 2nd Party are talking. 2nd party initiates ECT service and sends REFER message to TAS.
2. TAS sends refer message to S-CSCF.

NOTE: If TAS sends Re-invite message instead of refer message according to TISPAN group's latest conclusion, ICCF will transfer this re-invite towards ICS UE.

3. S-CSCF sends refer message to ICCF.
4. The ICCF notifies the UE-A through an ICCP message including the REFER target (i.e. address of 3rd Party).
5. The UE-A requests a call to the 3rd Party through an ICCP message.

6~8. ICCF sends 202 accept to 2nd party.

9~11. The ICCF sends an INVITE with MGW SDP of existing CS call leg to the 3rd Party.

12~14. 3rd party sends 200 OK to ICCF.

15. ICCF sends 200 OK to UE using ICCP protocol.

16. ICCF sends UPDATE/Re-invite to the MGCF to redirect the MGW to the 3rd Party.

17~19. ICCF sends notify 200 OK to 2nd party.

6.6.3.1.3.3 CONF service

Figure 6.6.3.1.3.3-1 provides an example flow of ICS UE executing CONF service via the CS access.

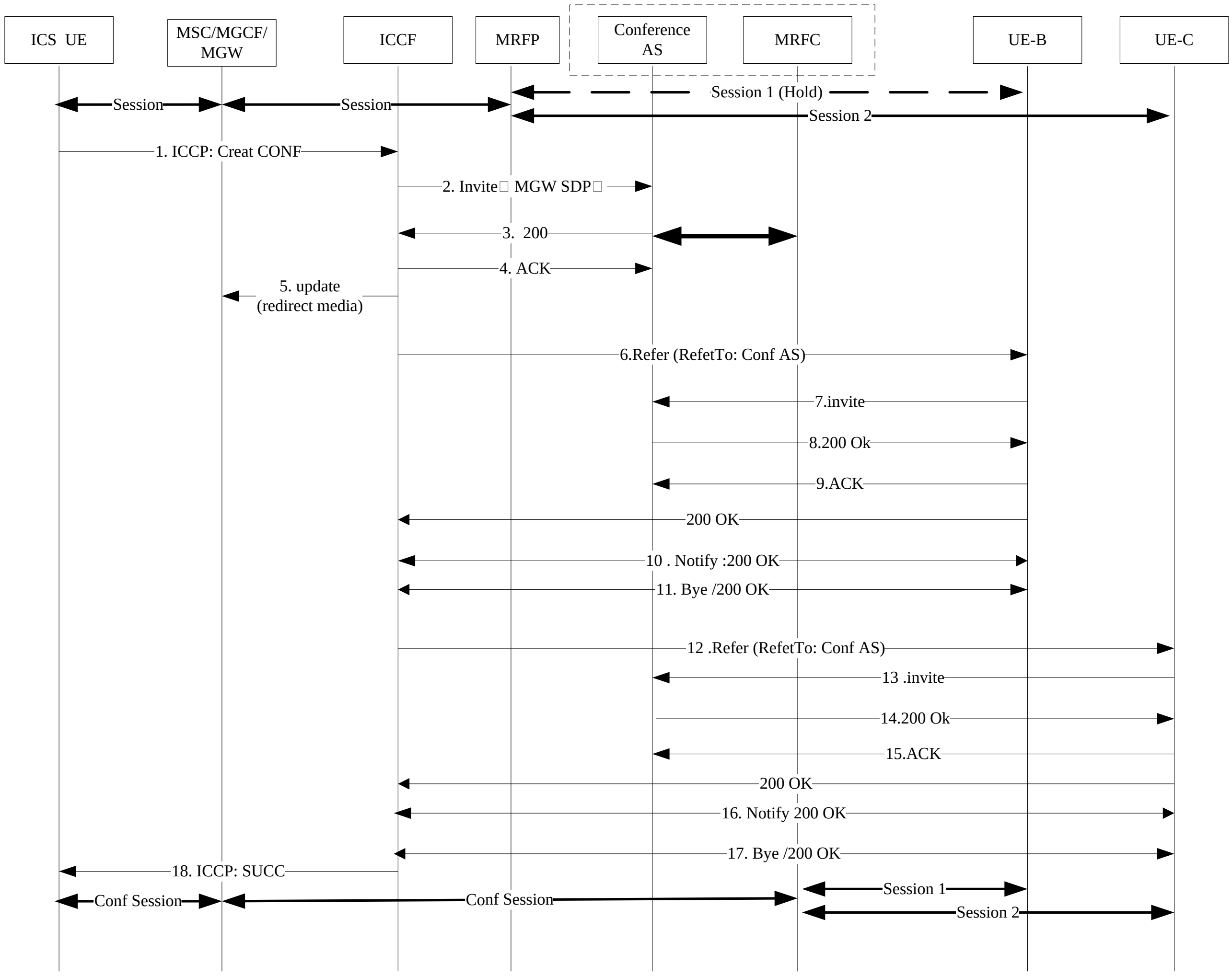


Figure 6.6.2.3.x-1 The flow of UE-A executing CONF service.

1. UE A have two sessions, the session with UE-B is hold, the session with UE-C is active. UE-A requests RUA to establish CONF using ICCP protocol.
2. ICCF sends SIP Invite to Conference AS including MGW SDP.
3. Conference AS responses 200 OK including Conference Focus URI.
4. ICCF sends ACK to Conference AS.
5. ICCF sends update to MGCF to redirect MGW media to the Conference.
6. ICCF sends Refer request towards UE-B to invite UE-B to the Conference.
7. UE-B sends SIP invite request towards Conference AS.
8. Conference AS responds 200 OK to UE-B.
9. UE-B sends ACK to Conference AS.
10. UE-B sends notify 200 OK to ICCF.
11. ICCF sends Bye to UE-B.
12. ICCF sends Refer request towards UE-C to invite UE-C to the Conference.
13. UE-C sends SIP invite request towards Conference AS.
14. Conference AS responds 200 OK to UE-C.

15. UE-C sends ACK to Conference AS.

16. UE-C sends notify 200 OK to ICCF.

NOTE 1: If ICCF is not inserted in PS session path, Notify message will be sent to UE.

17. ICCF sends Bye to UE-C.

18. ICCF sends Conference SUCC message to UE-A using ICCP protocol.

NOTE 2: AS an alternative, ICCF may sends refer message to Conference AS instead of sending to UE-B and UE-C, Conference AS to invite UE-B and UE-C to Conference.

6.6.3.1.4 Domain Transfer

6.6.3.1.4.1 Domain Transfer to IMS

The following figure provides an example flow for Domain Transfer to IMS when the user is engaged in a held voice session with the other end B and an active voice session with the other end C, when using ICS UE and the AS approach for the ICCF.

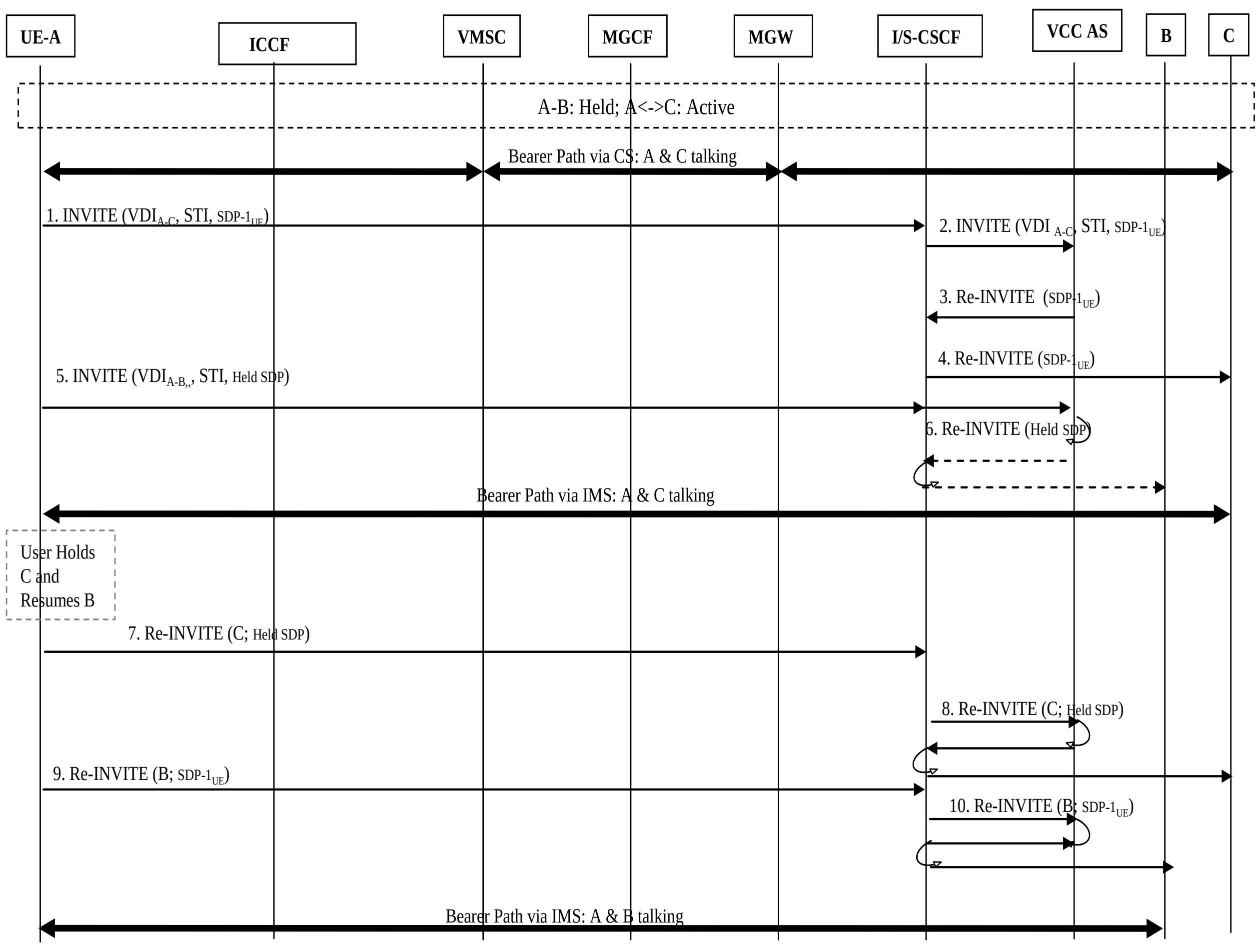


Figure 6.6.3.1.4.1-1: Domain Transfer to IMS

1. Upon detection of conditions requiring Domain Transfer, the UE initiates registration with IMS (if not already registered in IMS). Then, the UE-A initiates the Domain Transfer of the active session to IMS by sending an INVITE to the other end C. If the UE received the STI from the DTF during session establishment then the

domain transfer request contains an STI, otherwise, only the VDI, as defined in TS 23.206 [3], is used. The transfer session is created in the UE with the current state information of the session being transferred.

2. The INVITE is processed at the CSCF using standard procedures and the DTF of VCC AS is invoked as part of standard iFC processing at the CSCF.

3. The DTF of VCC AS uses Access Leg Update procedure to update the Access Leg and the Remote Leg associated with the other end C's session. The DTF of VCC AS extends a Re-INVITE toward the other end C for update of the SDP.

4. Standard Domain Transfer procedure at CSCF for extending the Re-INVITE toward the other end C. An IP bearer path is established between UE-A and the other end C as a result of the Re-INVITE processing at the other end C.

The DTF of VCC AS and UE subsequently release the other end C's Access Leg previously established via CS domain.

5. UE-A sends an INVITE for the held session to the other end B with a held SDP. If the UE received the STI from the DTF during session establishment then the domain transfer request contains an STI, otherwise, only the VDI, as defined in TS 23.206 [3], is used. The INVITE is processed at the CSCF using standard procedures and the DTF of VCC AS is invoked as part of standard iFC processing at the CSCF

   Step 5 may execute in parallel with procedures initiated by step 1.

6. The DTF of the VCC AS updates the Access Leg with the information received in the INVITE. This is needed for completion of the call control signalling path between the UE and the other end B via the Access Leg established with DTF of the VCC AS in IMS.

NOTE: An Access Leg update of the held session toward the other end B (updating of the Remote Leg of the B's session) may not be necessary at this point since active media is not needed for this session until B is "Resumed". However, this procedure may be carried out in parallel to completion of Domain Transfer of the active session toward the other end C as a UE implementation option.

   The DTF of VCC AS and UE subsequently release the other end B's Access Leg previously established via CS domain.

Editors Note 1: Release of pre-DT resources is FFS.

7. User subsequently Holds the other end C and Resumes the other end B. This results in a Re-INVITE with held SDP for C's session to be sent by UE-A.

8. The Re-INVITE with held SDP is communicated to the other end C and the session is Held.

9. A Re-INVITE for Resuming B's session is also sent by the UE-A.

10. The Re-INVITE is processed at the CSCF and the DTF of VCC AS for Access Leg Update of B's session resulting in establishing and/or updating of an IP bearer between the other end B and the UE-A (bearer is established if it was not established in the Held state as part of procedure initiated in step 6; bearer is updated if it was established in the Held state as part of procedure initiated in step 6).

6.6.3.1.4.2 Domain Transfer to CS

The following figure provides an example flow for Domain Transfer to CS when the user is engaged in a held voice session with the other end B and an active voice session with the other end C, when using ICS UE with I1-cs and the AS approach for the ICCF.

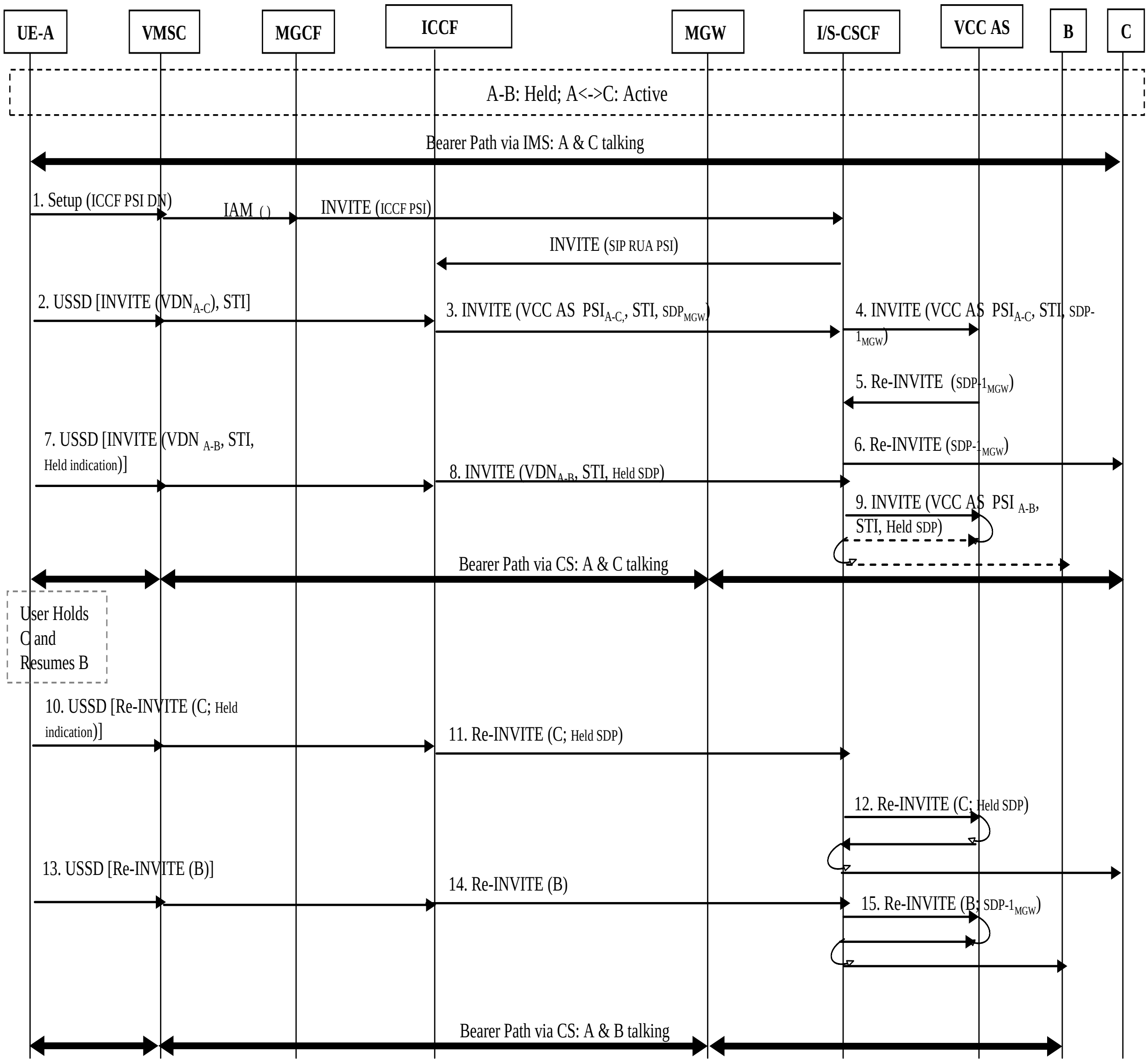


Figure 6.6.3.1.4.2-1: Domain Transfer to CS – I1-cs, AS approach for ICCF

1. Upon detection of conditions requiring Domain Transfer, the UE-A initiates location update with the CS network if not already done and initiates the Domain Transfer to IMS by establishing a Bearer Control Path with RUA of the ICCF through the VMSC.

2. UE-A sends information required to generate INVITE embedded in ICCP over USSD, for initiating the Domain Transfer of the active session with the other end C. If the UE received the STI from the DTF during session establishment then the domain transfer request contains an STI, otherwise only the VDN, as defined in TS 23.206 [3], is used. The transfer session is created in the UE with the current state information of the session being transferred.

Steps 1 and 2 may be combined for signalling optimizations.

The Bearer Control Signalling session established in the INVITE sent by S-CSCF in step 1 is identified at the ICCF by the caller's identity for correlation with the Session Control Signalling session transferred via the INVITE sent by the S-CSCF in step 3 as there can be only one Bearer Control Signalling session for an ICS UE at any given time. If the ICS UE has multiple IMS sessions for voice, a single circuit bearer is alternated between the multiple sessions.

NOTE 1: The ISUP link might (especially in roaming situations) not be able to transfer enough information to correlate the SIP INVITE. Under such circumstances the ICCF may need to interact with a CAMEL Service.

3. The information required to generate the INVITE is extracted from the USSD envelop by the CAAF of ICCF and propagated as SIP INVITE to IMS by the RUA of ICCF.

4. The INVITE is processed at the CSCF using standard procedures and the DTF of the VCC AS is invoked as part of standard iFC processing at the CSCF.

Editor's Note: It is FFS how to identify ME1's public user id relationship when ME1 and ME2 use the same public user id.

5. The DTF of the VCC AS uses Access Leg Update procedure to update the Access Leg and the Remote Leg associated with the other end C's session. The DTF of VCC AS extends a Re-INVITE toward the other end C for update of SDP.

6. Standard Domain Transfer procedure at CSCF for extending the Re-INVITE toward the other end C. A CS bearer path is established between UE-A and the other end C as a result of the Re-INVITE processing at the other end C. The DTF of the VCC AS and UE subsequently release the other end C's Access Leg previously established via IMS.

7. UE-A sends information required to generate INVITE embedded in ICCP over USSD, identifying the held session with the other end B with information indicating a held SDP. If the UE received the STI from the DTF during session establishment then the domain transfer request contains an STI, otherwise only the VDN, as defined in TS 23.206 [3], is used. step 7 could be combined with procedures initiated by Step 2.

8. The information needed to generate INVITE is extracted from the CS access signalling envelop by the CAAF of ICCF and propagated to as SIP INVITE to IMS by the RUA of ICCF.

9. The INVITE is processed at the CSCF using standard procedures and the DTF of VCC AS is invoked as part of standard iFC processing at the CSCF.

The DTF of VCC AS identifies this session as a held session and updates the Access Leg with the information received in the INVITE. This is needed for completion of the call control signalling path between the UE and the other end B via the Access Leg established with DTF of VCC AS via CS domain.

An Access Leg update of the held session toward the other end B, in other words, the updating of the Remote Leg of the B's session may not be necessary at this point since active media is not needed for this session until B is "Resumed". However, this procedure may be carried out in parallel to completion of Domain Transfer of the active session toward the other end C as a UE implementation option.

NOTE 2: The DTF of VCC AS must send re-INVITE also for the held session for continuation of the RTCP reports if previously being sent to the UE port.

The DTF of VCC AS and UE subsequently release the other end B's Access Leg previously established via IMS.

10. User subsequently Holds the other end C and Resumes the other end B. Information required to generate a Re-INVITE embedded in ICCP over USSD, with information indicating held SDP for C's session is sent by UE-A.

11. The information needed to generate Re-INVITE with held SDP is extracted from the CS access signalling envelop by the CAAF of ICCF and propagated as SIP INVITE to IMS by the RUA of ICCF.

12. The Re-INVITE with held SDP is communicated to the other end C and the session is Held.

13. The information needed to generate Re-INVITE embedded in ICCP over USSD, for Resuming B's session is sent by the UE-A. Note that step 13 could be combined with procedures initiated by step 10.

14. The information needed to generate Re-INVITE is extracted from the CS access signalling envelop by the CAAF of ICCF and propagated as SIP INVITE to IMS by the RUA of ICCF.

15. The Re-INVITE is processed at the CSCF and the DTF of VCC AS for Access Leg Update of B's session resulting in establishing and/or updating of an IP bearer between the other end B and the UE-A (bearer is established if it was not established in the Held state as part of procedure initiated in step 9; bearer is updated if it was established in the Held state as part of procedure initiated in step 9).

6.6.3.1.5 Managing Network Services related Information

6.6.3.1.5.1 Using I1-cs

The following figure provides an example flow for Managing Network Services related Information, when using ICS UE with I1-cs and the AS approach for the ICCF. The I5 reference point is used to manage network services related information from the ICCF. This does not preclude the use of other mechanisms for managing network services related information, e.g. using OSA or OA&M mechanisms; the details of these other mechanisms are out of scope of this document.

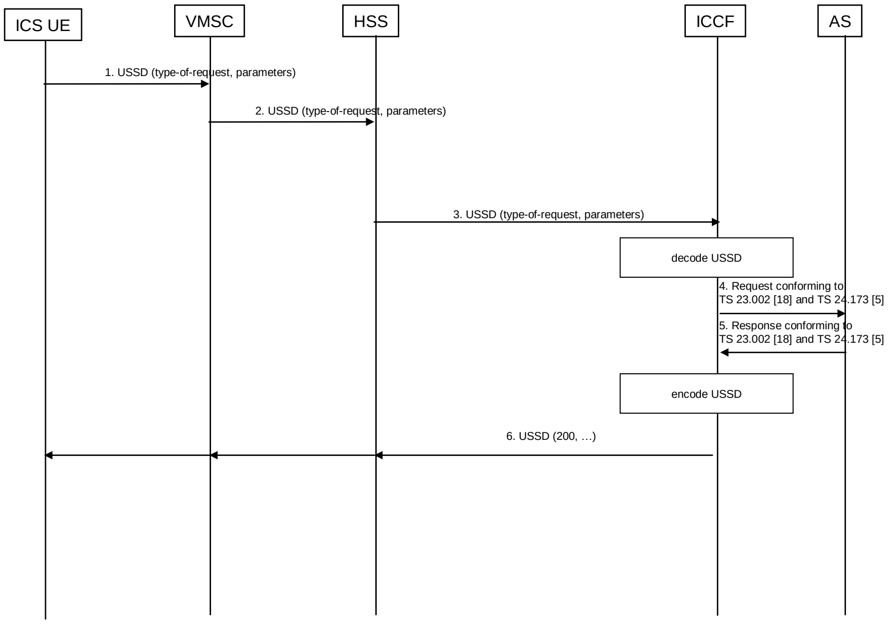


Figure 6.6.3.1.5.1-1: Managing Network Services related Information – using I1-cs

1. ICS UE initiates management of network services related information. The ICS UE encodes in ICCP sufficient information for the ICCF to use capabilities defined for the Ut reference point as defined in TS 23.002 [18] and TS 24.173 [5] for managing network services related information and sends it in a USSD message to the CAAF part of the ICCF.
2. Standard VMSC procedure to communicate the USSD message to the Home network.
3. Standard HSS procedure to communicate the USSD message to the associated service node

NOTE 1: The ICCF maybe statically assigned. Details are FFS.

4. The CAAF part of the ICCF terminates the USSD message. According to clause 6.2.2, the ICCF has the RPC part of the ICCF use capabilities defined for the Ut reference point as defined in TS 23.002 [18] and TS 24.173 [5] for managing network services related information.
5. The AS generates a response to the ICCF.
6. This step consists of a standard service node procedure to communicate the USSD message to the HSS, a standard HSS procedure to communicate the USSD message to the VMSC, and a standard VMSC procedure to communicate the USSD message to the ICS UE.

6.6.3.1.5.2 Using IP-CAN

As specified in TS 24.173 [5].

6.6.3.2 I1-cs: registered user solution – ISC model

6.6.3.2.1 IMS Registration via CS access

6.6.3.2.2 Origination, termination, Mid-call services and Domain Transfer

The information flows for Originations, Terminations, Mid Call services and Domain Transfers are the same as in clause 6.6.3.1, unless specified below.

6.6.3.2.3 Managing Network Services related Information

See clause 6.6.3.1.5, Managing Network Services related Information.

6.7 Architectural alternative: I1-cs IMS Adaptor approach

6.7.1 Signalling and bearer architecture for full duplex speech over CS access

This model is centred around the concept of emulating a standard IMS capable UE accessing IMS services via CS access, as an IMS end point, hence CS access specific adaptations are handled in the ICCF. In this approach, the ICCF acts as a light-weight P-CSCF (e.g. only includes a simplified IMS security and doesn't include policy handling towards the PCRF) towards the IMS core (see also figure 6.7.1-1). The ICCF is not impacted by additional IMS sessions established over IP-CAN in case a suitable IP-CAN is available.

Transparent CS Signalling, using USSD dialogues, is used to communicate needed session control signalling information from the UE using I1-cs via the MSC-S / MSC/VLR through the HSS (at initial registration in IMS and when UE is roaming in a visited domain), or directly from the MSC-S / MSC/VLR, when UE is roaming or at home, to the ICCF (serving MSC assumed to have a suitable USSD application for ICS). I1-cs is terminated in the CAAF of the ICCF and the CAAF performs necessary adaptation for the I1-cs when relaying the Session Control Signalling to/from the RUA which presents SIP UA behaviour on behalf of the UE toward IMS.

Editor's Note 1: IMS registration performed over the RUA leg is FFS.

Editor's Note 2: The IMS registration performed over the RUA leg and the security implications are for further study.

For establishment of IMS sessions via CS access, the UE establishes a media control signalling path with the SIP UA (RUA) within the ICCF by establishing a CS call toward the ICCF (the CAAF in the ICCF is bypassed). Different options do exist to route the CS call from the UE to the ICCF. In parallel, it establishes a session control signalling path with the USSD Handler in the CAAF within the ICCF using transparent CS Signalling using USSD as described above. The media control Signalling and session control Signalling are combined in the RUA at the ICCF for presentation of SIP UA behaviour for establishment of an IMS session.

The same principle applies for a terminating call to the UE, ie. I1-cs is used to carry signalling needed for establishment and/or for any required control of terminating IMS sessions from the ICCF to the UE, and media transport over CS access is either established or an already established one is reused.

The UE uses a session control signalling path via the ICCF to control the service related to the first session (e.g. mid call handling). Subsequent call related input is communicated to the ICCF using the session control signalling path; i.e. CS call setup procedures are not used for establishment of subsequent user sessions or for invocation of mid-call voice services. Upon Domain Transfer from CS access to PS access, the CS access related service state is released in both the ICCF and the UE.

Use of the I1-cs for session setup enables provisioning of MMTel bi-directional speech services to ICS UEs when using CS access with this model.

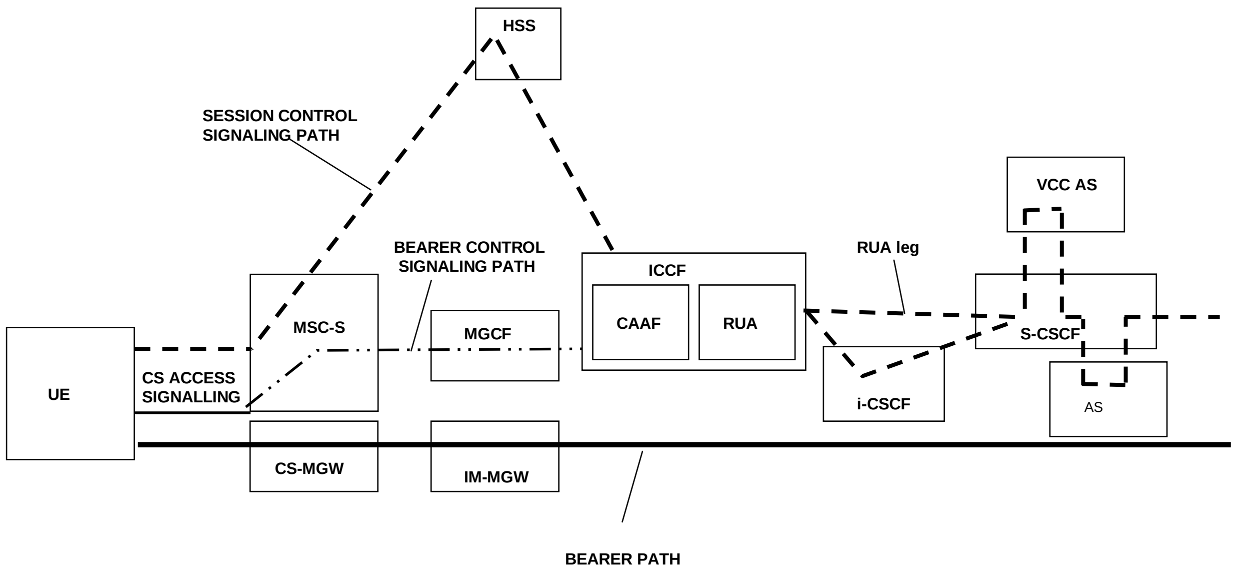


Figure 6.7.1-1: Signalling/Bearer Paths: USSD transport of ICCP with IMS Adaptor approach

NOTE 1: Some details omitted for brevity.

NOTE 2: In some cases the bearer control signalling path can carry session control signalling for IMS session as well.

6.7.1a Registration

6.7.1a.1 RUA Registration in IMS when attached to CS access

6.7.1a.1.1 Triggers for RUA Registration / De-registration

The following table contains possible triggers to start / stop RUA registration. In addition, the table contains information whether this trigger is of use for ICS UE or non ICS UE, respectively:

Table 6.7.1a.1.1-1: Possible triggers to start / stop RUA registration

Possible trigger to start / stop RUA registration	ICS UE	non ICS UE
Push by UE: using ICCP to ICCF	Yes	No
Push by network: Using CAMEL ph3 M-CSI (only attach / detach)	Yes	Yes
Push: new HSS functionality (attach / detach)	FFS	FFS
Pull: CAMEL ph2 ATI poll to retrieve attach / detach status	Yes	Yes
Registration Timeout in ICCF	Yes	Yes

Some of the above-listed triggers are combined into a solution for ICS UEs which can be enhanced to support non ICS UEs as well:

- To support ICS UEs: -
  - Registration trigger using ICCP from the UE to the ICCF right after IMSI attach; de-registration trigger via ICCP if possible before detach
  - Timeout in the ICCF to de-register in IMS (e.g. in case the UE could not de-register via ICCP).
    - If registration timeout occurs, and the user is engaged in a call, the ICCF re-registers the ICS user in IMS

  - If registration timeout occurs, and the user is not engaged in a call, the ICCF performs a Sh or MAP ATI poll to HSS to check CS status of an ICS user.
  - If an ICS user is detached, then the ICCF de-registers the ICS user in IMS.
  - If an ICS user is attached, then the ICCF re-registers the ICS user in IMS and starts a registration timer again.

  NOTE: The higher the value of the registration timeout timer is, the lower the signalling load.

- To support non ICS UEs, either of the following solutions should be applied:
  - M-CSI attach / detach trigger to inform the ICCF in case the UE has attached / detached (note: this requires CAMEL PH3 support in VPLMN and HPLMN)
  - The HSS pushes attach / detach trigger to inform the ICCF in case the UE has attached / detached (note: details of solution are FFS, especially how to detect detach in VPLMN)

6.7.1a.1.2 RUA Registration by the ICCF

The RUA in the ICCF performs a trusted registration on behalf of the UE. The solution for trusted registration by the RUA in the ICCF is similar to Early IMS security as specified in 3GPP Rel-6 TR 33.978 (see S2-071863 for details). In summary, the trusted registration allows the ICCF to register on behalf of a CS UE, based on the fact that it can be verified that the UE has been authenticated in the CS domain (this could be checked e.g. at the ICCF or at the HSS).

Similar to Early IMS, the RUA in the ICCF registers using a Temporary-IMPU based on the IMSI, which allows the I-CSCF and S-CSCF to derive the IMPI based on this T-IMPU.

When a register request is received without the Authorization-Header, the S-CSCF first checks the IP address in the "Contact" header field as well as the received IP address of the first Via header with the list of pre-configured ICCF IP addresses. If there is a match, it is considered as an ICS Registration and the UE is considered authenticated.

It has been concluded by SA3 that the overall procedures provide sufficient security for the purposes of ICS and does not impose any new threats and risks to the IMS system, assuming that the ICCF is within the same security domain as the S-CSCF and that it also exists within the same IMS trust domain.

The registered contact address is the ICCF, i.e., if the Request-URI would be matched, then a terminating call would be routed to the ICCF.

The RUA Registration provides the following information into the contact in the S-CSCF

- Feature tag to indicate capability audio only. A terminating SIP INVITE which is indicating other capabilities would not match the CS contact.
- Indicate in CS access characteristics (details are stage 3 issue). Possible solutions are:
  - Indicates CS access in P-Access-Network-Info as new access network type (e.g. "GERAN-cs", "UTRAN-cs").
  - New feature tag indicating CS access

NOTE: CS access characteristics is only set by ICCF.

To select between PS contact and CS contact bound to one IMPU, DSF will provide sufficient information in the request message sent to S-CSCF to indicate if it should select PS or CS contact. So with RUA registration there is no possibility that S-CSCF will fork to both PS and CS contact, i.e. the possibility to fork to one terminal twice. Standard measure can be applied here, such as Accept-Contact, Reject-Contact, etc.

On reception of a REGISTER request, the S-CSCF sends a third-party REGISTER request to each Application Server that matches the Filter Criteria sent from the HSS for the REGISTER request. If it is necessary to match only the initial filter criteria for ICCF registration but not for UE registration (or vice versa), the content of a PANI header or a contact headers could be used to differentiate those cases as criteria, e.g. if an AS always assumes that a UE is accessible via PS when registered, a third-party REGISTER request should not be sent to the AS when the UE is registered via ICCF from CS. But in the light of ICS principle to provide the IMS services both in CS access and PS access in the similar manner, the requirements to differentiate those cases have to be further studied and understood. The Application Server can use the RegEvent package as specified in RFC 3680 to see whether a user is IMS registered and to know all of the registered public IDs which are in the same implicit registered set.

NOTE: In case the ICS UE would also register in IMS over Gm reference point, then the S-CSCF maintains for both the registration over Gm and the RUA registration a contact with a unique IMPI/IMPU binding. Each contact can be de-registered individually.

6.7.1a.1.3 Originating / terminating call

CS Originating session:

If a session is originated in CS access, it is routed via PSI routing directly to the ICCF. The ICCF routes then the INVITE to the S-CSCF using standard call origination procedures.

Considering that a call originated in PS access does not need to involve ICCF in the originating session, ICCF is not included in originating iFC.

Followed the ICCF, the S-CSCF is trying to match the filter criteria for that SIP request according to their priority for other Application Servers. Only if the filter criteria are matched then the application server is invoked. The first AS to be invoked after ICCF for VCC capable ICS-UE would be DTF.

Terminating session:

The terminating session is routed first to S-CSCF using standard procedures. The S-CSCF invokes first other Application Servers if the corresponding filter criteria are matched. And ICCF is not included in terminating iFC. The last AS invoked via iFC performs access domain selection, And ICCF is only involved when CS access is selected. There are two cases:

- If CS access is selected, sufficient information (e.g. Accept-Contact with feature tag indicating CS) is indicated in the INVITE message only to select CS contact in the S-CSCF. The S-CSCF selects ICCF which contact from CS access is registered and route the INVITE message to ICCF.
- If PS access is selected, sufficient information (e.g. Reject-Contact with feature tag indicating CS) is indicated in the INVITE message only to select PS contact in the S-CSCF.

Editor's note: Alternative solution which does not need to perform access domain selection is FFS.

6.7.1a.1.4 RUA De-Registration by the ICCF

The ICCF will de-register the contact for the ICS UE if it receives a corresponding trigger, as discussed in clause 6.7.1a.1.1. In case the ICCF does not re-register on behalf of the ICS UE, the RUA registration will time out.

6.7.1a.1.5 Network De-Registration

In case the S-CSCF initiates network de-registration, as the ICCF has subscribed to reg-event package, the ICCF will be notified by the network and it can subsequently inform the ICS UE via USSD.

6.7.1a.1.6 RUA registration use cases

NOTE: These use cases are applicable to both IA model and AS model for I1-cs.

Use case 1: RUA registration:

This example exemplifies the case in which the trigger for registration is sent by UE (see also clause 6.6.2a.1.2.2 which describes the same use case).

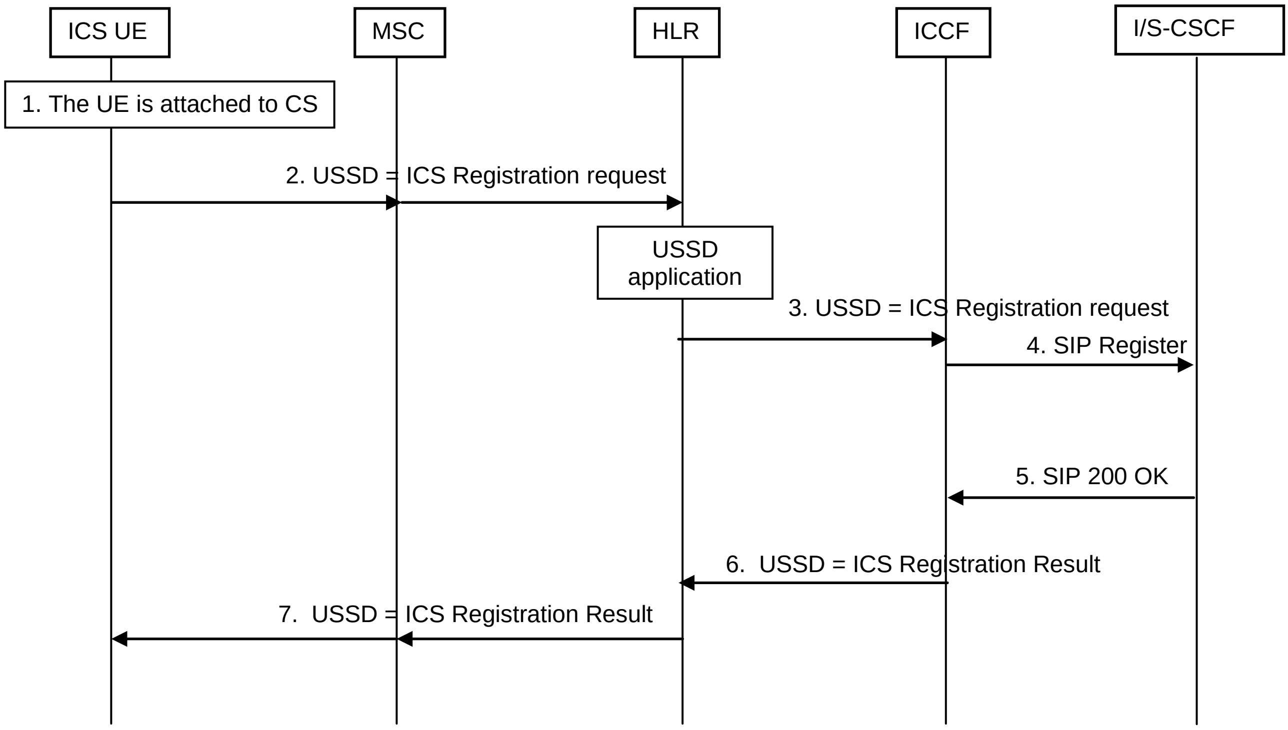


Figure 6.7.1a.1.6-1: RUA registration

1. ICS UE attaches to CS access.
2. Upon CS attachment, ICS UE sends ICS registration request towards HLR using USSD via MSC.
3. HLR forwards the request to targeted ICCF.
4. The RUA in ICCF composes the SIP REGISTER message based on the received USSD message as described in 6.7.1a.1.2, and sends it to IMS network.
5. IMS Registration is done. It should be noted that since ICCF is a trusted node, there is no need to return SIP 401 response. Instead SIP 200 OK is sent back to ICCF.
6. ICCF sends a USSD message back to HLR, informing the result of the registration request.
7. HLR sends the USSD message to the ICS UE via MSC.

Use case 2: RUA registration followed by IMS registration:

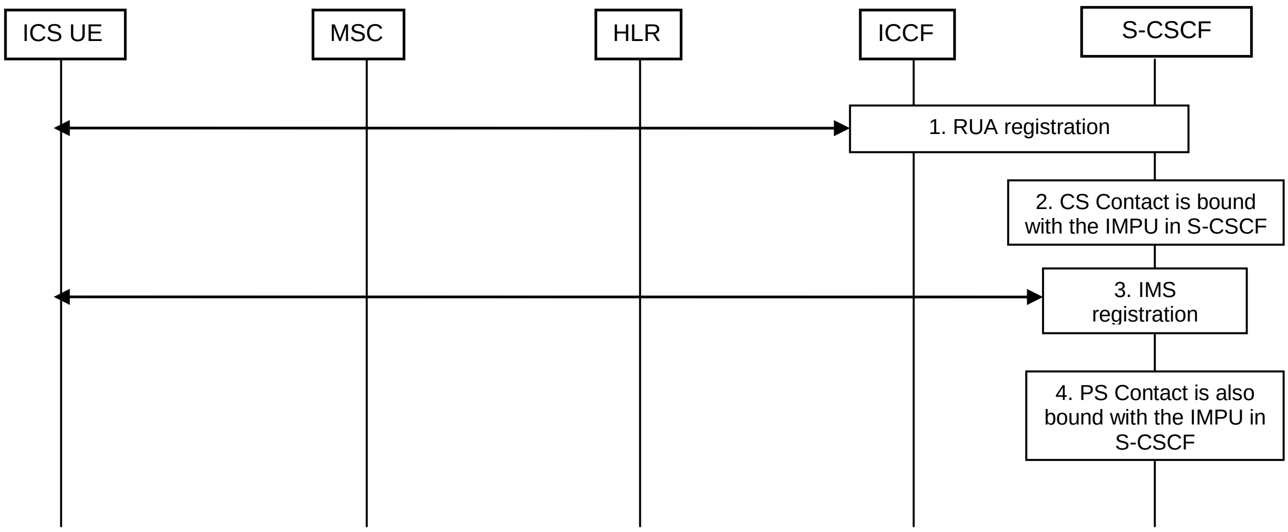


Figure 6.7.1a.1.6-2: RUA registration followed by IMS registration

1. When the ICS UE attaches to CS access, it does the RUA registration as shown in Use Case 1.

2. After the RUA registration, the CS contact, which points to the ICCF, is now bound with the IMPU of the ICS UE in S-CSCF.

3. Now the ICS UE obtains PS access, it executes the standard IMS registration in IMS.

4. After the IMS registration, the PS contact is now also bound with the IMPU of the ICS UE.

Use case 3: IMS registration followed by RUA registration:

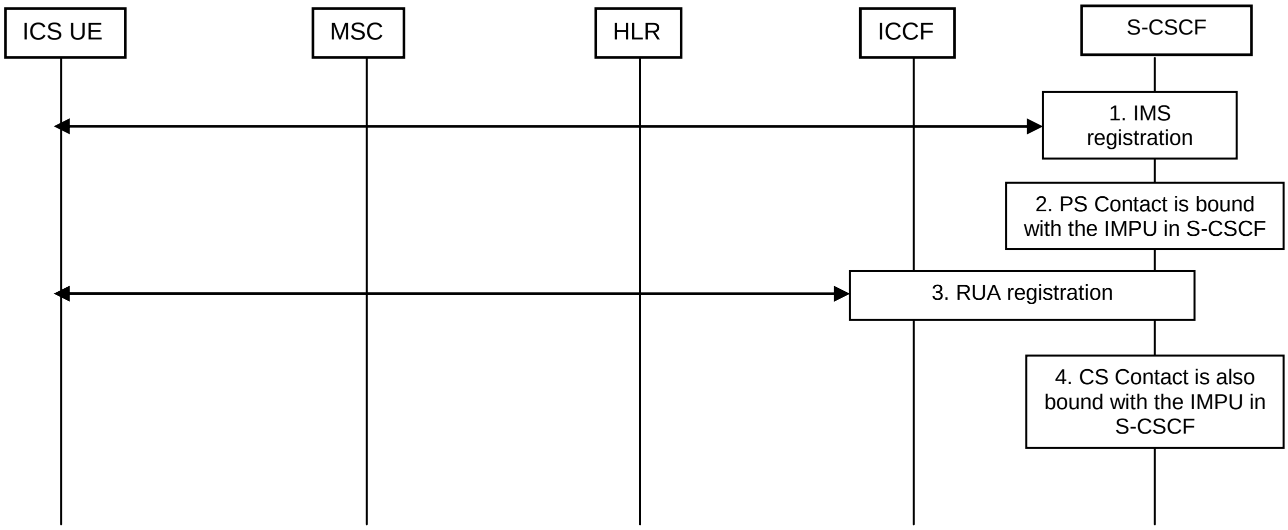


Figure 6.7.1a.1.6-3: IMS registration followed by RUA registration

1. When the ICS UE attaches to PS access, it does the IMS registration.

2. After the IMS registration, the PS contact is now bound with the IMPU of the ICS UE in S-CSCF.

3. Now the ICS UE obtains CS access, it executes the RUA registration as shown in Use Case 1.

4. After the RUA registration, the CS contact, which points to the ICCF, is now also bound with the IMPU of the ICS UE in S-CSCF.

Use case 4: RUA de-registration:

This example exemplifies the case in which the trigger for de-registration is sent by UE.

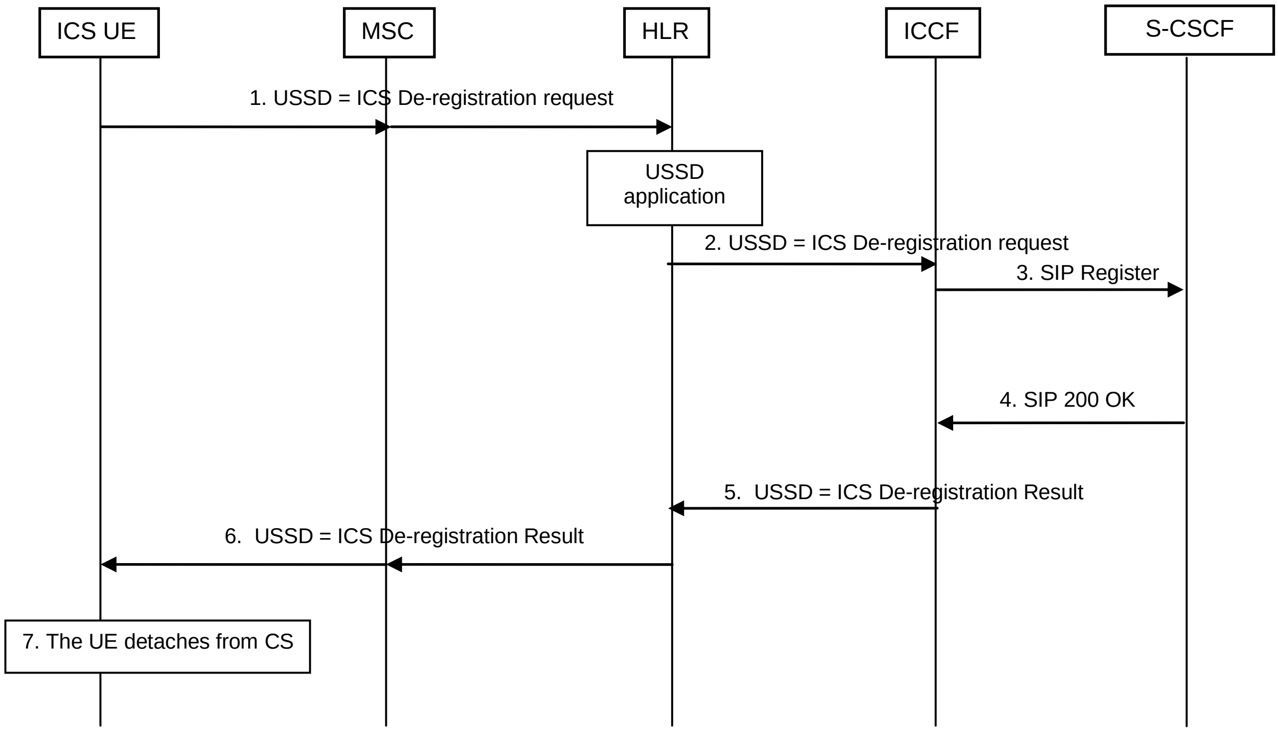


Figure 6.7.1a.6-4: RUA de-registration

1. ICS UE sends ICS de-registration request towards HLR using USSD via MSC.
2. HLR forwards the request to targeted ICCF.
3. The RUA in ICCF composes the SIP REGISTER message, with the Expire header's value set to zero, based on the received USSD message, and sends it to IMS network.
4. IMS deregistration is done. SIP 200 OK is sent back to ICCF.
5. ICCF sends a USSD message back to HLR, informing the result of the de-registration request.
6. HLR sends the USSD message to the ICS UE via MSC.
7. The ICS UE detaches from CS access.

6.7.2 Information flows

6.7.2.1 Origination

6.7.2.1.1 Use of USSD transport in IMS Adaptor approach for session control

The figure 6.7.2.1.1-1 provides an example flow for a call made by an ICS UE to UE-B in the IMS Adaptor approach when using USSD transport of ICCC, when ICCC is used for setup of all sessions.

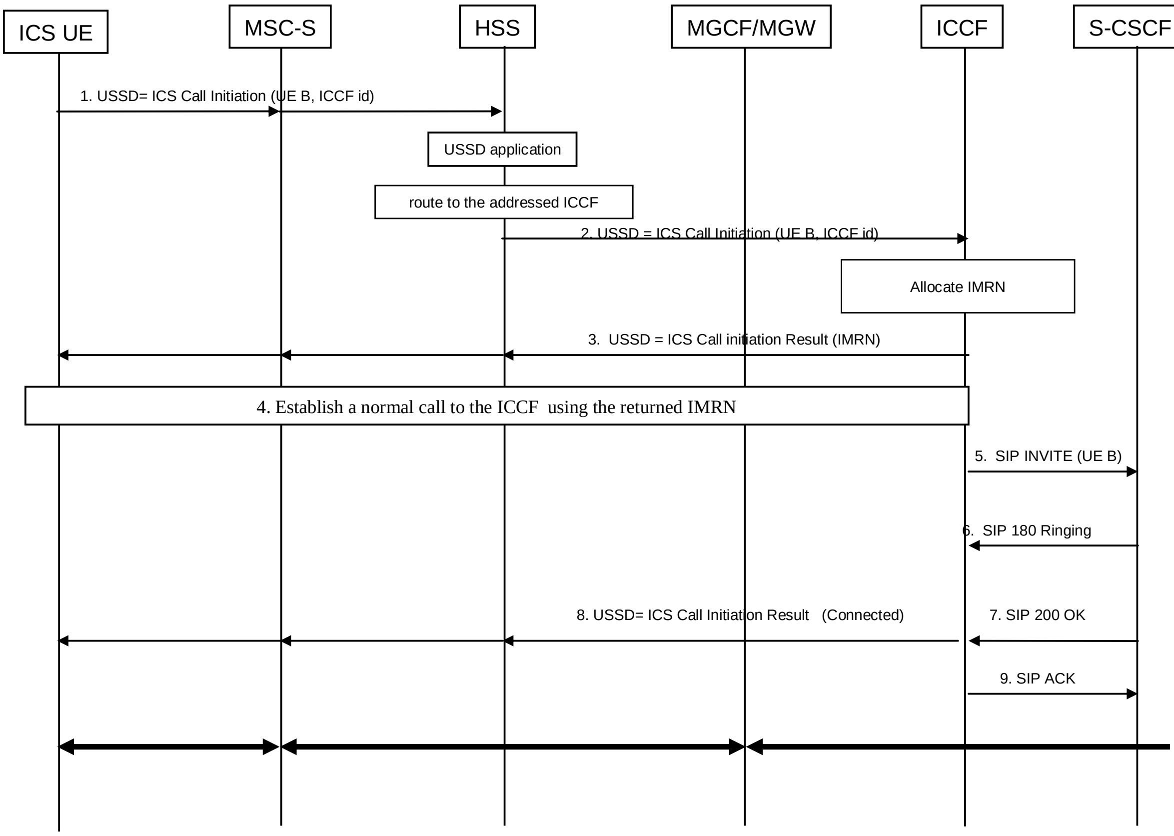


Figure 6.7.2.1.1-1: Use of USSD transport in IMS Adaptor approach for originating session control

1. The ICS UE initiates a call by sending an ICS Call Initiation Request inside a USSD dialog to the HSS containing B party address (SIP or Tel URI).

2. The USSD application within the HSS forwards the USSD Request to the ICCF instance addressed in the HSS. The ICCF shall store the received B Party Address's against an IMRN.

3. The ICCF allocates a roaming number if a media connection is not already established to the ICS UE. The ICCF returns the IMRN in an ICS Response within the USSD dialog. The USSD response is returned all the way to the ICS UE.

4. The ICS UE uses the IMRN to establish a normal CS call set up to the ICCF via MGCF. PSI routing is used to route the call from MGCF to ICCF.

5. Following that, ICCF initiates the call towards UE-B by sending a SIP INVITE to the S-CSCF.

6. ICCF receives SIP 180 ringing.

7. ICCF receives SIP 200 OK.

8. ICCF send an ICS Call initiation response to the ICS UE that includes the connected status.

9. ICCF sends an ACK to the UE-B and the media is now established end to end.

6.7.2.2 Termination

6.7.2.2.1 Use of USSD transport in IMS Adaptor approach for session control

Figure 6.7.2.2.1-1 provides an example flow for a call destined to an ICS UE when USSD transport for ICCP is used to support the setup of all terminating sessions.

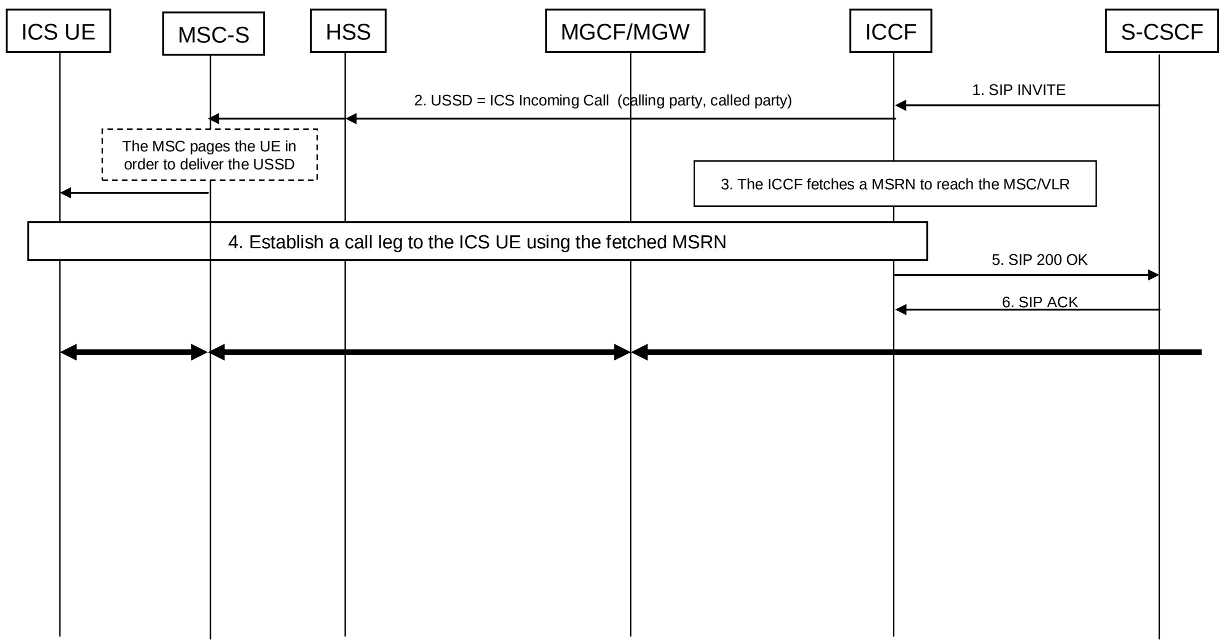


Figure 6.7.2.2.1-1: Use of USSD transport in IMS Adaptor approach for session terminations

1. An incoming SIP INVITE is received at the ICCF instance bound to the actual target ICS UE of the call.
2. ICCF initiates an ICS Incoming Call Request USSD to the MSC-S where the ICS UE is. The MSC pages the ICS UE. This step is optional.
3. ICCF then fetches a number such as MSRN if a media connection is not already established to the ICS UE.

Editor's note 1: The types of number to be used and how the number is allocated are FFS.

4. ICCF establishes a call leg using the MSRN (or other type of number) for that purpose. PSI routing is used to route the call from MGCF to ICCF.

Editor's note 2: The relationship and/or interaction between step 2 an 4 is FFS.

5. ICCF sends a SIP 200 OK to the S-CSCF.
6. S-CSCF returns a SIP ACK and the media path is now established end to end.

6.7.2.2.2 Use of USSD transport in IMS Adaptor approach for session control

Figure 6.7.2.2.2-1 provides an example flow for a call destined to an ICS UE when USSD transport for ICCP is used to support the setup of all terminating sessions.

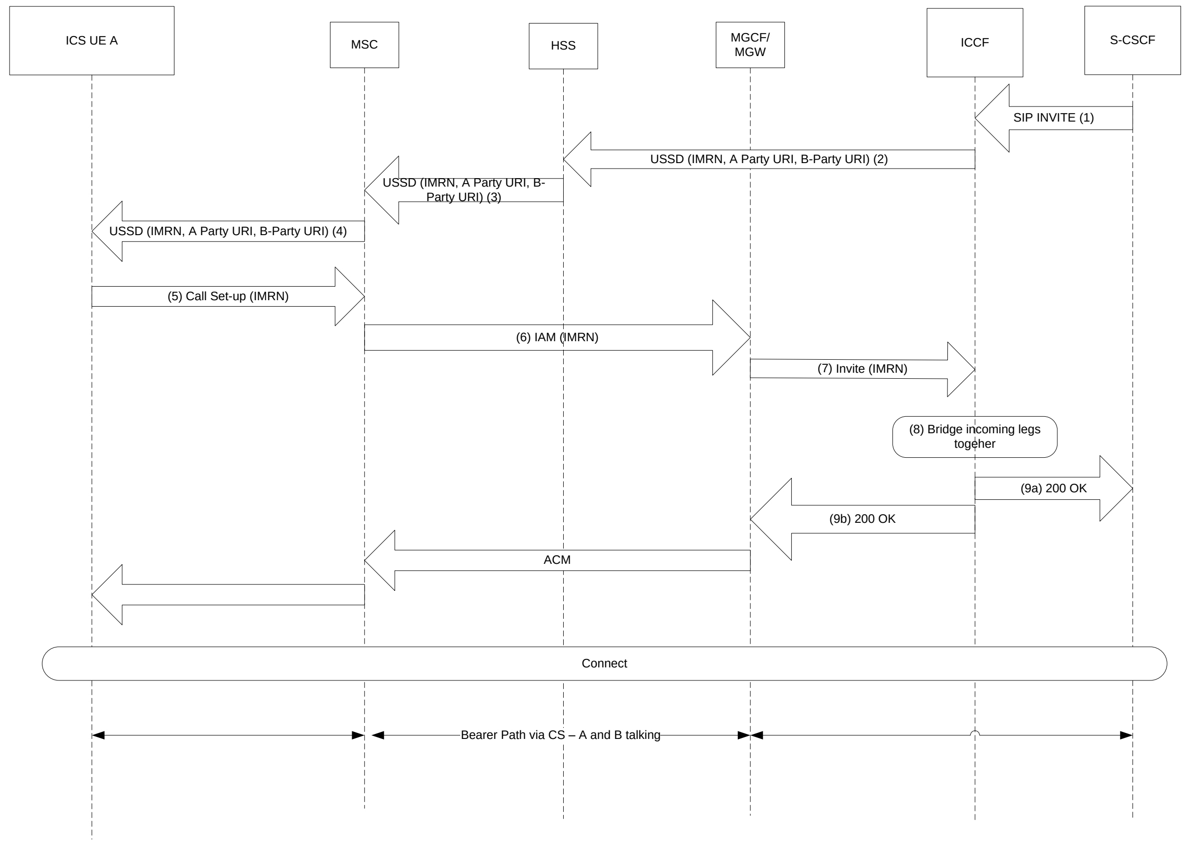


Figure 6.7.2.2.2-1: Use of USSD transport in IMS Adaptor approach for session terminations

1. An incoming SIP INVITE is received at the ICCF instance bound to the actual target ICS UE of the call. The ICCF shall store the information received in the INVITE such as but not limited to TO, FROM, CONTACT, PRIVACY INDICATION, GRUU, R-URI, SDP etc. Against this information shall be assigned an IMRN which is an E.164 number. The E.164 number or IMRN could be dynamic or may even be assigned sequentially. However the property of the IMRN shall be such that if the ICCF receives it back it shall be able to deduce the stored information and the corresponding incoming call leg.

2-4. ICCF initiates an ICS Incoming Call Request using USSD containing but not limited to: Flag to indicate its an ICS message, a flag to indicate it is for MT call, A Party – if allowed or indication that A-party ID is withheld, B-Party address – SIP or Tel URI, GRUU, the called ID so that the B party knows which public user ID it was called on to the MSC-S where the ICS UE is. The MSC pages the ICS UE, this step is optional. A USSD response is sent out towards the IMRN.

5. The UE shall originate a voice call in the CS domain according to "Information flow for an MO call" in TS 23.018 [3] using the assigned E.164 number to establish an Access Leg via the CS domain to the ICCF.

6. The originating call is processed in the CS network according to CS origination procedures described in TS23.206 clause 6.2.2 VCC UE Origination from CS Domain for routing to IMS. However the CAMEL triggers shall be deactivated due to CAMEL deactivation trigger information being provided in the MAP Insert Subscriber Data. The VMSC routes the call towards the user's home IMS network via an MGCF.

7. The MGCF initiates an INVITE towards the I-CSCF in the home IMS of the originating UE.

7b. The I-CSCF routes the INVITE to the ICCF on one of the following standard procedures specified in "PSI based Application Server termination - direct and PSI based Application Server termination - indirect" procedures in TS 23.228 [4].

NOTE 1: Direct routing to ICCF is shown, although routing via S-CSCF is also possible.

NOTE 2: For PSI, the ISC & Ma Reference Points are supported by the ICCF (see clause 6.2.2.2.2).

8. The ICCF analyzes the received E.164 number and correlates it with the stored information so that it can complete the incoming call and bridge the incoming call from the B party with the incoming call from the A

party. The E.164 number shall then be returned to the quarantine pool after which it maybe reused again at some suitable point

9a. ICCF returns a SIP 200 OK to the S-CSCF. Note: steps 9a and 9b can be executed in parallel or in a different order.

9b. ICCF returns a SIP 200 OK and the media path is now established end to end.

NOTE 3: charging records will need to be correlated. The IMRN could be used for this.

6.7.2.3 Mid-call services

6.7.2.3.1 General

Example information flows for mid-call services for the signalling/bearer architecture for full-duplex speech service with IMS centralized services over CS access are discussed in the following.

6.7.2.3.2 Mid-call event handling

Figure 6.7.2.3.2-1 provides an example flow when an ICS UE receiving services in IMS is engaged in a call with a B party puts B on hold and initiates a new call to a C party.

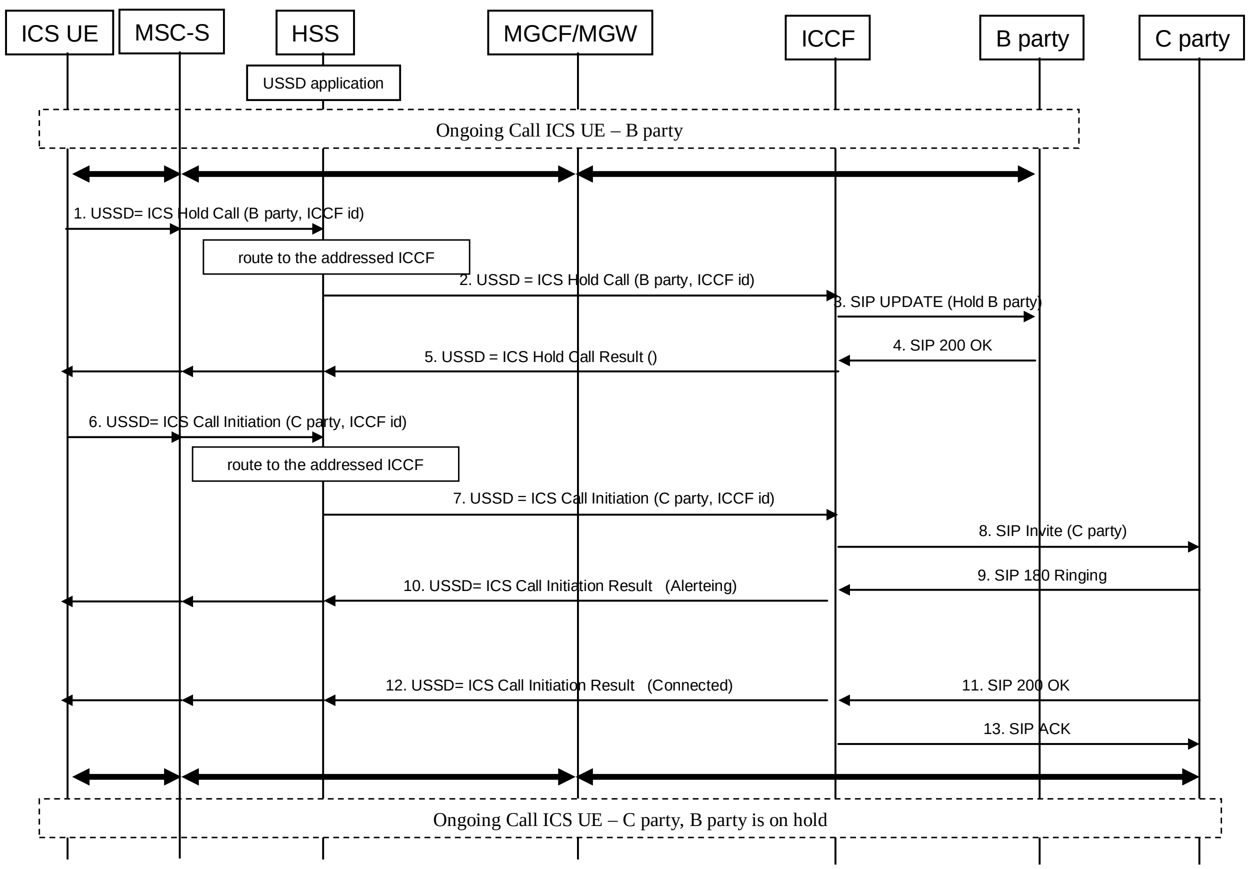


Figure 6.7.2.3.2-1: Mid-call event handling

1. The ICS UE sends an ICS Hold Request on B party inside a USSD dialog to the HSS.
2. The USSD application within the HSS routes the USSD Request to the appropriate ICCF.
3. The ICCF sends a SIP UPDATE to the B party to put the B party on hold.
4. ICCF receives a SIP 200 OK.

5. ICCF forwards an ICS Returned Result within a USSD Response to the ICS UE.

6. The ICS UE initiates a new call to the C party by sending an ICS Call Initiation Request within the USSD dialog to the HSS.

7. The HSS routes the USSD Request to the appropriate ICCF.

8. The ICCF sends a SIP INVITE to the C party.

9. ICCF receives SIP 180 ringing.

10. The ICCF sends an ICS Call Initiation Response to the ICS UE that includes the Alert status.

Editor's note 1: How the ICCF will then redirect the media to the appropriate bearer in the MGW, or if other means could be used, is FFS.

11. ICCF receives SIP 200 OK.

12. ICCF send an ICS Call Initiation Response to the C party that includes the connected status.

13. ICCF send an SIP ACK to the C party and the media is now established e2e between the ICS UE and C party.

Editor's note 2: When the ICCF puts B party on hold, normally RTCP should be continually by sent between B party, in order to keep the link alive. How to handle RTCP reports from the held B party is FFS.

6.7.2.4 Domain Transfer

Example information flows for domain transfers are shown below. It's assumed that the ICS UE stores the service states.

Editor's note 1: Support for multiple call continuity with mid call services is FFS.

Editor's note 2: The STID concept should be incorporated (S2-072037).

6.7.2.4.1 Single Call Continuity from CS access to PS access

The following figure provides an example flow for single call continuity between an ICS user and an IMS user from CS access to PS access.

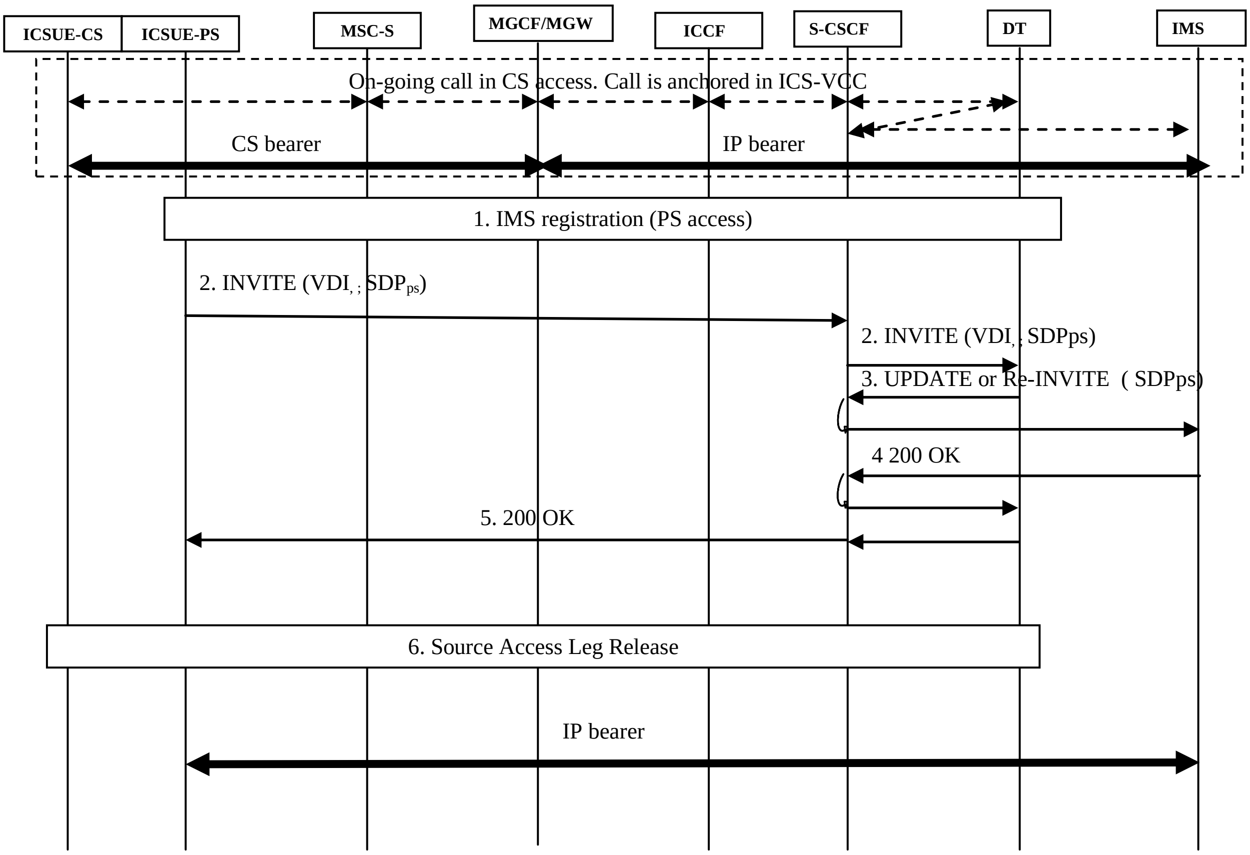


Figure 6.7.2.4.1-1: Single call continuity from CS access to PS access

1. If not already done so, the ICS UE registers in IMS from PS access to receive services.
2. The ICS UE initiates a SIP INVITE towards DTF using VDI to establish a target access leg in PS access and request call transfer. SDP includes UE address in PS access. The IMS core routes the SIP INVITE to the DTF where the source access leg from CS access is currently anchored.
3. The DTF sends a UPDATE or Re-INVITE to the remote end to request a change of address in the media connection.
4. The remote end returns a 200 OK to the DTF via IMS core. PSI routing is used to route the call from MGCF to ICCF.
5. The DTF in turn sends the 200 OK to the ICS UE via IMS core.
6. The DTF then clears the source access leg in CS access and the media connection is now established in PS access to the remote end.

6.7.2.4.2 Single call continuity from PS access to CS access

6.7.2.4.2.1 Use of ICCP to route a call on the transferring-in domain

The following figure provides an example flow for a single call continuity between a ICS user and a IMS user from CS access to PS access, where ICCC is used for setup of all sessions.

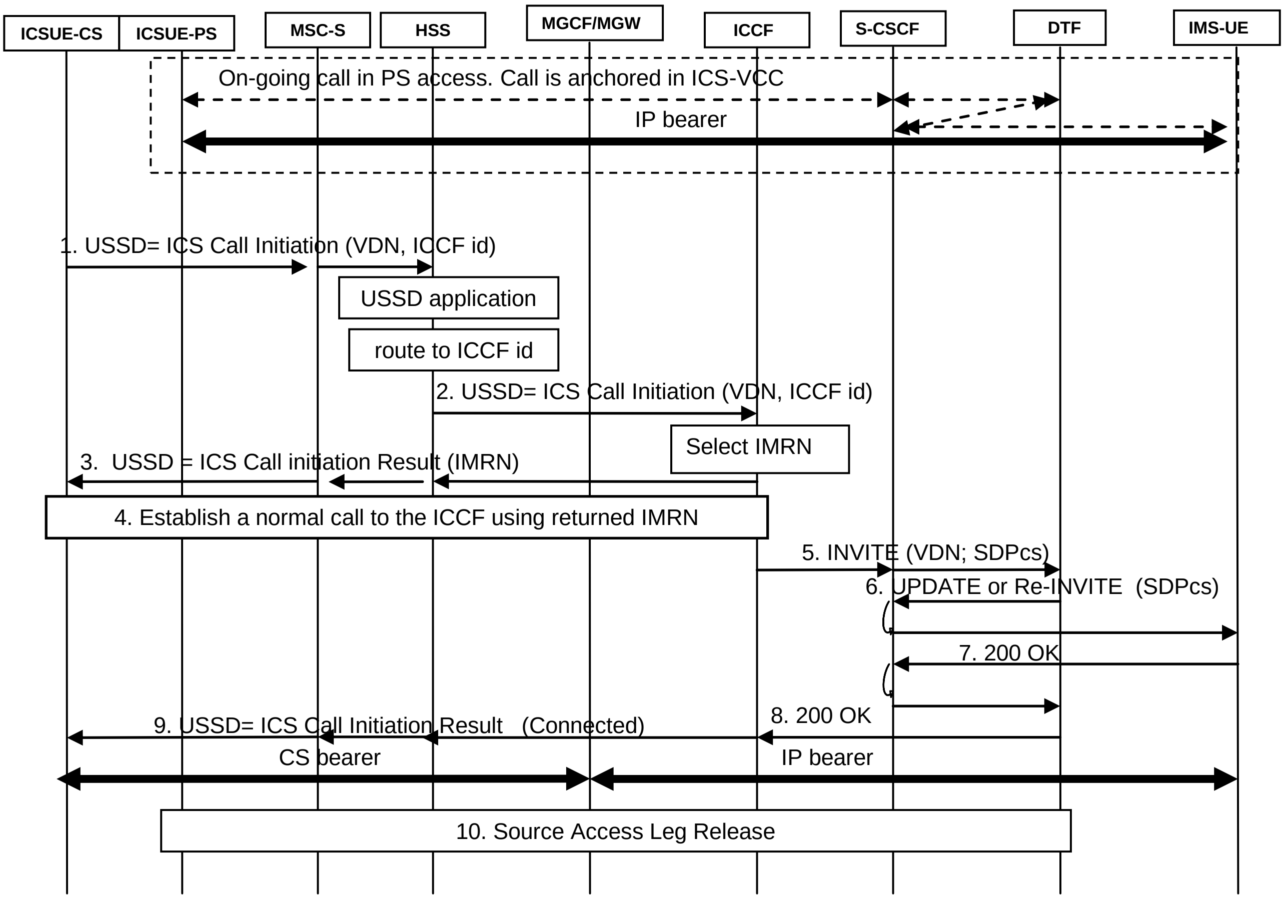


Figure 6.7.2.4.2.1-1: Single call continuity from PS access to CS access, use of ICCP to route a call

1. ICS UE initiates an ICS Call Initiation Request within a USSD dialog towards DTF using VDN to establish a target access leg in CS access and request call transfer.

2. The USSD application within the HSS forwards the USSD Request to ICCF.

3. The ICCF selects an IMRN if a media connection is not already established to the ICS UE. The ICCF returns the IMRN in an ICS response within the USSD dialog. The USSD response is returned all the way to the ICS UE.

4. The ICS UE uses the IMRN to establish a normal CS call set up to the ICCF via MGCF. PSI routing is used to route the call from MGCF to ICCF.

5. Following that, ICCF initiates a call to the VDN by sending a SIP INVITE to the IMS core which routes the INVITE towards the DTF where the source access leg from PS access is currently anchored. SDP includes MGW address in CS access.

6. The DTF sends a UPDATE or re-INVITE towards remote end via the IMS core to request a change of address in the media connection.

7. The remote end returns a 200 OK to the DTF.

8. The DTF in turn sends a 200 OK to the ICCF.

9. ICCF sends an ICS Call Initiation Result including the connected status back to the ICS UE within the USSD dialog.

10. The media connection is now established in CS access to the remote end and the DTF clears the source access leg in PS access.

6.7.2.4.2.2 Use of CAMEL to route a call on the transferring-in domain

The following figure provides an example flow for single call continuity between a ICS user and a IMS user from CS access to PS access, where CAMEL is used for routing a call to IMS.

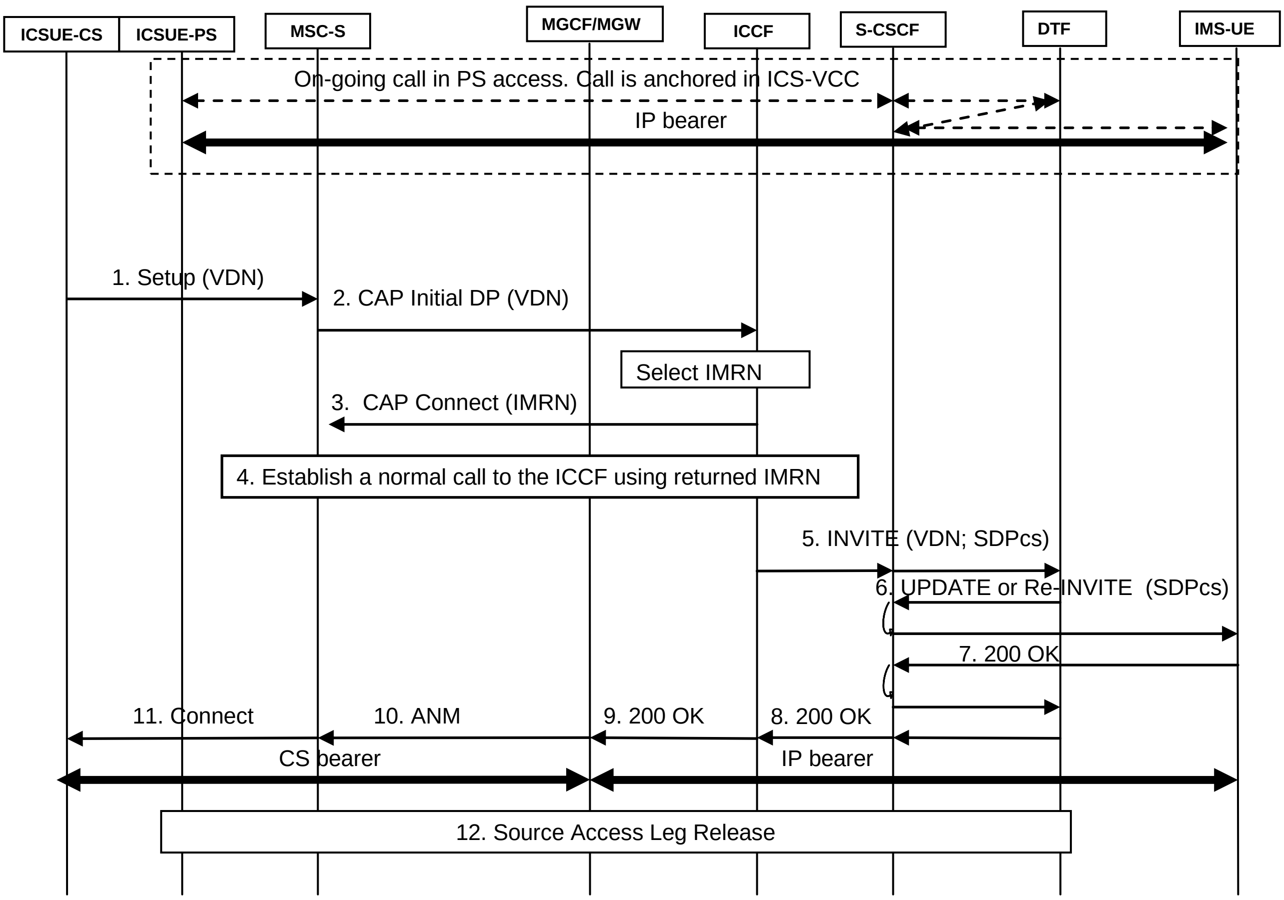


Figure 6.7.2.4.2.2: Single call continuity from PS access to CS access, use of CAMEL to route a call

1. The ICS UE initiates a normal CS call towards DTF using VDN to establish a target access leg in CS access and request call transfer.
2. The MSC-S sends a CAMEL originating trigger request to the ICCF to fetch an IMRN for ICCF.
3. The ICCF selects and returns an IMRN to the MSC-S.
4. The MSC-S routes a call to the IMRN to establish a normal CS call set up to the ICCF via MGCF.
5. Following that, ICCF initiates the call to the VDN by sending a SIP INVITE to the IMS core which routes the INVITE towards the DTF where the source access leg from PS access is currently anchored. SDP includes MGW address in CS access.
6. The DTF sends a UPDATE or re-INVITE towards remote end via the IMS core to request a change of address in the media connection.
7. The remote end returns a 200 OK to the DTF via IMS core.
8. The DTF in turn sends the 200 OK to the ICCF via IMS core.
9. The ICCF sends the 200 OK to the MGCF.
10. The MGCF sends the ANM to the MSC-S.
11. The MSC-S sends the Connect to the ICS UE.

12. The media connection is now established in CS access to the remote end and the DTF clears the source access leg in PS access.

6.7.2.4.3 Assignment of Session Transfer Identifier

See clause 6.6.2.

6.7.2.5 Managing Network Services related Information

See clause 6.6.3.1.5, Managing Network Services related Information.

6.8 Architectural alternative: L-CAAF-n approach

6.8.1 Signalling and bearer architecture

6.8.1.3 Calls established using CS bearers with non ICS UE

The signalling/bearer paths for an IMS session established via CS access for the L-CAAF-n approach is described in figure 6.8.1.3-1.

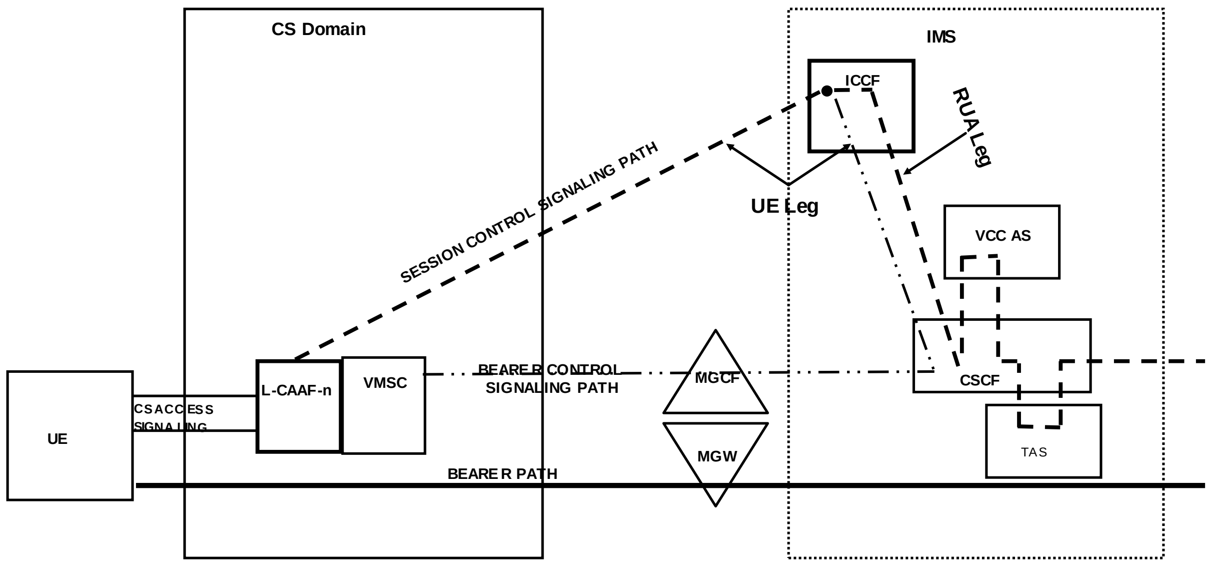


Figure 6.8.1.3-1: Signalling/Bearer Paths-non ICS UE- L-CAAF-n approach

CS Access Signalling is intercepted by the L-CAAF-n function, which establishes a session control signalling path to the ICCF. The L-CAAF-n performs necessary interworking with TS 24.008 [7]. The ICCF presents SIP UA behaviour on behalf of the UE toward IMS.

The UE establishes a standard CS call towards the CS core network; The L-CAAF-n intercepts the CS Access Signalling and establishes the Bearer Control Signalling path with the ICCF by establishing a CS call toward the ICCF. In parallel, it establishes the Session Control Signalling Path with the ICCF. The Bearer Control Signalling and Session Control Signalling stimuli are combined at the ICCF for presentation of SIP UA behaviour for establishment of an IMS session.

Use of the Session Control Signalling Path for session setup for this model enables the capability to provide all services exclusively by IMS.

Editor's note: A solution for call independent aspects such as user configuration and interworking/alignment of supplementary services data across CS domain and IMS is to be provided.

6.8.2 Information flows

6.8a Architectural alternative: Enhanced MSC Server approach

6.8a.1 Signalling and bearer architecture

6.8a.1.1 Calls established using CS bearers with non ICS UE

The Signalling/bearer paths for an IMS session established via CS access by non ICS UE with the enhanced MSC Server presenting a single SIP session is described in figure 6.8a.1.1-1.

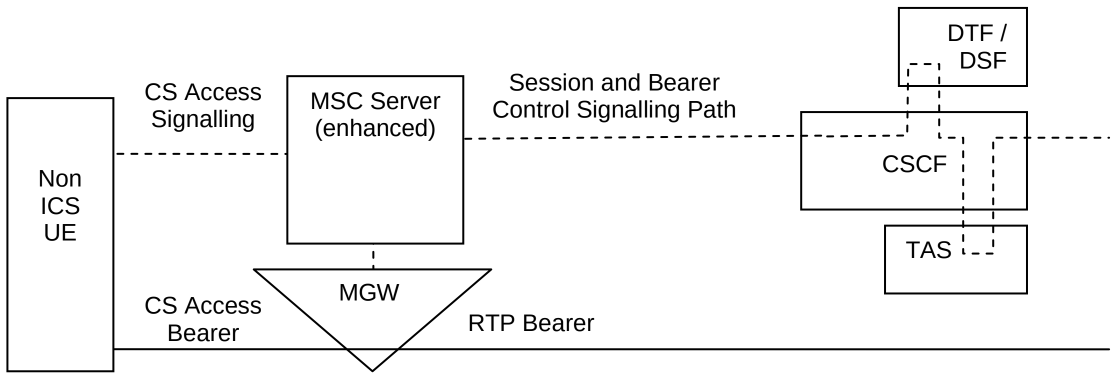


Figure 6.8a.1.1-1: Signalling/Bearer Paths-non ICS UE- Enhanced MSC Server approach

CS Access Signalling is terminated by the enhanced MSC Server, which establishes a single SIP session towards the IMS. The MSC Server performs necessary interworking with TS 24.008 [7]. The MSC Server presents SIP UA behaviour on behalf of the UE toward IMS.

Use of the direct SIP signalling between the MSC Server and CSCF for this model enables the capability to provide telephony services exclusively in IMS.

6.8a.1a IMS Address Discovery

Prior to initiating IMS registration on behalf of the UE, the MSC Server must discover the address of the appropriate recipient IMS node for the SIP REGISTER, e.g. I-CSCF or IBCF. The MSC Server derives a home domain name using the IMSI of the subscriber and performs a DNS query on this domain name, which returns the IP address(es) of the I-CSCF/IBCF. The structure of the derived domain name is a stage 3 issue. This domain name would effectively serve as a SIP URI for the domain name of the home network.

6.8a.2 Information Flows

6.8a.2.1 Registration

The following information flows depict the user registering in IMS via CS access for various idle-mode mobility scenarios.

Use Case 1: Initial IMS Registration via MSC Server:

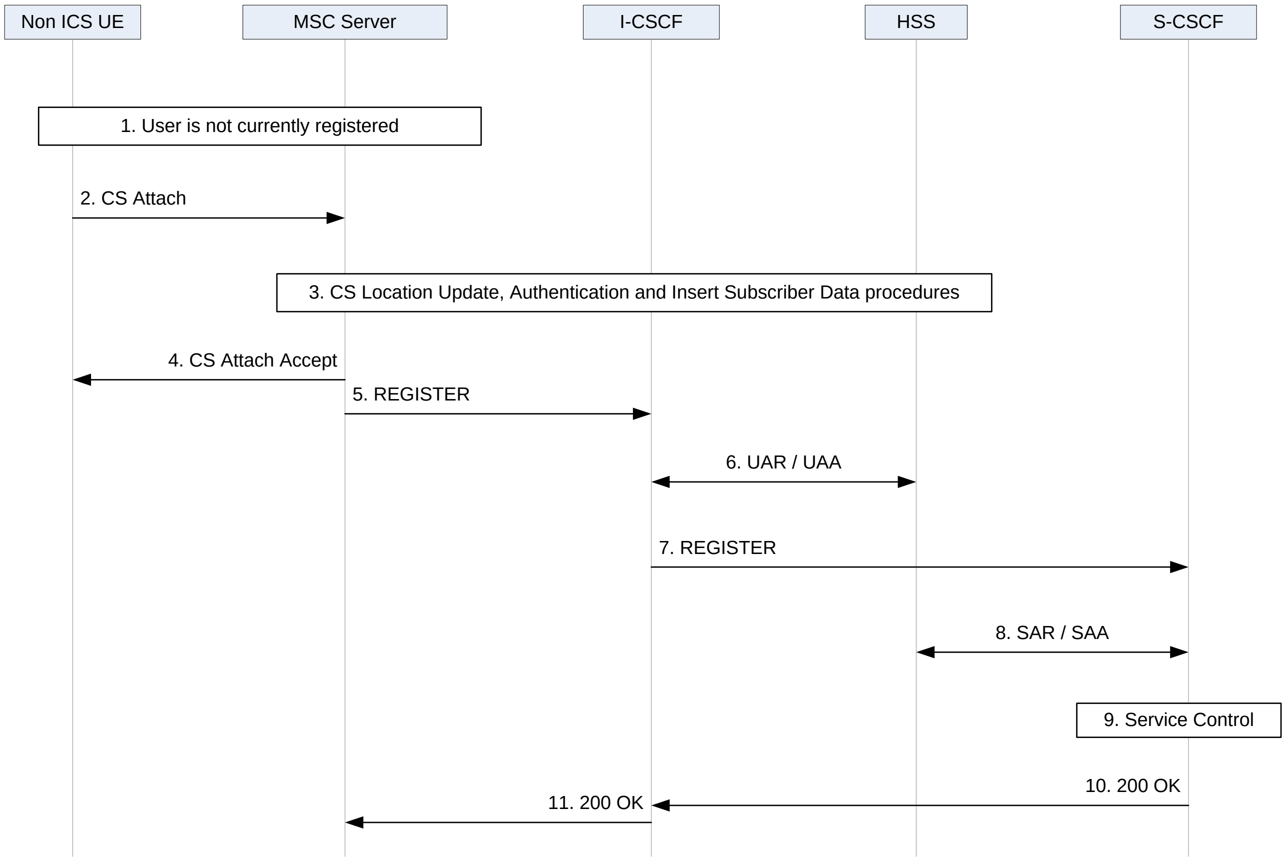


Figure 6.8a.2.1-1 Non ICS UE IMS registration via MSC Server

1. The UE is not registered in CS or IMS.
2. Standard CS Attach procedures are initiated towards the CS network.
3. Standard CS location update, authentication and insert subscriber data procedures.
4. A CS Attach Accept is returned to the UE.
5. Upon successful location update in CS, the MSC Server discovers the address of the appropriate I-CSCF/IBCF and sends a trusted SIP REGISTER to the IMS with the appropriate information (e.g. Temporary Public User Identity derived from IMSI, MSC Server address as the host portion of the "Contact" address, etc.). The REGISTER from the MSC Server does not include an Authorization header field or the header field values as required by RFC 3329. The principles of TR 33.803 shall be followed to allow this type of registration to co-exist with registrations which must be authenticated by the IMS. Network Domain Security (TS 33.210 [20]) can also be used to authenticate the MSC Server sending the REGISTER. This REGISTER also provides the following information in the Contact header:
   - Feature tag to indicate capability audio only. A terminating SIP INVITE which is indicating other capabilities would not match the CS contact.
   - Indication of CS access characteristics (details are stage 3 issue). Possible solutions are:
     - Indicates CS access in P-Access-Network-Info as new access network type (e.g. "GERAN-cs", "UTRAN-cs").
     - New feature tag indicating CS access.

Editor's note: How the MSC Server decides whether or not to initiate IMS registration for this subscriber is for further study.

6. Standard I-CSCF/HSS procedures for S-CSCF location/allocation. Note that the private identity is obtained from the IMSI, which is contained in the T-IMPU.

7. The I-CSCF forwards the REGISTER to the S-CSCF.

8. The S-CSCF identifies the REGISTER as a trusted registration from the MSC Server which is a trusted network node. The S-CSCF skips any further authentication procedures as it is assumed that the user has already been authenticated in the CS domain. The S-CSCF performs the SAR / SAA exchange with the HSS, resulting in the user status as "registered" and the S-CSCF name stored in the HSS. The S-CSCF stores the Contact address for each of the implicitly registered IMPUs.

NOTE: In case the user is already in the "registered" state for the same IMPI/IMPU but with a different contact address, and the previous registration has not expired, the S-CSCF performs the network initiated deregistration procedure for the old contact address.

9. Service control execution at the S-CSCF is performed according to the standard procedures.

10-11. A 200 OK response to REGISTER is returned. Note that IMS registration success/failure does not impact the CS attach status of the UE.

Use Case 2: IMS registration via new MSC Server:

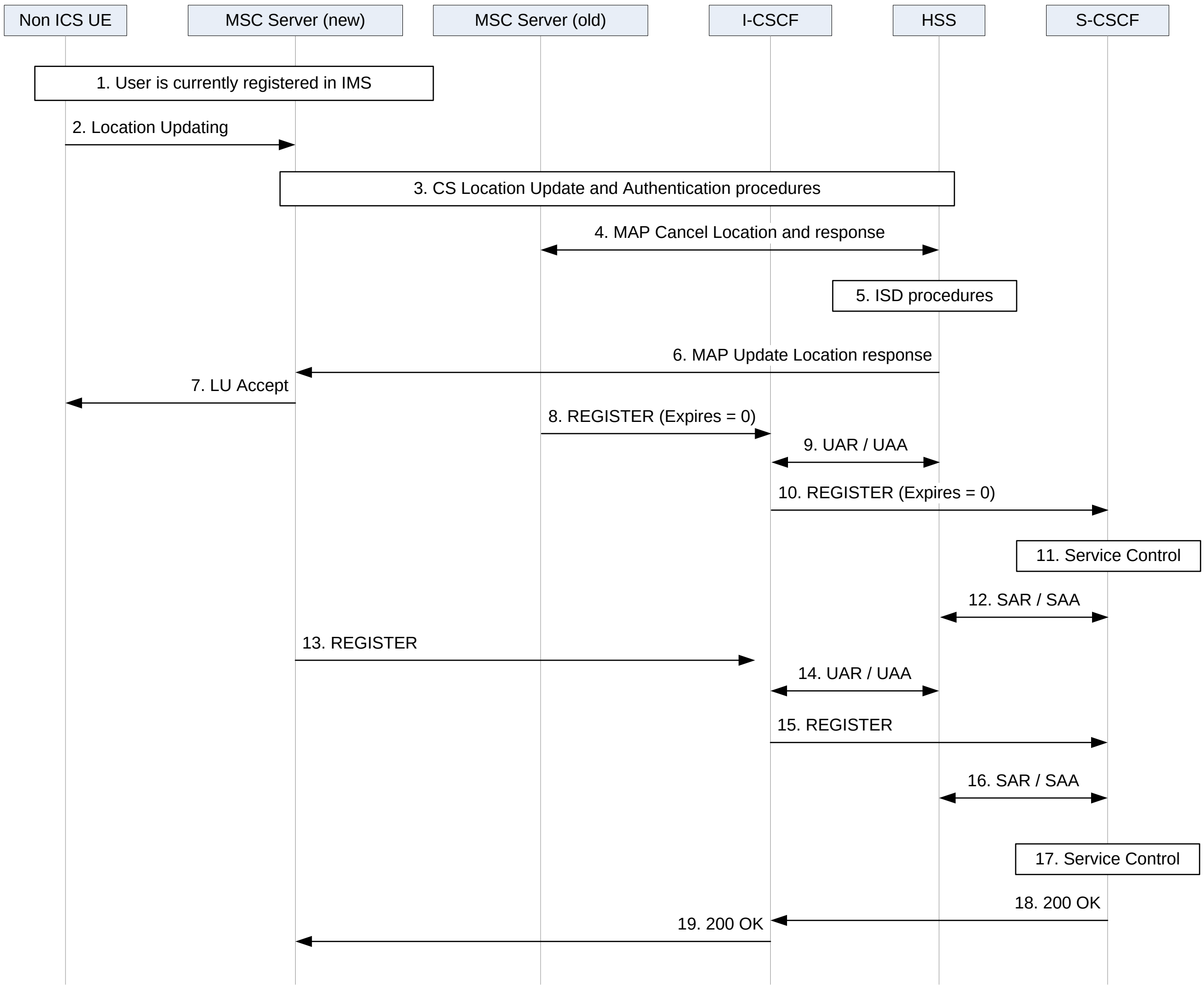


Figure 6.8a.2.1-2 Non ICS UE IMS registration via MSC Server

1. The UE is registered in CS and IMS.

2. Standard Location Updating procedures initiated towards the CS network.

3. Standard CS location update and authentication procedures.

4. The MAP Cancel Location procedures are completed between the HLR component of the HSS and the old VLR.

5. MAP Insert Subscriber Data procedures are completed towards the new VLR.

6. The HLR component of the HSS completes the MAP Update Location procedure with the new VLR.

7. A Location Updating Accept is sent to the UE.

8. The MSC Server initiates deregistration procedures towards IMS (REGISTER with Expires header value of 0). Step 8 can be triggered in parallel with step 5.

9. Standard I-CSCF/HSS procedures for S-CSCF location.

10. The I-CSCF forwards the REGISTER to the S-CSCF.

11. Service control execution at the S-CSCF is performed according to the standard procedures.

12. The S-CSCF identifies the REGISTER as a trusted registration from the MSC Server which is a trusted network node. The S-CSCF skips any further authentication procedures as it is assumed that the user has already been authenticated in the CS domain. The S-CSCF performs the SAR / SAA exchange with the HSS according to standard procedures.

13. The MSC Server discovers the address of the appropriate I-CSCF/IBCF and sends a trusted SIP REGISTER to the IMS as described in the previous use case.

14. Standard I-CSCF/HSS procedures for S-CSCF location/allocation. Note that the private identity is obtained from the IMSI, which is contained in the T-IMPU.

15. The I-CSCF forwards the REGISTER to the S-CSCF.

16. The S-CSCF handles the REGISTER as described in the previous use case.

17. Service control execution at the S-CSCF is performed according to the standard procedures.

18-19. A 200 OK to REGISTER is returned.

Use Case 3: UE registered in IMS via MSC Server roams to non-enhanced VMSC:

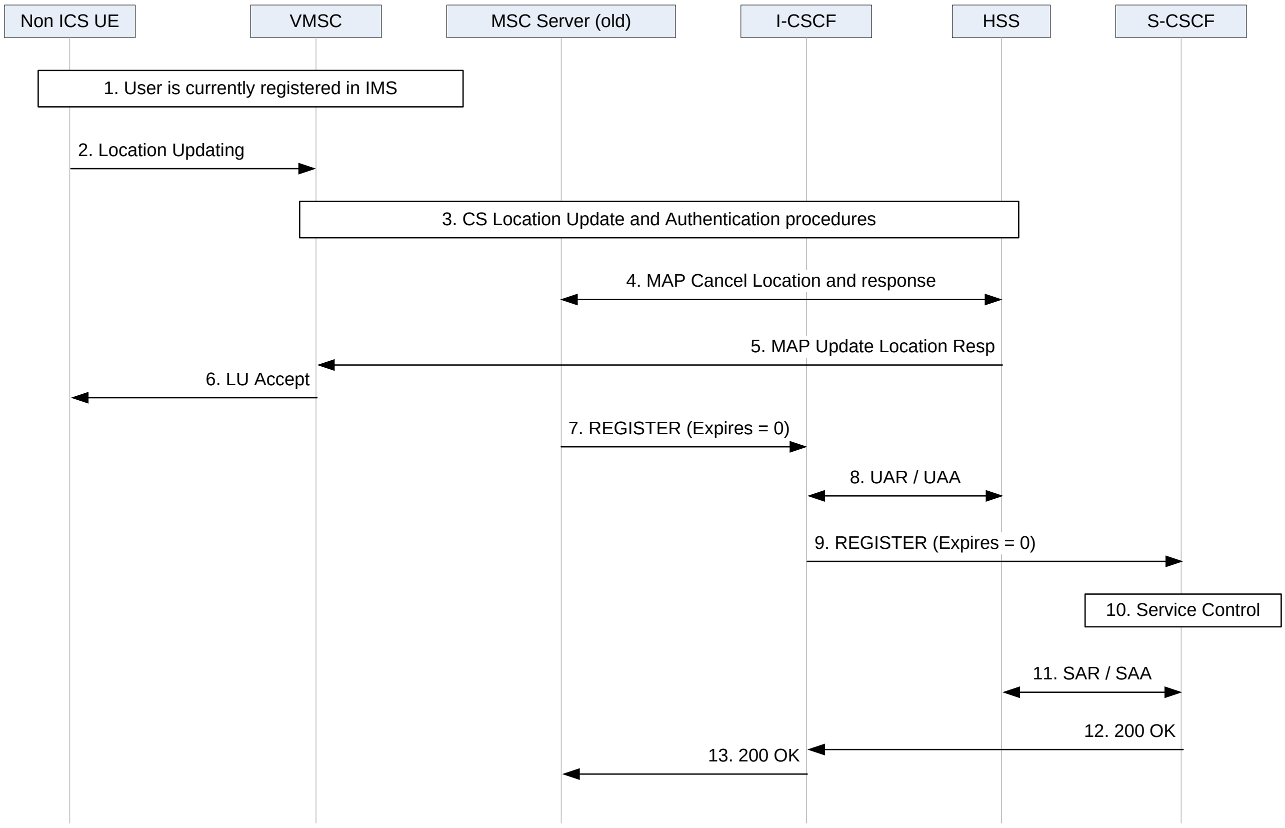


Figure 6.8a.2.1-3: Non ICS UE (registered at MSC Server) registration at Legacy VMSC

1. The UE is registered in CS and IMS.
2. Standard Location Updating procedures initiated towards the CS network.
3. Standard CS location update and authentication procedures.
4. The MAP Cancel Location procedures are completed between the HLR component of the HSS and the old VLR.
5. A MAP Update Location response is returned to the legacy VMSC.
6. Location Updating Accept is sent to the UE.
7. The MSC Server initiates deregistration procedures towards IMS (REGISTER with Expires header value of 0).
8. Standard I-CSCF/HSS procedures for S-CSCF location.
9. The I-CSCF forwards the REGISTER to the S-CSCF.
10. Service control execution at the S-CSCF is performed according to the standard procedures.
11. The S-CSCF identifies the REGISTER as a trusted registration from the MSC Server which is a trusted network node. The S-CSCF skips any further authentication procedures as it is assumed that the user has already been authenticated in the CS domain. The S-CSCF performs the SAR / SAA exchange with the HSS according to standard procedures. In this scenario, subsequent calls attempts to this UE shall be redirected to the CS domain via the MGCF, which is an example of "services related to unregistered state" as described in TS 23.228 [4].

12-13. A 200 OK to REGISTER is returned.

6.8a.2.2 Origination

The following information flow depicts IMS call origination performed via CS access.

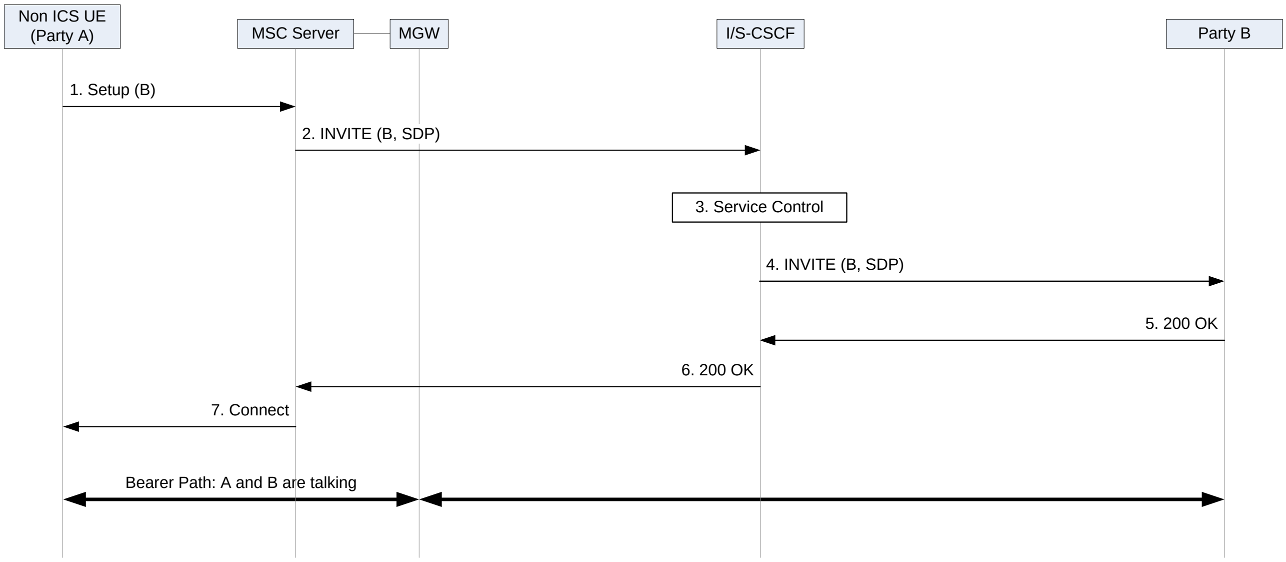


Figure 6.8a.2.2-1 Non ICS UE IMS origination at MSC Server

The non-ICS UE IMS origination procedure is as follows: -

1. The UE originates a voice call in the CS domain to the party-B using standard CS call setup procedures.
2. The MSC Server translates the setup message and initiates a SIP INVITE request addressed to a tel URI or, if directed by operator's local policy, to a SIP URI (using an E.164 address in the user portion and the setting user=phone), includes an initial SDP in the INVITE request, and forwards the request to the S-CSCF along the path determined upon the UE's most recent registration procedure.
3. The S-CSCF invokes service control using registered iFC.
4. The S-CSCF forwards the INVITE to the B-party UE.
5. The B-party UE accepts the session with a 200 OK response back to the S-CSCF.
6. The 200 OK is forwarded to the MSC Server.
7. The MSC Server translates the 200 OK and sends a connect message to the non ICS UE. The voice session can then be established between A-party and B-party.

6.8a.2.3 Termination

The following information flows depict IMS call termination made via CS access.

Use case 1: User is registered in IMS:

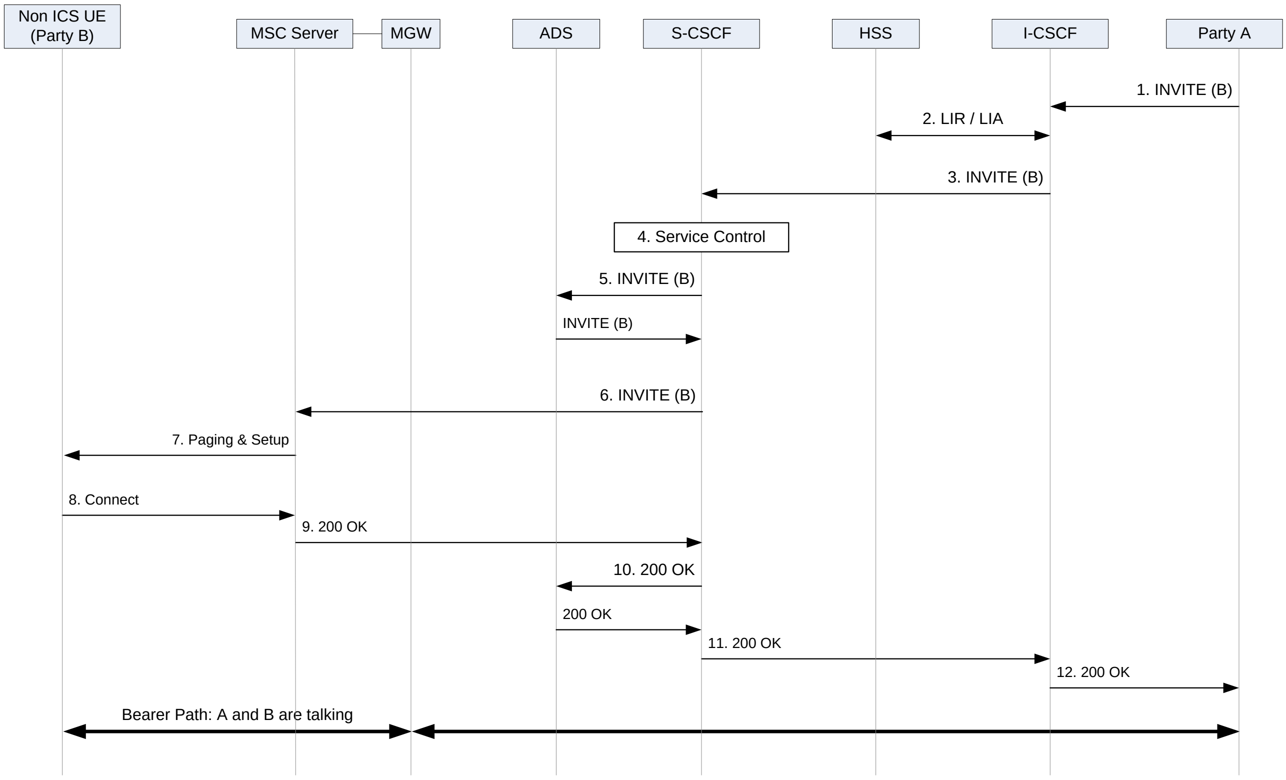


Figure 6.8a.2.3-1: Non ICS UE (registered in IMS) termination

The Non-ICS UE IMS (registered in IMS) termination procedure is as follows:

1. Party A sends an INVITE request for initiating a voice session with the Party B non ICS UE.
2. The I-CSCF in the terminating home network performs Location Query procedure with the HSS to acquire the S-CSCF address of Party B according to standard IMS procedures.
3. The I-CSCF forwards the INVITE request to the S-CSCF.
4. The S CSCF invokes service control using registered iFC according to standard IMS procedures.
5. The final iFC directs the INVITE to the access domain selection (ADS) function. The MSC Server is chosen. Note that the presence of the ADS in the iFC depends on the IMS subscription, it will not be required in all subscription scenarios.
6. The S-CSCF forwards the message to the MSC Server based on the contact address stored during registration, i.e. using standard S-CSCF procedures.
7. The MSC Server initiates the paging and sends a setup message to the Party B UE.
8. The Party B UE accepts the request and sends a connect message to the MSC Server.
9. The MSC Server translates the connect message into a 200 OK and sends it to the S-CSCF.
10. The S-CSCF sends the 200 OK to the ADS.
11. The S-CSCF sends the 200 OK to the I-CSCF.
12. The I-CSCF sends the 200 OK to the Party A UE.

Use case 2: User is not registered in IMS, call is redirected to the CS domain. This is considered a fallback solution to cover scenarios where the VMSC cannot register the UE in IMS.

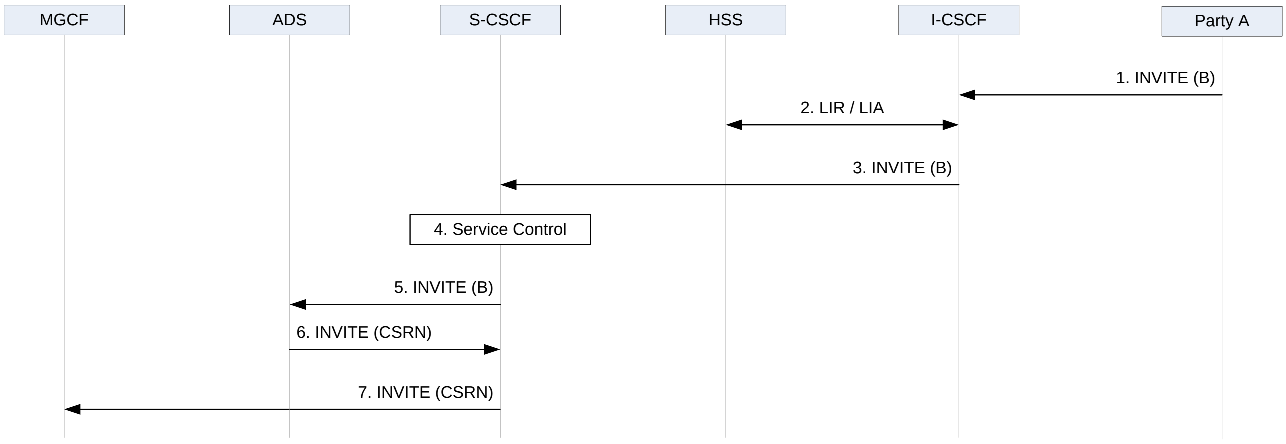


Figure 6.8a.2.3-2: Non ICS UE (not registered in IMS) termination

The Non-ICS UE IMS UE (not registered in IMS) termination is as follows:

1. The Party A UE sends an INVITE request for initiating a voice session with the Party B non ICS UE.
2. The I-CSCF in the terminating home network performs Location Query procedure with the HSS to acquire the S-CSCF address of Party B, according to standard IMS procedures.
3. The I-CSCF forwards the INVITE request to the S-CSCF.
4. The S-CSCF invokes service control using unregistered iFC.
5. The final iFC directs the INVITE to the access domain selection function. The user is not registered in IMS but attached to the CS network, so breakout to CS domain is chosen. A CSRN is fetched for routing to the CS domain.
6. The INVITE is returned to the S-CSCF.
7. The S-CSCF forwards the INVITE request to the BGCF. Termination of the call to Party B then follows standard CS procedures.

6.8a.2.4 Session Transfer

The following information flows illustrate session transfer scenarios involving CS access in networks enhanced with enhanced MSC Server capabilities.

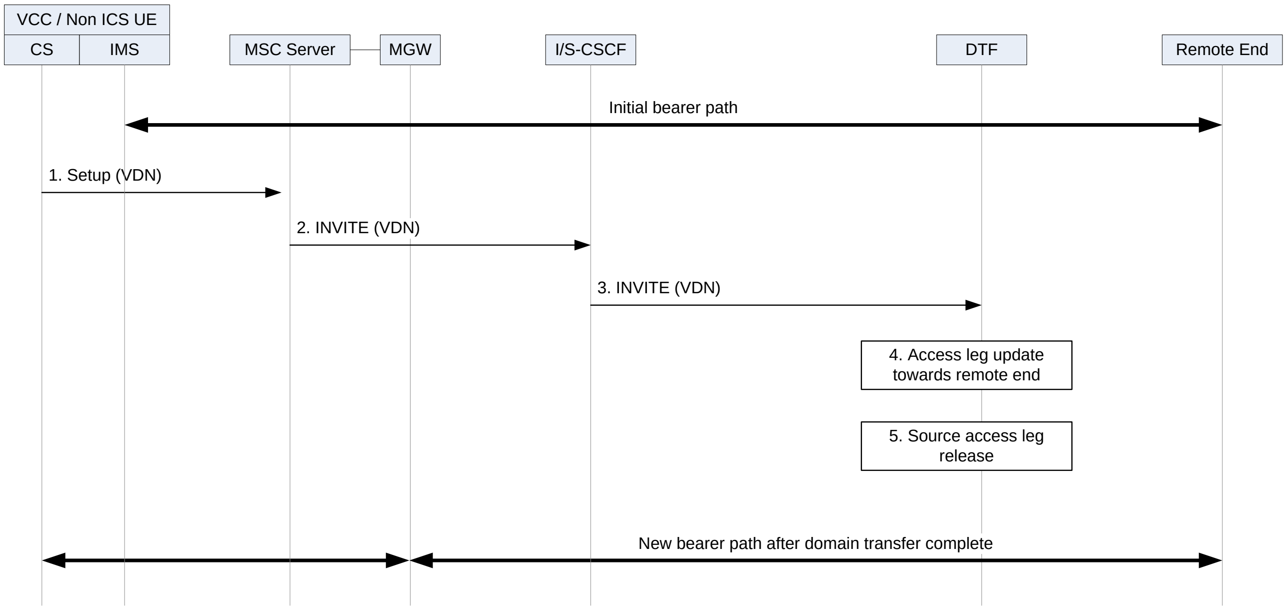


Figure 6.8a.2.4-1 Session Transfer IMS to CS

1. The VCC UE sends the Setup message towards the MSC with the VDN as the called party number.
2. The MSC Server initiates an INVITE towards the UE's home IMS, using routing information obtained during IMS registration. The VDN is carried in the Request-URI.
3. Originating iFC results in the INVITE being routed to the DTF. The DTF uses the VDN to recognize the INVITE request is for a session transfer and correlates this session with the existing session for VCC user.
4. The DTF function initiates Access Leg Update procedures towards the remote endpoint.
5. The DTF function initiates Source Access Leg Release procedures.

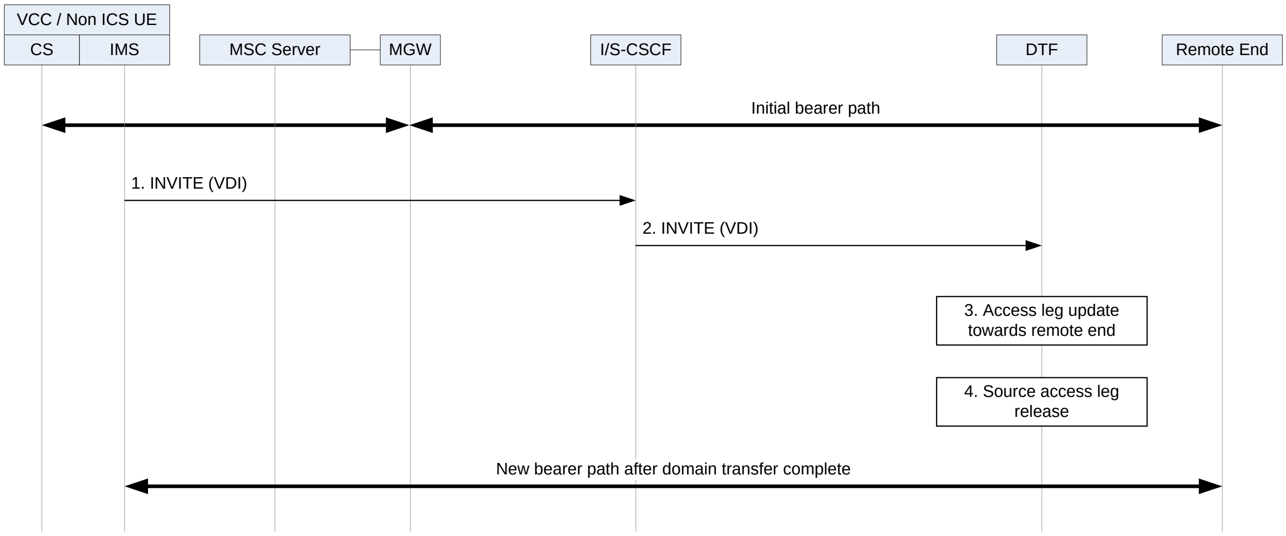


Figure 6.8a.2.4-2 Session Transfer CS to IMS

1. The VCC UE registers with IMS, if not already registered. It subsequently initiates an IMS originated session using a VDI.
2. Once the INVITE is returned to the S-CSCF, originating iFC results in routing of the IMS origination session to the DTF. The DTF detects the VDI and associates this INVITE request with an existing dialog.
3. The DTF function initiates Access Leg Update procedures towards the remote endpoint using standard VCC procedures.

4. The DTF function initiates Source Access Leg Release procedures using standard VCC procedures.

6.8a.3 Originated Service Domain Selection for enhanced MSC server

One of the tasks of the enhanced MSC Server is to decide whether a particular UE, which performs CS attach to the network, will receive services from CS domain or from IMS. Hence the enhanced MSC server has to perform originated Service Domain Selection (OSDS).

In order to perform OSDS, the enhanced MSC server needs to be able to differentiate between:

- Users which require enhanced MSC server functionality to receive services from IMS.
- Users which receive services from CS and which don't require enhanced MSC server functionality.

For that purpose, the enhanced MSC server needs to know whether an attaching user shall receive services from IMS and thus receive additional information about the subscriber. Since this is subscriber related information, it is stored in the HSS.

There are different candidate reference points which can be used by the enhanced MSC server to perform the differentiation: -

- MAP
- Sh
- Cx
- I6 using SIP register

These options for O-SDS are discussed in more detail below.

Option MAP:

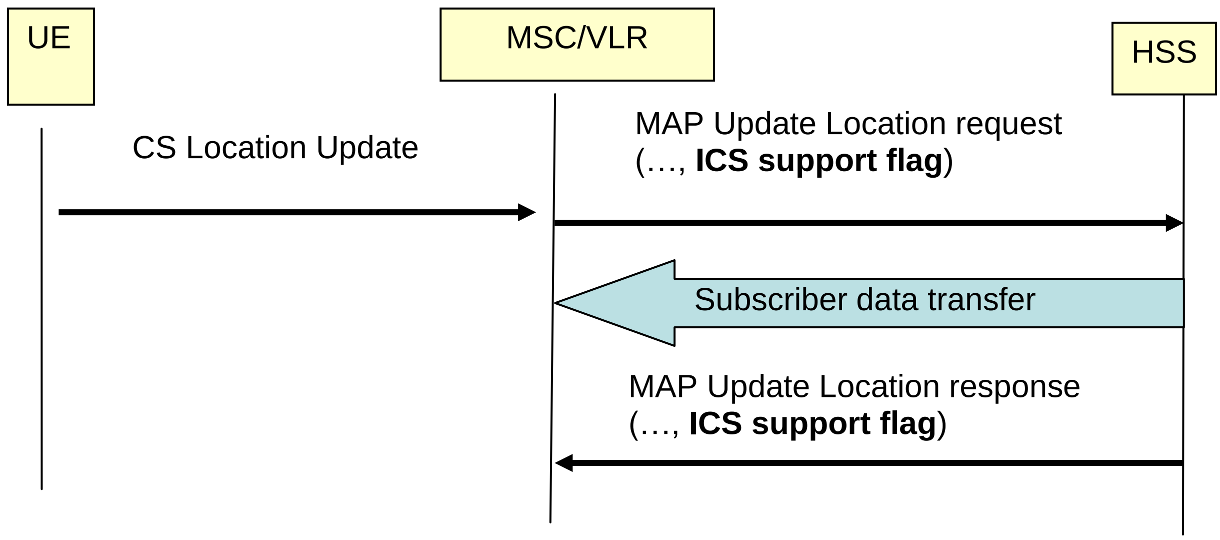


Figure 6.8a.3-1: ICS support flag in MAP

One possibility is to introduce a new flag into MAP Update Location to enable ICS capability negotiation between VPLMN and HPLMN.

MSC/VLR sets this flag in MAP Update Location if MSC/VLR has the enhanced MSC server capability as described in TR 23.892 and wishes to put corresponding subscriber under ICS control.

NOTE 1: If the flag is not present then this would be a clear indication to HSS that MSC/VLR does not support ICS.

HSS sets this flag in the Update Location response message If HSS recognizes new flag from MSC/VLR (i.e. ICS capable HSS) and wishes to put corresponding subscriber under ICS control. Note: if the flag is not present then this would be a clear indication to MSC that HSS does not support ICS and that this user will not require enhanced MSC server functionality. Based on the MAP functionality, Pre-Rel-8 HSS can ignore new introduced information element as unknown so that there is no harm to pre-rel8 HSS even if new ICS flag is received.

One other possibility is to introduce an indicator in the subscriber data which would indicate to the enhanced MSC server that the subscriber requires enhanced MSC server functionality. If the indicator is not present, then this would be a clear indication to enhanced MSC server that the subscriber does not require enhanced MSC server functionality. If the enhanced MSC server functionality is not present in the MSC, then this indicator has to be ignored (i.e. it should not cause unwanted behaviour).

Option Sh:

Prerequisites:

- An ICS specific Temporary Public User Identity is provisioned based on the IMSI (similar as to what is already defined in TS 23.003, but different in the user and/or domain part so as not to clash with UE registrations).

NOTE 2: The Sh does not use the IMPI in the protocol today.

- MSC-Server and HSS are in the same trust domain i.e. Sh is allowed. Note that today use of Sh is not specifically defined between different operators and so it *may* be required that restrictions need to be added in order to define what data is allowed to be sent to the visited operator. Of course, if two operators are part of the same trust domain, such functionality could of course be argued as unnecessary. Liaison with SA3 may be required here for guidance.

- MSC-Server must be able to find the address of the HSS to be contacted.

- A flag indicating that the user requires ICS enhanced MSC Server functionality is provisioned in the subscriber data.

The subscriber profile retrieval has to be performed for all users which are attaching to the CS network via this MSC. In order for the MSC-Server to be able to contact the correct HSS, the MSC-Server will need to know, at the very least, the address/FQDN of an SLF. As specified in TS 23.002, the MSC-Server could query the SLF to get the name of the HSS containing the required subscriber specific data.

Option Cx:

Same as for Option Sh, except that the enhanced MSC-Server uses the Cx interface.

Option SIP Register:

After performing successful location update procedures in the CS domain, the enhanced MSC-Server determines this subscriber to be a potential IMS subscriber and tries to register the user in IMS. IMS registration would only succeed for IMS subscribers which have the IMPI in the IRS in the HSS used during registration and fail otherwise. If the registration in IMS succeeds, the enhanced MSC-Server continues to handle this subscriber as IMS subscriber.

The routing of the registration message is handled by standard IMS routing.

Note that if the enhanced MSC-Server uses a different IMPI/IMPU during registration than the UE when doing IMS registration, the UE can be registered in IMS in addition to the enhanced MSC-Server and thus negating the need for support of IMS simultaneous registrations for the same IMPI/IMPU combination.

Whether or not there is a need to differentiate between ICS users which shall only be handled by enhanced MSC-Server and ICS users which shall only use ICS UE, is FFS. However, currently there are no requirements identified.

Conclusions on Different Options:

Overall, it can be concluded that the OSDS in the enhanced MSC server using a MAP approach would have the benefit of re-using existing agreements between two operators for the VLR to HLR/HSS interaction. On the other hand, the drawback could be seen as impacting current MAP protocol and legacy equipment such as the HLR. However, as clearly indicated, there would not be any backwards compatibility issues with pre-Rel-8 equipment.

The Sh option and Cx option provide the benefit of allowing the enhanced MSC server to act as an AS or CSCF (respectively), re-using the existing Sh/Cx interface protocols (the Cx might even require enhancements). One drawback is that yet another interconnect interface must be used (in addition to existing MAP and the required SIP interface) when the subscriber is roaming. There may also be potential issues with allowing Sh and Cx between networks that are not fully trusted. Where both networks are in the same trust domain, there is no issue.

The SIP register approach does not require a new interconnect interface, but instead tries to register appropriate subscribers in IMS and if this registration succeeds, the user is identified to receive his services from the IMS. Note that a pre-screening based on operator-policy in the enhanced MSC server can be performed to limit the number of users for which registration in IMS is tried.

Further on, SIP register can be used by enhanced MSC server to determine whether the attaching user is an IMS subscriber or not.

It has been concluded that the SIP register approach would be the appropriate way forward for the enhanced MSC server.

After performing successful location update procedures in the CS domain, the MSC determines this subscriber to be a potential IMS subscriber and tries to register the user in IMS. IMS registration would only succeed for IMS subscribers and fail for CS only users. If the registration in IMS succeeds, the MSC continues to handle this subscriber as IMS subscriber.

Editor's note 1: It is FFS how the MSC determines this subscriber to be a potential IMS subscriber.

The routing of the registration message is handled by standard IMS routing. Note that a pre-screening based on operator-policy in the enhanced MSC server can be performed to limit the number of users for which registration in IMS is tried.

Editor's note 2: Whether or not there is a need to differentiate between ICS users which shall only be handled by enhanced MSC server and ICS users which shall only use ICS UE is FFS.

6.8a.4 Co-existences of ICS UE and MSC Server approach

6.8a.4.1 Use of IMSI-derived IMPU/IMPI by both ICS UE and enhanced MSC Server

Assumptions

- ICS UE has IMSI.
- IMSI-derived T-IMPU-UE/T-IMPI-UE is used by ICS UE to register in IMS (no ISIM).
- IMSI-derived T-IMPU-N/T-IMPI-N is used by network function to register in IMS.

NOTE: T-IMPU-UE/T-IMPI-UE is different from T-IMPU-N/T-IMPI-N.

Therefore this user has in his subscriber profile in the HSS the following private IDs:

- T-IMPI-UE (derived from IMSI).
- T-IMPI-N (derived from IMSI).

The user has (at least) the following public IDs:

- Tel-URI
- SIP-URI
- T-IMPU-UE (based on IMSI) – barred
- T-IMPU-N (based on IMSI) – barred

These are grouped into Implicit Registration Set (IRS) as follows and as shown in the following figure:

- Implicit Registration Set:
  - T-IMPU-UE (based on IMSI).
  - T-IMPU-N (based on IMSI).

- Tel-URI (based on MSISDN).
- SIP-URI.

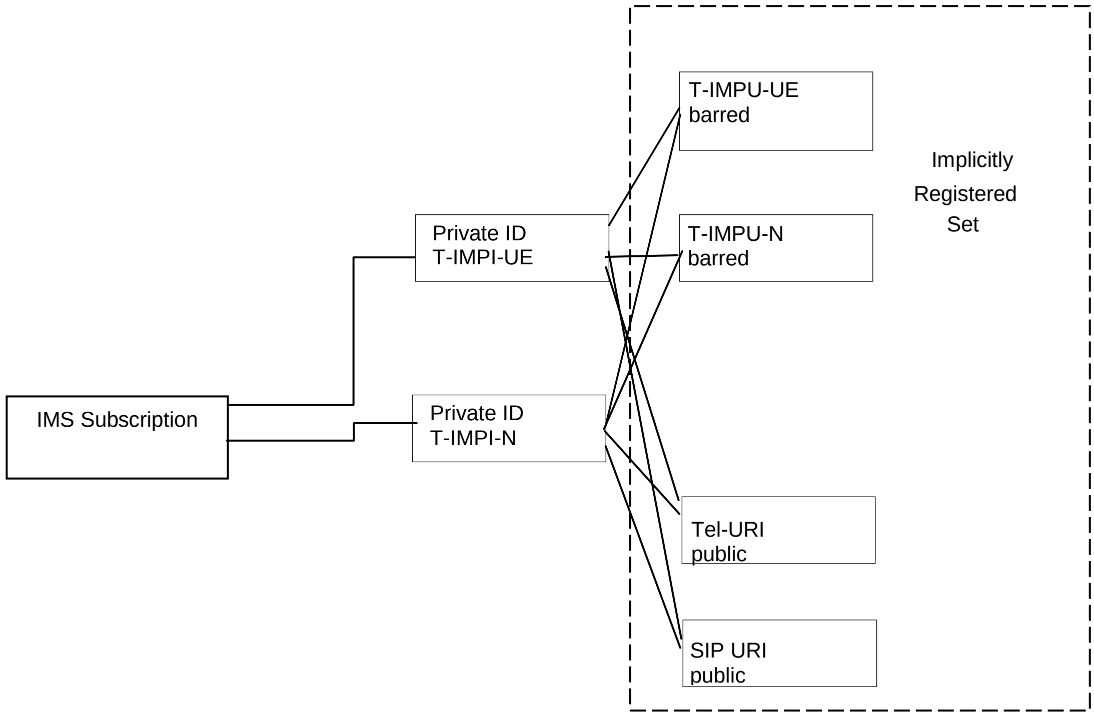


Figure 6.8a.4.1-1: Implicit Registration Set - IMSI-derived IMPU/IMPI by both ICS UE and enhanced MSC Server

Using this Implicit Registration Set configuration, the user can have multiple, simultaneous registrations in CS and in PS as there will be different IMPI/IMPU combinations from the enhanced MSC server and from the ICS UE.

6.8a.4.2 ISIM registered UE and enhanced MSC Server

Today, there are recommendations that suggest that if a UE includes both USIM and ISIM, and registers in IMS, the ISIM shall be used.

Assumptions:

- User has a UICC including both a USIM and an ISIM.
- ICS UE uses credentials stored in ISIM for registration in IMS.
- IMSI-derived T-IMPU-N/T-IMPI-N is used by network function to register in IMS.

Therefore this user has in his subscriber profile in the HSS the following private IDs:

- IMPI (as stored on the ISIM)
- T-IMPI-N (derived from IMSI)

The user has (at least) the following public IDs: -

- Tel-URI
- SIP-URI
- T-IMPU-N (based on IMSI) – barred

These are grouped into Implicit Registration Set as follows and as shown in the following figure:

- Implicit Registration Set:
  - T-IMPU-N (based on IMSI).
  - Tel-URI (based on MSISDN).
  - SIP-URI.

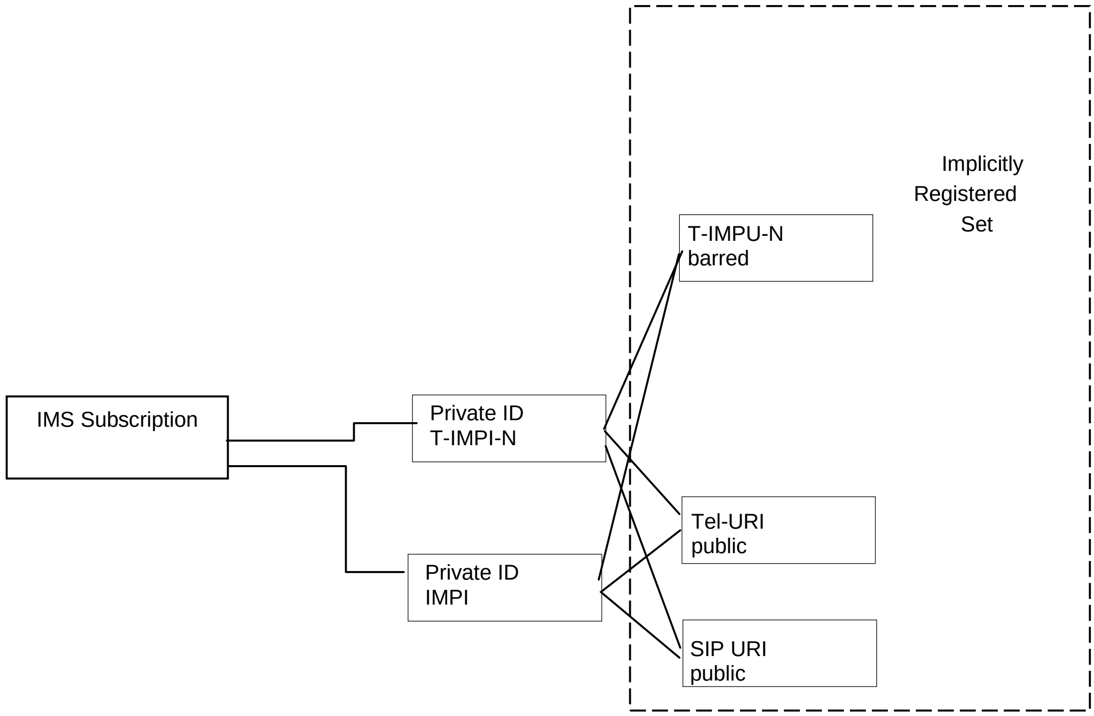


Figure 6.8a.4.2-1: Implicit Registration Set - ISIM registered UE and enhanced MSC Server

Using this Implicit Registration Set configuration, the user can have multiple contacts with different IMPI/IMPU combinations from the enhanced MSC server, and from a UE having an ISIM.

When the user is using a non-ICS UE, the enhanced MSC will be able to register the user based on the network derived IMPU/IMPI (based on the IMSI).

If an ISIM registered UE is used and the user becomes registered through both PS and CS, the IMS registration through PS can use the identities on ISIM (i.e., identities not derived from the IMSI), while the CS registration is based on the IMSI instead. The correlation between the CS and PS will be possible then based on the shared public user identities (and in particular that the IMSI derived identity belongs to the same Implicit Registration Set as the ISIM IMPU).

6.8a.4.3 Co-existence of ICCF and enhanced MSC server

The definition of non-ICS UE reveals that two classes of UEs are covered, namely:

- non-ICS UE with VCC capability (i.e., could be seen as a Rel-7 UE); and
- non-ICS UE without VCC capability (which could be any UE pre Rel-7 or even Rel-7 UEs without VCC capability).

UEs without VCC capability are supported by the enhanced MSC server approach without any doubt, but of course do not support domain transfer. UEs with VCC Rel-7 capability are supported as well, however, since the enhanced MSC server is Rel-8 functionality, also Rel-8 VCC or even MMSC implementation will be necessary, as it was always stated that changes to TS 23.206 are necessary. Also it has been clarified that all limitations of VCC Rel-7 especially with regard to mid-call handling apply as well.

When now looking at ICS UEs, which have the possibility to use ICCC for additional signalling, two possibilities can be differentiated for the signalling between UE and CN:

A Both TS 24.008 [7] and ICCC are used:

A-1 enhanced MSC server + ICCF.

A-2 I1-cs termination in enhanced MSC server.

B Only TS 24.008 [7] is used.

It has to be noted that without ICCC the additional session transfer capabilities especially regarding mid-call and support of subsequent calls as under study in ICS will not be enabled.

Since it has been clarified in TR 23.893 [19] that MMSC users which use CS access for media transmission are ICS users at the same time, no special consideration for MMSC users must be given from an enhanced MSC server perspective.

Obviously, in case A.1) the ICCF has to be used in order to support both I1-cs and I1-ps. However, in case of an originated session, the ICCF would receive from the enhanced MSC server an originated SIP INVITE, whereas it would receive from the MGCF a terminated SIP INVITE. One advantage of A.1 is that ICCF and enhanced MSC Server can evolve independently.

Since the USSD signalling for I1-cs is passing the enhanced MSC server anyway, it is one option to terminate I1-cs in the enhanced MSC server; this case is denoted as A.2 in the following. One advantage with A.2 is that USSD for I1-cs does not need to be transported over the NNI. Support for I1-ps would still require ICCF to be included in the signalling path. Note that A.2 could be seen as a deployment option for I1-cs support.

In case B), when the UE is not using ICCC, then either we fall back to VCC Rel 7 limitations or additional functionality in the DTF is required. The DTF would have to identify the active call and the call on hold, and in any case not more than 2 calls are allowed in parallel for the same IMPU, making it impossible to support concepts in which multiple UE share the same IMPU.

The following table summarize the above-provided analysis.

UE Capability:	**Signalling between CN and UE; 24.008 always via enhanced MSC server**	**MMSC Requirements:**	**Limitations:**
Pre-Rel-7 UE	24.008 only	N/A	No Domain Transfer possible
VCC (Rel-7) UE	24.008 only	"Normal" Rel-8	VCC possible with MMSC with its identified limitations
ICS & MMSC (Rel-8) UE	A) 24.008 + I1-cs A.1) enhanced MSC server + I1-cs termination in ICCF A.2) I1-cs termination in enhanced MSC server	A) "Normal" Rel-8	A) None
	B) 24.008 only	B) DTF needs to identify the active call.	B) only 2 calls for the same IMPU (1 active, 1 hold)

Case B) is limited compared to Case A, but Case B can support VCC Rel-7 behaviour. In order to support more than Rel-7 behaviour, case A) needs to be considered. Case A) is also covering the case in which an operator wants to deploy both support for ICS UE and non ICS UE in one PLMN. This combination would also enable to easily fallback from ICS UE capability to enhanced MSC server support in case of SIM swapping between ICS UE and a non-ICS UE.

6.8a.5 Subscription to the registration event package

After successful initial IMS registration on behalf of the UE, the eMSC-S shall be able to subscribe to the registration event package as described in RFC 3680 [16]. The eMSC-S shall automatically refresh this subscription as necessary.

The actions taken by the eMSC-S upon receiving a network-initiated deregistration indication from IMS should be consistent with UE handling as described in TS 24.229 [6].

Network-initiated deregistration from IMS shall only affect the status of the UE's registration in IMS, it shall not impact CS registration status.

In the event that IMS re-registration is not performed, fallback mechanisms as documented in clause 6.20.1 shall be utilized for centralization of services in IMS for non ICS UE.

6.9 CAMEL redirection of CS call to IMS

6.9.1 Call origination using CAMEL redirection of CS call to IMS

For non-ICS UE and as an option for each of the above ICCC solutions, initial call origination can be implemented using CAMEL triggers to re-route the first ICS call towards the ICCF. Figure 6.9.1-1 provides an example flow for a call made by an ICS UE-A to the other end B using CAMEL to redirect the CS calls to IMS, with AS approach of RUA of ICCF implementation.

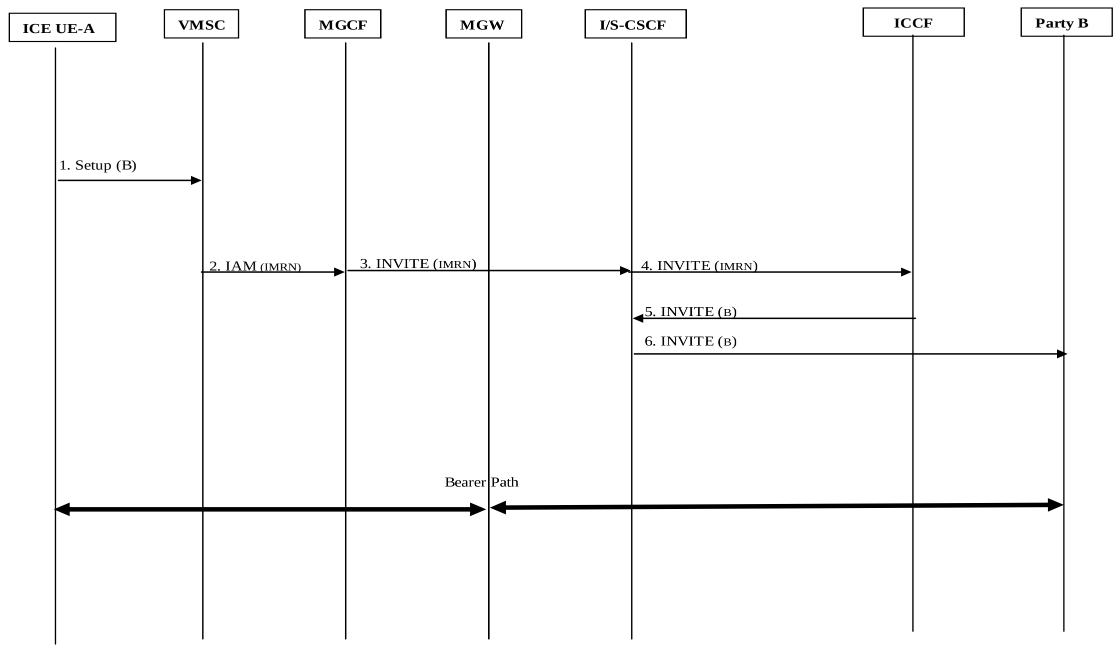


Figure 6.9.1-1: ICS origination: CAMEL Redirection to IMS

1. ICS UE-A initiates a call to user B. The ICS UE initiates standard CS procedures for establishing a CS originated call.

2. Standard VMSC procedure for CS origination. CAMEL origination triggers are used at the VMSC to reroute the call to RUA of ICCF by retrieving an IMRN which is allocated as part of the CAMEL origination trigger processing as described in TS 23.206 clause 6.2.2.2.

Editor's Note 1: How Line ID services (e.g. temp OIR) are executed when using CS call control for initial session setup is FFS.

Editor's Note 2: The relationship and inter-working with VCC is FFS.

3. Standard MGCF procedure for PSTN origination.

4. Standard CSCF procedure for PSI termination.

5. The RUA of the ICCF invokes a B2BUA, terminating the UE Leg and the other originating the RUA Leg for presentation of IMS session toward other end B on behalf of ICS UE-A.

6. Standard IMS originated session processing at the CSCF.

Editor's Note 3: If subsequent call/service control uses SIP via I1-ps, both parties must be able to establish the PS session control signalling in the call setup phase and in middle of the voice call when the service needs to be invoked. How to associate the existing CS bearer with the subsequent control signalling sent via I1-ps is FFS.

6.9.2 Use of CS call control procedures for first session setup, CAMEL used to redirect CS calls to IMS

For each of the ICCC based ICS solutions, it is also possible to use standard CS call control procedures for setup of the first UE session with CAMEL used for redirection of the first user session to IMS. In this case, the ICCC is used for subsequent session set up and control of mid call services. SIP is used with the RUA providing SIP UA behaviour on behalf of the UE for control of all user sessions. The Bearer Control Signalling path is established between the UE and the RUA by redirecting the CS call toward the RUA using CS redirection techniques such as the CAMEL origination triggers.

If a supplementary service needs to be invoked for the first user session, the UE uses ICCC to control the service related to the first session. The UE uses the ICCC for establishment and service control of the second user voice session. CS call setup procedures are not used for establishment of subsequent user sessions or invocation of mid-call voice services. The RUA maintains the SIP/SDP state machine. Upon Domain Transfer, the service state is released in the UE. The RUA is therefore inserted in the session path for IMS sessions established for a dual mode UE for synchronization of service data post Domain Transfer to CS when using this model.

Call based (temporary) line identification services shall be provided as assisted by CS domain with this model.

Editor's Note: The need for distributed service configuration for some Supplementary Services e.g. Line ID services with this model is FFS.

6.9a Deriving ICS specific IMPI/IMPU

For an ICS UE or a non-ICS UE, when it has no ISIM but USIM and performs a registration via PS access and when the network performs another registration on behalf of the UE, they may use the same IMPI and IMPU derived from the IMSI, resulting in different Contact for each registration. This is the reason why multiple registration support is needed in this case. Multiple registration support will be available in 3GPP R8 also due to other requirements (e.g. MMSC), still an alternative solution would be desirable if it does not rely on multiple registration support. The following describes such an alternative. Such a solution for enhanced MSC Server approach has been elaborated in clause 6.8a.4.

Instead of using the same IMPI/IMPU derived from the IMSI as the UE, the network should use different IMPI and IMPU derived from the IMSI when it performs registration on behalf of the UE (see also clause 6.8a.4.1) Such as <IMSI>@ics.mnc<MNC>.mcc<MCC>.3gppnetwork.org is used by the network while the UE uses <IMSI>@ims.mnc<MNC>.mcc<MCC>.3gppnetwork.org as the IMPI in the REGISTER message, according to the rules in TS 23.003. How the format is exactly defined is stage 3 issue. And these two pairs of IMPI/IMPU shall belong to the same implicit registration set, and the two temporary IMPUs should be barred for SIP non-registration procedures, as indicated in TS 23.228 [4]. This would make a solution independent on multiple registration function, while it works the same in all the other aspects.

6.10 Requirements on ICCP when using CS access

6.10.1 General

ICCP is used within the context of the I1-cs architectural alternatives to support ICS enabled clients receiving IMS services over a CS access. It needs to be transported within USSD, and as such it provides a singular transport solution for all GSM/UMTS CS access networks with an ICS enabled UE.

6.10.2 ICCP functional requirements

- The ICCP shall support an ICS capable UE to perform presentation services like OIP and TIP if needed to present SIP URI.
- The ICCP shall support the mid-call handling capabilities, such as call hold, resume, etc.
- The ICCP shall support an ICS capable UE (if it is necessary) to identify and request the individual call for service execution, e.g. to identify the call to be transferred.
- The ICCP may support an ICS capable UE to perform communication services setting modifications.

NOTE: Identification of an individual call is necessary if the ICCF is not involved in the path over PS access.

Editor's note: Whether or not additional ICCP functional requirements are needed for initial registration, re-registration and de-registration, for call initiation, for domain transfer and for call identification is FFS.

6.10.3 ICCP Non-functional requirements

- The number of bytes required to transfer the ICCP must be minimized so that it can fit with a single USSD where possible.
- The processing to be performed over ICCP must be minimized so that no noticeable latency in call set up vs. a normal CS call is to be experienced by end-users.

6.11 Media Handling

In the architecture alternatives described, the ICCF provides SIP UA behaviour on behalf of the UE towards IMS, and thereof controls all subsequent session set up and mid call services. To achieve this, the ICCF needs not only handle the session control signalling but also control the media accordingly, as it's the only entity besides ICS UE who owns the knowledge of the ongoing sessions. For example, for holding an existing session at mid call and subsequently originating a new session, The ICCF has to give instruction to toggle between the sessions and direct the media to correct destination.

During mid call service Hold, according to TS 26.114, RTCP packets should be sent continuously as "remote end-point aliveness information" to the held remote end. Meanwhile if UE originates a new session towards another remote end, a new media path would be needed from the UE to the new remote end. In this case ICCF can utilize MRFC/MRFP to handle both media paths. MRFC/MRFP can be either linked in statically or dynamically.

The conclusion is to link in MRFC/MRFP dynamically.

Editor's Note: how MRFC/MRFP is invoked from ICCF is FFS.

6.12 ICS Data Management

6.12.0 Introduction

ICS presents situations where the subscriber may use either the CS or IMS domain, for instance, to set a call forwarding value. Note that the services are centralized in IMS, so the call forwarding is a service provided by IMS. The following sections detail solutions which allow this to be done.

ICS presents situations where the CS and IMS network must share service information. This allows a subscriber to use either domain, for instance, to set a call forwarding value. The following sections detail solutions which allow this to be done.

If the service sets between IMS and CS domains need to be aligned, then the service set in IMS has to be limited to the CS telephony supplementary service set and vice versa, since there are services in IMS which cannot be mapped to existing CS supplementary services and since there are CS supplementary services which are not defined in IMS. . All of the solution alternatives shall consider only TS11 applicable supplementary services and shall exclude services that do not have a corresponding service in MMTel (e.g. BOIC and BIC when roaming).

The following sub-clauses discuss the alternatives for ICS Data Management.

NOTE: The following rules are independent whether the operator uses the enhanced MSC solution, the ICS UE solution or a combination of both. The following alternatives apply regardless of the conclusion on which entity should perform the settings conversion between MMTel and CS telephony supplementary services.

6.12.1 Alternative 1: Different data models are not allowed under the same subscription

Alternative 1 is illustrated in the figure below:

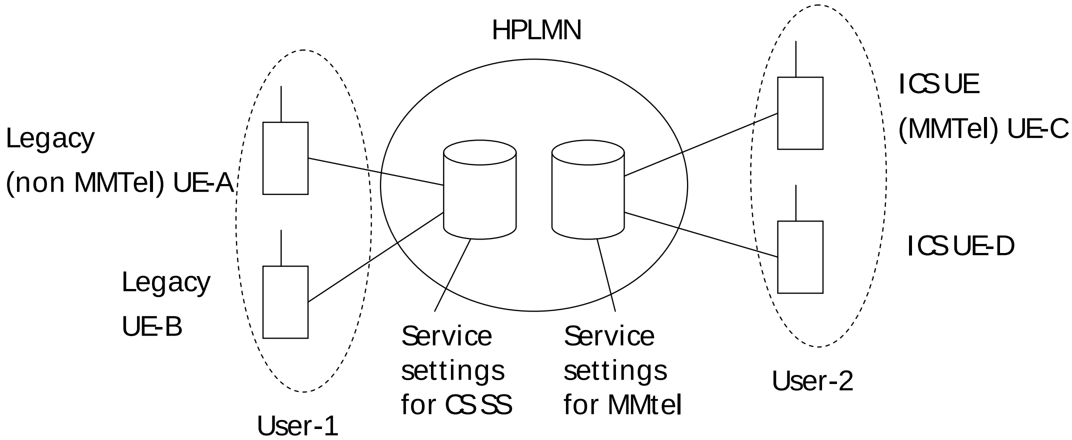


Figure 6.12.1-1: Alternative 1

In this alternative, it is not allowed to use the same subscription with an ICS UE (MMTel UE) and a non ICS UE (non MMTel UE):

- A user needs to use two separate subscriptions when using an ICS UE and a non ICS UE, or
- The operator needs to restrict MMTel service for all users.

Both ensure that the service setting modification requests can always be converted back and forth between the end user's device and network.

This alternative does not require any standardization work.

6.12.2 Alternative 2: One-time upgrade

Alternative 2 is illustrated in the figure below.

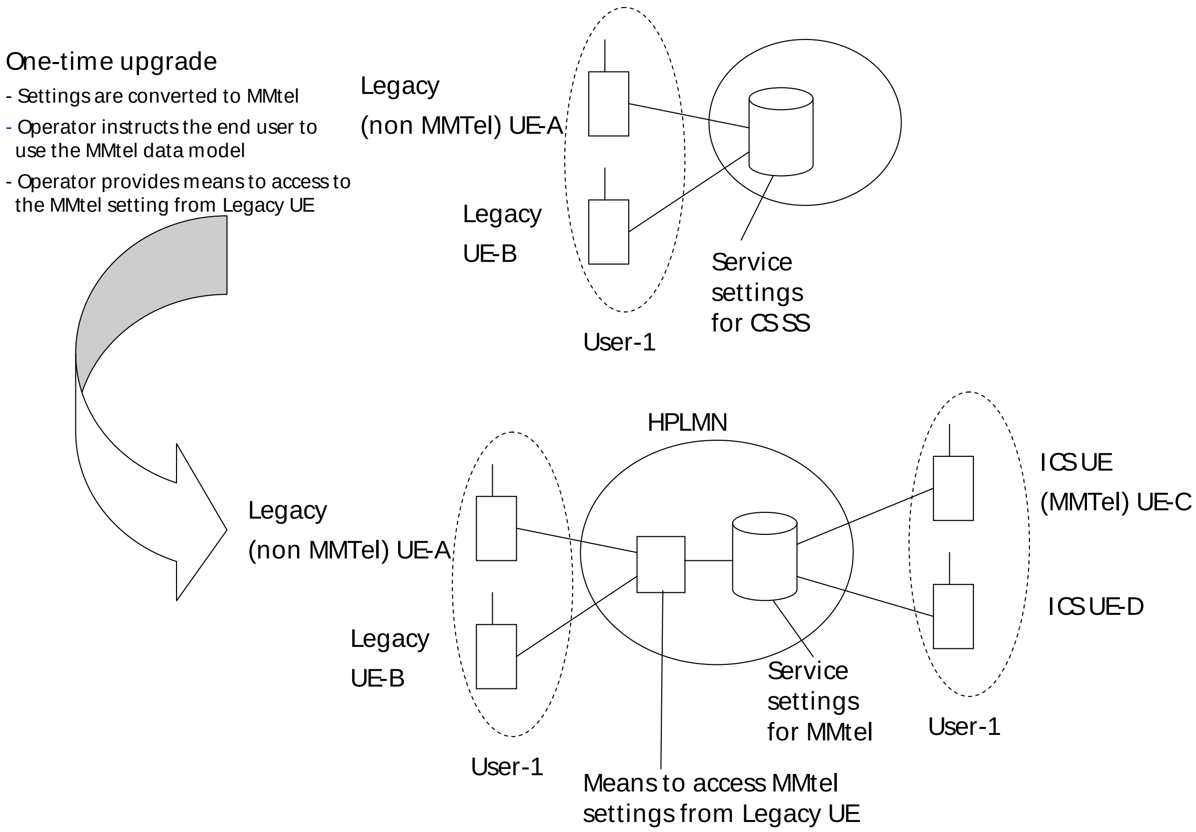


Figure 6.12.2-1: Alternative 2

For clarity reasons the User-2 is not shown in the figure, but the one-time upgrade is performed per subscriber, i.e. User-2 may continue to use CS telephony supplementary service data model while User-1 has been upgraded to use MMTel data model.

In this alternative, it is allowed to use the same subscription with an ICS UE (MMTel) and a non ICS UE (non MMTel UE) with the following restrictions:

- A user who uses a non ICS UE (non MMTel UE) only and has not used an ICS UE (MMTel UE) under the same subscription, uses the CS telephony supplementary service data model, and the service data conversion between CS domain and IMS is therefore possible.

- A user who begins to use the MMTel service data model (e.g. takes an MMTel UE into use under the same subscription), results in the behaviour that the settings in IMS may not be able to be converted from/to CS domain. For this reason, the network stores a subscriber specific flag that indicates which data model is in use. When the operator switches the user to the MMTel service data model, the operator sets the flag to indicate that MMTel service data model is in use, and from now on the network needs to block the CS telephony supplementary service setting commands from the UE. At the time of upgrade to MMTel service model, the operator must also instruct the end user not to use the CS telephony supplementary service settings UI anymore, but to use either an MMTel settings UI in the MMTel UE, or other access to the MMTel settings the operator may provide. This upgrade from CS telephony supplementary service data model to MMTel data model is performed only once per subscriber. The actual trigger for the upgrade does not need to be standardized.

Editor's Note: It is FFS what is the mechanism to block the CS telephony supplementary service setting commands in the network. At least blocking the SS commands in HLR or use of CAMEL service may be considered. This may depend on the decision which entity is responsible to convert the settings between CS telephony supplementary services and MMTel.

- In order to access the MMTel service settings from non MMTel UE after the upgrade, the operator may provide e.g. a Ut service settings client implemented in J2ME application, SIM toolkit application, or WAP/web page. Such client may be pushed to the non MMTel UE at the point of upgrade, with the instructions to the end user how to use them. It is up to the operator to decide what kind of mechanism is provided to its user to access the MMTel service settings. It also depends on the legacy UE, what kind of means it supports. This will not be standardized.

NOTE: The drawback of this alternative is that the CS telephony supplementary service settings UI cannot be used after the one-time upgrade to MMTel services.

6.12.3 Alternative 3: Convert the settings whenever possible

Alternative 3 is illustrated in the figure below.

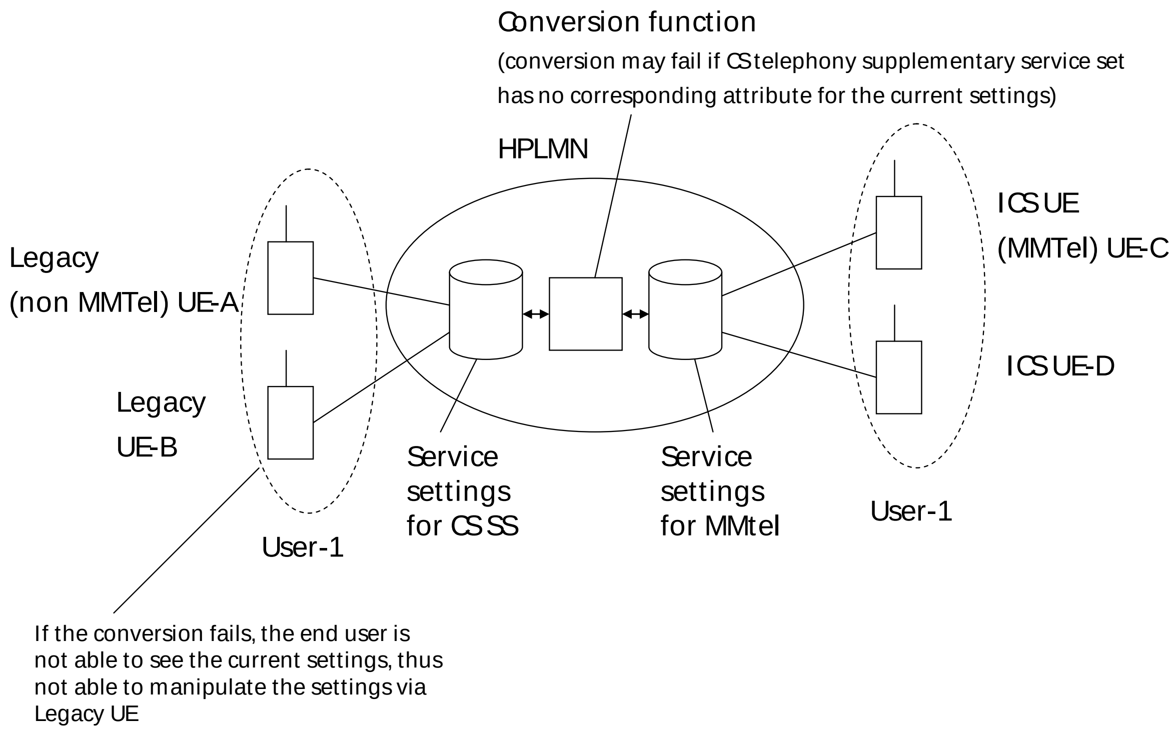


Figure 6.12.3-1: Alternative 3

In this alternative it is allowed to use the same subscription with an ICS UE (MMTel) and a non ICS UE (non MMTel UE) with the following restrictions:

- The service settings are converted between CS domain and IMS whenever the settings data in IMS allows it.
- Whenever the user submits a service data to MMTel settings that is not possible to convert to CS domain, the conversion is not performed. If the end user now tries to retrieve the service settings in CS while using a legacy UE, the network returns a "network error" indication to the user. Thus the end user may not be able to know the reason why the settings are not available. The end user might not even try to modify the settings after such network error. But if the end user anyhow now tries to modify the settings (e.g. activate the CFU), there are two alternatives; either the network replaces the corresponding settings in the current MMTel settings with the new value, or the network blocks the modification which cannot be unambiguously mapped to the current MMTel settings.

Editor's Note: It is FFS whether the conversion and replacement functions should be standardized, or left implementation specific.

- Whenever the user modifies the MMTel service settings such that they can be converted to CS telephony supplementary service settings again, the user is able to use a legacy UE for retrieving and modifying service settings.

NOTE: This alternative would lead to inconsistent user experience, because retrieving/modifying the services from the legacy UE works in some cases and in some cases not. The user may not be aware why the service settings occasionally are not available to the legacy UE.

6.12.4 Option: Limited MMTel Service Set

A possible complementary to the alternatives above would be to standardize an XML schema for a subset of MMTel service set equal to CS telephony supplementary service set. The settings from a UE that implements the support for this schema can be converted back and forth with CS telephony supplementary service settings. A limited version of MMTel implemented in the UE ensures an access to the service settings of the CS telephony supplementary service model over the Ut reference point.

6.12.5 Non-transparent data solution alternative

This solution proposes to extend the Sh interface to allow an IMS AS to request CS information.

Currently, the Sh interface supports transferring service data to IMS AS, but it does not support transferring CS service data to IMS AS. The IMS AS cannot update the CS service data in HLR/HSS either. This solution extends the Sh interface to support this functionality.

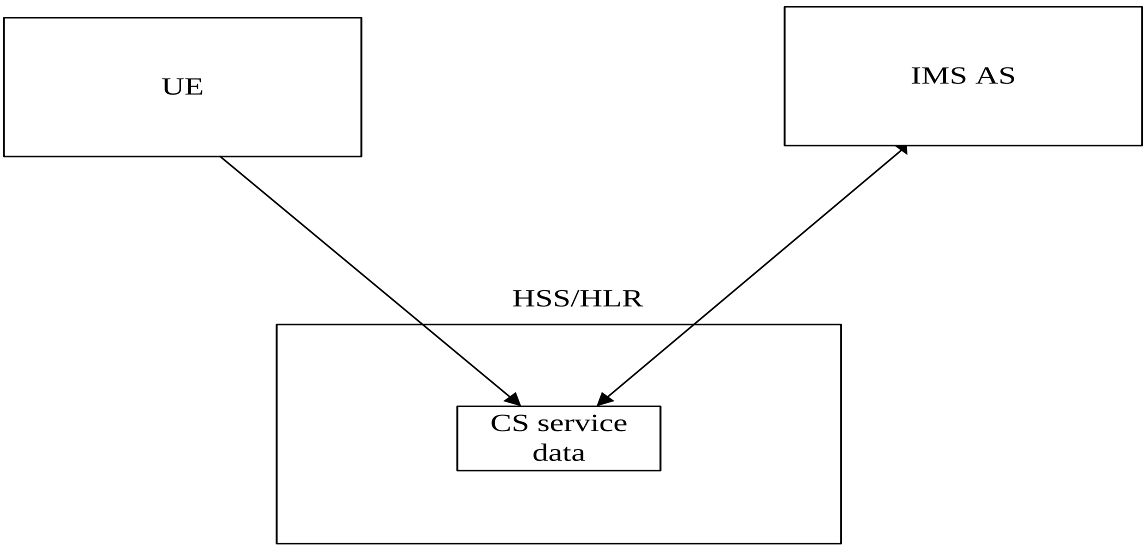


Figure 6.12.1-1: Non-transparent data solution

The following extensions for the Sh interface are necessary:

- Add a new data Ref value in the Sh-Subscribe message allowing the IMS AS to subscribe the CS service data change in HSS/HLR;

- Add a new data Ref value in the Sh-Update message used when IMS AS changes CS service data.

Once the Sh interface is modified, the HSS/HLR can use the information flow shown in Figure 6.12.1-2 to change the CS service data stored in IMS AS

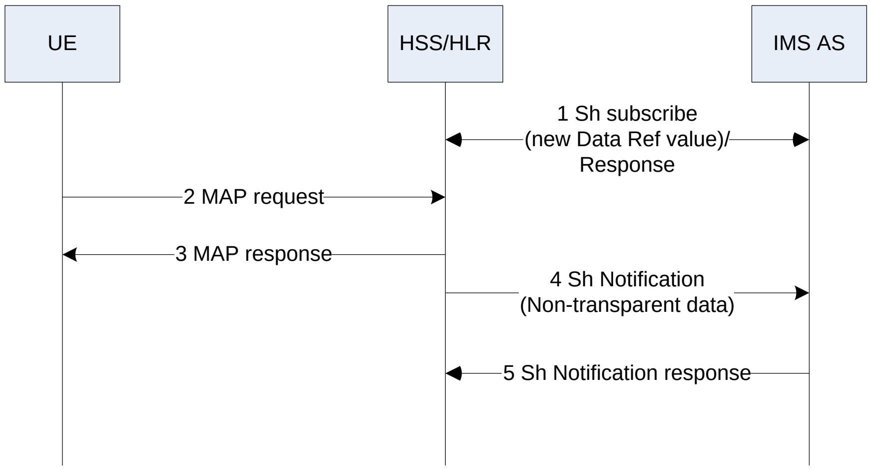


Figure 6.12.1-2: Using extended Sh interface for IMS AS to subscribe CS supplementary data change in HSS/HLR

The IMS AS can also use the modified Sh interface to modify the CS service data stored in HSS/HLR as shown in the flow of Figure 6.12.1-3.

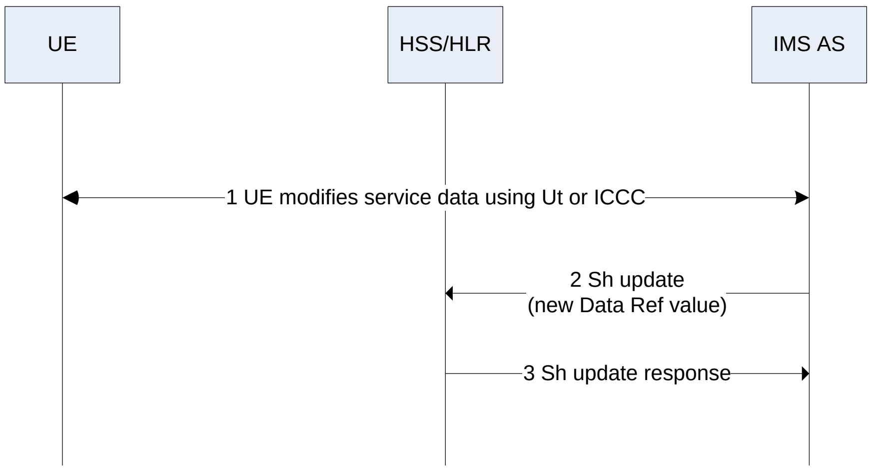


Figure 6.12.1-3: Using extended Sh interface to transfer CS supplementary data from IMS AS to HSS/HLR

Overall, this alternative has the disadvantage of requiring modifications to extend the Sh interface. However, the solution also has the following benefits:

- The extended Sh interface can be shared by other Application Servers.
- This approach maintains the integrity of the principle of transparent data. Since the CS related data is a new data type, it is not considered transparent data. The HSS/HLR would be expected to define and use this information, as opposed to transparent data which the HSS is not supposed to understand or modify.

NOTE: The definition of XML data structure used in Sh message and the detail of the new added data reference value needs stage 3 work.

6.12.6 Transparent data solution alternative

This second solution allows the Sh interface to remain the same, but violates the principle that the HSS not understand or modify any of the transparent data. This solution has the HSS/HLR create a copy of the CS service data but in a format which is more suitable as transparent data (e.g. XML). The HSS/HLR and any Application Servers can use and share this particular file of transparent data using current Sh interface message without any extensions. This approach is shown in Figure 6.12.2-1.

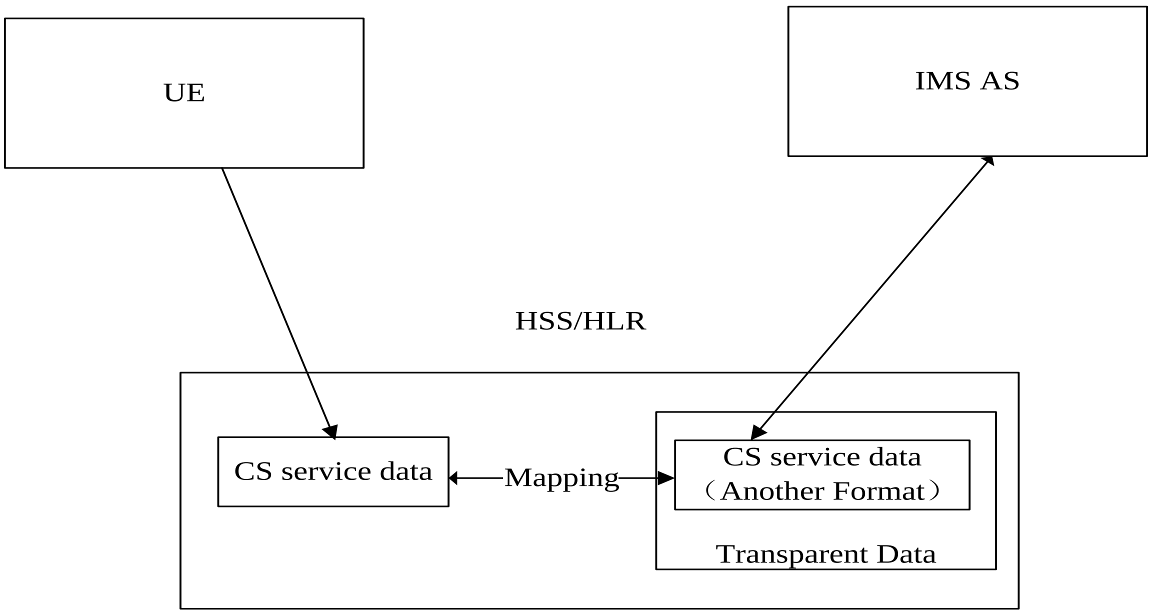


Figure 6.12.2-1: Transparent data solution

In this solution, HSS/HLR converts the CS service data to the related transparent data and back again as needed. The HSS/HLR needs to be aware of the data structure of this specific transparent data file, and continues to be ignorant of all the other transparent data files. However, it does violates the principle that HSS/HLR is not aware of any of the transparent data structures.

Figure 6.12.2-2 shows how this would work when the UE modifies the service data via the IMS AS. The IMS AS is assumed to have a current copy of the service data. The UE's modification of this data is made in the AS, and then sent via an Sh update to the specific transparent data file known to the HSS. The HSS will acknowledge this transfer in Step 3, and then copy any changes from the transparent data file to the CS copy of the file.

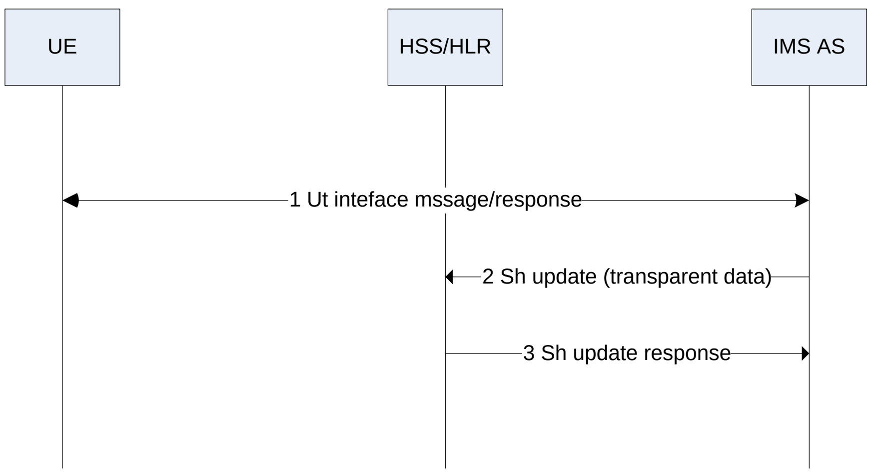


Figure 6.12.2-2: Information flow for IMS AS changes HSS/HLR's CS service data

This approach also supports changes made by the CS network in much the same way as the first alternative. The messaging for a CS based change is shown in Figure 6.12.2-3.

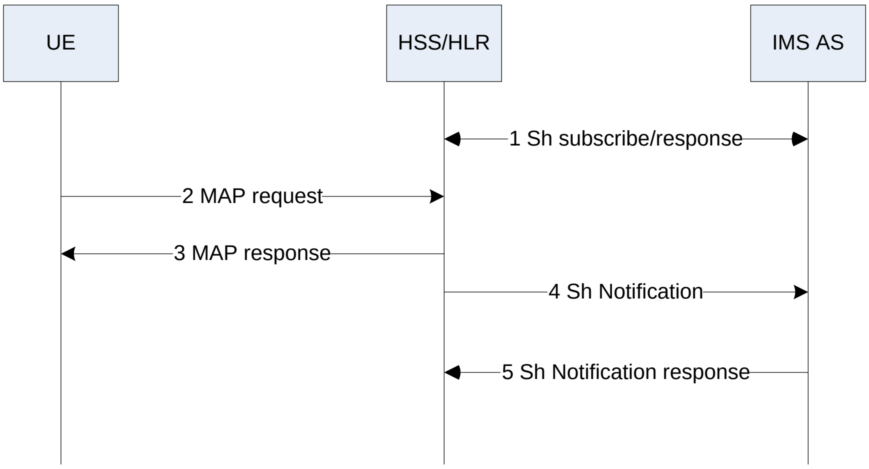


Figure 6.12.2-3: Information flow for HSS/HLR updates IMS AS's CS service data

The HSS/HLR will copy the CS changes to the transparent data before it notifies IMS AS in Step 4.

The advantage of this alternative is the Sh interface can be used without modification. However the HSS will be violating the principle that the HSS does not understand or use the contents of any transparent data file.

Editor's Note: Whether New Data Service Indication value needs to be added in Sh-Subscribe and Sh-Notification message is FFS.

NOTE 1: Data synchronization between these two copies of data is implementation specific.

NOTE 2: The definition of XML data structure need stage 3's work.

6.12.7 Data validation discussion

HLR already has the functionality for CS service data's validation. Both of the proposed solutions can reuse this part of functionality if the rules for the data validation do not change when the service is moved to IMS.

It is possible that the data validation rules will change, and this opens the question of whether the HLR/HSS is modified to support the new rules, or if the IMS AS does the data checking. In cases where the IMS AS is now used to check the data, the CS transactions will need to send the data to the IMS AS first before the CS transaction completes. If the IMS AS accepts the data as provided, the CS transaction can complete successfully. If the IMS AS rejects the data, then the CS transaction will complete with a failure.

The IMS AS modifications via the Ut interface has a similar issue if the HSS does the data validation. The data will need to be sent to the HSS to determine if the settings are allowed, and the IMS AS will need to check to see the HSS data transaction completes successfully before it completes the transaction to the UE over the Ut interface. The procedures used in order to have either the HSS or the IMS AS approve the data is FFS.

6.13 Service Support for ICS UE, using I1-ps and I1-cs approach, ICCC for call setup

6.13.1 Line ID Services (OIP, OIR, TIP, TIR)

With the RUA of the ICCF providing SIP UA behaviour on behalf of the ICS UE, these services are provided in IMS. ICCC is used for communication of Line ID information between the IMS and the UE for IMS sessions using CS voice media.

6.13.2 Communication Diversion Services

6.13.2.1 Communication Diversion services exclusively controlled in home IMS (CFU, CFNL)

With the ICCF providing the SIP UA behaviour on behalf of the ICS UE, these services are provided in IMS.

6.13.2.2 Communication Diversion services requiring participation of serving network (CFNRy, CFNRc and CFB)

With the RUA of the ICCF providing SIP UA behaviour on behalf of the ICS UE, these services are provided in IMS. ICCC may be used for indication of the service triggers to IMS, another option is use of Release cause code generated by the CS network nodes. In addition, CFNRc can be triggered by using the MAP failure code response generated by the CS network nodes.

6.13.2.2.1 Communication Forwarding on mobile subscriber not reachable (CFNRc) over I1-CS

6.13.2.2.1.1 CFNRc for no paging response

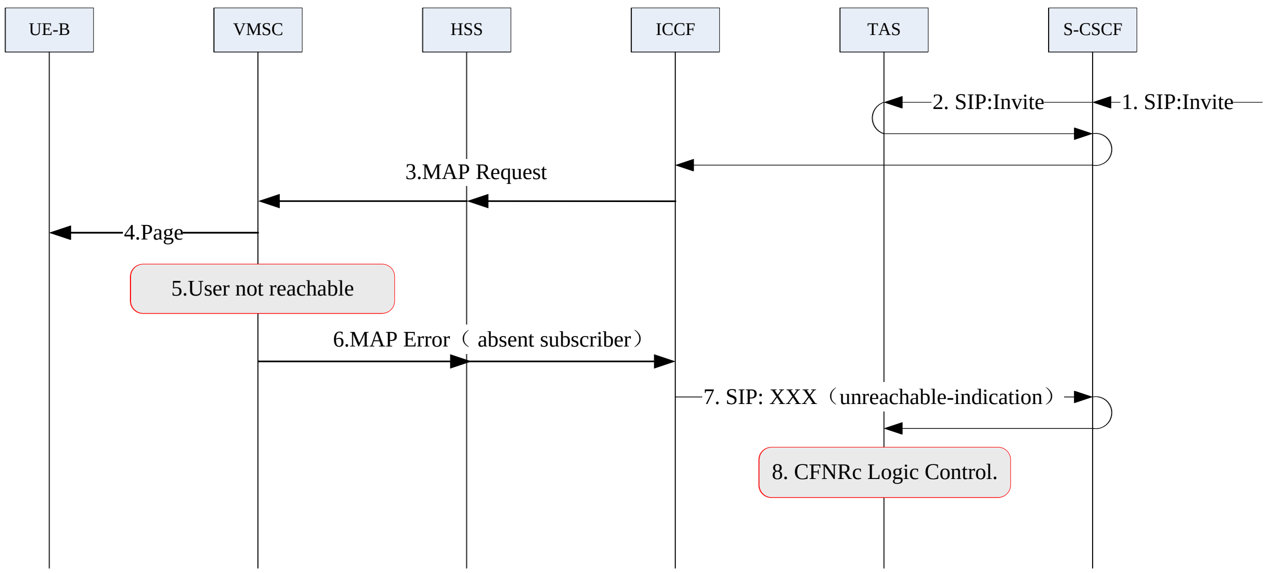


Figure 6.13.2.2.1.1-1: CFNRc for no paging response over I1-cs

1. An incoming SIP INVITE is received at the S-CSCF of the B party.
2. The S-CSCF forwards the INVITE to the TAS and ICCF.
3. The ICCF (acting as a B2BUA) sends a MAP request to HSS and the HSS forwards the MAP request to the VMSC.
4. VMSC sends page message to UE-B.

5. No response to the page message, VMSC determines that the UE-B is not reachable.

6. VMSC sends a MAP Error message to ICCF, with the absent subscriber failure code.

7. ICCF converts the MAP Error message into SIP response with unreachable indication according to the failure code, and sends the SIP message to TAS via S-CSCF.

8. According the SIP message and service profile data, TAS executes the CFNRc logic.

6.13.2.2.1.2 CFNRc for the IMSI detach

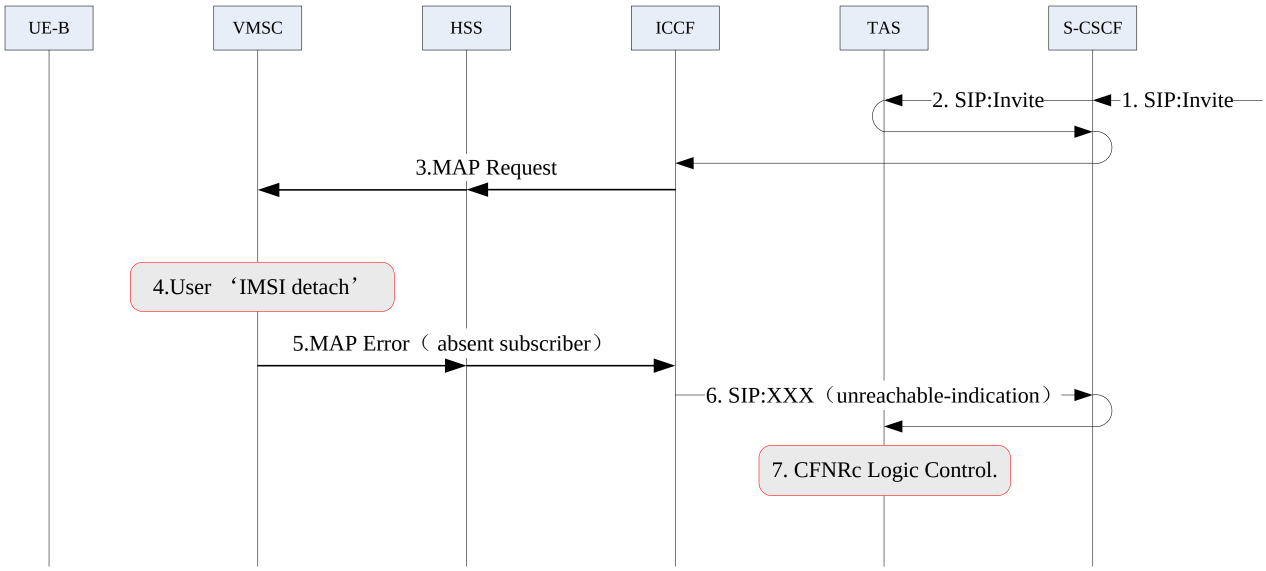


Figure 6.13.2.2.1.2-1: CFNRc for the IMSI detach over I1-cs

1. An incoming SIP INVITE is received at the S-CSCF of the B party.

2. The S-CSCF forwards the INVITE to the TAS and ICCF.

3. The ICCF (acting as a B2BUA) sends a MAP request to the HSS and the HSS forwards the MAP request to the VMSC.

If the subscriber absent information is already known by HSS, HSS will send a MAP error message with the absent subscriber failure code to the ICCF directly. Then the call flow will skip to step 6.

4. VMSC gets the subscriber 'IMSI detach' from the subscriber data of VLR.

5. VMSC sends a MAP Error message to ICCF, with the absent subscriber failure code.

6. ICCF converts the MAP Error message to a SIP response with unreachable indication according to the failure code, and sends the SIP message to TAS via S-CSCF.

7. TAS executes the CFNRc logic.

6.13.2.2.2 Communication Forwarding on Busy User (CFB)

6.13.2.2.2.1 Communication Forwarding on Busy User (CFB) over I1-cs

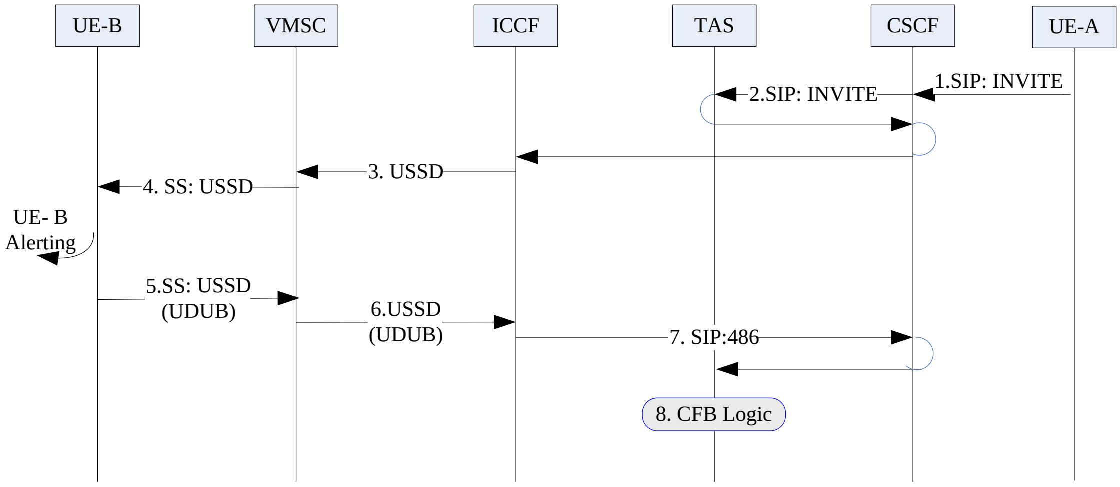


Figure 6.13.2.2.3.1-1: CFB Service over I1-cs

1. An incoming SIP INVITE is received at the S-CSCF of the B party from the A party.
2. The S-CSCF forwards the INVITE to the TAS and ICCF.
3. The ICCF sends an incoming Call Request in USSD message to the VMSC over I1-cs.
4. The VMSC forwards the USSD message to B party.
5. After UE-B alerting, it rejects this call due to busyness, and sends UDUB (User Determined User Busy) in a USSD message to the VMSC.
6. The VMSC forwards this USSD message to the ICCF.
7. The ICCF creates SIP 486 expressing B party is busy, and sends the SIP 486 to the TAS via the CSCF.
8. The TAS processes the SIP 486 and executes the CFB logic.

6.13.2.2.2.2 Communication Forwarding on Busy User (CFB) over I1-ps

The CFB call flow over I1-ps follows the procedures specified in ETSI TS 183 004 [A1.4].

6.13.3 Communication Deflection

With the RUA of the ICCF providing SIP UA behaviour on behalf of the ICS UE, these services are provided in IMS. ICCC is used for indication of the service triggers to IMS.

6.13.3.1 Communication Deflection (CD) over I1-cs

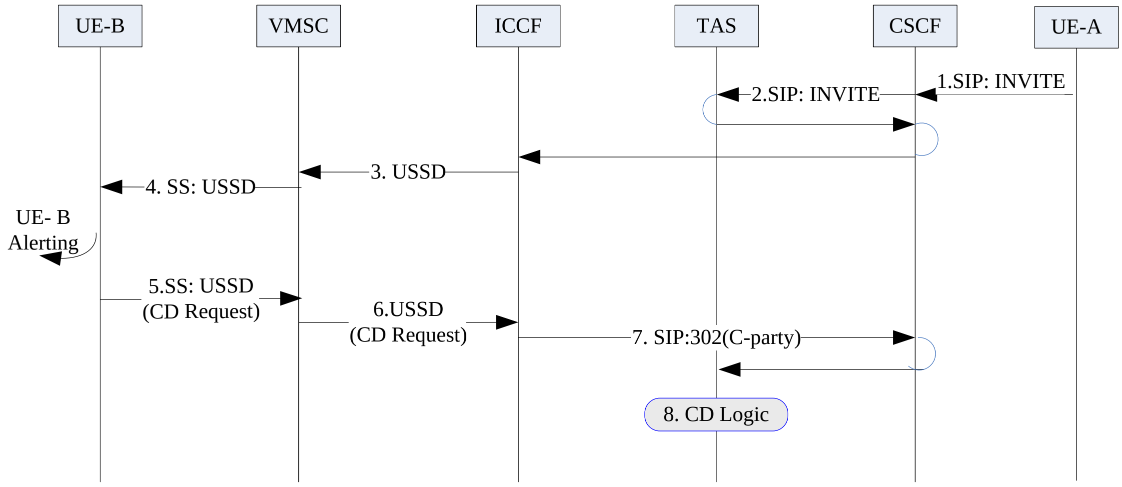


Figure 6.13.3.1-1: CD Service over I1-cs

1. An incoming SIP INVITE is received at the S-CSCF of the B party from the A party.
2. The S-CSCF forwards the INVITE to the TAS and ICCF.
3. The ICCF sends an incoming Call Request in USSD message to the VMSC over I1-cs.
4. The VMSC forwards the USSD message to B party.
5. After UE-B alerting, it sends CD request with C party number in USSD message to the ICCF.
6. The VMSC forwards the USSD message to the ICCF.
7. The ICCF converts the Communication Deflection request into SIP 302 which contains SIP or Tel URI of C party. Then the ICCF sends SIP 302 to the TAS via the CSCF.
8. The TAS executes the Communication Deflection logic.

6.13.3.2 Communication Deflection (CD) over I1-ps

The CD call flow over I1-ps follows the procedures specified in ETSI TS 183 004[A1.2].

6.13.4 Communication Barring

With the RUA of the ICCF providing SIP UA behaviour on behalf of the ICS UE, these services are provided in IMS.

6.13.5 Mid call services (Communication Hold, CW, Conf, Communication Transfer)

With the RUA of the ICCF providing SIP UA behaviour on behalf of the ICS UE, these services are provided in IMS. ICCC is used for communication of service control signalling between the IMS and the UE for IMS sessions using CS voice media.

6.13.6 Session continuity

The Voice Call Continuity for ICS UEs capable of VCC is based on TS 23.206. The session continuity of ICS video call would require DTF to be enhanced with capability of anchoring multimedia call which is under study in MMSC WI. TS 23.206 modification are required.

6.13.7 User configuration of Supplementary Services

Service data management and control of user configuration of services data is provided by IMS.

6.13.8 Serving Domain Considerations

Table 6.13.8-1 below provides recommendation for serving domain for the services discussed in this clause:

Table 6.13.8-1: Service set for ICS UE with I1-ps and I1-cs, use of ICCC

Service Capability	Serving Domain
Supplementary Services	
Originating Identification Presentation	IMS
Originating Identification Restriction	IMS
Terminating Identification Presentation	IMS
Terminating Identification Restriction	IMS
Communication Diversion (CFU, CFNL)	IMS
Communication Diversion (CFNR, CFB)	IMS
Communication Deflection	IMS
Call Wait	IMS
Communication Hold	IMS
Communication Barring	IMS
Conference	IMS
Explicit Communication Transfer	IMS
Service Continuity	
Basic Service Continuity	IMS [1]
Service Continuity on non mid call services	IMS [1]
Service Continuity with mid call services	IMS [1]
Other capabilities	
Call Independent Supplementary Services Operations	IMS
[1] Applicable to ICS UEs capable of VCC.	

6.14 Service Support for ICS UE using I1-cs approach, CS call control with CAMEL for call setup

When using the I1-cs approach with CS call setup for ICS UEs, the ICCF provides a SIP UA behaviour on behalf of the ICS UE for CS sessions established using standard CS procedures, and redirected to IMS using techniques similar to the procedures defined for VCC in TS 23.206. Since the solution uses standard CS procedures for sessions established over CS access, it is limited in its service offering as discussed below.

6.14.1 Line ID Services (OIP, OIR, TIP, TIR)

Since standard session setup procedures are used to establish the ICS UE sessions, with the VMSC controlling session origination and session delivery from/to the ICS UEs, Line ID services configuration in the HLR component of the HSS for appropriate service execution at the VMSC is required. Some control of these services may also be provided in IMS with the use of ICCC.

6.14.2 Communication Diversion Services

6.14.2.1 Communication Diversion services exclusively controlled in home IMS (CFU, CFNL, CFNRy, CFNRc)

With the ICCF providing the SIP UA behaviour on behalf of the ICS UE, these services are provided in IMS.

Editor's note: It is FFS how CFNR is provided in IMS.

6.14.2.1.1 Communication Forwarding Unconditional (CFU)

Incoming call will be redirected to the pre-configured target by TAS if the CFU service has been actived. This service is independent of the access network type used.

6.14.2.1.2 Communication Forwarding on Not Logged-in (CFNL)

Editor Note: CFNL will depend on the result of CS registration discussion, and it is regardless of the implement of ICS termination.

6.14.2.1.3 Communication Forwarding on Subscriber Not Reachable (CFNRc)

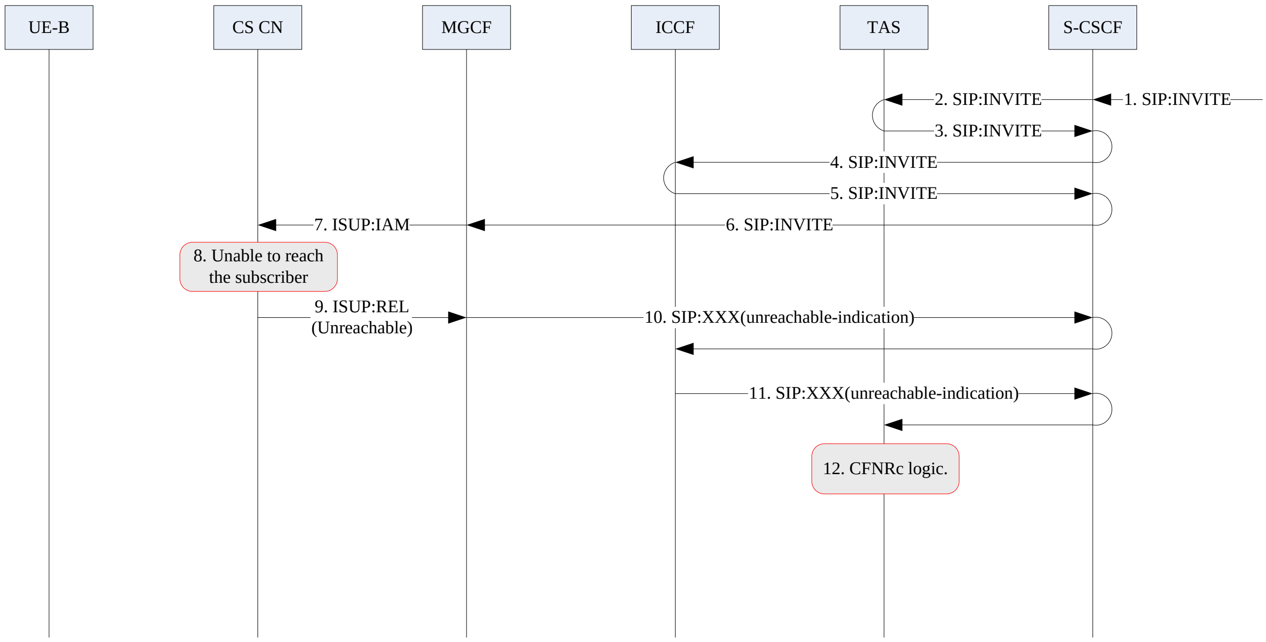


Figure 6.14.2.1.3-1: CFNRc Service

1. An incoming SIP INVITE is received at the S-CSCF of the B party.

2-4. The S-CSCF forwards the INVITE to the TAS and ICCF.

5-7. The ICCF generates an INVITE to the UE-B via CS domain.

8-9. An ISUP RELEASE with cause "unreachable" will be return from CS CN while the subscriber is unreachable.
The cause will be #18(no user responding) when fail to paging UE-B or #20(subscriber absent) when MSC got SRI response with "subscriber absent" from HLR.

10-12. The RELEASE message will be convert to SIP response with unreachable indication by MGCF and forwards to TAS for execute CFNRc service logic.

NOTE 1: The release cause return from CS-CN in step-9 will be #18(no user responding) when fail to paging UE-B or #20(subscriber absent) when MSC got SRI response with "subscriber absent" from HLR. According TS 29.163 [18], MGCF will map to SIP 480 for ISUP REL with #18(no user responding)/#20(subscriber absent) and #19 (no answer from the user)/#21(call rejected).

In TISPAN WI03075 ETSI TS 183 004, TAS will execute CFNRc service if the response message is 408(Request timeout), 503(Service unavailable) or 500(Server internal error).

So TAS can't distinguish the really release cause from the response SIP message. There may be some possible mechanisms to resolve this problem: 1) ICCF checks the call session's state and then convert SIP 480 to the correct SIP response. 2) TAS checks the call session's state and then executes the correct service logic. 3) MGCF separate SIP response by CS release cause. Which mechanisms will be recommended for standardization is stage 3 work.

NOTE 2: Option 2 needs special TAS treatment.

NOTE 3: Option 3 requires MGCF upgrade to Rel-8.

6.14.2.1.4 Communication Forwarding on No Reply (CFNRy)

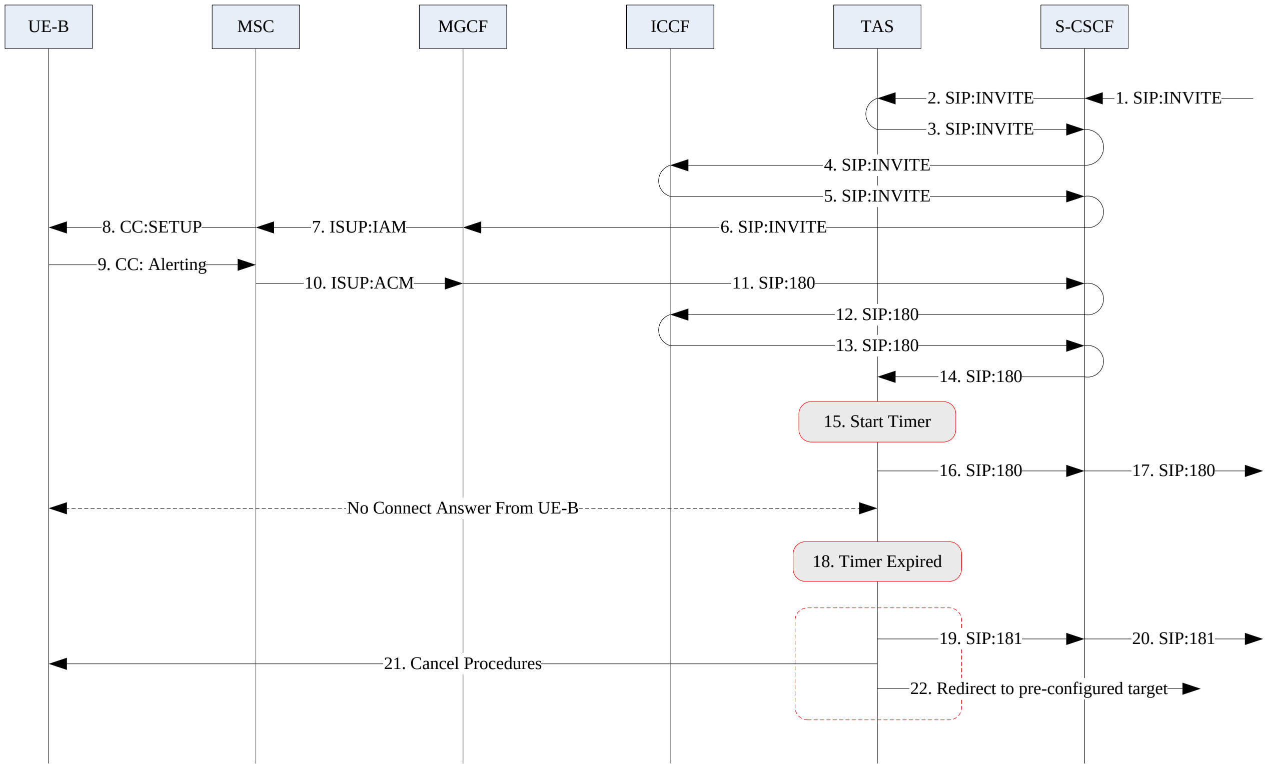


Figure 6.14.2.1.4-1: CFNRy Service

This call flow applies for both the I1-cs and the I1-ps approach.

1. An incoming SIP INVITE is received at the S-CSCF of the B party.

2-4. The S-CSCF forwards the INVITE to the TAS and ICCF.

5-8. The ICCF generates an INVITE to the UE-B via CS domain. If the message is to be routed to CS, the IMS entity (e.g. in case the ICCF sends the SRI message to get the MSRN towards the HLR) will insert the indication to the SRI message in order to suppress the announcement in the MSC if the call cannot be successfully established later.

9-14. The UE-B sends back an ALERTING indication.

15-17. When got an alerting from UE-B, TAS would start the non-reply timer, and forwards SIP 180 to next hop.

18. After ALERTING, UE-B has not any other requests or responses, so the non-reply timer in TAS would not be stopped until expired, and CFNRy service will be triggered.

19-23. The TAS executes CFNRy logic, sends an 181 SIP message to remote original caller party to indicate that call is forwarding, cancels the call establishing with UE-B, and redirects the call to pre-configured target like UE-C or voice mail.

NOTE 1: Within the MSC there should be no MT CAMEL trigger for "No Answer" and "Busy" configured.

NOTE 2: In case the timer expires and the MSC does not play an announcement, the MSC should at least release the call.

6.14.2.2 Communication Diversion services requiring participation of serving network (CFB)

Some control of these services may be provided in IMS. Since standard session setup procedures are used to establish the ICS UE sessions, with the VMSC controlling session origination and session delivery from/to the ICS UEs, some specific functions may also required at the VMSC.

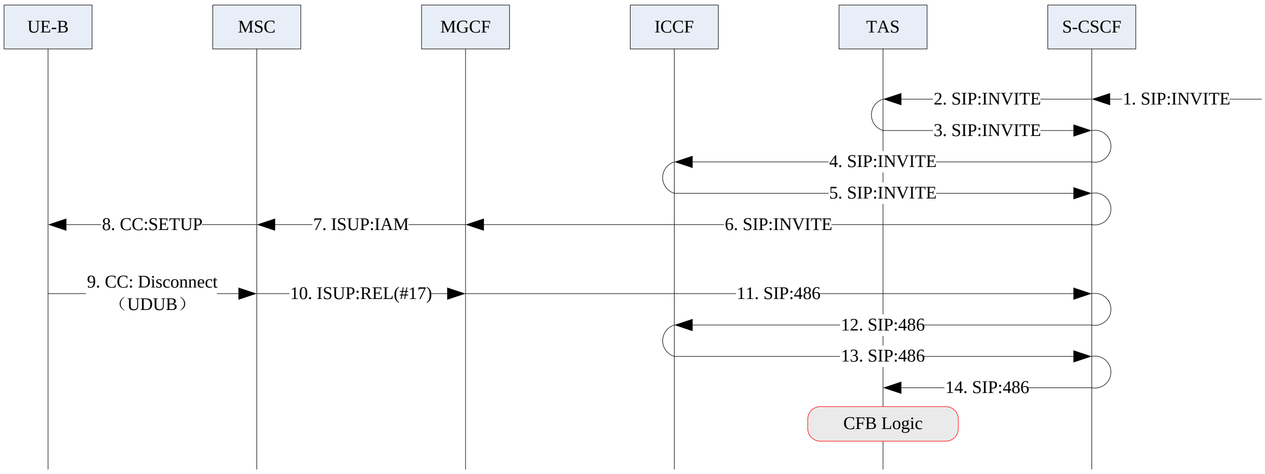


Figure 6.14.2.2-1: CFB Service

This call flow applies for both the I1-cs and the I1-ps approach.

1. An incoming SIP INVITE is received at the S-CSCF of the B party.

2-4. The S-CSCF forwards the INVITE to the TAS and ICCF.

5-8. The ICCF generates an INVITE to the UE-B via CS domain. If the message is to be routed to CS, the IMS entity (e.g. in case the ICCF sends the SRI message to get the MSRN towards the HLR) will insert the indication to the SRI message to suppress the announcement in the MSC if the call cannot be successfully established later.

9-10. The UE-B disconnects the call and returns a cause: UDUB, then the MSC will release the incoming call with a cause "User Busy" in RELEASE message.

11. MGCF will convert the RELEASE message with failure cause "User Busy"(#17) to a 486(Busy Here) SIP message and forwards to next hop.

12-14. The 486 Busy Here SIP response will be returned to TAS finally, and the CFB logic would be executed by the TAS.

NOTE: In case the timer expires and the MSC does not play an announcement, the MSC should at least release the call.

6.14.3 Communication Deflection

With the RUA of the ICCF providing SIP UA behaviour on behalf of the ICS UE, these services are provided in IMS. ICCC is used for indication of the service triggers to IMS.

6.14.4 Communication Barring

With the RUA of the ICCF providing SIP UA behaviour on behalf of the ICS UE, these services are provided in IMS.

6.14.5 Mid call services (Communication Hold, CW, Conf, Communication Transfer)

With the RUA of the ICCF providing SIP UA behaviour on behalf of the ICS UE, these services are provided in IMS. ICCC is used for communication of service control signalling between the IMS and the UE for IMS sessions using CS voice media.

6.14.6 Session continuity

The Voice Call Continuity for ICS UEs capable of VCC is based on TS 23.206. The session continuity of ICS video call would require DTF to be enhanced with capability of anchoring multimedia call which is under study in MMSC WI . TS 23.206 modification are required.

6.14.7 User configuration of Supplementary Services

Service data management and control of user configuration of services data is provided by the IMS for services controlled in IMS and by the CS domain for services controlled in CS domain.

6.14.8 Serving Domain Considerations

Table 6.14.8-1 below provides recommendation for serving domain for the services discussed in this clause:

Table 6.14.8-1: Service set for ICS UE with and I1-cs, use of CAMEL

Service Capability	Serving Domain
Supplementary Services	
Originating Identification Presentation	IMS[1]
Originating Identification Restriction	IMS[1]
Terminating Identification Presentation	IMS[1]
Terminating Identification Restriction	IMS[1]
Communication Diversion (CFU, CFNL, CFNR)	IMS
Communication Diversion (CFB)	IMS[2]
Communication Deflection	IMS
Call Wait	IMS
Communication Hold	IMS
Communication Barring	IMS
Conference	IMS
Explicit Communication Transfer	IMS
Service Continuity	
Basic Service Continuity	IMS[3]
Service Continuity on non mid call services	IMS[3]
Service Continuity with mid call services	IMS[3]
Other capabilities	
Call Independent Supplementary Services Operations	IMS/CS[4]
[1] Exclusive control in IMS of this service is not possible, configuration of the CS domain service is required. May require ICCC for control of service in IMS. [2] Some specific VMSC functions may be required such as use of VT-CSI or ISUP Release cause. The exact mechanism is FFS. [3] Applicable to ICS UEs capable of VCC. [4] Service data management provided by the IMS for services controlled in IMS and by CS domain for services controlled in CS domain.	

6.15 Domain selection function to ICS

6.15.1 Access Domain Selection (ADS)

6.15.1.1 ADS for terminating sessions (TADS)

When TADS-IMS selects the IMS access (e.g. selected I1-ps or just using IP-CAN) to deliver the termination session, the TADS in the UE can indicate to the TADS-IMS whether voice session should be delivered via CS access and whether I1-ps or I1-cs should be used for continuing the session setup.

The following figure depicts the relationship between TADS-IMS and TADS in UE.

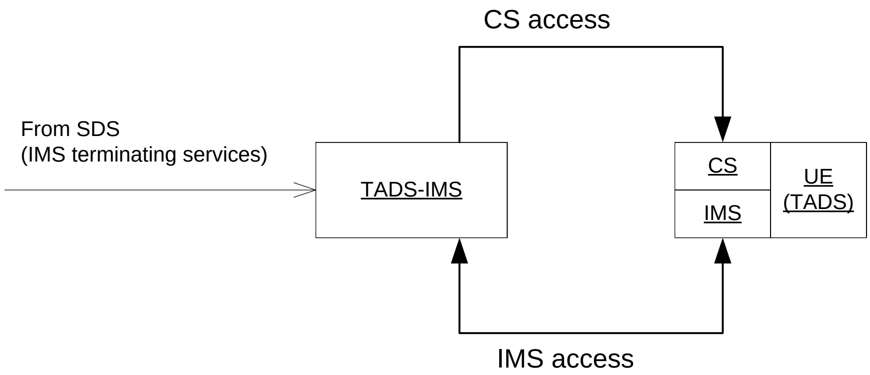


Figure 6.15.1.1-1: Relationship between TADS-IMS and TADS in UE

The TADS in the UE may use the following information to determine the response to TADS-IMS.

- ICS capable IMS network.
- Access network currently on.
- User preference and operator policy.
- Any other information that the UE can be used to determine the media capabilities of the IP-CAN.

NOTE: If the TADS-UE determines the session (both media and session control signalling) can be delivered over the IP-CAN then the UE would continue the session setup as defined for IMS.

6.15.1.2 TADS-UE Scenarios

The following figure 6.15.1.2-1 shows a high level call flow between TADS in the IMS and in the ICS UE for selection ICS termination.

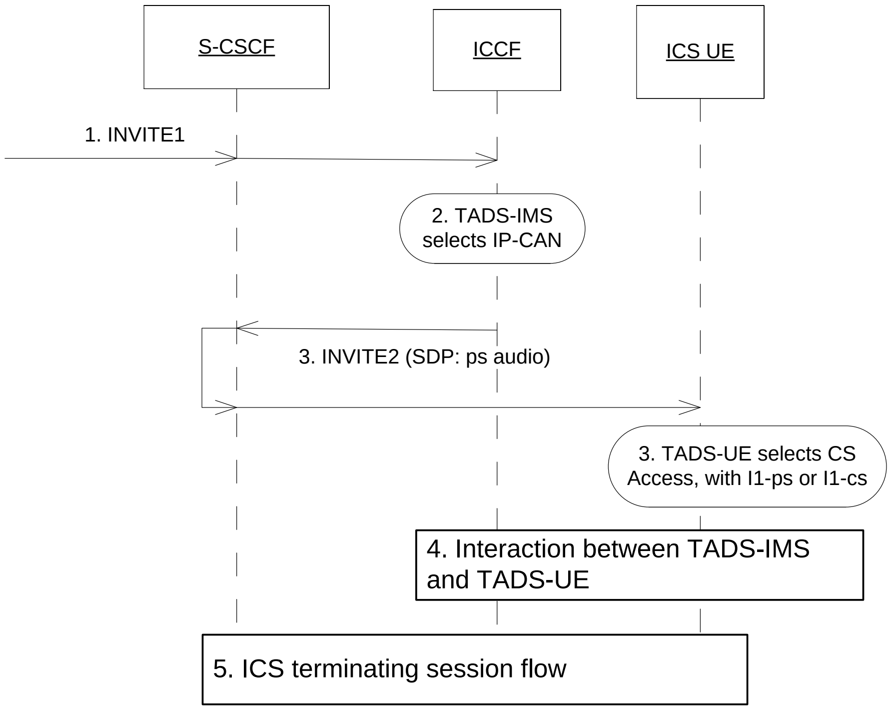


Figure 6.15.1.2-1: TADS in the IMS and in the ICS UE for selection ICS termination

1. An incoming SIP INVITE from the originating party and is forwarded to ICCF.
2. ICCF invokes the TADS (TADS-IMS) and it selects the IP-CAN to deliver the session. An INVITE2 is generated toward the ICS UE.
3. When ICS UE receives the INVITE2, the TADS in the UE determine that the IP-CAN is not suitable for voice media and it needs to use the CS access for voice. It also determines whether I1-cs or I1-ps (e.g. based on DTM or multiRab case) can be used for session signalling.
4. TADS-UE indicates to TADS-IMS that voice needs to be delivered over CS media and which session signalling is to be used (I1-cs or I1-ps).

Editors Note: It is FFD on how step4 and 5 are combined to minimize the time delay for call setup.

ICCF performs the ICS termination based on the procedure defined in clause 6.5 (for I1-ps) or clause 6.6 (i1-cs).

6.16 Impact on IMS

6.17 Impact on CS Core Network

6.18 Impact on UE

ICS UE shall be able to learn whether both HPLMN and the VPLMN supports ICCC; i.e. whether I1-ps and/or I1-cs or neither are supported. UE needs this information e.g. to determine whether it should use the ICCC in call initiations

Editor's Note: How the UE learns whether the HPLMN and VPLMN supports ICCC or not is FFS.

Non-ICS UE does not need to learn the home or visited network support for ICS.

6.19 Fallback Mechanisms

6.19.1 Fallback for L-CAAF-n and enhanced MSC solutions

6.19.1.1 Introduction

It is possible that a UE could move on to an MSC that is either not enhanced for ICS or does not have an associated L-CAAF-n. This can occur when a network has a partial roll-out of an L-CAAF-n or enhanced MSC solution, or when roaming outside of the HPLMN. Therefore, a default behaviour needs to be specified in order for the HPLMN to still provide some kind of service to the subscriber.

6.19.1.2 Possible Solutions

When there is no ICS functionality available at an attached MSC, the following fallback solutions are possible:

1. Fallback to redirection to IMS. That is, to redirect all originated calls to the IMS by using CAMEL. This of course will not provide for mid-call services in IMS, and also means that the subscriber will be present in CS, but unregistered in IMS. Terminating services will be in IMS, using redirection from the GMSC in the HPLMN.

2. Fallback to CS:

   2.1. MO calls handled in CS (i.e. MSC in VPLMN), MT calls handled in IMS. Hence originating or mid-call services will be in CS domain and terminating services in IMS.

   2.2. MO calls and MT calls handled in CS. For incoming IMS calls, the T-SDS (Terminating Service Domain Selection function) will need to be enhanced to be made dynamic as currently it is only statically configured.

3. Fall back to UE based ICS solution. This is not considered a valid alternative, since if the UE supports ICS, and home network supports ICCC, then the ICS UE will use ICCC, and the UE will not rely on ICS function in VMSC. If the ICCC is not available, the ICS UE has to disable its ICS functionality, thus falling back to UE based ICS solution is impossible in any case.

Option 2.2, above, will require fairly substantial work in order to provide a technical realisation: possibly too much work for the actual benefit it brings. Option 3 is not applicable.

Option 1 provides the most functionality out of all options, but of course relies on CAMEL support in the VPLMN.

Option 2.1 will have a fairly simple technical realisation and is probably the best option to use when there is no CAMEL in the VPLMN (a fairly rare case in most roaming agreements between most operators, but certainly not an impossibility). This option may also be used if the subscriber is using only services that are standardised already as a Supplementary Service, but the HPLMN was provisioning these in IMS as part of a network-based ICS architecture.

6.19.1.3 Conclusion

Options 1 and 2.1 should be defined in the network based ICS solution. It should be an operator option (e.g. by CAMEL provisioning) when option 1 should be used and when option 2.1 should be used in VPLMNs with CAMEL. In VPLMNs with no CAMEL, option 2.1 should always be used.

In the above cases, the HLR/HSS should download to the VMSC/VLR the service data that enables the execution of the required services in the CS domain.

Option 2.2 is too complex in relation to their actual use and benefits they will bring. Option 3 is not applicable.

6.20 ICS capability exchange between the ICCF and the ICS UE

6.20.1 ICS capability exchange between the ICCF and the ICS UE

The ICS UE establishes a dialogue with the ICCF for ICS capability exchange upon network attachment. This may be achieved by using a new SIP event package or USSD exchange between the ICS UE and the ICCF.

The following information is exchanged as part of this dialogue:

- UE's ICS capability.
- ICCC capability of the HPLMN and VPLMN, more specifically, type of ICCC expected for use with the current network attachment.

6.20.2 ICS Association between the ICCF and the IMSC

The ICS Association is established between the ICCF and an IMSC when a ICS user attaches to an IMSC.

The IMSC discovers the following as part of this exchange:

- Control domain for the served user; the control domain is IMS if the Location Update is performed for an ICS user and operator policy allows delivery of ICS to the ICS user via the serving CS network (e.g. service provider policy allows ICS only when the ICS user is not internationally roaming).
- UE's ICS capability; the IMSC provides the UE identity to request this capability.
- ICCC capability of the VPLMN, more specifically, type of ICCC expected for use with the current network attachment.

6.20.2.1 Alternatives for establishing ICS Association between ICCF and IMSC

The following options are possible for implementation of the ICS Association between the ICCF and the IMSC.

Option 1: Diameter exchange upon CS Attach

Figure 6.20.2.1-1. below illustrates use of Diameter to establish the ICS Association between the ICCF and the IMSC upon CS Attach.

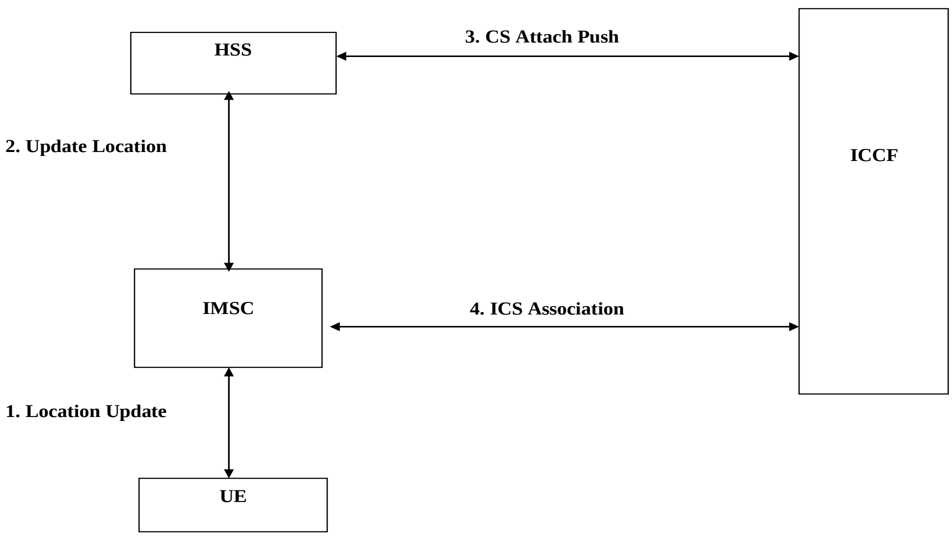


Figure 6.20.2.1-1: ICS Association using Diameter

The HSS maintains the ICS subscription information, as well as subscriber location to aid in determining the IMSC contact address as part of the subscriber data. Upon Location Update from an IMSC, this information is pushed down to the ICCF over the Sh or MAP interface. With the use of this information, the ICCF initiates a dialogue with the IMSC to establish the ICS Association. A new Diameter interface between the ICCF and the IMSC is used for establishment of the ICS Association (Step 4 in the figure).

After the ICS Association has been established for a ICS UE, the IMSC initiates a SIP REGISTER for the UE if it's a non ICS UE or a ICS UE using I1-cs.

Option 2: USSD exchange upon CS Attach

The IMSC uses UE initiated USSD procedure toward the ICCF upon CS Attach to establish the ICS Association. Figure 6.20.2.1-1 below illustrates use of USSD to establish the ICS Association between the ICCF and the IMSC upon CS Attach.

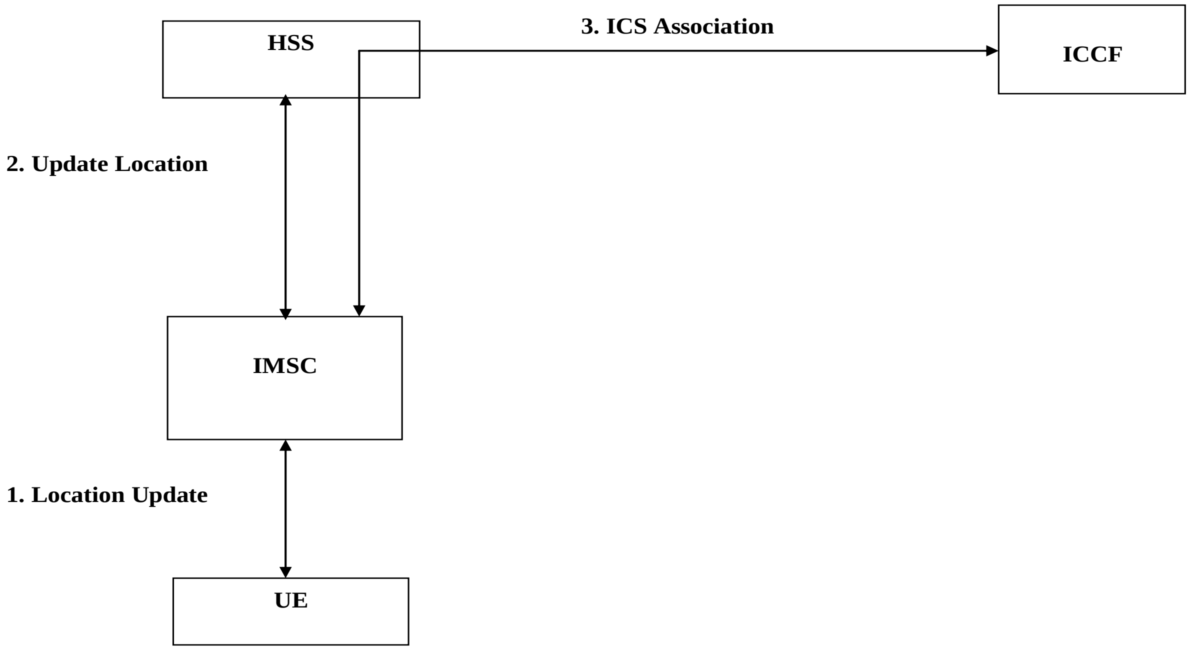


Figure 6.20.2.1-2: ICS Association using USSD

After the ICS Association has been established for a ICS UE, the IMSC initiates a SIP REGISTER for the UE if it's a non ICS UE or a ICS UE using I1-cs.

This option eliminates the need for MAP/Sh enhancements and/or a new Diameter interface between the ICCF and the IMSC.

Option 3: Use of SIP REGISTER upon CS Attach

Upon CS Attach, the IMSC establishes the ICS Association as part of the SIP REGISTER; SIP extensions may be required to exchange ICS information as part of IMS Registration via CS access.

This option eliminates the need for new interface or USSD but requires local VPLMN configuration of ICS subscription for ICS users as the IMSC shall only redirect ICS users to IMS.

7 Conclusion

7.1 Interim conclusion on registered services to ICS users when using I1-cs

With I1-cs, two solutions are described in the TR on how to enable registered services (services invoked as a result of registered iFC) to ICS users registered in IMS via CS access:

- I1-cs: registered user solution: SIP Register performed by the ICCF; and
- I1-cs: registered user solution: CS Registration status push from the HSS.

During the course of the study, it has been recognized that both methods require standardization effort.

Registered services are also available for the ICS user using I1-cs when the user is registered in IMS via an IP-CAN using the IMS SIP registration procedure defined in TS 23.228 [4]. When not registered in IMS via an IP-CAN, or in absence of I1-cs mechanisms for registration via CS access, the unregistered user solution described in the TR supports many of the requirements with use of standard IMS procedures.

Finally, the study has shown that both solutions have their advantages and disadvantages. However, it is believed that the benefits of the two registered user solutions are outweighed by the cost of supporting them and so they are not recommended for further specification. Thus no I1-cs mechanism for registration is to be specified.

7.2 Conclusion on ICS architecture solutions

Different approaches have been identified as possible solutions for enablement of ICS for ICS UE. The SIP Application Server approach with I1-ps and I1-cs (described in clauses 6.5 and 6.6) is recommended as the working assumption for ICS UE.

The use of Mw reference point for the ICCF (described in clause 6.7) is not recommended to be specified in normative specifications.

Different approaches have been identified as possible solutions for enablement of ICS for non ICS UEs. The approach of the enhanced MSC-Server (described in clauses 6.3.3 and 6.8a) is recommended as the baseline for non ICS UEs in the normative specification, along with the associated fallback mechanisms (described in clause 6.19).

The L-CAAF-n approach for non ICS UE (clause 6.3.1) is *not* recommended to be specified in normative specifications.

7.3 Conclusion on use of ICCC

Two different methods of call control have been documented for initiation of ICS UE sessions established using CS bearers:

- Method 1: Using ICCC (SIP with I1-ps or ICCP with I1-cs) in conjunction with TS 24.008 [7] signalling.
- Method 2: Using CAMEL based redirection to IMS in conjunction with TS 24.008 [7] signalling.

Both of these methods are recommended for standardization of ICS UE session originations and terminations when using CS bearers.

Operator policy may control the use of I1-cs or I1-ps.

ICCC has to be used for mid-call signalling and to setup subsequent sessions.

7.3.1 Conclusion on the use of I1-ps

7.3.1.1 Conclusion on the use of I1-ps to initiate the first ICS UE session

To always use SIP with I1-ps in conjunction with TS 24.008 [7] signalling

7.3.1.2 Conclusion on the use of I1-ps for Domain Transfer

I1-ps is always used for Domain Transfer from a VoIP capable access network to CS access network.

7.3.2 Conclusion on the use of I1-cs

7.3.2.1 Conclusion on the use of I1-cs to initiate the first ICS UE session

Operator policy may decide on one of the following two alternatives:

EITHER

- To always use ICCP with I1-cs in conjunction with TS 24.008 [7] signalling.

OR

- To use CAMEL signalling in conjunction with TS 24.008 [7] signalling for call origination, if no IMS parameters (such as SIP URI) need to be sent by the originating party.

- To use standard CS call termination if no IMS parameters (such as SIP URI) need to be sent to the terminating party.
- To use ICCP with I1-cs in conjunction with TS 24.008 [7] signalling if IMS parameters (such as SIP URI) need to be exchanged.

The Setup message used in CS origination procedures should contain the B-Party number whenever an E.164 number is dialled so that the MSC can analyse the B-party digits and so detect any undetected emergency calls.

7.3.2.2 Conclusion on the use of I1-cs for Domain Transfers

- Domain transfer may be performed without I1-cs when the ICS UE is involved in only one session and when a STI was not assigned to the session by the DTF (as specified in clause 6.6.2) and with no supplementary services (e.g. HOLD) invoked
- Domain transfer is performed with use of I1-cs when the ICS UE is involved in more than one session or in a session for which a STI was assigned by the DTF (as specified in clause 6.6.2) or when a supplementary service (e.g. HOLD) is invoked.

7.4 Conclusion on centralization of conditional call forwarding

Three different solutions have been proposed for the centralization of conditional call forwarding services in IMS when the call has been delivered using CS call control. These solutions are discussed in this TR, and a brief overview of them has been provided below:

Solution 1: Use default PSIs

In this solution, Conditional CF is configured in IMS with default service configuration in the CS domain. The Conditional CF services are configured in the HLR. The different forwarded number can be set in HLR to indicate the type of CF. The VMSC redirects the call to IMS for appropriate handling of the service in IMS.

Solution 2: Use MAP suppression-of-announcements

In this solution, Conditional CF is configured and provisioned in IMS domain exclusively, according to IMS Centralized Service principle. It provides means to suppress the announcement in CS domain at the MSC for CFNR/CFB. MGCF provides the appropriate responses to TAS on behalf of the ICS user in CS domain.

Solution 3: Use CAMEL O-CSI

In this solution, Conditional CF is configured in IMS with default service configuration in the CS domain so that the VMSC invokes the CF service in the CS domain upon detection of CF triggers. The O-CSI camel service is also assumed to be configured in HLR especially as one natural assumption is that ICS users are likely to have VCC capability. It is used to redirect session control to IMS for processing of CF.

These three solutions may be supported standalone or may be combined with each other if necessary.

It is recommended to document these solutions and if necessary, combinations of these solutions as implementation options in the technical specification for IMS Centralized Services.

7.5 Conclusion on ICS UE receiving home IMS services via an IMSC

When using an ICS UE, the ICCF in the home IMS network as defined in clause 6.2.2.2 of this document shall provide the ICS control function, with an IMSC providing access to the ICCF.

The conclusions of this TR that apply to ICS UE (clauses 7.2, 7.3 and 7.4) are applicable when using an IMSC as well as when using CS network not enhanced for non ICS UE.

The IMSC applies the same procedures for registration in IMS regardless of the ICS capability of the UE . Whether or not the IMSC applies the registration procedure for ICS user is based on configuration as described in 6.8a.3 Originated Service Domain Selection for enhanced MSC server.

7.6 Conclusion on handling of multiple media for IMS Centralized Services

Speech media of a session may be transmitted over CS access. Bi-directional video may be transmitted multiplexed with speech over CS access (BS30) only in case UTRAN is used as access network and based on operator policy and UE capabilities. Video and other media components of a session may also be transmitted via PS access.

When bi-directional video is transmitted multiplexed with speech over CS access, for the SIP Application Server approach with I1-ps and I1-cs, the interworking between H.245 and SIP/SDP shall be done by the MGCF and IM MGW as described in TS 29.163 [18]; for the enhanced MSC-Server approach, similarly, the enhanced MSC server and MGW shall provide the interworking between H.245 and SIP/SDP.

Service continuity for multimedia call is handled within the MMSC study.

7.7 Conclusion on use of ICCC for CS terminations

When using I1-ps, the solution that describes the use of CS-Origination procedures to set up the bearer control signalling session is recommended for standardization.

When using I1-cs or when not using ICCC, the solution that describes the use of standard CS-Termination procedures to set up the bearer control signalling session is recommended for standardization.

7.8 Conclusion on relation between ICS and Service Continuity

The conclusion regarding the relation between ICS and Service Continuity with regard to domain / session transfer is as follows:

- ICS TS shall include the following aspects from TR 23.892:
    - Functionality required for centralization of services in the solutions for ICS UE and non ICS UE (i.e. enhanced MSC Server solution).
    - Communication of parameters required to identify sessions for domain transfer / session transfer between the ICS UE and ICCF using either I1-ps or I1-cs (the parameter as such are transmitted to / from DTF and hence the detail of which parameter is transmitted needs to be documented in the Service Continuity TS).
    - Description of the internal ICCF capability to merge / split two SIP sessions in the case of I1-ps on its own, namely the SIP session over the session control signalling path via PS access and the SIP session for the bearer control signalling path
- Service Continuity TS shall include the following aspects from TR 23.892:
    - The functionality required for IMS Service Continuity requests from / to CS access in the UE and the AS (e.g. session transfer function).
        - Transfer of one or more bi-directional speech sessions in order to enable domain transfer while Call Hold/Resume, Conference or Explicit Communication Transfer is active.

        NOTE 1: Session transfer between different accesses and partial session transfer is discussed in TR 23.893 [19] and is out of scope of this ICS work.

        NOTE 2: Session split/merge function needed for the user when media is carried simultaneously over two accesses is discussed in TR 23.893 [19] and is out of scope of this ICS work.
    - The IMS Session Continuity procedures (e.g. anchoring of originating and terminating calls, termination and domain transfer).
    - Definition of session transfer related parameters like VDI/VDN, STI, ST info.
    - How to assign parameters needed to identify sessions and media for session transfer.

- It is recommended to not describe how the parameter is being transmitted between the UE and the network when using a CS access for media transmission (i.e. when to use I1-cs and when to use I1-ps) and to leave that detail for the ICS TS.

NOTE 3: Finalization of the ICS TS is not dependent on the Service Continuity TS, even though there will be cross-references between the two TS.

7.9 Conclusion on the relationship between ICS and MMSC

Both ICS and MMSC specify functions which are provided by a SIP application server. These functions may be collocated as optional functions in a single SIP application server. The interfaces and interactions between them are not specified in this release.

7.10 Conclusion on T-ADS

The T-ADS has two functions, the selection function and the execution function. The selection function needs to be able to select for a terminating session amongst the following:

- Registration in IMS by the ICS or non-ICS UE for speech or speech and video via PS for media via PS ("PS contact")
- Registration in IMS by the ICS UE via PS for media via CS ("I1-ps contact")
- Registration in IMS by the ICS UE via PS for non-speech media and UE is reachable in CS (IMS registered for non speech only, hence speech services and or speech/video use I1-cs; "non-speech contact")
- Registration in IMS by the IMSC for media via CS ("IMSC contact")
- No registration in IMS but ICS or non-ICS UE reachable in CS ("unregistered", for I1-cs and in fallback cases for IMSC)

If either only "PS contacts" or only "IMSC contact" for one UE are possible in the S-CSCF, then standard contact selection in the S-CSCF is sufficient and T-ADS is not needed.

The selection and execution function of the T-ADS are provided by the same AS.

The execution function of the T-ADS is in the last IMS application server in the terminating iFC.

For a network which implements the ICCF, the ICCF is the last AS.

For a network requiring T-ADS but which does not implement the ICCF, the last AS is not specified (one candidate is MMSC AS, as defined in TR 23.893 [19]).

NOTE: The AS which implements the ICCF may be the same AS which implements the MMSC.

T-ADS should be described in the normative specification in TS 23.292 [21].

Annex A: ICS Service examples with I1-ps

A.1 Customized Alerting Tone

Customized alerting tone (also known as customized ring tone) is a service where the callee is able to customize the alerting tone (typically a music song) that is played to the caller while the callee is alerted. This service can be and is offered in today's CS networks by routing the call via a special announcement server, that connects the caller to the announcement while the callee is alerted. IMS can extend this service so that the callee sends the music clip (either the music clip itself or a reference to the clip is sent in a SIP method) to the caller, and caller's device plays the clip locally to the caller. This service is currently under discussion also in TISPAN NGN R2, and thus a possible candidate for 3GPP MMTel in Rel-8.

The next example illustrates how the service is currently offered in the IMS using mechanisms defined in RFC 3261, when both A and B party use IP-CAN to access IMS services.

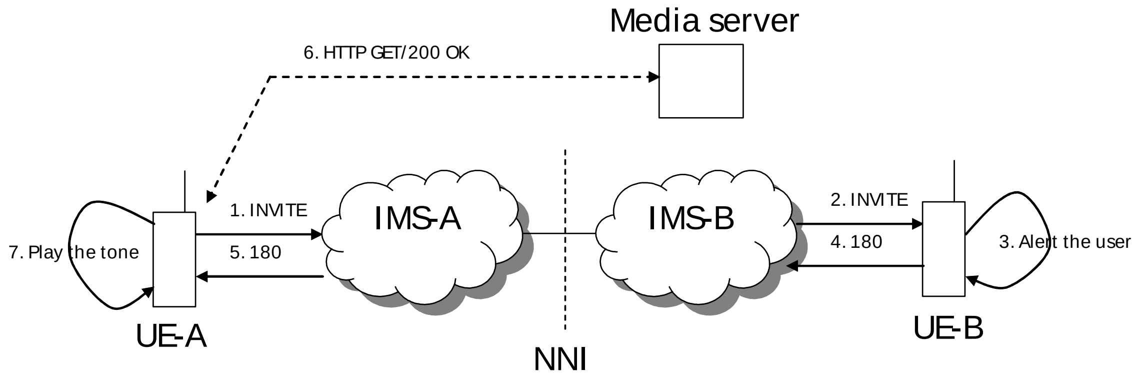


Figure A.1-1

1. The caller calls to callee and sends INVITE towards UE-B
2. INVITE is routed to UE-B
3. UE-B alerts the user.
4. Callee has pre-configured the UE-B to send a customized ringing tone towards the caller. The ringing tone may be different depending on e.g. the caller's ID. UE-B sends 180 Ringing response towards UE-A. The response contains an Alert-Info header, which carries a reference to the ringing tone, the actual ringing tone is either pre-configured in the Media Server, or carried in the body of the 180 Ringing.
5. UE-A receives the 180 Ringing.
6. Optionally UE-A retrieves the ringing tone from the Media Server if reference is used.
7. UE-A plays the ring tone to the caller.

In the next figure we show how the same service can be offered when UE-A is in ICS system using CS access.

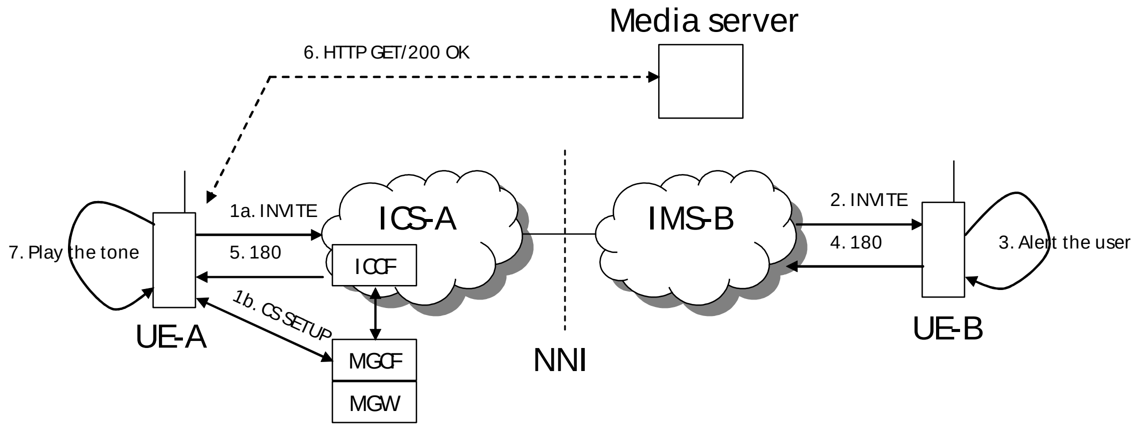


Figure A.1-2

1a The caller calls to callee and sends INVITE towards UE-B

1b The caller initiates a CS call towards the MGCF/MGW.

Rest of the flow is identical with the previous flow where the caller was accessing the IMS services via IP-CAN:

2. INVITE is routed to UE-B
3. UE-B alerts the user.
4. Callee has pre-configured the UE-B to send a customized ringing tone towards the caller. The ringing tone may be different depending on e.g. the caller's ID. UE-B sends 180 Ringing response towards UE-A. The response contains an Alert-Info header, which carries a reference to the ringing tone, the actual ringing tone is either pre-configured in the Media Server, or carried in the body of the 180 Ringing.
5. UE-A receives the 180 Ringing.
6. Optionally UE-A retrieves the ringing tone from the Media Server if reference is used.
7. UE-A plays the ring tone to the caller.

In similar manner, also the UE-B may use CS access via ICS to access the IMS services. The ringing tone in this example can be a short audio clip or e.g. a midi file. Also other media can be offered in similar manner, e.g. a video ringing tone, as long as the size of the media file is feasible to be transferred during the call setup phase.

As a conclusion, this service can be offered exactly in the same manner in ICS than in pure IMS. There is no impact to the SIP protocol or IMS procedures regarding this service due to introduction of ICS.

A.2 Instant Picture Presentation

In this example, the caller (UE-A) wants to send a "near real time" picture as part of session setup to UE-B (e.g. want to show the callee her new hair style). Basically, she just took a picture of herself and then places the call to the callee. UE-A encodes the picture into a feasible size to be included in the SIP INVITE. Image can be carried in INVITE using the Call-Info header with purpose=icon, the image can either be referenced in the media server, or carried as such in the body of the INVITE. Using reference in this example does not make too much sense because the caller will have to place it in a server first prior to the call; thus, delaying the whole call setup routine to callee.

In the next figure we show how the service can be offered when UE-A is in ICS system using CS access.

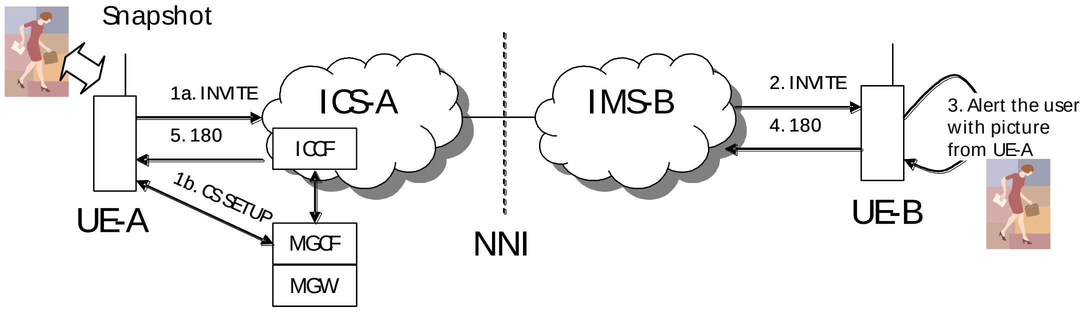


Figure A.2-1

1a The caller calls to callee and sends INVITE towards UE-B along with the snapshot.

1b The caller initiates a CS call towards the MGCF/MGW.

2. INVITE is routed to UE-B.

3. UE-B alerts the user with the snapshot of caller

4. UE-B sends 180 Ringing response towards UE-A.

5. UE-A receives the 180 Ringing.

In similar manner, also the UE-B may use CS access via ICS to access the IMS services.

As a conclusion, this service can be offered exactly in the same manner in ICS than in pure IMS. There is no impact to the SIP protocol or IMS procedures regarding this service due to introduction of ICS.

A.3 Call Reject with customized reason

This service is similar to the customized alerting tone, but here the additional multimedia content is rendered to caller related to the call release in the setup phase. In the next figure we show how the service can be offered exactly as in SIP/IMS, even while the callee (UE-B) is using ICS and CS access to access the IMS services.

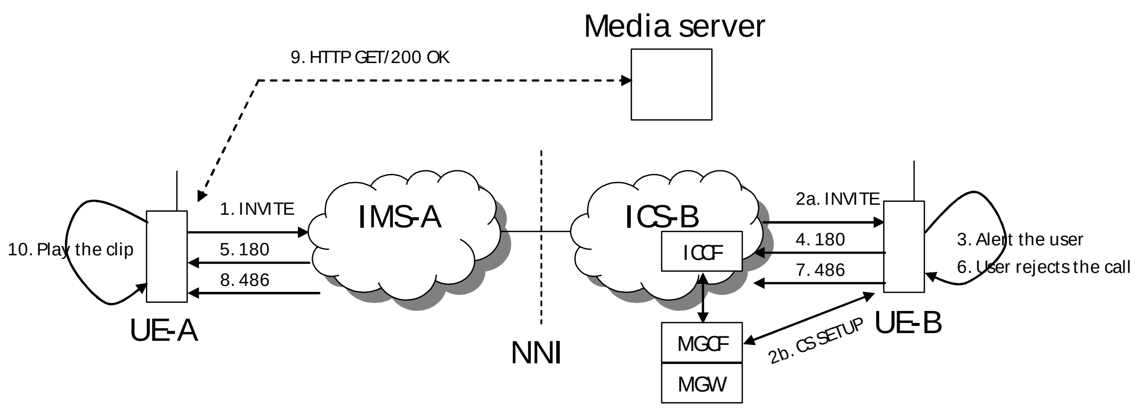


Figure A.3-1

1. The caller calls to callee and sends INVITE towards UE-B.

2a The INVITE is routed to the UE-B.

2b ICCF initiates a CS SETUP via MGCF towards the UE-B.

3. UE-B alerts the callee.

4. UE-B sends 180 Ringing towards the UE-A.

5. UE-A receives the 180 Ringing.

6. Callee decides to reject the call.

7. Callee has pre-configured to the UE-B a few predefined error responses, like "I will call you later", "On vacation", "In a meeting, is it urgent?", etc. Callee selects one of these responses and the UE-B attaches the response to the 486 Busy Here. As described in RFC 3261, the additional error response can be carried in Error-Info header of 486, either as a reference, or as a direct content in the body of the response. The additional error information can also be e.g. a short audio clip. UE-B sends 486 Busy Here response towards the UE-A.

8. UE-A receives the 486 Busy Here.

9. Optionally UE-A retrieves the additional error information from the Media Server if reference is used.

10. UE-A plays the error information to the caller.

In similar manner, also the UE-A may use CS access via ICS to access the IMS services. As a conclusion, this service can be offered exactly in the same manner in ICS than in pure IMS. There is no impact to the SIP protocol or IMS procedures regarding this service due to introduction of ICS.

A.4 Caller's Location

In this service example the caller's UE fetches its location and offers it in the call setup towards the callee. Callee can present the caller's location in the screen of the UE-B e.g. using a street map, which is either stored locally in the UE-B, or retrieved from a map database in the network.

In the next figure we show how the service can be offered exactly as in SIP/IMS, even while the callee (UE-B) is using ICS and CS access to access the IMS services.

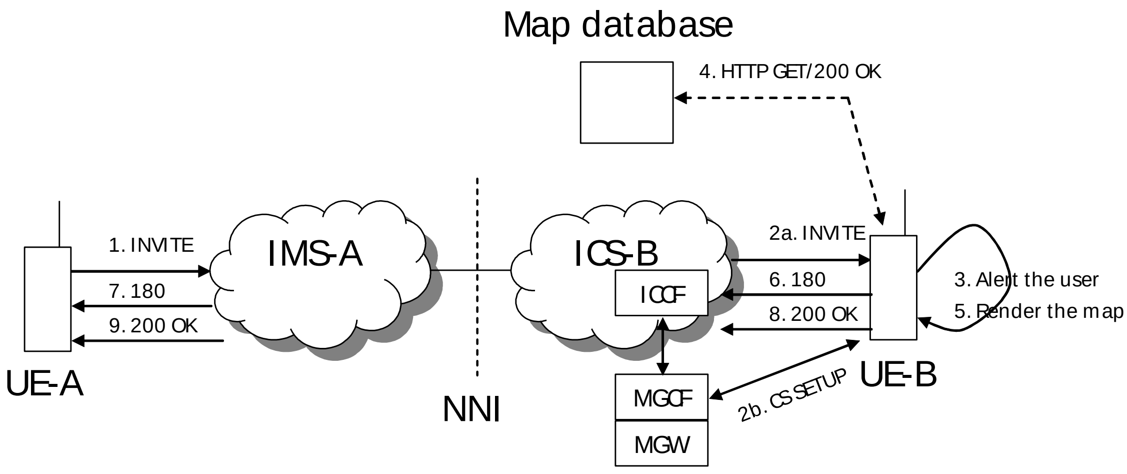


Figure A.4-1

1. The caller calls to callee and sends an INVITE towards UE-B. The INVITE includes a Geolocation header as described in SIP Location Conveyance I-D (draft-ietf-sip-location-conveyance-07) and the location of the user carried in PIDF-LO (RFC 4119) in the body of the INVITE. The PIDF-LO element carries the location of the UE-A. The UE-A may have retrieved its location e.g. by using its integrated GPS receiver, as a civic address from DHCP server as described in RFC 4676, or by other means. Also the IMS-A may add the location of the UE-A on behalf of the UE, if the network supports location retrieval function and thus is able to locate the UE-A.

2.a The INVITE is routed to the UE-B.

2.b ICCF initiates a CS SETUP via MGCF towards the UE-B.

3. UE-B alerts the callee.

4. Optionally the UE-B uses the location of the caller to retrieve e.g. a street map from a network map database, if the UE-B does not have integrated map database.

5. UE-B renders the map data to the screen, along with the current location of the caller.

6. UE-B sends 180 Ringing towards the UE-A (can be sent in parallel with steps 3-5).

7. UE-A receives the 180 Ringing.

8. Callee decides to answer the call. UE-B sends 200 OK towards UE-A.

9. UE-B receives the 200 OK for INVITE.

In similar manner, also the UE-A may use CS access via ICS to access the IMS services. Also UE-B may provide his/her location to UE-A using similar procedure and PIDF-LO element e.g. in the body of 180 Ringing response. As a conclusion, this service can be offered exactly in the same manner in ICS than in pure IMS. There is no impact to the SIP protocol or IMS procedures regarding this service due to introduction of ICS.

A.5 Ad-hoc conferencing

This example shows how the multimedia conferencing service can be offered in ICS. IMS Multimedia conferencing (TS 24.147) is part of 3GPP R7 Multimedia Telephony service (TS 24.173). It defines SIP procedures to create a multiparty conference session, add and remove users to/from the session, add/remove media to/from the multiparty session, receive notifications on other users in the conference session, etc. In this example we show how all of these standard procedures can be offered also via the CS access, while the user is accessing the IMS services via ICS. In this example user A initiates the conference session and is using CS access, user B is invited participant using CS access, and user C is invited participant using IP-CAN access.

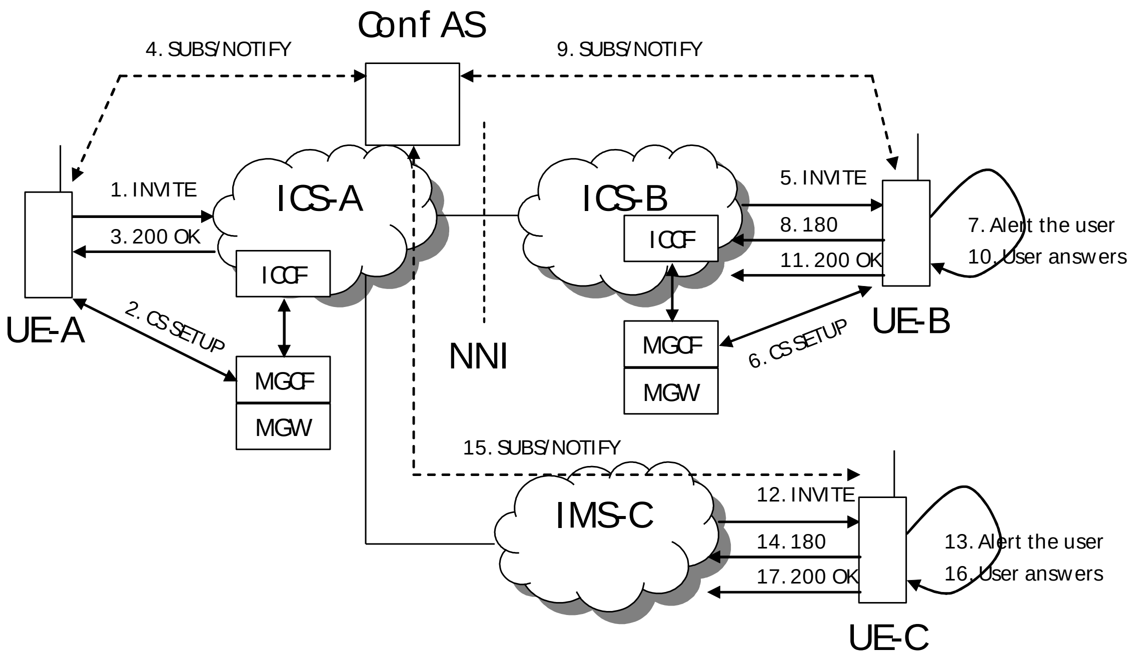


Figure A.5-1

1. The user A wants to create an ad-hoc voice conference with users B and C. User A selects the B and C from his/her local phonebook and selects "create a conference call" in the UI. UE-A sends an INVITE towards the conference server, the identities of B and C are carried in the body of the INVITE, as described in draft-ietf-sip-uri-list-conferencing-01 (part of TS 24.173). User A may also add a subject to the conference session, e.g. "Urgent, meeting today is moved up 2 hrs which is now!".

2. UE-A initiates a CS SETUP via MGCF towards the ICCF.

3. Conference AS (may be part of TAS) receives the INVITE. The AS creates the conference session and responds with 200 OK. UE-A receives the 200 OK for INVITE. 200 OK carries in the Contact header the SIP URI of the conference session and the isfocus parameter.

4. UE-A subscribes for the conference state event package as described in RFC 4575 (part of TS 24.173). The notification indicates to the UE-A whenever the invited user is added to the conference, whether some invited user fails to accept the conference invitation (e.g. is busy), and later on during the conference session, whenever another user is added or old user leaves the session. The event package may also carry information about the supported medias in the conference session, it may indicate that the conference AS supports e.g. voice, video, and messaging. UE-A may update the information on the UI regarding the conference state, participants, etc, whenever it gets the notification from the conference AS.

NOTE: SUBSCRIBE/NOTIFY is not routed through the ICCF.

5. Conference AS invites the participants B and C to the session. AS sends INVITE towards user B. UE-B receives an INVITE which carries in the Contact header the SIP URI of the conference session and the isfocus parameter.

6. ICCF initiates a CS SETUP via MGCF towards the UE-B.

7. UE-B alerts the user B. UE-B may indicate to the user B that this is a multiparty session.

8. UE-B sends 180 Ringing towards the conference AS.

9. UE-B subscribes for the conference state event package. The notification indicates to the UE-B who are the other invitees in the conference session, whether some invited user fails to accept the conference invitation (e.g. is busy), and later on during the conference session, whenever another user is added or old user leaves the session. The event package carries also the subject of the session, "Urgent, meeting today is moved up 2 hrs which is now!". The event package may also carry information about the supported medias in the conference session, it may indicate that the conference AS supports e.g. voice, video, and messaging. UE-B may update the information on the UI regarding the conference state, participants, etc, whenever it gets the notification from the conference AS.

10. User B answers the call.

11. UE-B sends 200 OK for INVITE towards the conference AS. Conference AS updates the status of the user B in the conference event package and delivers it to the other participants via NOTIFY.

12-17. Conference AS invites the user C to the conference. This can be performed in parallel with steps 5-11.

In the next example flow the user A adds user D to the conference session. User D uses IP-CAN to access IMS services.

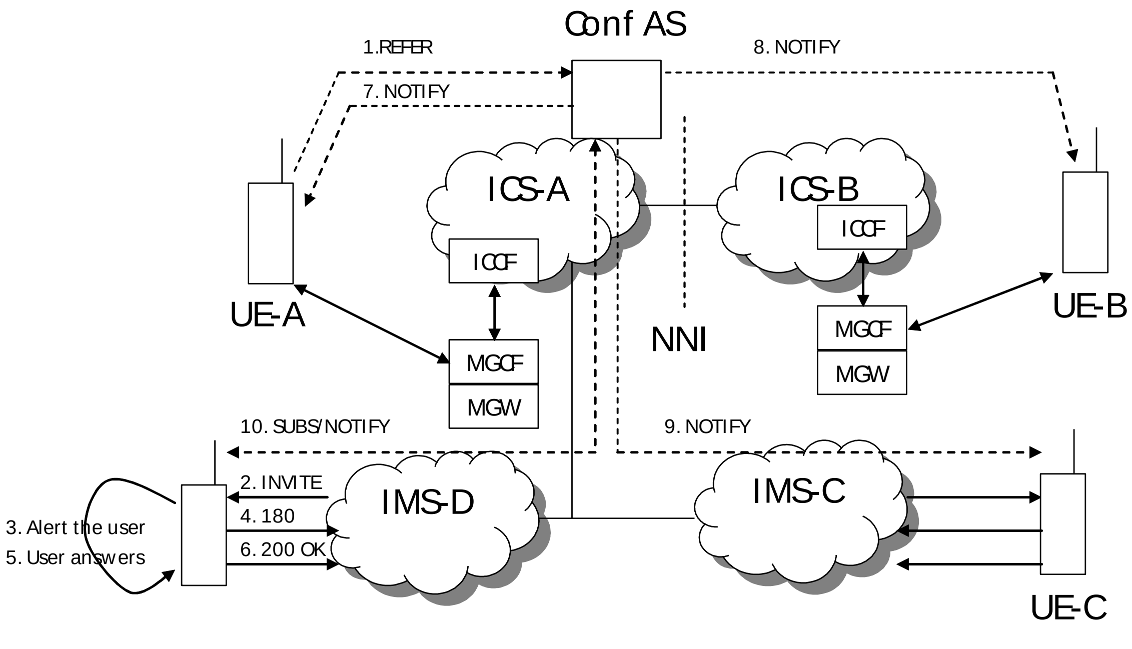


Figure A.5-2

1. The user A wants to add user D to the conference session. User A select the user D from his/her phonebook and selects "add to the conference". UE-D sends REFER towards the conference AS. The REFER contains the identity of the user D in the Refer-To header. AS invite the user D to the session, the AS sends INVITE towards the UE-D.

NOTE: REFER is not routed through the ICCF.

2. UE-D receives the INVITE. It carries in the Contact header the SIP URI of the conference session and the isfocus parameter.
3. UE-D alerts the user D.
4. UE-D sends 180 Ringing towards the conference AS. UE-D subscribes the conference state event package and gets notified on the other participants in the conference session (not shown in the figure). The event package carries also the subject of the session, "Urgent, meeting today is moved up 2 hrs which is now!". The event package may also carry information about the supported medias in the conference session, it may indicate that the conference AS supports e.g. voice, video, and messaging.
5. User D answers the call.
6. UE-D sends 200 OK for INVITE towards the conference AS.
7. Conference AS sends NOTIFY to UE-A to indicate that user D was successfully added to the conference session.
8. Conference AS sends NOTIFY to UE-B to indicate that user D was successfully added to the conference session.
9. Conference AS sends NOTIFY to UE-C to indicate that user D was successfully added to the conference session.
10. Conference AS sends NOTIFY to UE-D to indicate the other participants of the conference session.

In the next example flow we show how user A is able to add another media (e.g. messaging) to the conference session.

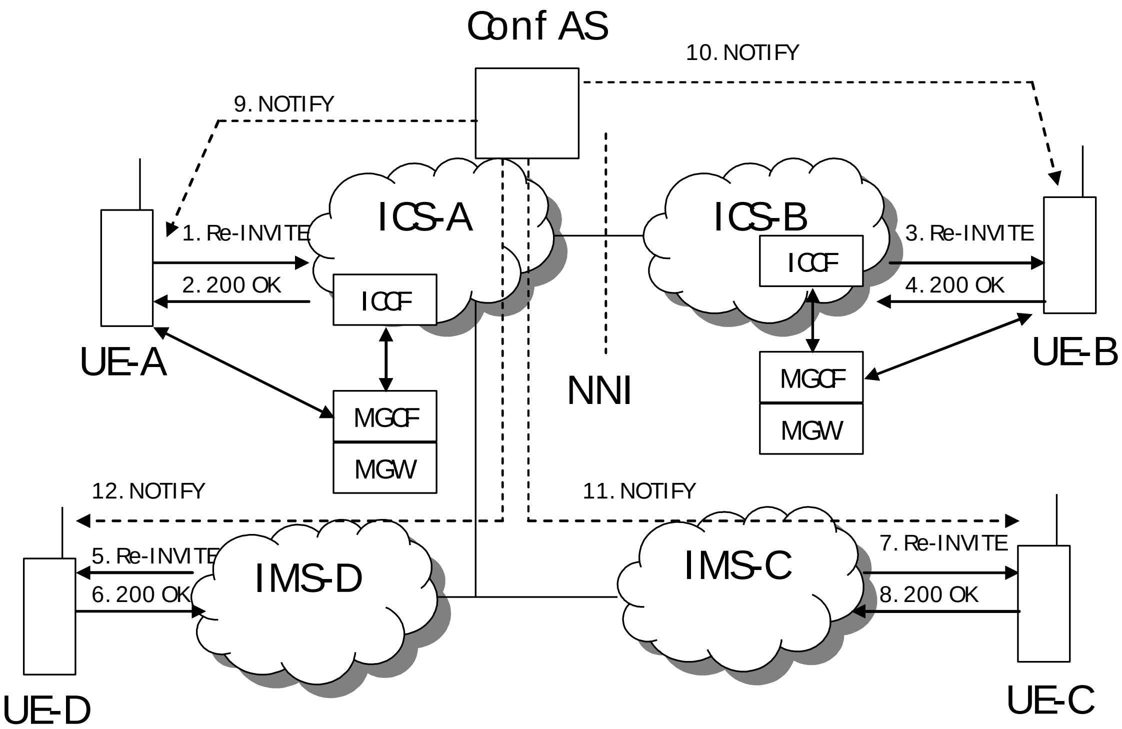


Figure A.5-3

1. The user A noticed earlier via the conference state event package that the conference AS supports messaging media. User A wants to add a messaging media to the conference session in order to share a few important notes he/she has made during the conference call. UE-A sends a Re-INVITE with SDP offer indicating an MSRP media.

2. Conference AS accepts the SDP offer by sending 200 OK for INVITE with SDP answer.

3. Conference AS sends a Re-INVITE indicating an MSRP media towards UE-B.

4. UE-B accepts the MSRP media.

5. Conference AS sends a Re-INVITE indicating an MSRP media towards UE-D.

6. UE-D accepts the MSRP media.

7. Conference AS sends a Re-INVITE indicating an MSRP media towards UE-C.

8. UE-C accepts the MSRP media.

9. Conference AS sends a NOTIFY to UE-A to indicate that the user's B, C and D have accepted the MSRP media.

10. Conference AS sends a NOTIFY to UE-B to indicate that the user's A, C and D have accepted the MSRP media.

11. Conference AS sends a NOTIFY to UE-C to indicate that the user's A, B and D have accepted the MSRP media.

12. Conference AS sends a NOTIFY to UE-D to indicate that the user's A, B and C have accepted the MSRP media. From now on, all participants are able to send and receive MSRP messages in the conference session.

As a conclusion, this service can be offered exactly in the same manner in ICS than in pure IMS. There is no impact to the SIP protocol or IMS procedures regarding this service due to introduction of ICS.

Annex B: Information flow for mid call service using MRFC/MRFP

The following figures B.1-1 and B.2-1 show two examples for dynamic link-in of MRFC/MRFP invoked by the ICCF. Figure B.3-1 shows an example for MRFC/MRFP invoked by the TAS.

B.1 Information flow for using MFRP by ICCF for held call only

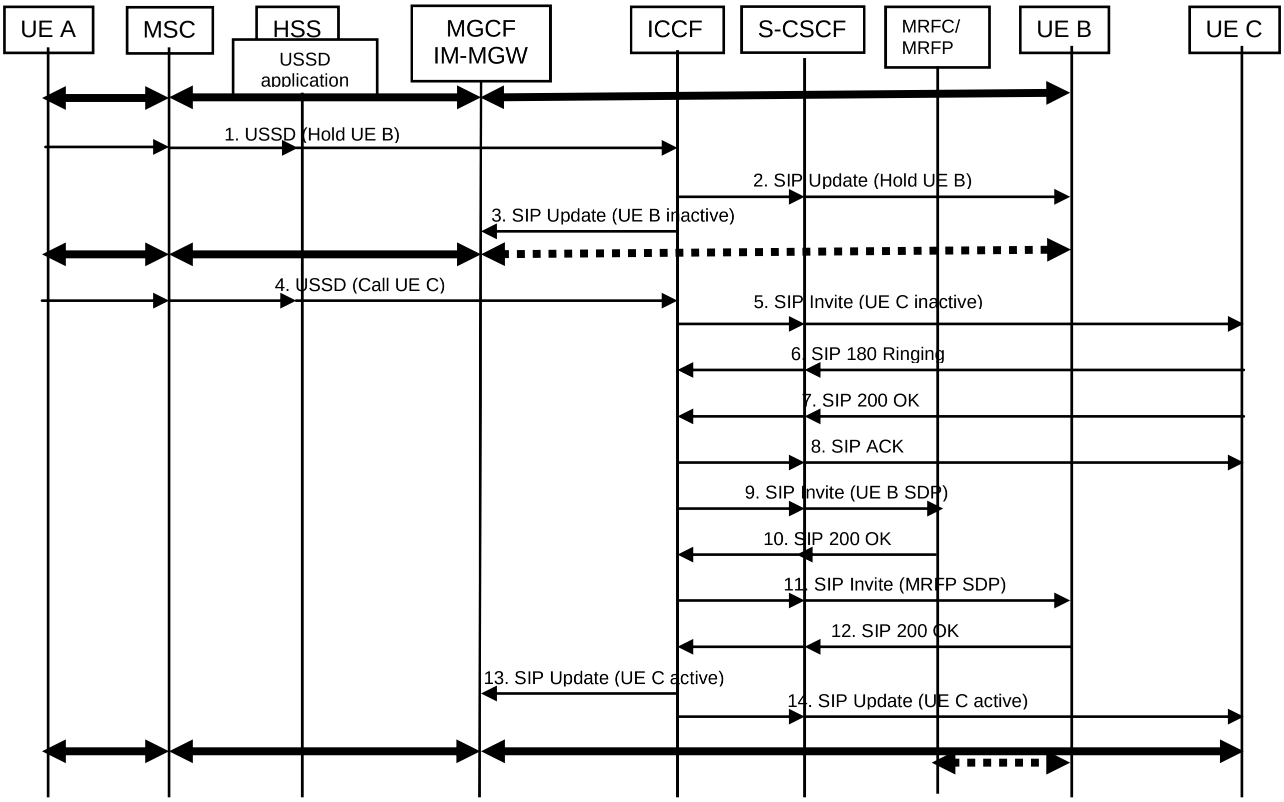


Figure B.1-1: Media handling in Hold and Originating Session Setup

The precondition of the procedure is that there is an active session before UE-A and UE-B.

1. UE-A sends an ICS Hold Request inside a USSD dialog to HSS via MSC, and HSS routes the request to the addressed ICCF.

2. ICCF sends a SIP Update via S-CSCF to hold UE-B.

3. ICCF sends a SIP Update to MGCF to set the media into inactive. Now only RTCP is sent to UE-B from IM-MGW.

4. UE-A now sends an ICS Request inside a USSD dialog to HSS via MSC, in order to initiates a call to UE C, and HSS routes the request to the addressed ICCF.

5. ICCF sends a SIP Invite via S-CSCF to UE C, with the media set to inactive.

6. UE C sends a SIP 180 Ringing via S-CSCF to the ICCF.

7. UE C sends a SIP 200 OK via S-CSCF to the ICCF.

8. ICCF sends a SIP ACK via S-CSCF to UE C. Now the session is established but the media transfer is not allowed.

9. ICCF then sends a SIP Invite via S-CSCF to MRFC/MRFP to prepare a dialog for UE-B.

10. MGCF/MRFP sends a SIP 200 OK via S-CSCF to the ICCF.

11. ICCF sends a SIP Re-Invite via S-CSCF to UE-B to update the media path.

12. UE-B sends a SIP 200 OK via S-CSCF to the ICCF.

13. ICCF sends a SIP Update to MGCF to activate and redirect the media to UE C.

14. ICCF sends a SIP Update to UE C to activate the media. Now the media path is established e-2-e between UE-A and UE C, and RTCP is sent from MRFP to UE-B.

B.2 Information flow for using MFRP for both active and held call

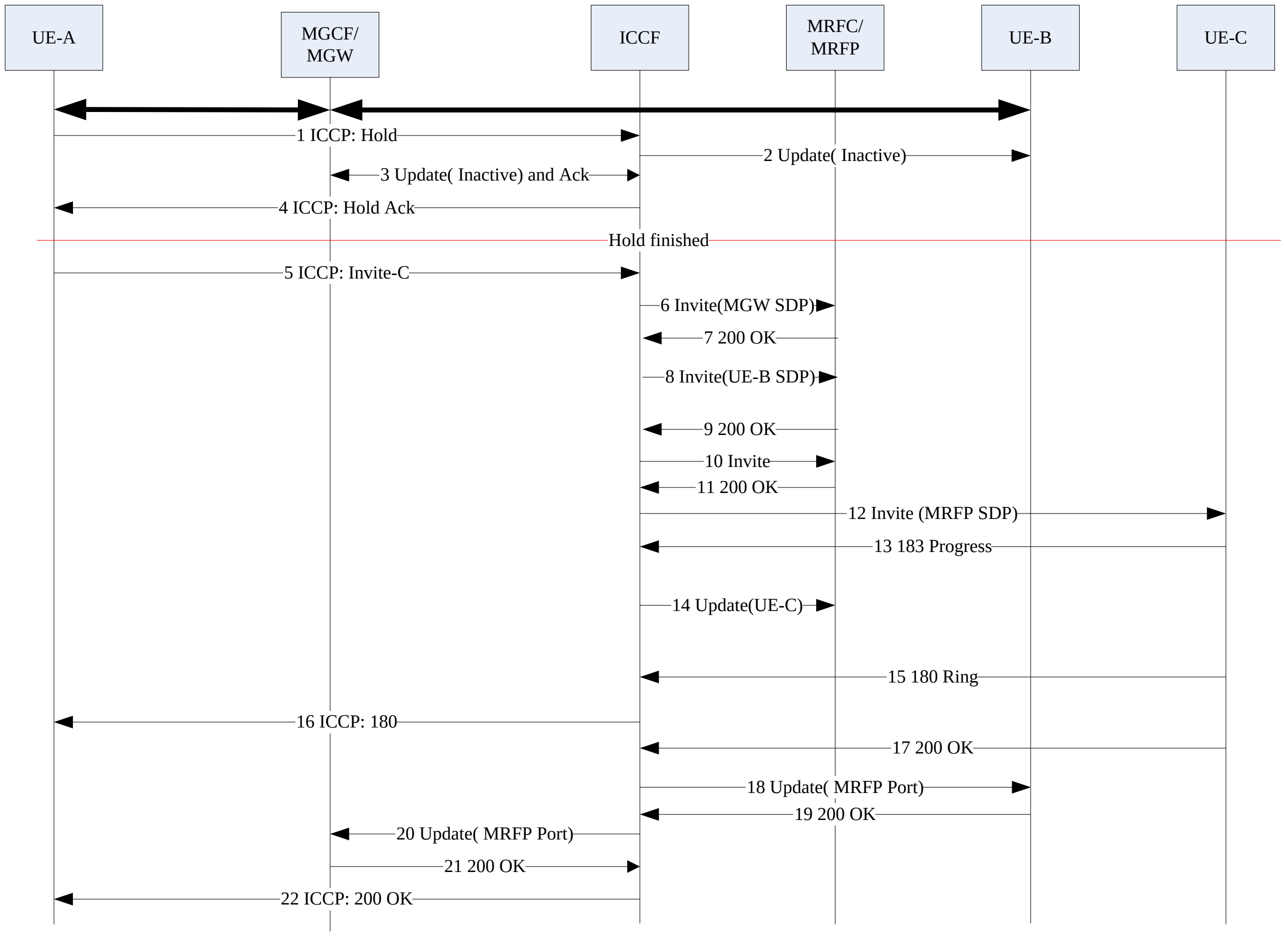


Figure B.2-1: Information flow for mid call using MRFC/MRFP

1) UE-A initiates Hold request towards ICCF using ICCP protocol.

2) ICCF sends SIP Update request to UE-B to hold UE-B's media.

3) ICCF sends SIP Update request to MGCF to hold MGW's media.

4) ICCF sends Hold ack response towards UE-A.

NOTE: The session between UE-A and UE-B is successfully put on hold.

5) UE-A sends Invite request towards ICCF, the called party is UE-C.

6) ICCF sends SIP Invite request to MRFC including media information of MGW.

7) MRFC responses SIP 200 OK including media information of MRFP.

8) ICCF sends SIP Invite request towards MRFC including UE-B's media information.

9) MRFC responses SIP 200 OK including media information of MRFP.

10) ICCF sends SIP Invite to MRFC with null media information.

11) MRFC responses SIP 200 OK including media information of MRFP.

12) ICCF sends SIP Invite towards UE-C including MRFP's media information return in step 11.

13) UE-C responses SIP 183 response including UE-C's media information.

14) ICCF sends SIP Update request to MRFC to change the media information related to UE-C.

15) UE-C sends SIP 180 response.

16) ICCF sends 180 response using ICCP towards UE-A.

17) UE-C sends SIP 200 OK response.

18) ICCF sends SIP Update request to UE-B to change UE-B's media path to MRFP.

19) ICCF receives SIP 200 OK response.

20) ICCF sends SIP update request towards MGCF to active bearer of UE-A.

21) ICCF receives SIP 200 OK from MGCF.

22) ICCF sends 200 Ok response to UE-A using ICCP.

The consequent call retrieve (retrieve call with UE-B, hold call with UE-C) flow is not shown in this figure for reasons of simplification.

B.3 information flow for using MFRP by TAS for held call

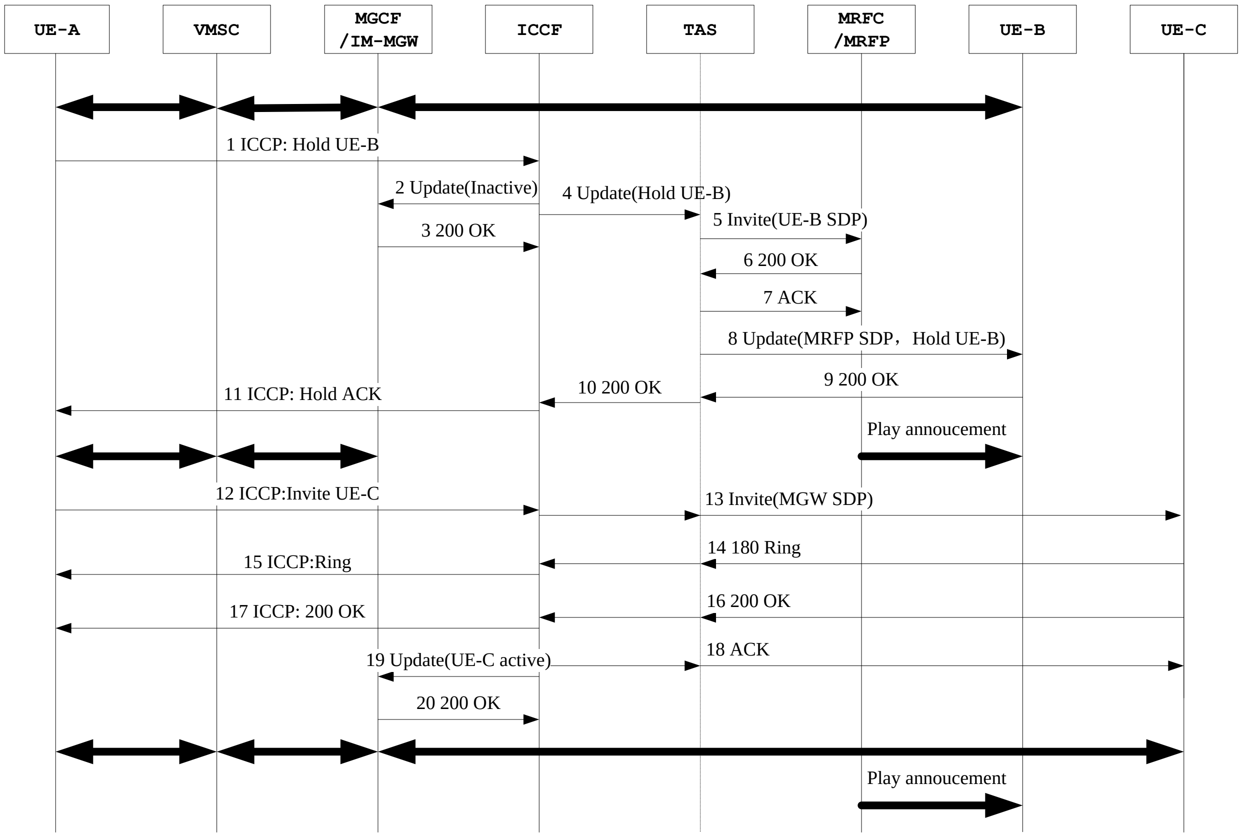


Figure B.3-1: Information flow for mid call using MRFC/MRFP by TAS

The precondition of the procedure is that there is an active session between UE-A and UE-B. The TAS is remained in the RUA leg.

1) UE-A initiates a Hold request towards ICCF using ICCP protocol.
2) ICCF sends a SIP Update request to MGCF to set MGW's media inactive.
3) MGCF responses a SIP 200 OK back to ICCF
4) ICCF sends a SIP Update request to TAS via S-CSCF to hold UE-B's media.
5) TAS sends a SIP Invite request including media information of UE-B to MRFC to play announcement to UE-B.
6) MRFC responses a SIP 200 OK including media information of MRFP.
7) TAS sends a SIP ACK to MRFC.
8) TAS sends a SIP Update request including media information of MRFP to UE-B via S-CSCF to change UE-B's media path to MRFP.
9) UE-B responses a SIP 200 OK to TAS via S-CSCF. Now MRFP plays announcement to UE-B.
10) TAS responses a SIP 200 OK to ICCF via S-CSCF.
11) ICCF sends a Hold ACK response towards UE-A.
12) UE-A sends an Invite request towards ICCF using ICCP protocol, the called party is UE C.
13) ICCF sends a SIP Invite towards UE C including MGW's media information via S-CSCF and TAS.
14) UE C sends a SIP 180 response back to ICCF via S-CSCF and TAS.

15) ICCF sends a 180 response using ICCP towards UE-A.

16) UE C sends a SIP 200 OK response back to ICCF via S-CSCF and TAS.

17) ICCF sends a 200 Ok response to UE-A using ICCP.

18) ICCF sends a SIP ACK to UE C via S-CSCF and TAS.

19) ICCF sends a SIP Update request towards MGCF to active and redirects the media to UE-C.

20) MGCF responses a SIP 200 OK to ICCF.

Now the media path is established between UE-A and UE C, and the announcement is played to UE-B.

NOTE: For the clarity of the call flow, the S-CSCF is not shown in the figure. The messages between ICCF and TAS should pass through the S-CSCF, as well as the messages between TAS and MRFC, UE-B (UE C) and TAS etc.

Annex C:
Comparative analysis of solutions for non ICS UE support

A CS network enhancement approach using L-CAAF-n in the visited CS network and an approach using CAMEL redirection have been documented as two different alternatives for ICS support of non ICS UEs. Table Y.1 below provides a comparative analysis for the two alternatives.

Table C-1: L-CAAF-n vs. CAMEL solution for non ICS UE support

Analysis criteria	L-CAAF-n approach	CAMEL approach
Call Setup Capabilities		
Calls to/from users addressable by E.164 numbers.	Supported.	Supported.
Calls to/from users addressable only by SIP-URIs	Call originations from ICS user to a SIP URI is not possible due to UE limitations; CLI not presented for call terminations to ICS user when SIP URI is used as caller's identity.	Call originations from ICS user to a SIP URI not possible; CLI not presented for call terminations to ICS user when SIP URI is used as caller's identity.
Supplementary Service Capabilities		
Line ID services	Service controlled exclusively in IMS. Presentation of SIP URI, and richer modes such as picture/business card caller id as CLI not possible.	Require service configuration in the HLR. Presentation of SIP URI, and richer modes such as picture/business card caller id as CLI not possible.
Communication Diversion services	Service controlled exclusively in IMS.	CFU and CFNL provided exclusively in IMS. IMS may not be able to distinguish between CFB, CFNRy and CFNRc, so further study is required.
Call Hold, Conf, CW, ECT, Call Deflection	Service controlled exclusively in IMS.	Service controlled in CS domain.
Supplementary Service Data Management		
Service data configuration	Service data configuration in IMS. Details of this is FFS.	Service data configuration in IMS for services controlled in IMS and in CS domain for services controlled in CS domain. Details of this is FFS.
User configuration of service data	CISS interworked with Ut in the L-CAAF-n. Details of this is FFS.	Requires a solution for interworking CISS with IMS. Details of this is FFS.
Service data synchronization	Requires service data synchronization for roaming into CS networks not upgraded with L-CAAF-n. Details of this is FFS.	Requires a solution for static and dynamic service data synchronization for services which are configured in IMS and CS domain such as the Line ID services. Details of this is FFS.
VCC Capabilities [only applicable to VCC capable UE]		
Basic Service Continuity	Supported.	Supported.
Service Continuity with non mid call services	Supported.	Supported.
Service Continuity with mid call services	Not possible due to UE limitations.	Not possible due to UE limitations.
Infrastructure Impact		
IMS Core Network impact	Requires AS and HSS enhancements for service data synchronization and user configuration of service data for support of roaming to networks not upgraded with L-CAAF-n	Requires AS and HSS enhancements for service data synchronization and user configuration of service data.
CS Core Network impact	Requires CS core network enhancements for L-CAAF-n. Requires HLR enhancements for service data synchronization, and user configuration of service data for support of roaming to networks not upgraded with L-CAAF-n.	Requires HLR enhancements for service data synchronization, and user configuration of service data. Requires an interface between the gsmSCF and the ICCF for communication of call related data such as Original Called Party Number for redirection of CS calls to IMS.
Other criteria		
Support of Emergency calls	IMS Control of Emergency calls established via PS or CS access may be possible but solutions for IMS control of Emergency calls are FFS. Support of service continuity for Emergency calls not possible due to UE limitations.	IMS Control of Emergency calls established via CS access not possible. Support of service continuity for Emergency calls not possible due to UE limitations.

Annex D: MGW Realization for the Enhanced MSC Server approach

Option 1: New MGW and reference point

In this option, a new logical MGW function would be defined to directly interwork CS access interfaces (A and IuCS) with the Mb reference point. This new MGW (i.e. different than the CS-MGW and IM-MGW) would be controlled by the MSC Server via a new reference point I7. While the MGW function and reference point would be new, they would be realized by a combination of the CS-MGW and IM-MGW and the combined Mc and Mn reference points respectively.

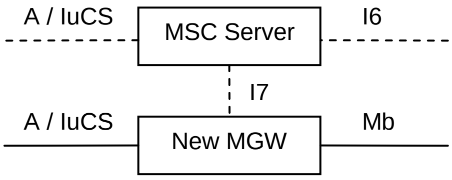


Figure D-1

This option involves the most impact to 3GPP specifications.

Option 2: MSC Server support for IMS-MGW

In this option, the Mn reference point would be added to the MSC Server, allowing it to directly control the IMS-MGW. This effectively decouples the Mn reference point from the MGCF function, as Mn would now be applicable to multiple call control functions which interwork CS and IMS bearers. The bearer interworking required by this architecture would involve both the CS-MGW and IMS-MGW functions, which could obviously be combined into a single physical node in implementation. The I7 reference point is realized by using both the Mc and Mn reference points.

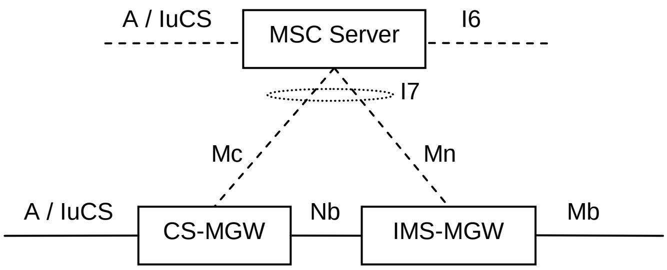


Figure D-2

Option 3: Co-location of MSC Server and MGCF

In this option, the call control function is assumed to be a combination MSC Server and MGCF, allowing it to directly control the CS-MGW and IMS-MGW functions, which could obviously be combined into a single physical node in implementation.

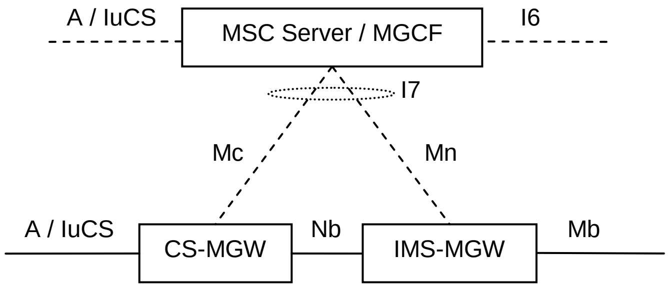


Figure D-3

While this option provides the Mn interface to the MSC Server/MGCF, it is misleading as MGCF functionality (i.e. interworking between the Mg reference point and either ISUP or BICC) as a whole is not required by the eMSC-S.

Option 4: Reuse Mc changes being introduced for SIP-I

R8 is adding SIP-I to the Nc reference point, which is also resulting in changes to the Mc reference point. This will allow a MGW to interwork CS access bearers and RTP bearer (which does not use NbUP framing). This is the interworking required by this architecture, therefore this R8 work can be leveraged.

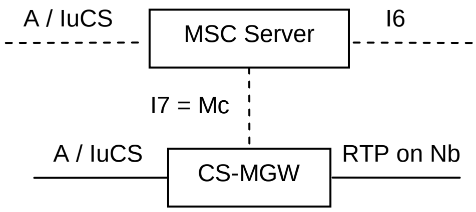


Figure D-4

These Mc changes are being specified in TS 23.231 (stage 2) and TS 29.231 (stage 3).

While this work in R8 is expected to provide the CS-MGW with the changes required to meet the user plane requirements of this architecture, the CS-MGW remains a CS domain node and these changes are being made to the Nb reference point. i.e. the CS-MGW is not being enhanced to support the Mb reference point.

Annex E:
Change history

Change history							
Date	**TSG #**	**TSG Doc**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2008-03	SP-39	SP-080102	-	-	MCC editorial update for presentation to TSG SA for Approval	1.5.2	2.0.0
2008-03	SP-39	-	-	-	MCC Update after TSG SA approval to Rel-8	2.0.0	8.0.0
2008-03					validation of keywords, drafting rules conformity, correction of title	8.0.0	8.0.1