*3GPP2 X.S0042-A*
*Version 1.0*
*Date: August 2008*

# *Voice Call Continuity between IMS and Circuit Switched Systems*

# Revision History

| Revision | Description | Date |
| --- | --- | --- |
| Rev. 0 v1.0 | Initial publication | October 2007 |
| Rev. A v1.0 | First revision; Generalized to include EUTRAN-HRPD inter-technology handoffs. | August 2008 |

Voice Call Continuity between IMS and Circuit Switched Systems

# CONTENTS

# LIST OF FIGURES

# LIST OF TABLES

# FOREWORD

(This foreword is not part of this Standard.)

“Shall” and “shall not” identify requirements to be followed strictly to conform to this document and from which no deviation is permitted. “Should” and “should not” indicate that one of several possibilities is recommended as particularly suitable, without mentioning or excluding others, that a certain course of action is preferred but not necessarily required, or that (in the negative form) a certain possibility or course of action is discouraged but not prohibited. “May” and “need not” indicate a course of action permissible within the limits of the document. “Can” and “cannot” are used for statements of possibility and capability, whether material, physical or causal.

# 1 Introduction

## 1.1 Scope

Voice call continuity allows users to move voice calls between the Circuit-Switched (CS) domain and the Multimedia Domain (MMD), connected through different Internet Protocol - Connectivity Access Networks (IPCANs) (e.g., WLAN, HRPD, E-UTRAN). This document defines an inter-technology Domain Transfer (DT) call model which supports procedures that allow a mobile subscriber to perform DT for the following cases:

– from a multimedia session with a voice component to a 1x CS voice session, and,

– for the case where the UE is equipped with dual radios (e.g., WLAN), from a 1x CS voice session to a multimedia session with a voice component.

It is the goal of this specification to allow the core network to know as precisely as possible the current accessibility of the User Equipment (UE) and to deliver services efficiently across the appropriate access network while minimizing the impact on the legacy systems.

The UEs in this specification include two kinds - one is single radio, dual mode handset (e.g., HRPD/1x), and the other is dual radio, dual mode handset (e.g., WLAN/1x). Single radio handsets are assumed to be incapable of simultaneous full-duplex radio communications with both a packet radio access network and a 1x radio access network, i.e., they have only one radio transmitter and may have one or two radio receivers. Dual radio handsets are assumed to be capable of being in simultaneous full-duplex communication with both a packet data access network and a 1x radio access network, i.e., they have two radio transmitters and two radio receivers.

Supplementary services and supplementary services continuity e.g., Call Waiting, Call Transfer, 3WC, Call Hold) are outside the scope of this document.

The Stage 2 of this document specifies the interactions and signaling flows between a new functional entity in the MMD network called the VCC Application Server (VCC AS) and the:

- Serving-Call/Session Control Function (S-CSCF)
- Home Location Register (HLR)
- Home Subscriber Server (HSS)
- Single Radio Terminal
- Dual Radio Terminal
- Mobile Switching Center (MSC)
- Media Gateway Control Function (MGCF)

The Stage 3 specification in this document provides the protocol details for voice call continuity between the MMD based on the Session Initiation Protocol (SIP) and the Session Description Protocol (SDP) and the 3GPP2 CS domain which uses MAP and ISUP.

This document makes no VCC specific enhancements to SIP, SIP events or SDP, beyond those specified in [MMD Part-4].

This document is applicable to UEs, AS and MGCF providing voice call continuity capabilities.

# 2 References

## 2.1 Normative References

The following standards contain provisions which, through reference in this text, constitute provisions of this Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below. ANSI and TIA maintain registers of currently valid national standards published by them.

- 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 3GPP2 document, a non-specific reference implicitly refers to the latest version of that document in the same release as the present document.


[A.S0008] 3GPP2 A.S0008-C, “Interoperability Specification (IOS) for High Rate Packet Data (HRPD) Radio Access Network Interfaces with Session Control in the Access Network”.

[A.S0009] 3GPP2 A.S0009-C, “Interoperability Specification (IOS) for High Rate Packet Data (HRPD) Radio Access Network Interfaces with Session Control in the Packet Control Function”.

[ANSI ISUP] ANSI T1.113: “Telecommunications Signalling System No. 7 (SS7) – Integrated Services Digital Network (ISDN) User Part (ISUP)”.

[C.S0005] 3GPP2 C.S0005-D v2.0: “Upper Layer (Layer 3) Signaling Standard for cdma2000 Spread Spectrum Systems – Release D”, October 2005.

[C.S0082] 3GPP2 C.S0082-0: “Circuit Services Notification Application Specification for cdma2000 High Rate Packet Data”

[ITU ISUP] ITU-T Recommendations Q.761to Q.764 (2000): "Specifications of Signalling System No.7 ISDN User Part (ISUP)".

[MMD Part-4] 3GPP2 X.S0013-004: "IP Multimedia Call Control Protocol based on SIP and SDP; Stage 3".

[MMD Part-10] 3GPP2 X.S0013-010: “All-IP Core Network Multimedia Domain - IP Multimedia Subsystem Sh Interface; Signaling Flows and Message Contents – Stage 2”

[MMD Part-11] 3GPP2 X.S0013-011: “All-IP Core Network Multimedia Domain: Sh Interface Based on Diameter Protocols Protocol Details - Stage 3”

[RFC 3261] IETF RFC 3261 (June 2002): “SIP: Session Initiation Protocol”, June 2002.

[RFC 3840] IETF RFC 3840: “Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)”, August 2004.

[WIN] 3GPP2 N.S0013: “WIN Phase 1”

[X.S0004-540] 3GPP2 X.S0004-540: "MAP Operations Signaling Protocols"

[X.S0004-550] 3GPP2 X.S0004-550: "MAP Parameters Signaling Protocols"

[X.S0004-630] 3GPP2 X.S0004-630: "Basic Call Processing"

[X.S0004-641] 3GPP2 X.S0004-641: "Mobile Application Part (MAP) - SMS"

[X.S0050] 3GPP2 X.S0050-0, "Session Initiation Protocol (SIP) to ISDN User Part (ISUP) Interworking"

[E-UTRA] 3GPP TS 36.331 "E-UTRA; Radio Resource Control (RRC); Protocol specification".

[SRVCC] 3GPP TS 29.277: "Optimised Handover Procedures and Protocol between EUTRAN access and non-3GPP accesses (S102); Stage 3"

## 2.2 Informative References

[S.R0087-A] S.R0087-A: "cdma2000 – WLAN Interworking"

[S.R0108] S.R0108-0: "HRPD-cdma2000 1x Interoperability for Voice and Data - System Requirements"

# 3 Definitions, Symbols and Abbreviations

This section contains definitions, symbols and abbreviations that are used throughout the document.

## 3.1 Definitions

#### 1x CS

The Legacy Circuit Switched signaling system

#### VCC AS

An entity that:

1. assists in domain selection for terminating services to a terminal that is 1x CS registered, or IMS registered, or both.
2. assists in DTs between IMS VoIP and 1x CS voice. Depending on the IP-CAN technology used to connect to MMD, the handoff may be supported in both directions (e.g. WLAN VoIP to/from 1x CS) or only in one direction (e.g. HRPD VoIP to 1x CS).

#### Domain Transfer (DT)

Transfer of a voice call from the CS domain to MMD while maintaining an active session or transfer of a voice media from MMD to the CS domain while maintaining an active session.

#### Retarget

A SIP request is retargeted when the original "target" indicated by the Request-URI is changed to a new "target" by changing the Request-URI.

### 3.1.1 Symbols and Abbreviations

| | |
|---|---|
| ANSI | American National Standards Institute |
| AN | Access Networkt |
| AS | Application Server |
| B2BUA | Back-to-Back User Agent |
| BGCF | Border Gateway Control Function |
| BS | Base Station |
| CS | Circuit-Switched |
| CSCF | Call/Session Control Function |
| CSNA | Circuit Services Notification Application |
| DT | Domain Transfer |
| E-UTRA | Evolved UMTS Terrestrial Radio Access |
| E-UTRAN | Evolved UMTS Terrestrial Radio Access Network |
| HLR | Home Location Register |
| HRPD | High Rate Packet Data |
| HSS | Home Subscriber Server |

| | |
|---|---|
| iFC | initial Filter Criteria |
| IMRN | IMS Routing Number |
| IMS | IP Multimedia Subsystem |
| IMSI | International Mobile Subscriber Identity |
| IP-CAN | IP-Connectivity Access Network |
| IPv6 | Internet Protocol version 6 |
| ISUP | ISDN User Part |
| I-CSCF | Interrogating-CSCF |
| MAP | Mobile Application Part |
| MC | Message Center |
| MDN | Mobile Directory Number |
| MGCF | Media Gateway Control Function |
| MGW | Media Gateway |
| MMD | Multimedia Domain |
| MSC | Mobile Switching Center |
| PCM | Pulse Code Modulation |
| P-CSCF | Proxy-CSCF |
| PDIF | Packet Data Interworking Function |
| PDSN | Packet Data Serving Node |
| PSI | Public Service Identity |
| PSTN | Public Switched Telephone Network |
| RTP | Real-time Transport Protocol |
| SCP | Service Control Point |
| SDP | Session Description Protocol |
| SIP | Session Initiation Protocol |
| SMS | Short Message Service |
| S-CSCF | Serving-CSCF |
| TDM | Time Division Multiplexing |
| TLDN | Temporary Local Directory Number |
| TRK | trunk |
| UDP | User Datagram Protocol |
| UE | User Equipment |
| UMTS | Universal Mobile Telecommunications System |
| URI | Uniform Resource Identifier |
| UTC | Coordinated Universal Time |
| VCC | Voice Call Continuity |
| VCC AS | Voice Call Continuity Application Server |
| VDN | VCC Domain Transfer Directory Number |
| VDI | VCC Domain Transfer URI |
| VoIP | Voice over IP |
| VLR | Visitor Location Register |
| WIN | Wireless Intelligent Network |
| WLAN | Wireless Local Area Network |

# 4 Stage 2

## 4.1 Architecture Reference Model

Figure 1 illustrates the architecture reference model to support Voice Call Continuity, including IMS-CS DTs, call origination and call termination. Only those MMD or CS network entities or interfaces supporting VCC are shown.

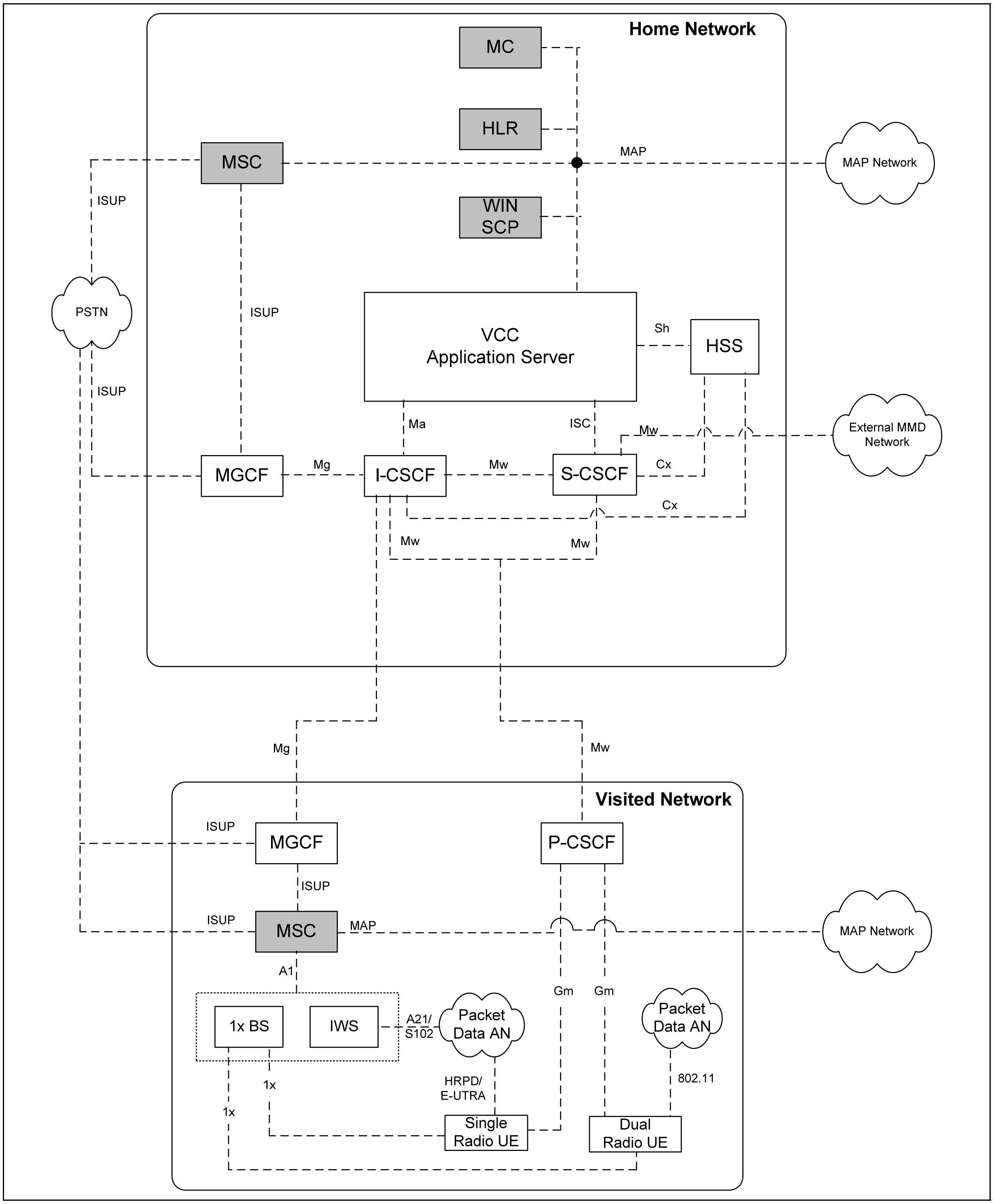


***Figure 1*** **VCC Architecture Reference Model**

Voice Call Continuity introduces a new VCC Application Server (VCC AS) functional entity in the MMD network and relevant reference points for communication with the CS and IMS functional entities. The VCC AS makes use of existing CS and IMS functional entities and reference points.

The VCC Application Server comprises two main functions:

- assists in terminating services to a terminal that is 1x CS registered and/or IMS registered
- is involved in voice call setup signaling to facilitate VoIP-to-1x CS voice call DTs and for dual radio UEs, also facilitates 1x CS voice call to VoIP DTs

The VCC AS is anchored in the call signaling path of all voice calls originated from, or terminated to, a VCC UE that is IMS registered and tuned to a packet data access network, or 1x CS registered and tuned to 1x. It has the following signaling interfaces:

- VCC AS / S-CSCF (ISC)

The VCC AS serves as a SIP Back-to-Back User Agent (B2BUA) that interfaces to the S-CSCF via an ISC SIP signaling interface.

- VCC AS / I-CSCF (Ma)

The VCC AS interfaces to an I-CSCF via an 'Ma' SIP signaling interface. This interface is used to anchor the VCC AS in the call path by sending SIP request from I-CSCF directly to the VCC AS.

- VCC AS / HLR (MAP)

The VCC AS interfaces to the 1x CS HLR using MAP in order to obtain routing information for terminating voice calls to a UE via the 1x CS network.

- VCC AS / HSS (Sh)

The VCC AS also interfaces to the HSS via an Sh interface [MMD Part-10] using the Diameter protocol to transfer data between the VCC AS and HSS.

- VCC AS / WIN SCP (MAP)

The VCC AS interfaces to the WIN SCP using the MAP protocol in order to provide routing information for 1x voice call originations and terminations and to anchor the VCC AS in these calls. The WIN SCP may be integrated with the VCC AS or may be a standalone network element.

## 4.2 Signaling Flows for Registration

This section of the specification illustrates signaling flows for registration. The registration model supports a "dual registration", i.e., the mobile may be registered in either the IMS network, in the CS network (i.e., in the HLR), or in both.

Registration is the procedure by which the UE is authorized and granted permission to access and utilize either, the IMS network, the CS network, or both. IMS Registration provides limited real-time information (e.g., only at the time of registration) as to the status of access network connectivity. There is a defined mechanism (see section 4.2.3) by which the UE can inform the VCC AS as to the loss of packet data access connectivity if the UE is CS access connected.

### 4.2.1 IMS Registration

In support of IMS-CS voice call continuity procedures, a dual mode handset that wishes to originate an IMS VoIP call or have a voice call delivered to it via IMS must perform IMS registration, including third party registration with the VCC AS by the S-CSCF. This informational flow does not take into account security features such as user authentication. Figure 2 illustrates the scenario where terminal UE 1 performs an IMS registration.

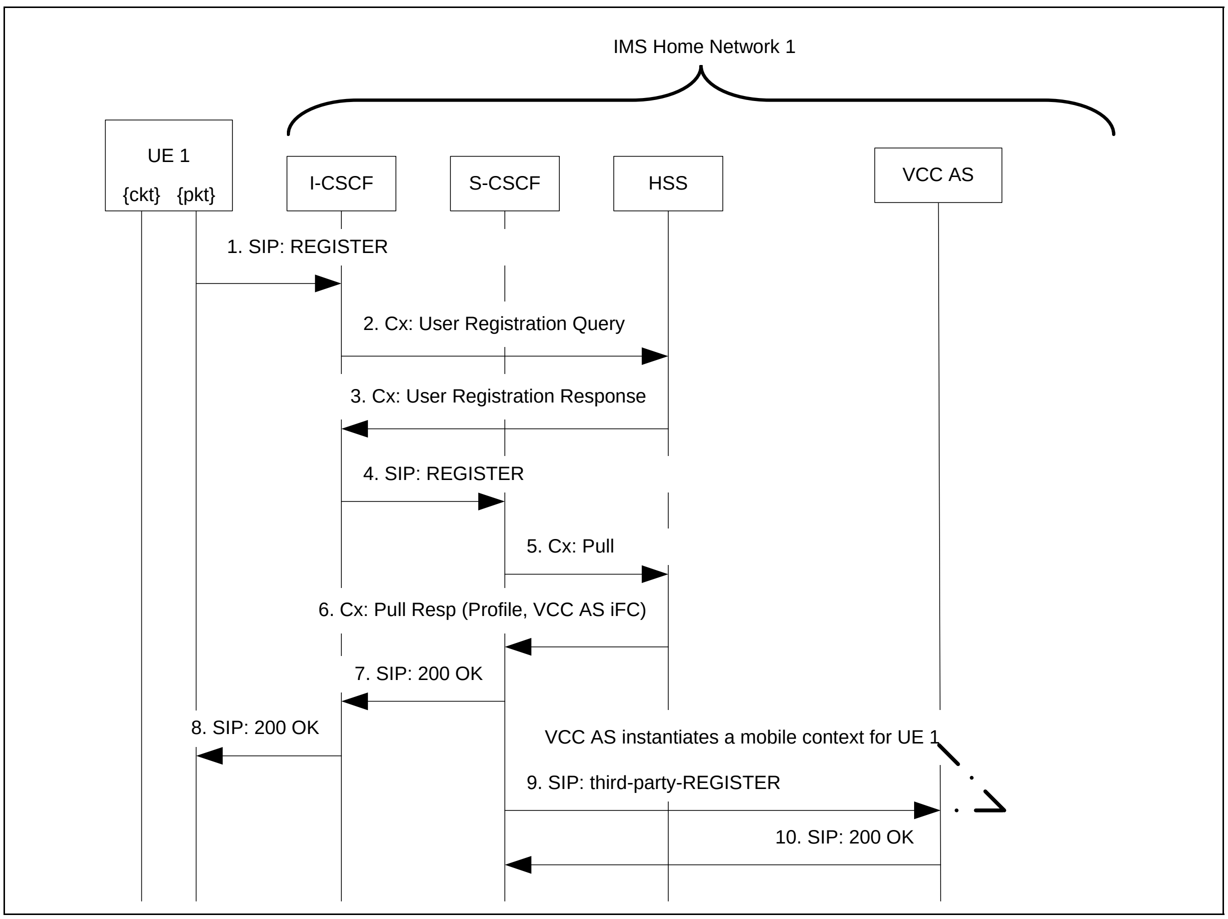


***Figure 2*** **IMS registration**

1. UE 1 sends a SIP *REGISTER* message to the I-CSCF via the P-CSCF (not shown).

2. The I-CSCF sends a Cx *User Registration Query* message to the HSS to obtain the SIP URI of the S-CSCF.
3. The HSS returns a Cx *User Registration Response* message to the I-CSCF with the S-CSCF URI.
4. The I-CSCF sends the *REGISTER* message to the S-CSCF.
5. The S-CSCF sends a Cx *Pull* message to the HSS.
6. The HSS returns a Cx *Pull Resp* message to the S-CSCF with the user's profile information, as well as the VCC AS iFC.
7. The S-CSCF sends a SIP *200 OK* to the I-CSCF.
8. The I-CSCF sends a *200 OK* via the P-CSCF (not shown) back to UE 1.
9. The S-CSCF executes a SIP third-party registration with the VCC AS to indicate that UE 1 has just registered with the S-CSCF.
10. The VCC AS instantiates a context for UE 1 and sends a *200 OK* back to the S-CSCF.

### 4.2.2 1x CS Registration

Figure 3 illustrates the scenario where UE 1 performs 1x CS registration.

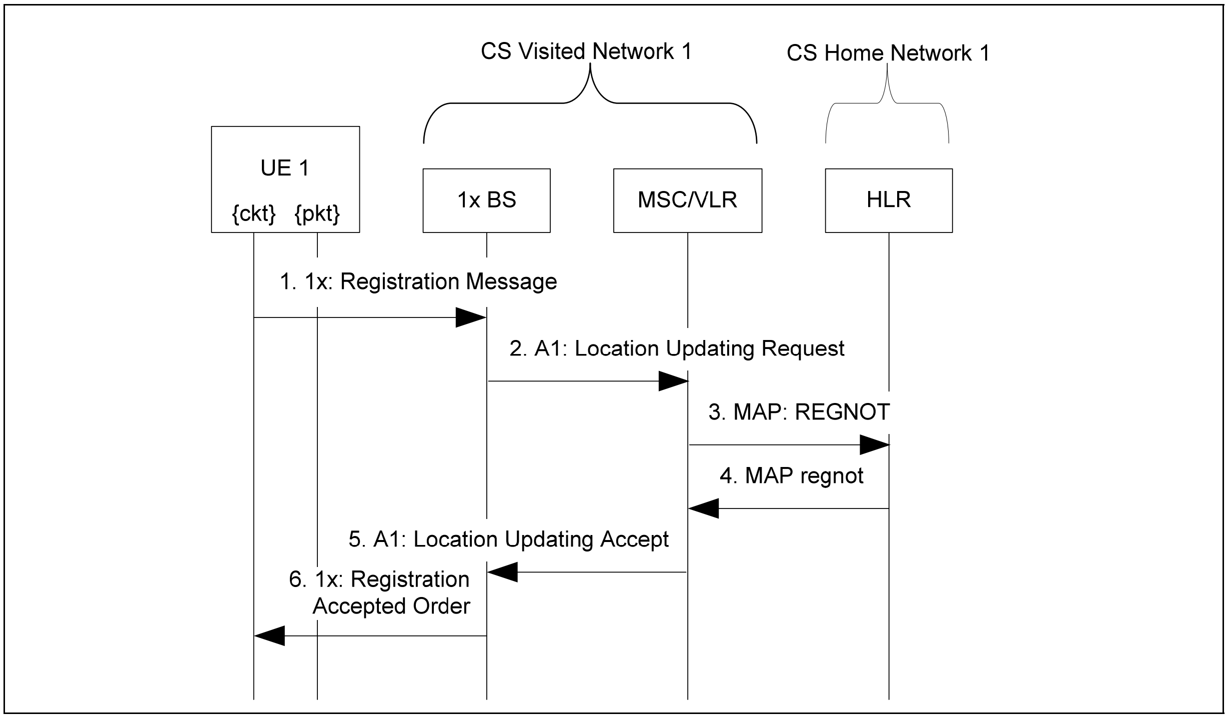


***Figure 3*** **1x CS registration**

1. UE 1 sends a 1x *Registration* Message to the 1x BS.
2. The 1x BS sends an AI *Location Updating Request* to the MSC/VLR.
3. The MSC/VLR sends a MAP *REGNOT* message to the HLR to update the location pointer for UE 1.
4. The HLR sends a MAP *regnot* message back to the MSC/VLR.
5. The MSC/VLR sends an A1 *Location Updating Accept* message to the 1x BS.
6. The 1x BS sends a 1x *Registration Accepted Order* message back to UE 1.

### 4.2.3 Domain Availability Notification

The procedure described in this section is used by the UE to send an indication to the VCC AS that it is currently attached only to the CS domain.

When a UE was last IMS registered via a packet data air-interface and the UE detects the loss of the VoIP capable packet data air-interface and 1x CS air-interface is available, the UE registers (if necessary) with the 1x network and sends an SMS addressed to the VDN associated with the VCC AS.

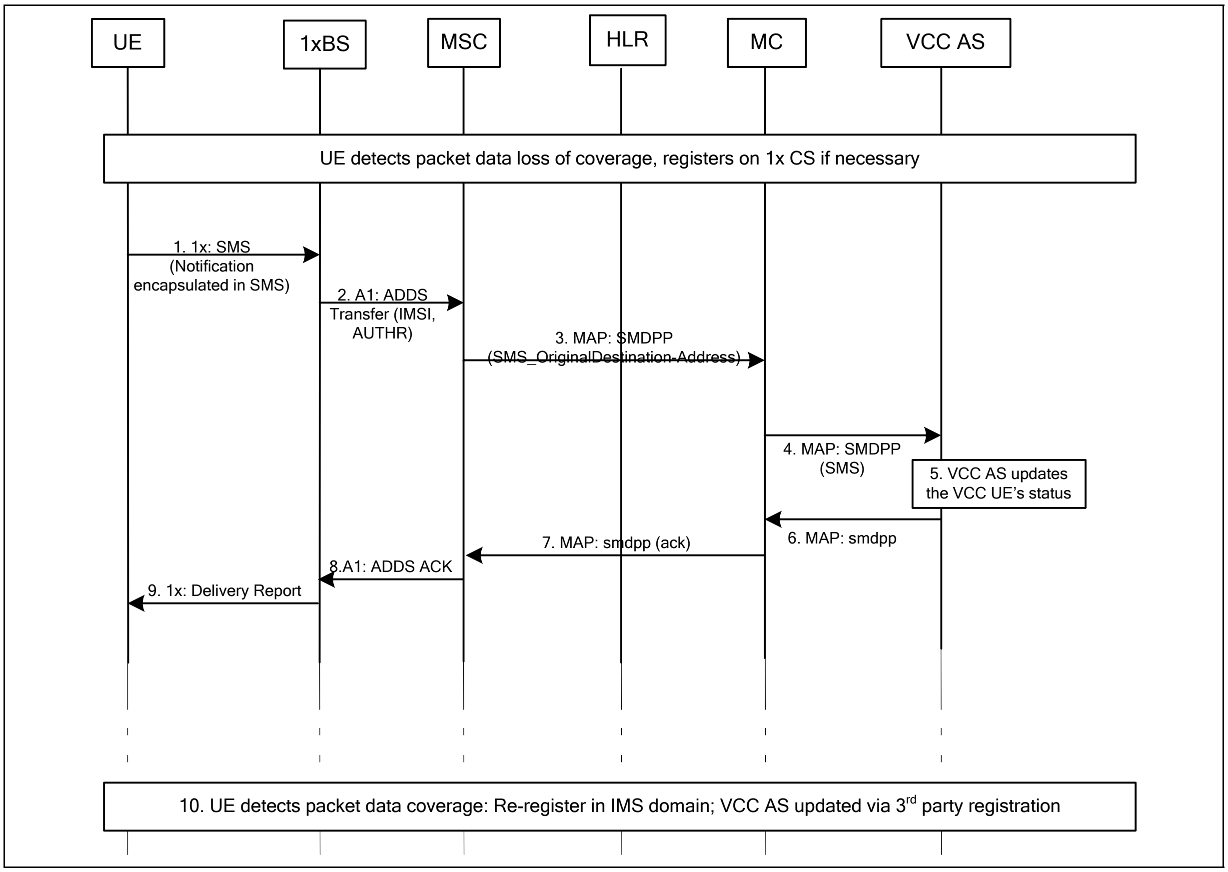


***Figure 4*** **UE Initiated Notification after 1x CS Registration**

1. On detecting loss of VoIP capable packet data coverage, the UE registers on 1x CS if necessary. The UE encapsulates the notification update in a SMS message addressed to the VCC AS (i.e., address to the VDN associated with the VCC, which is provisioned at the UE).
2. An A1 *ADDS_transfer* message is sent from the 1xBS to the Visited MSC.
3. The Visited MSC forwards the SMS message to the MC. Optionally, the MSC may send the SMS message directly to the VCC AS [X.S004-641].
4. On receipt of a SMS message, the MC may perform a local database lookup based on the VCC E.164 contained in the Original Destination Address number that maps to the VCC AS address. The MC forwards the MAP *SMDPP* message to the VCC AS using Global Title Translation (GTT) or Point Code (PC) routing.

5. The VCC AS updates the domain availability of the VCC UE in order to deliver all incoming voice calls to the UE on 1x CS.

6. The VCC AS responds to the delivered SMS message with a positive acknowledgement. If the MSC sent the SMS message directly to the VCC AS, the VCC AS would send the response directly to the MSC [X.S0004-641].

7-9. The delivery report message is forwarded through the MC/MSC to the originating UE.

10. When packet data access with VoIP capability becomes available again, the UE shall perform an IMS re-registration and the VCC AS is updated via 3rd party registration (as described in Section 4.2.1) so that future calls can be delivered over IMS.

## 4.3 Signaling Flows for Call Origination

This section of the specification illustrates signaling flows for call origination.

### 4.3.1 IMS VoIP Call Origination with VCC AS Anchoring

This section describes an IMS VoIP Call Origination that anchors the VCC AS into the call in preparation for a possible VCC DT to a CS voice call.

NOTE: Although the scenario provides an example of a VoIP call origination, an originating MMD session with other media types (i.e., video) may be anchored since audio media can be added before the DT occurs.

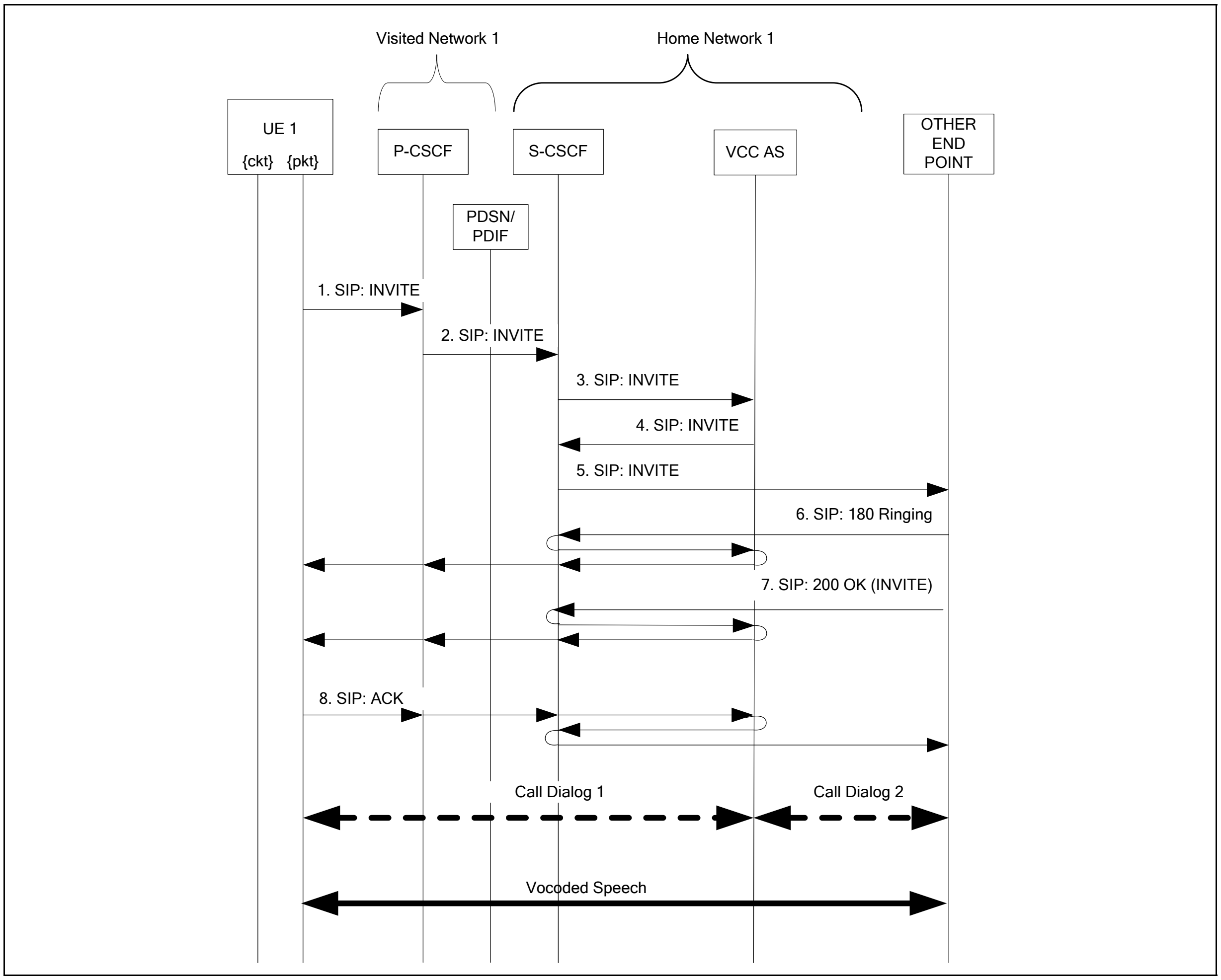


***Figure 5*** **IMS VoIP Call Origination with VCC AS Anchoring**

1. UE 1 sends a SIP *INVITE* message, containing an initial SDP, to the P-CSCF in order to initiate a VoIP call via IMS.
2. The P-CSCF sends the *INVITE* to the S-CSCF.
3. The *INVITE*, based upon the initial Filter Criteria (iFC) is forwarded to the VCC AS.

4. The VCC AS, acting as B2BUA, sends an *INVITE* on behalf of UE 1 to the destination defined in the *INVITE* in Step 1. The *INVITE* is sent to the S-CSCF. The VCC AS will remain in the call session signaling path for the remainder of the call and assist in any future DT of the VoIP call to a 1x CS voice call as needed.

5. The S-CSCF sends the *INVITE* to the Other End Party (OEP).

6. The OEP returns a SIP *180 Ringing* message to the VCC AS that is sent back along the signaling path to UE 1.

7. The OEP answers the call and sends a SIP *200 OK* message to the VCC AS that is sent back along the signaling path to UE 1.

8. UE 1 responds with a SIP *ACK* that is sent through the signaling path towards the OEP. The VCC AS reception of the *ACK* from the UE 1 completes the establishment of call dialog 1 between UE 1 and the VCC AS. The OEP's reception of the *ACK* from the VCC AS, completes the establishment of call dialog 2 between the VCC AS and the OEP.

## 4.3.2 CS Call Origination with VCC AS Anchoring

### 4.3.2.1 WIN Based Solution

This section describes a CS Voice Call Origination that anchors the VCC AS into the call in preparation for a possible VCC DT to IMS VoIP. Figure 6 illustrates the call flow that uses a WIN SCP to route the CS voice call to the VCC AS. The WIN SCP function may be performed by a separate entity, or optionally (not shown), it may be performed as an integrated function within the VCC AS.

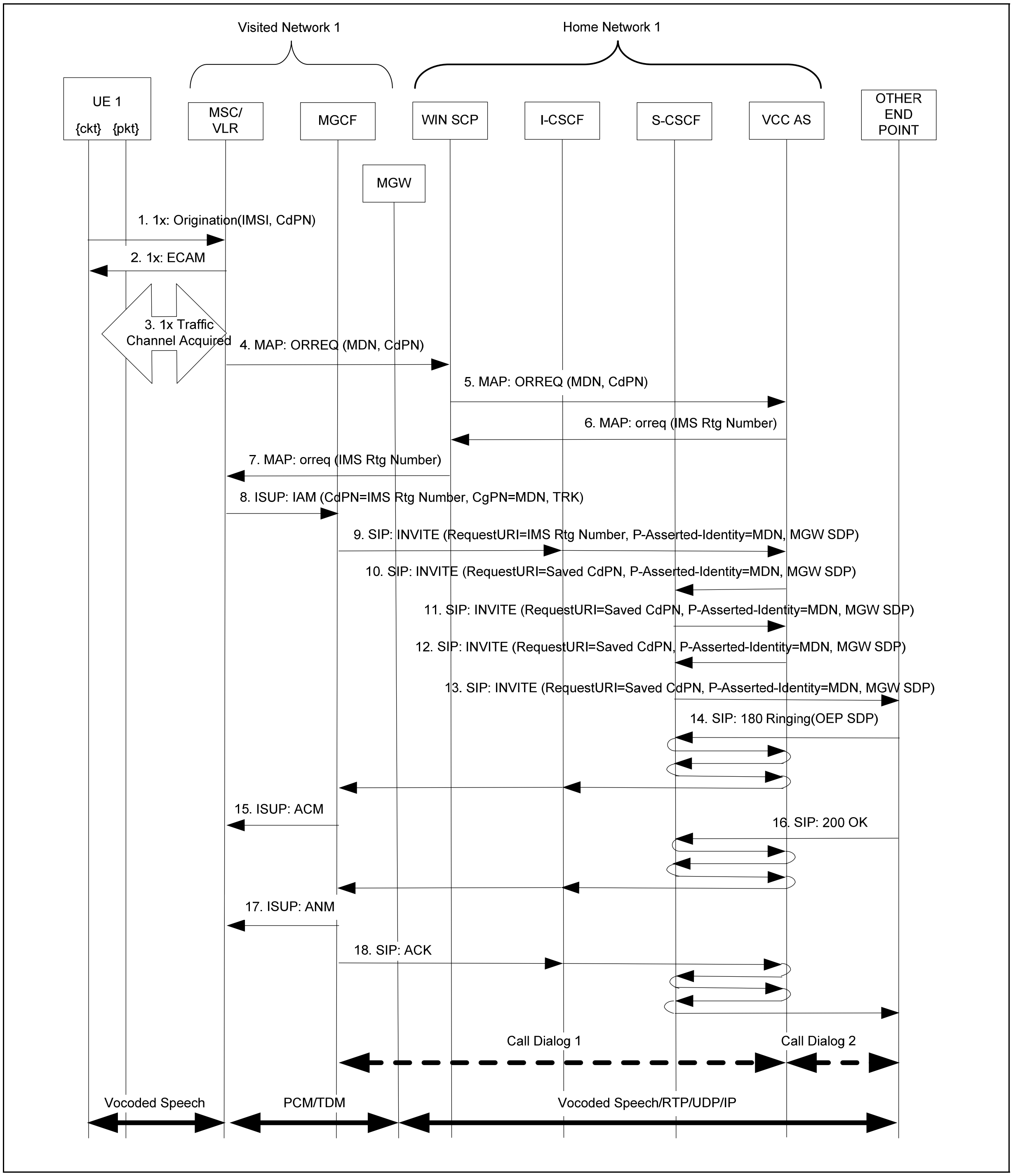


***Figure 6*** **CS Call Origination with VCC AS Anchoring (WIN based)**

1. UE 1 originates a call from the 1x CS MSC.

   NOTE: Steps 4 – 7 are shown using the MAP *ORREQ* operation. Optionally, a post digit analysis trigger using the MAP *ANLYZD* operation may be used instead to obtain the IMS Routing Number and to anchor the call in IMS and to obtain the IMS Routing Number.

2. The MSC/VLR sends a 1x *Channel Assignment* to UE 1.

3. UE 1 acquires the 1x traffic channel.
4. The MSC invokes a call origination WIN trigger to obtain routing information. The MSC sends a MAP *ORREQ* message, containing the Calling Party MDN and the Called Party Number (CdPN), to the WIN SCP. Optionally, the MSC/VLR may send the *ORREQ* message directly to a VCC AS that has an integrated WIN SCP function and step 5 is skipped.
5. The WIN SCP sends the *ORREQ* message on to the VCC AS. This initiates VCC call anchoring.
6. The VCC AS stores the MDN and CdPN for later call setup and returns a MAP *orreq* response message with the IMS Routing Number (E.164) of the VCC AS as the routing information for this call.
7. The WIN SCP returns the *orreq* message to the MSC/VLR. Step 7 is skipped if the VCC AS has an integrated WIN SCP function.
8. The MSC/VLR sends an ISUP *IAM* message to the MGCF in the serving network, containing UE 1's routing information (i.e., IMS Routing Number in the CdPN field and UE 1's MDN in the CgPN) and a PCM/TDM trunk to be connected between the MSC and the MGW.
9. The MGCF requests that the MGW connect to the PCM/TDM trunk and allocate an ephemeral termination to be connected to the far end. The MGCF then sends the SIP *INVITE* with the IMS Routing Number (i.e., VCC AS PSI) in the Request URI header and UE 1's MDN in the P-Asserted-Identity header to the I-CSCF. The I-CSCF queries the HSS in order to determine the next hop in the routing path for the PSI. In this case, it is the VCC AS.
10. The VCC AS acting as a B2BUA creates an *INVITE* request containing MGW-SDP and the tel URI of the UE, generated from the MDN. The VCC AS adds the "orig" parameter to the topmost Route header of the INVITE request. The VCC AS sends the *INVITE* to the S-CSCF for IMS Processing.
11. Based on filter criteria, the S-CSCF sends a *INVITE* message to the VCC AS containing UE 1's routing information as received in the previous step.
12. The VCC AS sends the *INVITE* back to the S-CSCF.
13. Upon completion of filter criteria processing, the S-CSCF sends the *INVITE* to the Other End Point (OEP) (IMS user or PSTN MGCF/MGW).
14. The OEP responds with a SIP *180 Ringing* message, which is sent back along the signaling path and is received at the MGCF. It contains the OEP SDP.
15. The MGCF modifies the MGW ephemeral termination with the remote OEP SDP and sends an ISUP *ACM* message to the MSC/VLR.
16. The OEP answers the call and sends a SIP *200 OK* message back to the MGCF back along the signaling path.
17. The MGCF sends an ISUP *ANM* message to the MSC/VLR.
18. The MGCF also sends a SIP *ACK* message back to the OEP along the signaling path. This completes the establishment of call dialog 1 between the MGCF and the VCC AS and call dialog 2 between the VCC AS and the OEP.

### 4.3.2.2 Non-WIN Based Solution

NOTE: The VCC Subscriber is subscribed to an Origination Trigger service with the trigger type of “All calls”.

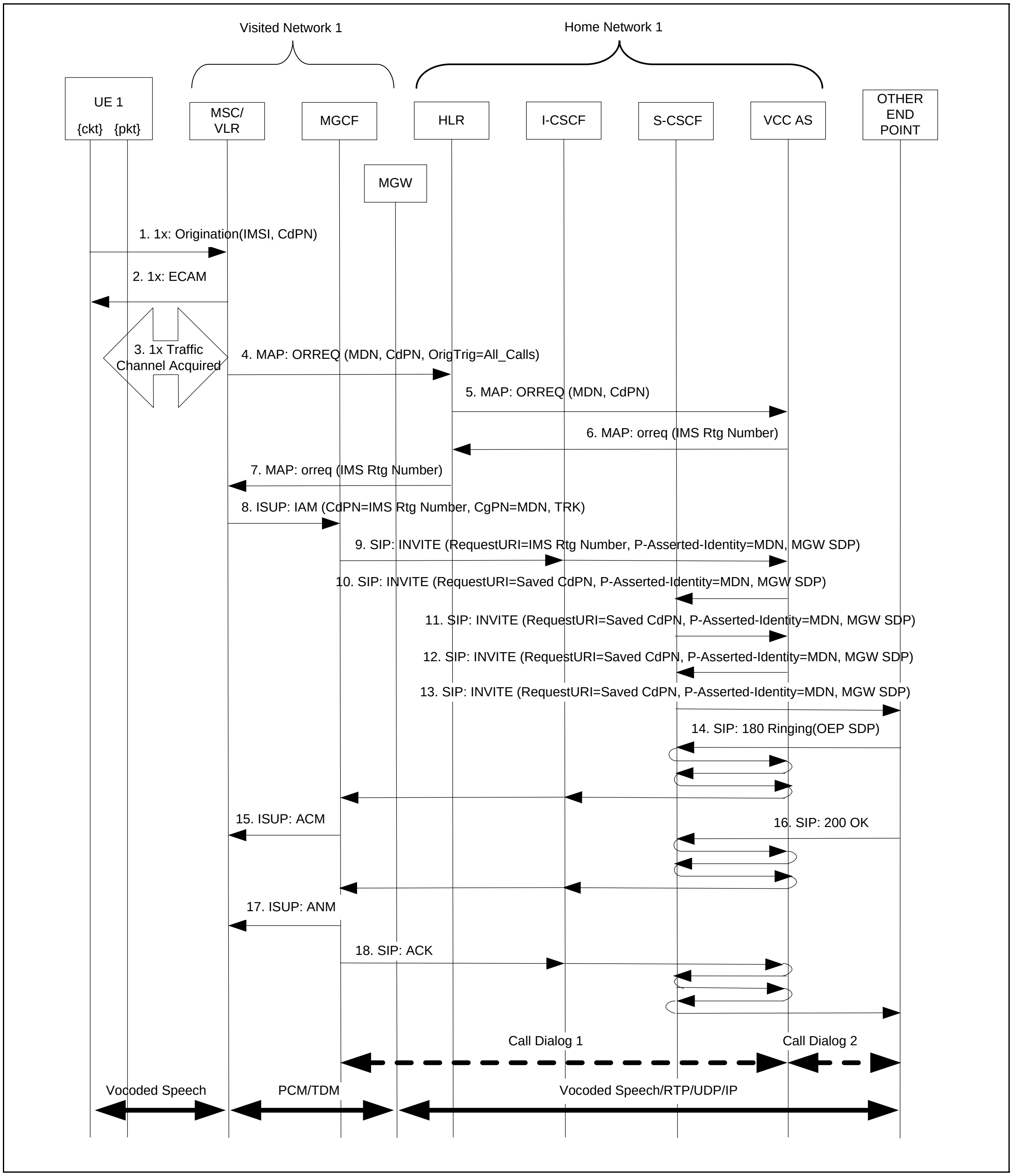


***Figure 7*** **CS Call Origination with VCC AS Anchoring (Non-WIN based)**

1. UE 1 originates a call from the 1x CS MSC.

2. The MSC/VLR sends a 1x *Channel Assignment* to UE 1.
3. UE 1 acquires the 1x traffic channel.
4. The MSC invokes a call origination trigger to obtain routing information from the HLR. The MSC sends a MAP *ORREQ* message, containing the Calling Party MDN and the Called Party Number (CdPN), to the HLR.
5. Upon receiving the *ORREQ* message, the HLR sends an *ORREQ* message to VCC AS. This initiates VCC call anchoring.
6. The VCC AS stores the MDN and CdPN for later call setup and returns a MAP *orreq* response message with the IMS Routing Number (E.164) of the VCC AS as the routing information for this call.
7. The HLR returns the *orreq* message to the MSC/VLR.
8. The MSC/VLR sends an ISUP *IAM* message to the MGCF in the serving network, containing UE 1's routing information (i.e., IMS Routing Number in the CdPN field and UE 1's MDN in the CgPN) and a PCM/TDM trunk to be connected between the MSC and the MGW.
9. The MGCF requests that the MGW connect to the PCM/TDM trunk and allocate an ephemeral termination to be connected to the far end. The MGCF then sends the SIP *INVITE* with the IMS Routing Number (i.e., VCC AS PSI) in the Request URI header and UE 1's MDN in the P-Asserted-Identity header to the I-CSCF. The I-CSCF queries the HSS in order to determine the next hop in the routing path for the PSI. In this case, it is the VCC AS.

The VCC AS acting as a B2BUA creates an INVITE request containing MGW-SDP and the tel URI of the UE, generated from the MDN. The VCC AS adds the "orig" parameter to the topmost Route header of the INVITE request. The VCC AS sends the INVITE to the S-CSCF for IMS Processing.

10. The VCC AS sends the *INVITE* to the S-CSCF for IMS processing.
11. Based on filter criteria, the S-CSCF sends an *INVITE* message to the VCC AS containing UE 1's routing information as received in the previous step
12. The VCC AS sends the *INVITE* back to the S-CSCF.
13. Upon completion of filter criteria processing, the S-CSCF sends the *INVITE* to the Other End Point (OEP) (IMS user or PSTN MGCF/MGW).
14. The OEP responds with a SIP *180 Ringing* message, which is sent back along the signaling path and is received at the MGCF. It contains the OEP SDP.
15. The MGCF modifies the MGW ephemeral termination with the remote OEP SDP and sends an ISUP *ACM* message to the MSC/VLR.
16. The OEP answers the call and sends a SIP *200 OK* message back to the MGCF back along the signaling path.
17. The MGCG sends an ISUP *ANM* message to the MSC/VLR.
18. The MGCF also sends a SIP *ACK* message back to the OEP along the signaling path. This completes the establishment of call dialog 1 between the MGCF and the VCC AS and call dialog 2 between the VCC AS and the OEP.

## 4.4 Signaling Flows for Call Delivery

This section of the specification illustrates signaling flows for call delivery. Domain selection for delivery of terminating services to a hybrid mobile (e.g., HRPD/1x or WLAN/1x) may be based on current information about the user's registration status, radio network connectivity and may also take into account the operator's policy and subscriber's preference.

### 4.4.1 Voice Call Delivery on 1x CS

This section describes the delivery of a voice call to a subscriber that has performed 1x CS registration with the CS network as described in Section 4.2.2. Figure 8 illustrates a signaling flow for the scenario where a terminal that is 1x CS registered receives an IMS VoIP call that is delivered to the subscriber as a CS voice call. It is assumed that the VCC AS determines that UE 1 is connected to the 1x CS network. The initial SIP *INVITE* comes from an Other End Point (OEP), which may be another IMS, an MGCF, or other SIP-capable entity.

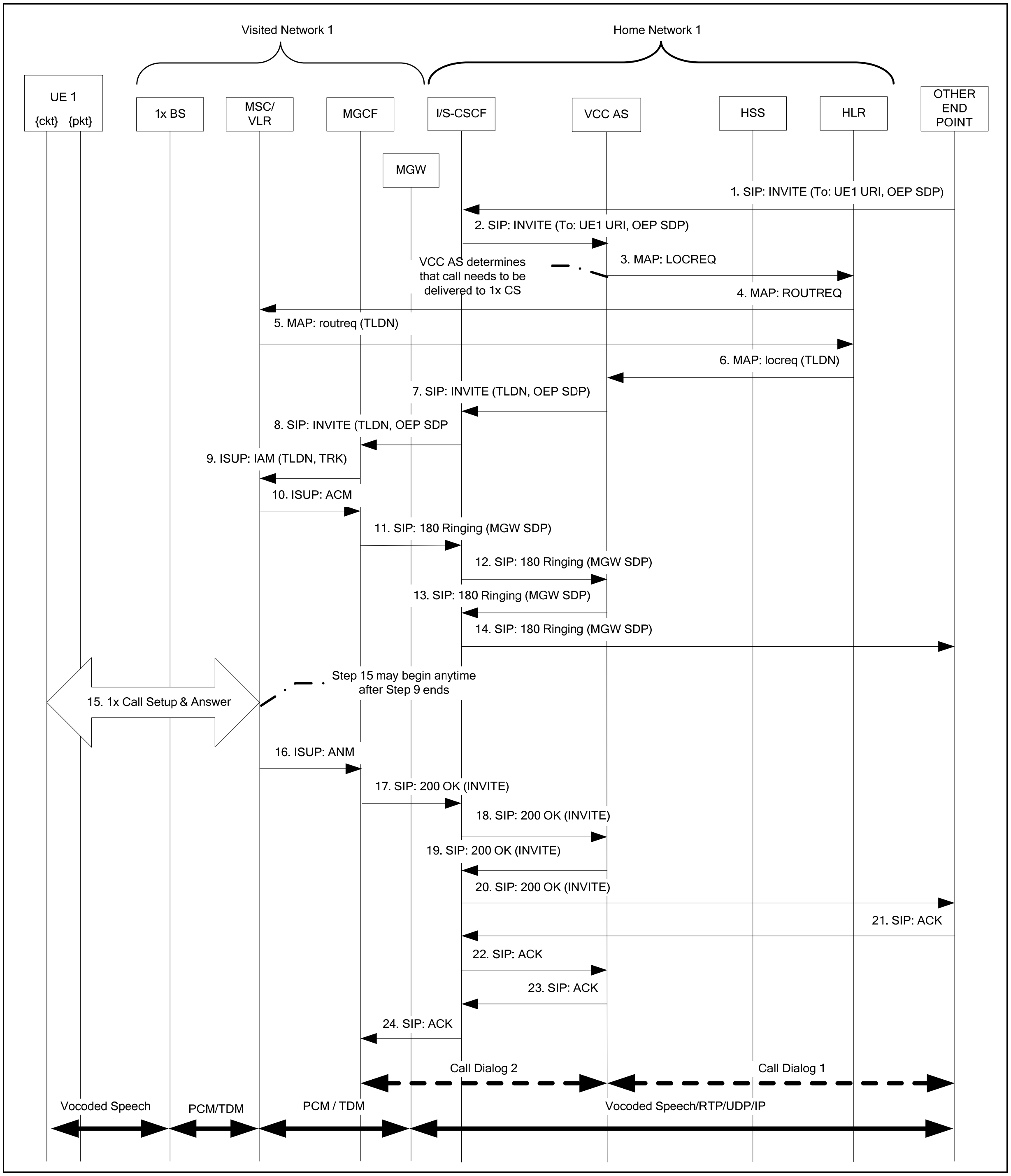


***Figure 8*** **Voice call delivery to 1x CS**

1. The Other End Point (OEP) sends a SIP *INVITE* to the I/S-CSCF in the IMS home network of UE 1.

2. Based on the filter criteria, the I/S-CSCF sends the *INVITE* to the VCC AS. This initiates VCC AS call anchoring.

3. The VCC AS determines that the call needs to be delivered to 1x CS and sends a MAP *LOCREQ* message to the HLR to determine UE 1's location and routing information.

4. The HLR sends a MAP *ROUTREQ* message to the MSC/VLR to determine UE 1's routing information.

5. The MSC/VLR sends a MAP *routreq* message back to the HLR containing UE 1's routing information (i.e., TLDN).

6. The HLR sends a MAP *locreq* message back to the VCC AS containing UE 1's routing information (i.e., TLDN).

7. The VCC AS sends an *INVITE* message back to the I/S-CSCF containing UE 1's routing information (i.e., TLDN).

8. The I/S-CSCF sends an *INVITE* message to the MGCF containing UE 1's routing information (i.e., TLDN). The interaction between the S-CSCF and BGCF, and MGCF and BGCF are not shown for brevity.

9. The MGCF requests that the MGW be configured with an ephemeral termination to be connected to the OEP and a physical PCM trunk termination, TRK, to be connected to the MSC/VLR. The MGCF sends an ISUP *IAM* message to the MSC/VLR. Note that if the MGCF/MGW does not have TDM trunks connecting it to the MSC/VLR, then the call may be routed to the PSTN and then to the MSC/VLR.

10. The MSC/VLR returns an ISUP *ACM* message to the MGCF

11. The MGCF sends a SIP *180 Ringing* message to the I/S-CSCF. It contains the SDP for the MGW ephemeral termination.

12. The I/S-CSCF sends the *180 Ringing* to the VCC AS.

13. The VCC AS sends the *180 Ringing* to the I/S-CSCF.

14. The I/S-CSCF sends the *180 Ringing* to the OEP.

15. Anytime after Step 9 has ended, a call is setup between the MSC/VLR, the 1x BS and UE 1. UE 1 answers the call.

16. When the call is answered, the MSC/VLR sends an ISUP *ANM* message to the MGCF.

17. The MGCF sends a SIP *200 OK* message to the I/S-CSCF.

18. The I/S-CSCF sends a *200 OK* message to the VCC AS.

19. The VCC AS sends a *200 OK* message to the I/S-CSCF.

20. The I/S-CSCF sends a *200 OK* message to the OEP.

21. The OEP sends a SIP *ACK* message to the I/S-CSCF.

22. The I/S-CSCF sends the *ACK* to the VCC AS. This completes the establishment of SIP call dialog 1 between the VCC AS and the OEP.

23. The VCC AS sends the *ACK* to the I/S-CSCF.

24. The I/S-CSCF sends the *ACK* to MGCF. This completes the establishment of SIP call dialog 2 between the VCC AS and the MGCF.

Post-conditions:

There is now a voice call setup between the UE and the OEP via the 1x CS network. SIP call dialog 1 for this voice call is illustrated by a heavy dashed double arrow between the

VCC AS and the OEP. SIP call dialog 2 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the MGCF. The voice bearer path is illustrated by heavy solid double arrows connecting the UE, MSC MGW and the OEP.

## 4.4.2 Voice Call Delivery on IMS

This section describes the delivery of a voice call to a subscriber that has IMS registered for IMS services as described in Section 4.2.1 Figure 9 illustrates a signaling flow for the scenario where a terminal receives a voice call that is delivered to the subscriber as an IMS VoIP call.

It is assumed that the VCC AS determines that UE 1 is connected to the IMS network. The initial INVITE comes from an OEP, which may be another IMS, an MGCF, or other SIP-capable entity.

NOTE 1: The procedures for how the VCC AS determines the call delivery path is outside the scope of this document.

NOTE 2: Although the call delivery scenario is an example of a VoIP call, MMD sessions with other media types (i.e., video) may be anchored since audio media can be added before the DT occurs.

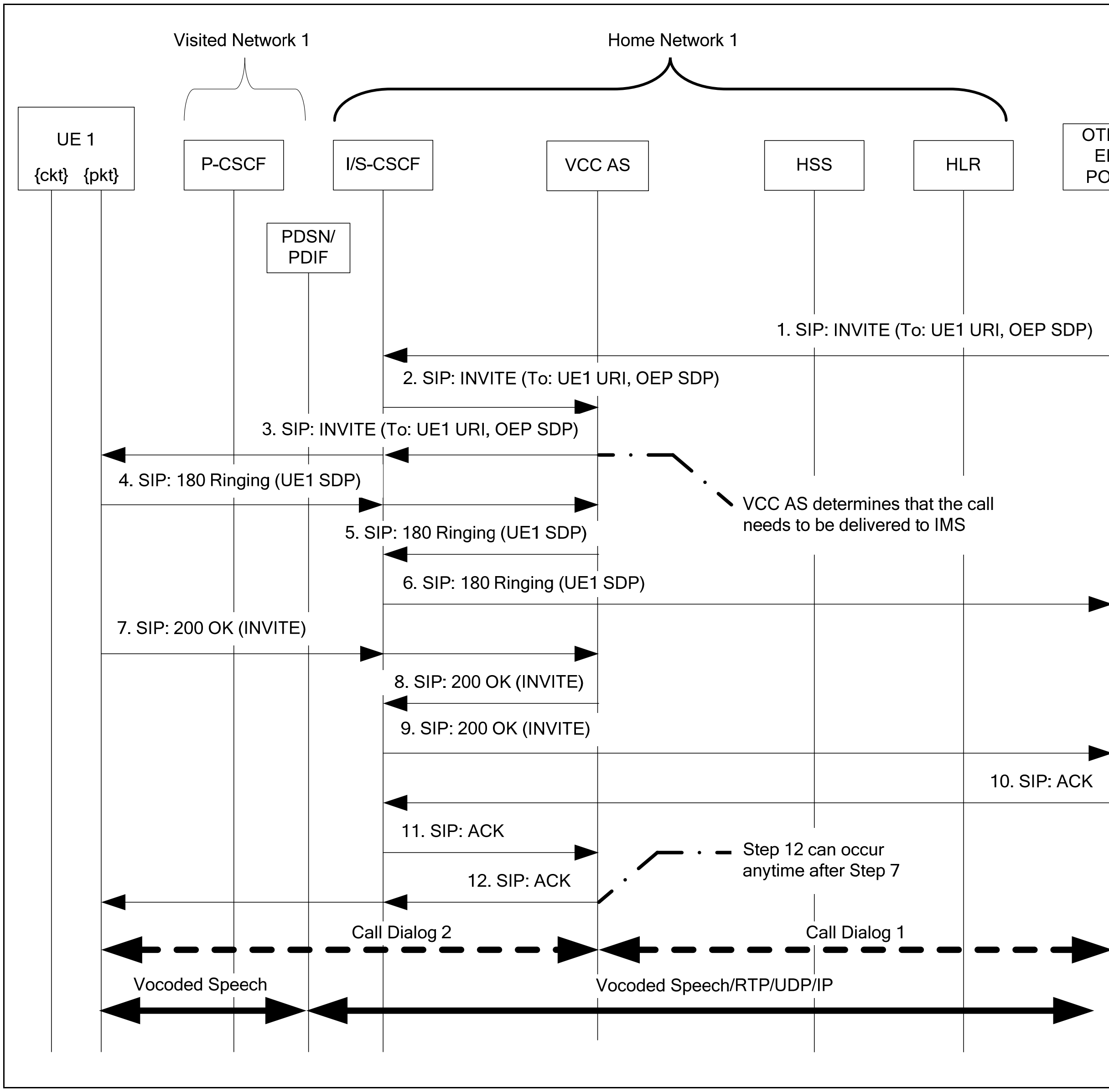


***Figure 9*** **Voice call delivery to IMS**

1. The Other End Point (OEP) sends a SIP *INVITE* to the I/S-CSCF in the IMS home network of UE 1.
2. Based on the filter criteria, the I/S-CSCF sends the *INVITE* to the VCC AS, which acts as a SIP B2BUA, in order to establish SIP call dialog 1 between the OEP and the VCC AS. This initiates VCC AS call anchoring.
3. The VCC AS, which acts as a SIP B2BUA, determines that the call need to be delivered to IMS and generates the *INVITE* to the I/S-CSCF serving UE 1, and the I/S-CSCF sends the *INVITE* to UE 1 via the P-CSCF (not shown), in order to establish SIP call dialog 2 between the VCC AS and UE 1.
4. UE 1 returns a SIP *180 Ringing* message with the SDP of the bearer to the I/S-CSCF, via the P-CSCF (not shown), and the I/S-CSCF sends it to the VCC AS.

5. The VCC AS sends the *180 Ringing* message to the I/S-CSCF.

6. The I/S-CSCF sends the *180 Ringing* message to the OEP.

7. UE 1 sends a SIP *200 OK* message to the I/S-CSCF, via the P-CSCF (not shown), and the I/S-CSCF sends it to the VCC AS.

8. The VCC AS sends the *200 OK* message to the I/S-CSCF.

9. The I/S-CSCF sends the *200 OK* message to the OEP.

10. The OEP returns a SIP *ACK* message to the I/S-CSCF.

11. The I/S-CSCF sends the *ACK* to the VCC AS. This completes the establishment of SIP call dialog 1 between the OEP and the VCC AS.

12. Anytime after Step 7, the VCC AS sends the *ACK* message to UE 1 via the I/S-CSCF and the P-CSCF (not shown). This completes the establishment of SIP call dialog 2 between UE 1 and the VCC AS.

Post-conditions:

There is now voice call setup between UE 1 and the OEP via the IMS network. SIP call dialog 1 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the OEP. SIP call dialog 2 for this voice call is illustrated by a heavy dashed double arrow between UE 1 and the VCC AS. The voice bearer path is illustrated by heavy solid double arrows connecting the UE, PDSN/PDIF and the OEP.

### 4.4.3 MDN Homed on 1x CS Redirected to IMS using WIN Triggers

This scenario assumes that the ISUP call termination is routed to the home or gateway MSC. The call must be redirected to IMS for service processing and anchoring before being delivered to the VCC subscriber. Figure 10 illustrates the call flow that uses a WIN SCP to route the CS voice call to the VCC AS. The WIN SCP function may be performed by a separate entity, or optionally (not shown), it may be performed as an integrated function within the VCC AS.

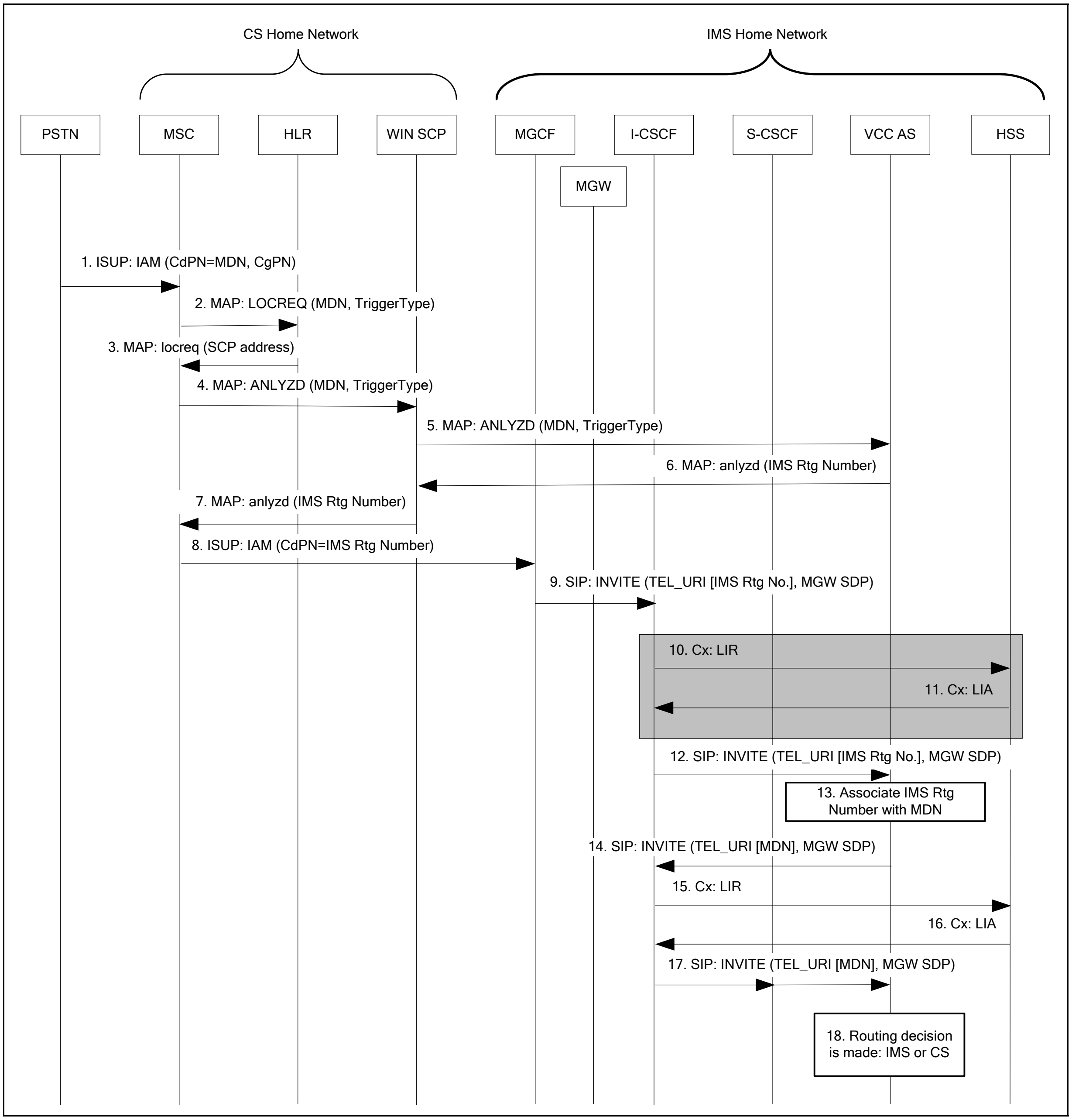


***Figure 10*** **MDN Homed on 1x CS Redirected to IMS using WIN Triggers**

1. A call termination message and the dialed UE address digits (i.e., mobile directory number - MDN) are received by the home MSC.

2. The MSC sends a MAP *LOCREQ* to the HLR to obtain a routing number. The *LOCREQ* messagae contains the dialed called party number (MDN) received in Step 1.

3. The HLR responds with a MAP *locreq* to the MSC. The *locreq* contains the address of the WIN SCP.

4. The MSC sends a MAP *ANLYZD* to the WIN SCP. The *ANLYZD* message contains the dialed called party number (MDN) received in Step 1. Optionally, the MSC may send the *ANLYZD* message directly to a VCC AS that has an integrated WIN SCP function and step 5 is skipped.

5. The WIN SCP forwards the *ANLYZD* message to the VCC AS.

6. The VCC AS stores the MDN and associates it with the subscriber. The VCC AS returns a MAP *anlyzd* message containing the IMS Routing Number, which is an E.164 temporary routing number associated with the VCC AS. The E.164 temporary routing number is locally associated with the called party as identified by the MDN. The E.164 temporary routing number is also a Public Service Identity (PSI) for the VCC AS that is homed in the IMS Home Network of the subscriber.

7. The WIN SCP returns the *anlyzd* message to the MSC. Step 7 is skipped if the VCC AS has an integrated WIN SCP function.

8. The MSC generates an ISUP *IAM* with the Called Party Number set to the IMS Routing Number. The *IAM* is routed to a MGCF within the IMS home network of the subscriber.

9. The MGCF sends a SIP *INVITE* request including a TEL URI generated from the IMS Routing Number and also containing the MGW-SDP to the configured I-CSCF.

10. Optionally, the I-CSCF sends a Cx *LIR* (*Location Information Request*) to the HSS to determine the routing information, i.e., the address of the AS hosting the VCC PSI.

11. In response to the *LIR*, the HSS returns a Cx *LIA* (*Location Information Answer*) containing the VCC AS address.

12. The I-CSCF forwards the *INVITE* with the TEL URI to the VCC AS.

13. The VCC AS uses the IMS Routing Number to make the association with the MDN received in the *ANLYZD* message (Step 5).

14. The VCC AS acting as a B2BUA creates an *INVITE* request containing MGW-SDP and the TEL URI of the UE, generated from the MDN. The VCC AS sends the INVITE to the I-CSCF for IMS processing.

15. The I-CSCF sends a *LIR* to the HSS.

16. In response to the *LIR*, the HSS returns a *LIA* containing the S-CSCF assigned to the user.

17. The I-CSCF sends the *INVITE* containing the TEL URI to the assigned S-CSCF. The S-CSCF using the iFC routes the *INVITE* to the VCC AS.

18. Upon receiving the *INVITE*, the VCC AS determines where the call request should be routed. The call flow then continues at step 3 of Figure 8 (if the call is to be delivered to the CS domain), or at step 3 of Figure 9 (if the call is to be delivered to the IMS domain).

### 4.4.4 MDN Homed on IMS using Local Number Portability

In this scenario, a UE with an MDN that was ported and rehomed in the IMS Domain. This option may be desirable in networks that support Number Portability.

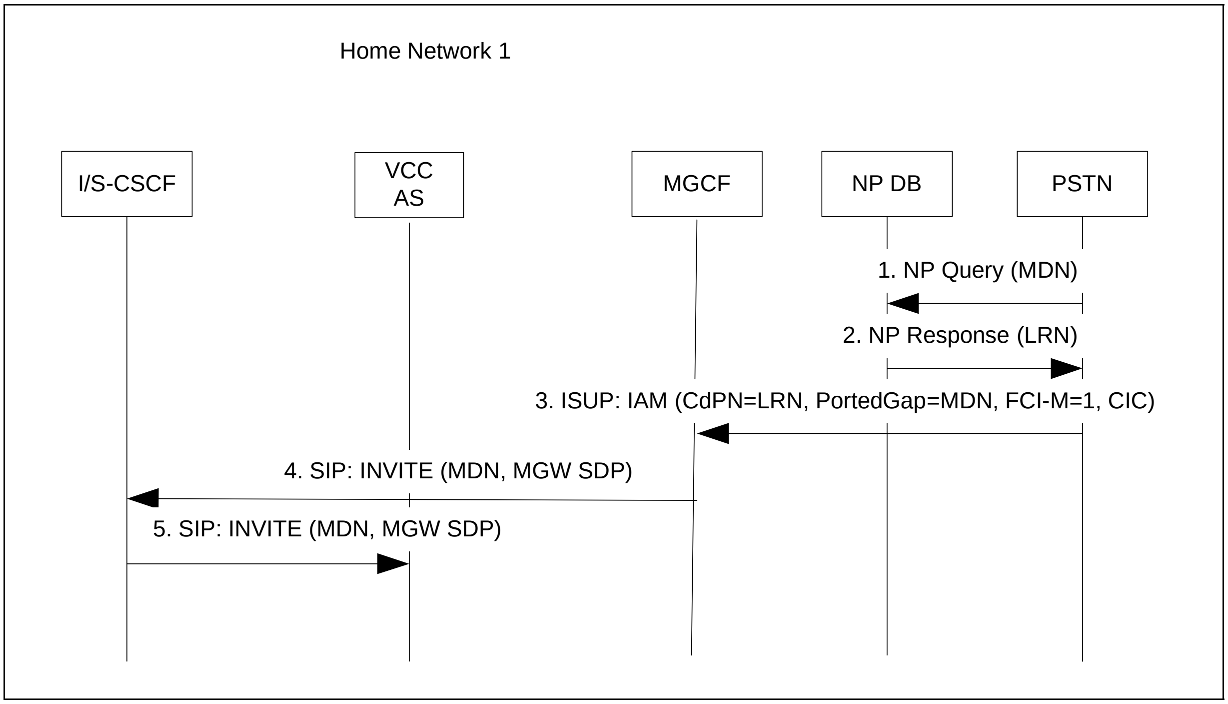


***Figure 11*** **MDN Homed on 1x CS Redirected to IMS using LNP**

1. PSTN queries the Number Portability database.
2. The response includes a Local Routing Number (LRN) that enables the call to be redirected to IMS.
3. The PSTN sends an ISUP *IAM* to the MGCF in the home network including the LRN, the original called MDN in the ISUP Ported Gap informational element and the FCI bit M set to indicate a ported DN.
4. The MGCF detects the ported indicator and uses the original called MDN in the Request URI of the of the SIP *INVITE* towards the I-CSCF where the HSS is queried for the UE's S-CSCF.
5. Based on iFC, the S-CSCF sends the *INVITE* message to VCC AS. The call flow then continues at step 3 of Figure 8 (if the call is to be delivered to the CS domain), or at step 3 of Figure 9 (if the call is to be delivered to the IMS domain).

### 4.4.5 MDN Homed on 1x CS Redirected to IMS without WIN support

This scenario assumes that the ISUP call termination is routed to the home or gateway MSC. The call must be redirected to IMS for service processing and anchoring before being delivered to the VCC subscriber. Figure 12 illustrates the call flow that uses HLR to route the CS voice call to the VCC AS.

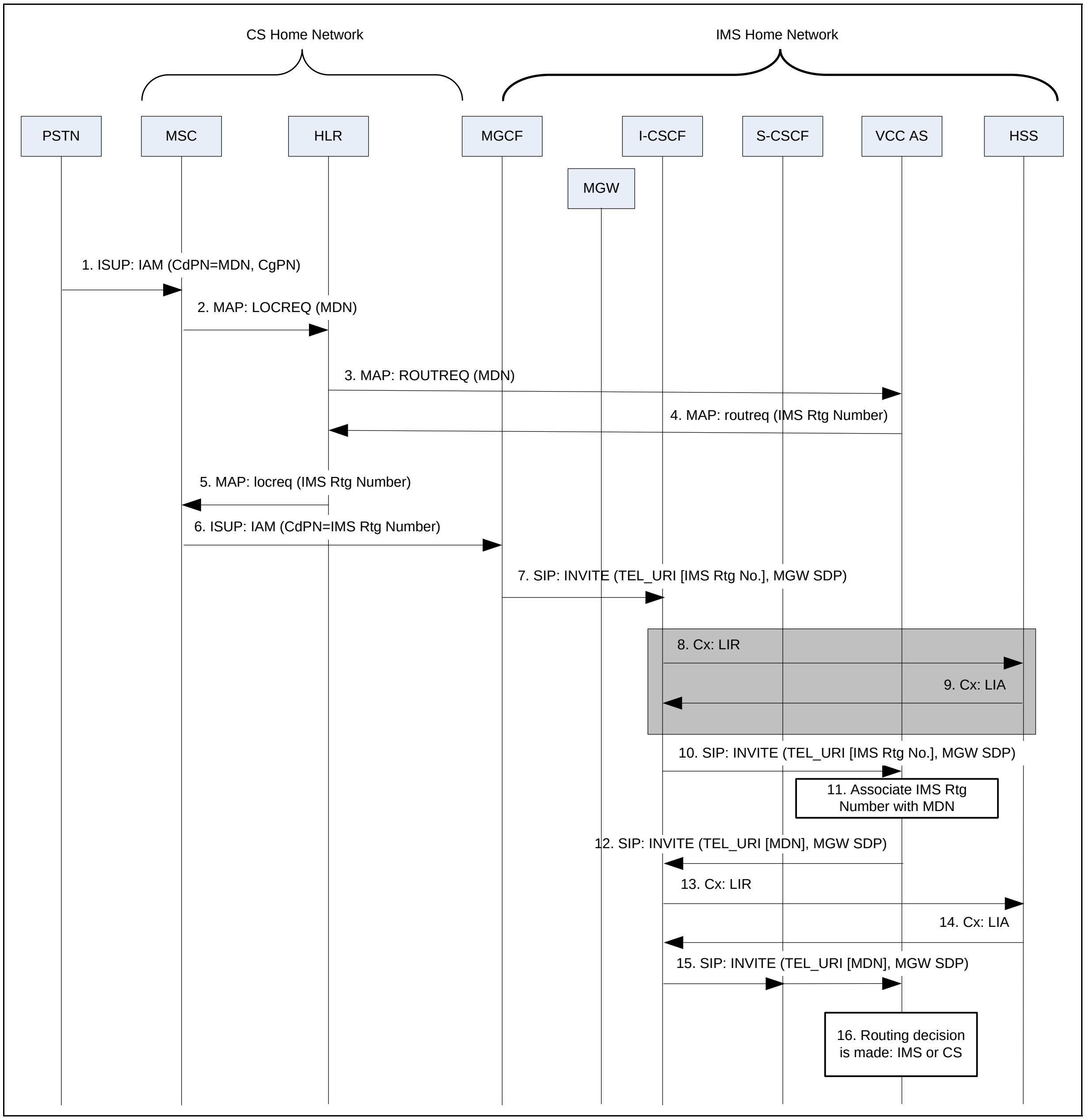


***Figure 12*** **MDN Homed on 1x CS Redirected to IMS without WIN**

1. A call termination message and the dialed UE address digits (i.e., mobile directory number - MDN) are received by the home MSC.

2. The MSC sends a MAP *LOCREQ* to the HLR to obtain a routing number. The *LOCREQ* message contains the dialed called party number (MDN) received in step 1.

3. The HLR sends the MAP *ROUTREQ* message to the VCC AS which contains the dialed Called Party Number (MDN).

   NOTE: If the subscriber is attached to the 1x CS network and the VCC AS wants to deliver the incoming call over the CS network, the VCC will send another *LOCREQ* message to the HLR (following step 16). The HLR behavior after getting these two *LOCREQ* messages differs based on the source address of these messages e.g., MSCID or MSCIN.

4. The VCC AS stores the MDN and associates it with the subscriber. The VCC AS returns a MAP *routreq* message containing the IMS Routing Number, which is an E.164 temporary routing number associated with the VCC AS. The E.164 temporary routing number is locally associated with the called party as identified by the MDN. The E.164 temporary routing number is also a Public Service Identity (PSI) for the VCC AS that is homed in the IMS Home Network of the subscriber.

5. The HLR returns the MAP *locreq* message to the MSC which contain the IMS Routing Number.

6. The MSC generates an ISUP *IAM* with the Called Party Number set to the IMS Routing Number. The IAM is routed to a MGCF within the IMS home network of the subscriber.

7. The MGCF sends a SIP *INVITE* request including a TEL URI generated from the IMS Routing Number and also containing the MGW-SDP to the configured I-CSCF.

8. Optionally, the I-CSCF sends a Cx *LIR* (*Location Information Request*) to the HSS to determine the routing information, i.e., the address of the AS hosting the VCC PSI.

9. In response to the *LIR*, the HSS returns a Cx *LIA* (*Location Information Answer*) containing the VCC AS address.

10. The I-CSCF forwards the *INVITE* with the TEL URI to the VCC AS.

11. The VCC AS uses the IMS Routing Number to make the association with the MDN received in the *ROUTREQ* message (Step 3).

12. The VCC AS acting as a B2BUA creates an *INVITE* request containing MGW-SDP and the TEL URI of the UE, generated from the MDN. The VCC AS sends the INVITE to the I-CSCF for IMS processing.

13. The I-CSCF sends a *LIR* to the HSS.

14. In response to the *LIR*, the HSS returns a *LIA* containing the S-CSCF assigned to the user.

15. The I-CSCF sends the *INVITE* containing the TEL URI to the assigned S-CSCF. The S-CSCF using the iFC routes the *INVITE* to the VCC AS.

16. Upon receiving the *INVITE*, the VCC AS determines where the call request should be routed. The call flow then continues at step 3 of Figure 8 (if the call is to be delivered to the CS domain), or at step 3 of Figure 9 (if the call is to be delivered to the IMS domain).

## 4.5 Single Radio Domain Transfer: VoIP-to-1x CS Voice

This section of the specification illustrates signaling flows for VoIP-to-1x CS voice single radio DT. The signaling shown between the UE and the MSC for the flows in this section does not represent actual signaling messages. One should look to the appropriate references to see the signaling details.

Figure 13 Figure 14 and Figure 15 illustrate the three-step process involved during the VoIP-to-1x CS voice single radio DT procedure.

In Step 1, the bearer path for UE is routed through the visiting network packet data network. Note that the VCC AS has already been put into the call flow signaling path during session setup as illustrated in Figure 5 or Figure 9 The P-CSCF can be either in the Visited Network for UE 1 or the Home Network for UE 1. Also, the SIP signaling between UE 1 and the P-CSCF traverses the packet data network but is shown as a dotted-line directly between UE 1 and the P-CSCF for brevity.

In Step 2, the MSC and MGCF in the Visited Network for UE 1 in conjunction with the VCC AS in the Home Network for UE 1 execute procedures to put the MGW in the Visited Network for UE 1 into the bearer path between UE 1 and the far end.

In Step 3, UE 1 performs a DT to the 1x CS network. Subsequently, the bearer path between UE 1 and the MSC in the Visited Network is routed between the MSC and the far end via the MGW in the Visited Network for UE 1.

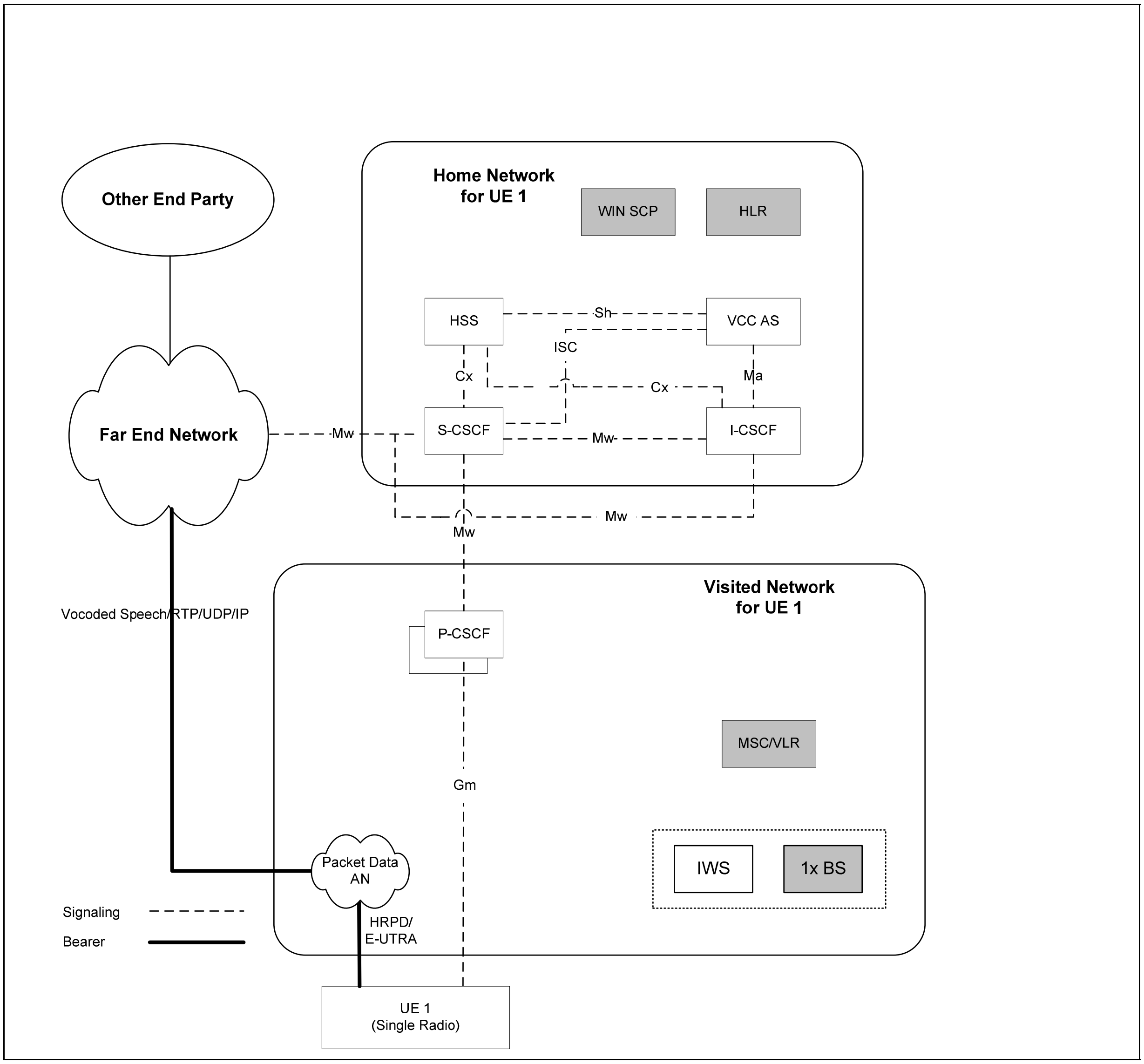


***Figure 13*** **VoIP-to-1x CS voice Single Radio DT - Step 1: Before MGW is put into path**

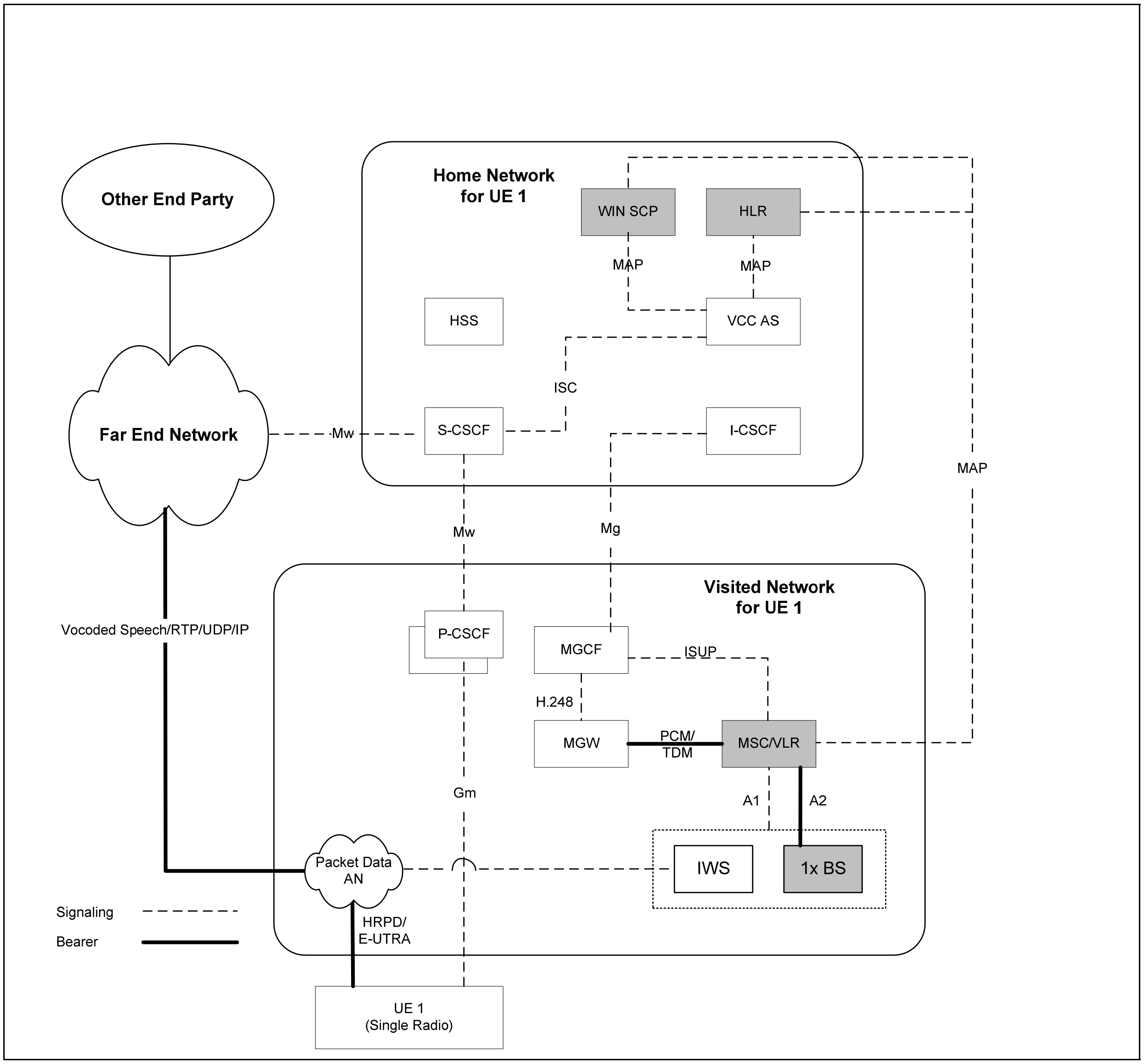


***Figure 14*** **VoIP-to-1x CS voice Single Radio DT - Step 2: Before DT to 1x**

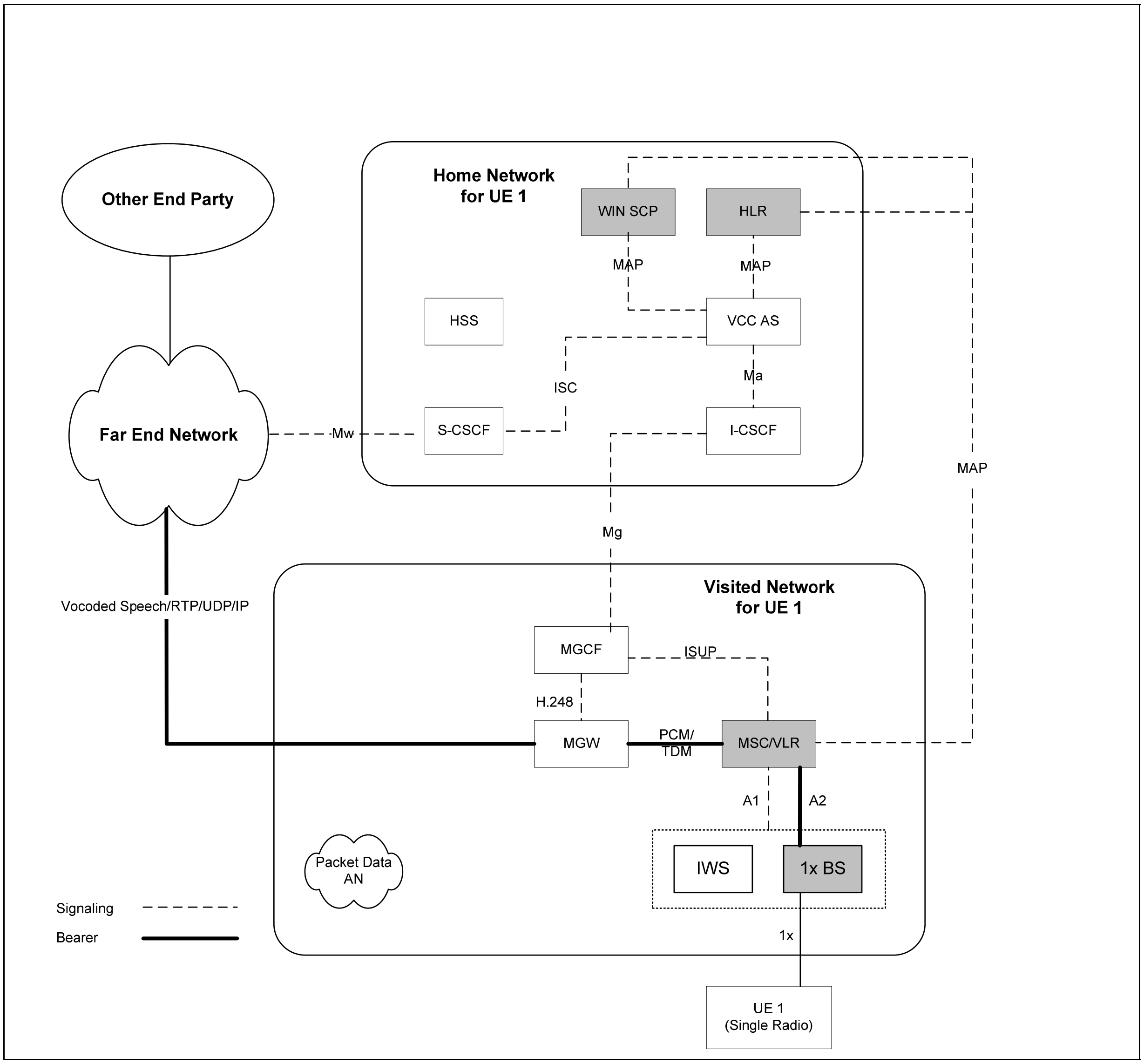


***Figure 15*** **VoIP-to-1x CS voice Single Radio DT - Step 3: After DT to 1x performed**

### 4.5.1 Single Radio VoIP-to-1x CS Voice DT

Figure 16 illustrates a detailed call flow for the single radio VoIP-to-1x CS voice DT procedure.

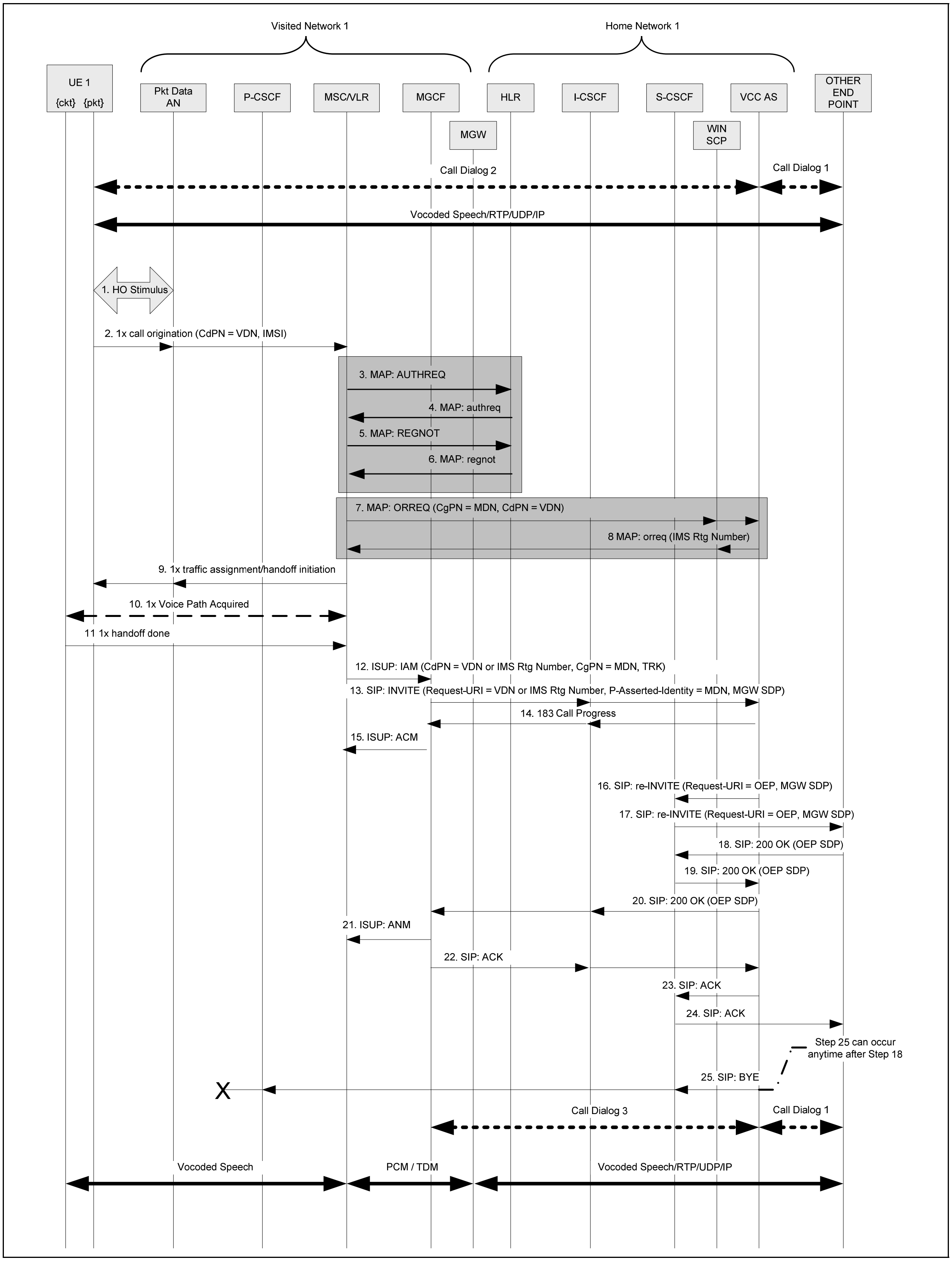


***Figure 16*** **Single Radio VoIP-to-1x CS voice DT**

Pre-condition:

It is assumed that initially there is an IMS VoIP call setup between a single radio, dual mode UE 1 and the Other End Point (OEP). SIP call dialog 1 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the OEP. SIP call dialog 2 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and UE 1. The voice bearer path is illustrated by a heavy solid double arrow between UE 1 and the OEP.

1. UE 1 and the packet data AN interact to initiate a DT. See [A.S0008] and [A.S0009] or SRVCC] for signaling details.

2. UE 1 sends a 1x *call origination* to the MSC/VLR via the packet data AN (and optionally, the 1x BS) and includes the VDN. The specific messages and any acknowledgements are not shown for brevity. See [A.S0008] and [A.S0009] or [SRVCC] for signaling details.

   NOTE 1: steps 3-6 are optional, depending on whether the UE 1 has previously been 1x CS registered and authenticated.

3. The Visited MSC/VLR may initiate a 1x registration procedure on behalf of UE 1. The Visited MSC sends a MAP *AUTHREQ* message to UE 1's HLR to authenticate UE 1 prior to allowing registration and prior to allocating a 1x traffic channel to UE 1.

4. UE 1's HLR responds by sending an MAP *authreq* message to the Visited MSC.

5. The Visited MSC sends an MAP *REGNOT* message to UE 1's HLR.

6. UE 1's HLR responds by sending an MAP *regnot* message to the Visited MSC.

   NOTE 2: Steps 7-8 are shown using the MAP *ORREQ* operation. Optionally, a post digit analysis trigger using the MAP *ANLYZD* operation may be used instead to obtain routing information for the DT.

   NOTE 3: If either origination triggers are not supported by the MSC/VLR or origination triggers are not armed for this subscriber, proceed to Step 9.

7. Once the visited MSC/VLR has obtained the service profile for the originating subscriber (i.e., by Step 4), the Visited MSC/VLR invokes a call origination trigger to obtain routing information. The Visited MSC/VLR sends a MAP *ORREQ* message to the WIN SCP (or to the HLR), containing the Calling Party Number (MDN) of UE 1 (derived from the IMSI) and the Called Party Number from the call origination. The WIN SCP (or HLR) sends the *ORREQ* message on to the VCC AS. Optionally, the Visited MSC/VLR may send the *ORREQ* message directly to a VCC AS that has an integrated WIN SCP function.

8. The VCC AS determines that this is a DT scenario based on the VDN in the Called Party Number (and the Calling Party Number) in the *ORREQ* message, and then allocates an IMS Routing Number, which is an E.164 temporary routing number associated with this DT. The VCC AS then sends back the MAP *orreq* message to WIN SCP (or HLR), which returns the *orreq* message to the MSC/VLR. Optionally, the VCC AS has an integrated WIN SCP function and sends the *orreq* message directly to the MSC/VLR.

9. Anytime after Step 2 the MSC/VLR sends a *1x traffic assignment/handoff initiation* to UE 1 via the packet data AN and the packet data air interface. This instructs UE 1 to perform the handoff and acquire the 1x traffic channel. See [A.S0008] and [A.S0009] or [SRVCC] for signaling details.

10. The 1x BS acquires UE 1's reverse traffic channel and the voice path is established with the MSC.

11. UE 1 responds by sending a 1x *handoff done* (via the 1x BS) to the MSC/VLR. See [A.S0008] and [A.S0009] or [SRVCC] for signaling details.

12. The Called Party Number is either the dialed digit string in the 1x *call origination* message (Step 2) or, in the event that origination triggering resulted in the allocation of an IMRN, the IMRN (Step 8). The translation of Called Party Number performed by the Visited MSC/VLR results in an ISUP *IAM* message being routed to either an MGCF in the serving network or to the PSTN, based upon operator policy. The Visited MSC includes the E.164 MDN of UE 1 in the Calling Party Number field of the *IAM*.

    After receiving the 1x *handoff done* message (Step 11), the MSC/VLR progress the call by sending the *IAM* towards the destination.

13. The Visited MGCF requests the MGW to create two terminations. The first termination is a TDM connection between the Visited MGW and the Visited MSC. The second termination is an RTP/UDP/IP ephemeral termination.

    The Visited MGCF sends a SIP *INVITE* message via the I-CSCF to the VCC AS containing the Request-URI, a P-Asserted-Identity, and an SDP offer. The Request-URI is based upon the *IAM* Called Party Number, the P-Asserted-Identity is based upon the *IAM* Calling Party Number, and the SDP offer is based upon the Visited MGW SDP information.

14. The VCC AS examines the P-Asserted-Identity header of the *INVITE* (see Step 13) to determine which subscriber is performing the VoIP-to-1x CS voice DT. Note that the VCC AS has already been put into the call flow signaling path during session setup. If a SIP dialog has been established between the user identified in the P-Asserted-Identity and a remote user the VCC AS sends a 183 Call Progress to the Visiting MGCF.

15. The Visiting MGCF [X.S0050] creates and returns an ISUP *ACM* message to the MSC/VLR.

16. If a SIP dialog has been established between the user identified in the P-Asserted-Identity (see Step 13) and a remote user, the VCC AS determines the OEP of the ongoing call. The VCC AS creates a SIP *re-INVITE* message with the Request-URI set to the OEP and an SDP offer based upon the Visited MGW SDP information. The VCC AS sends the *re-INVITE* to the S-CSCF.

17. The S-CSCF forwards the *re-INVITE* to the far end network.

18. The Other End Point (OEP) (IMS user or PSTN MGCF/MGW) modifies its RTP bearer termination with the Visited MGW SDP and responds with a SIP *200 OK* message to the S-CSCF containing an SDP answer with the OEP SDP

19. The S-CSCF forwards the *200 OK* to the VCC AS.

20. The VCC AS sends a *200 OK* message via the I-CSCF to the Visited MGCF containing an SDP answer with the OEP SDP information.

21. The Visited MGCF requests modification of the Visited MGW ephemeral termination with the OEP SDP information and instructs the Visited MGW to reserve/commit Remote resources. The Visited MGCF sends an ISUP *ANM* message.

22. Anytime after the *200 OK* is received, the Visited MGCF sends a SIP *ACK* message via the I-CSCF to the VCC AS. This completes the establishment of SIP call dialog 3 between the MGCF and the VCC AS. The VCC AS records the location of UE 1 as present in the 1x CS domain.

23. Anytime after the VCC AS receives the *200 OK*, it sends an *ACK* message to the S-CSCF.

24. The S-CSCF forwards the *ACK* to the OEP.

25. Anytime after Step 18, the VCC AS sends a SIP *BYE* message to UE 1 via the S-CSCF to release SIP call dialog 2 between UE 1 and the VCC AS. Note that since UE 1 is already on a 1x traffic channel it will not respond to this *BYE*.

Post-conditions:

There is now voice call setup between UE 1 and the OEP via the 1x CS network. SIP call dialog 1 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the OEP. SIP call dialog 3 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the MGCF. The voice bearer path is illustrated by heavy solid double arrows connecting the UE 1, MSC, MGW and the OEP.

## 4.5.2 Single Radio VoIP-to-1x CS voice DT (PD Indicator)

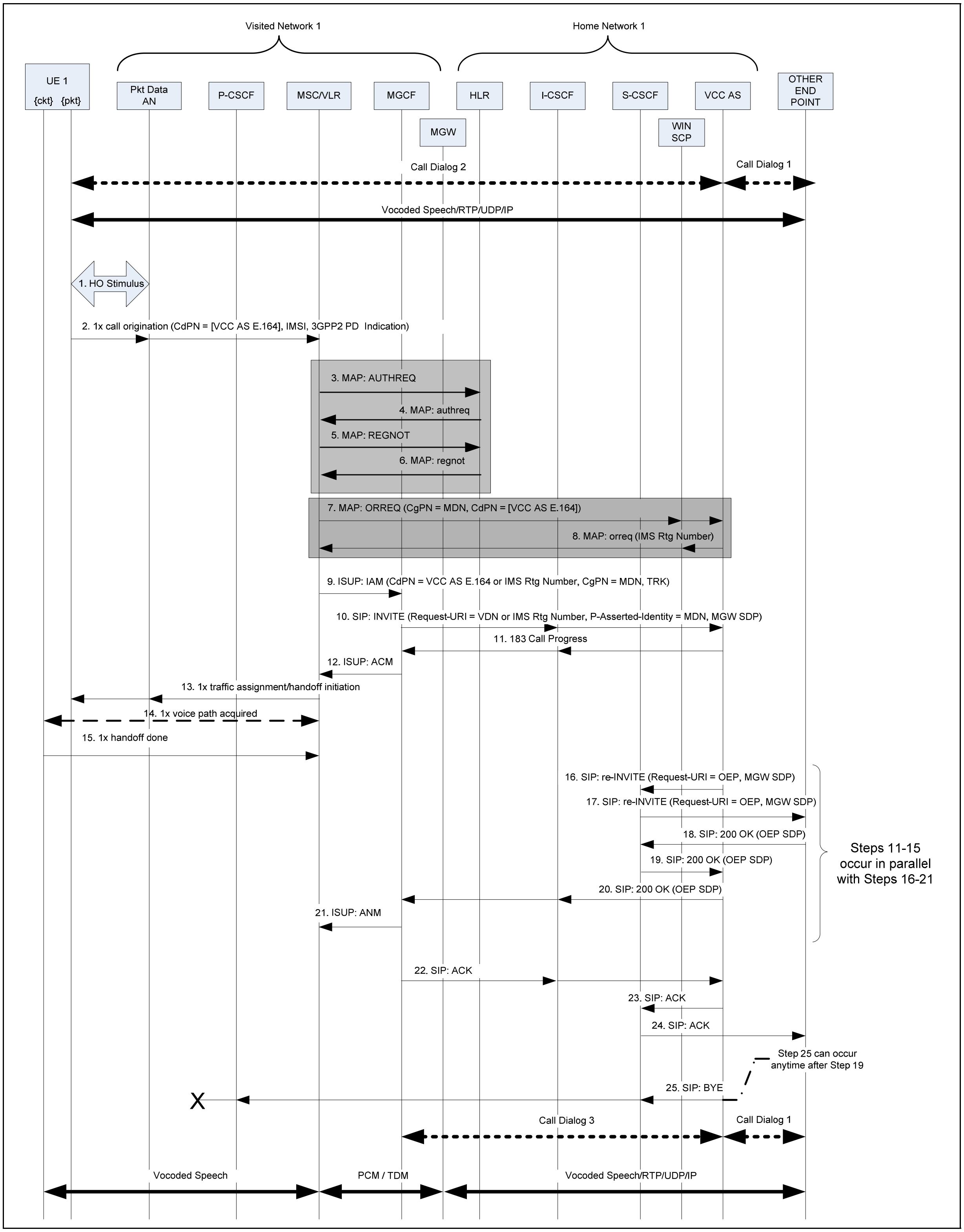


***Figure 17*** **Single Radio VoIP-to-1x CS voice DT (PD Indicator)**

Pre-conditions:

It is assumed that initially there is an IMS VoIP call setup between UE 1 and the Other End Point (OEP). SIP call dialog 1 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the OEP. SIP call dialog 2 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and UE 1. The voice bearer path is illustrated by a heavy solid double arrow between UE 1 and the OEP.

1. UE 1 and the packet data AN interact to initiate a DT. See [A.S0008] and [A.S0009] or [SRVCC] for signaling details.

2. UE 1 sends a 1x *call origination*[1] to the MSC/VLR via the packet data AN and IWS (and optionally, the 1x BS) and includes the VDN and an optional PD indicator. The specific messages and any acknowledgements are not shown for brevity. See [A.S0008] and [A.S0009] or [SRVCC] for signaling details.

   Note, steps 3-6 are optional, depending on whether the UE 1 has previously been 1x CS registered and authenticated.

3. The Visited MSC/VLR may initiate a 1x registration procedure on behalf of UE 1. The Visited MSC sends an MAP *AUTHREQ* message to UE 1's HLR to authenticate UE 1 prior to allowing registration and prior to allocating a 1x traffic channel to UE 1.

4. UE 1's HLR responds by sending an MAP *authreq* message to the Visited MSC.

5. The Visited MSC sends an MAP *REGNOT* message to UE 1's HLR.

6. UE 1's HLR responds by sending an MAP *regnot* message to the Visited MSC.

   Note: Steps 7-8 are shown using the MAP *ORREQ* operation. Optionally, a post digit analysis trigger using the MAP *ANLYZD* operation may be used instead to obtain routing information for the DT.

   Note 2: If either origination triggers are not supported by the MSC/VLR or origination triggers are not armed for this subscriber, proceed to Step 9.

7. Once the visited MSC/VLR has obtained the service profile for the originating subscriber (i.e., by Step 4), the Visited MSC/VLR invokes a call origination trigger to obtain routing information. The Visited MSC/VLR sends a MAP *ORREQ* message to the WIN SCP (or to the HLR), containing the Calling Party Number (MDN) of UE 1 (derived from the IMSI) and the Called Party Number from the call origination. The WIN SCP (or HLR) sends the *ORREQ* message on to the VCC AS. Optionally, the Visited MSC/VLR may send the *ORREQ* message directly to a VCC AS that has an integrated WIN SCP function.

8. The VCC AS determines that this is a DT scenario based on the VDN in the Called Party Number (and the Calling Party Number) in the ORREQ message, and then allocates an IMS Routing Number, which is an E.164 temporary routing number associated with this DT. The VCC AS then sends back the MAP *orreq* message to WIN SCP (or HLR),

---

[1] The CSNA protocol is used to deliver a 1x *Call Origination* message from UE 1 via the HRPD AN.

which returns the *orreq* message to the MSC/VLR. Optionally, the VCC AS has an integrated WIN SCP function and sends the *orreq* message directly to the MSC/VLR..

9. The MSC creates an ISUP IAM. The Called Party Number is either the dialed digit string in the 1x *call origination* message (Step 2) or, in the event that origination triggering resulted in the allocation of an IMRN, the IMRN (Step 8). The Visited MSC includes the E.164 MDN of UE 1 in the Calling Party Number field of the ISUP IAM. The translation of Called Party Number performed by the Visited MSC/VLR results in an ISUP *IAM* message being routed to either an MGCF in the serving network or to the PSTN, based upon operator policy.

10. The Visited MGCF requests the MGW to create two terminations. The first termination is a TDM connection between the Visited MGW and the Visited MSC. The second termination is an RTP/UDP/IP ephemeral termination.

    The Visited MGCF sends a SIP *INVITE* message via the I-CSCF to the VCC AS containing the Request-URI, a P-Asserted-Identity, and an SDP offer. The Request-URI is based upon the *IAM* Called Party Number, the P-Asserted-Identity is based upon the *IAM* Calling Party Number, and the SDP offer is based upon the Visited MGW SDP information.

    NOTE: Steps 11-15 occur in parallel to steps 16-21.

11. The VCC AS examines the P-Asserted-Identity header of the *INVITE* (see Step 10) to determine which subscriber is performing the VoIP-to-1x CS voice DT. Note that the VCC AS has already been put into the call flow signaling path during session setup. If a SIP dialog has been established between the user identified in the P-Asserted-Identity and a remote user the VCC AS sends a SIP *183 Call Progress* to the Visiting MGCF

12. The Visiting MGCF [X.S0050] creates and returns an ISUP *ACM* message to the MSC/VLR.

13. In parallel with sending the *IAM* (step 9), or after receiving the *ACM* (step 12) [X.S0004-630], the MSC/VLR sends a 1x *traffic assignment/handoff initiation* to UE 1 via the packet data AN and the packet data air interface. This instructs UE 1 to perform the handoff and acquire the 1x traffic channel. See [A.S0008] and [A.S0009] or [SRVCC] for signaling details.

14. The 1x BS acquires UE 1's reverse traffic channel and the voice path is established with the MSC.

15. UE 1 responds by sending a 1x *handoff done* (via the 1x BS) to the MSC/VLR. See [A.S0008] and [A.S0009] or [SRVCC] for signaling details.

16. Based upon the P-Asserted-ID (see Step 10), the VCC AS determines the OEP of the ongoing call. The VCC AS creates a SIP *re-INVITE* message with the Request-URI set to the OEP and an SDP offer based upon the Visited MGW SDP information. The VCC AS sends the *re-INVITE* to the S-CSCF.

17. The S-CSCF forwards the *re-INVITE* to the far end network.

18. The Other End Point (OEP) (IMS user or PSTN MGCF/MGW) modifies its RTP bearer termination with the Visited MGW SDP and responds with a SIP *200 OK* message to the S-CSCF containing an SDP answer with the OEP SDP

19. The S-CSCF forwards the *200 OK* to the VCC AS.

20. The VCC AS sends a *200 OK* message via the I-CSCF to the Visited MGCF containing an SDP answer with the OEP SDP information.

21. The Visited MGCF requests modification of the Visited MGW ephemeral termination with the OEP SDP information and instructs the Visited MGW to reserve/commit Remote resources. The Visited MGCF sends an ISUP *ANM* message.

22. Anytime after the *200 OK* is received, the Visited MGCF sends a SIP *ACK* message via the I-CSCF to the VCC AS. This completes the establishment of SIP call dialog 3 between the MGCF and the VCC AS. The VCC AS records the location of UE 1 as present in the 1x CS domain.

23. Anytime after the VCC AS receives the *200 OK*, it sends a *ACK* message to the S-CSCF.

24. The S-CSCF forwards the *ACK* to the OEP.

25. Anytime after Step 19, the VCC AS sends a SIP *BYE* message to UE 1 via the S-CSCF to release SIP call dialog 2 between UE 1 and the VCC AS. Note that since UE 1 is already on a 1x traffic channel it will not respond to this *BYE*.

Post-conditions:

There is now voice call setup between UE 1 and the OEP via the 1x CS network. SIP call dialog 1 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the OEP. SIP call dialog 3 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the MGCF. The voice bearer path is illustrated by heavy solid double arrows connecting the UE 1, MSC, MGW and the OEP.

# 4.6 Dual Radio Domain Transfer: VoIP-to-1x CS Voice

This section of the specification illustrates signaling flows for dual mode, dual radio VoIP-to-1x CS voice DT. The signaling shown between the UE and the MSC for the flows in this section does not represent actual signaling messages. One should look to the appropriate references to see the signaling details.

Figure 18 Figure 19 and Figure 20 illustrate the three-step process involved during the DUAL RADIO VoIP-to-1x CS voice DT procedure.

In Step 1, the bearer path between UE 1 and the far end is routed between the packet data access network in the Visited Network for UE 1 and the far end network. Note that the VCC AS has already been put into the call flow signaling path during session setup as illustrated in Figure 5 or Figure 9  The P-CSCF can be either in the Visited Network for UE 1 or the Home Network for UE 1. Also, the SIP signaling between UE 1 and the P-CSCF traverses the packet data access network but is shown as a dotted-line directly between UE 1 and the P-CSCF for brevity.

In Step 2, the MSC and MGCF in the Visited Network for UE 1 in conjunction with the VCC AS in the Home Network for UE 1 execute procedures to put the MGW in the Visited Network for UE 1 into the bearer path between UE 1 and the far end.

In Step 3, UE 1 performs a DT to the 1x CS network. Subsequently, the bearer path between UE 1 and the MSC in the Visited Network for UE 1 is routed between the MSC and the far end via the MGW in the Visited Network for UE 1.

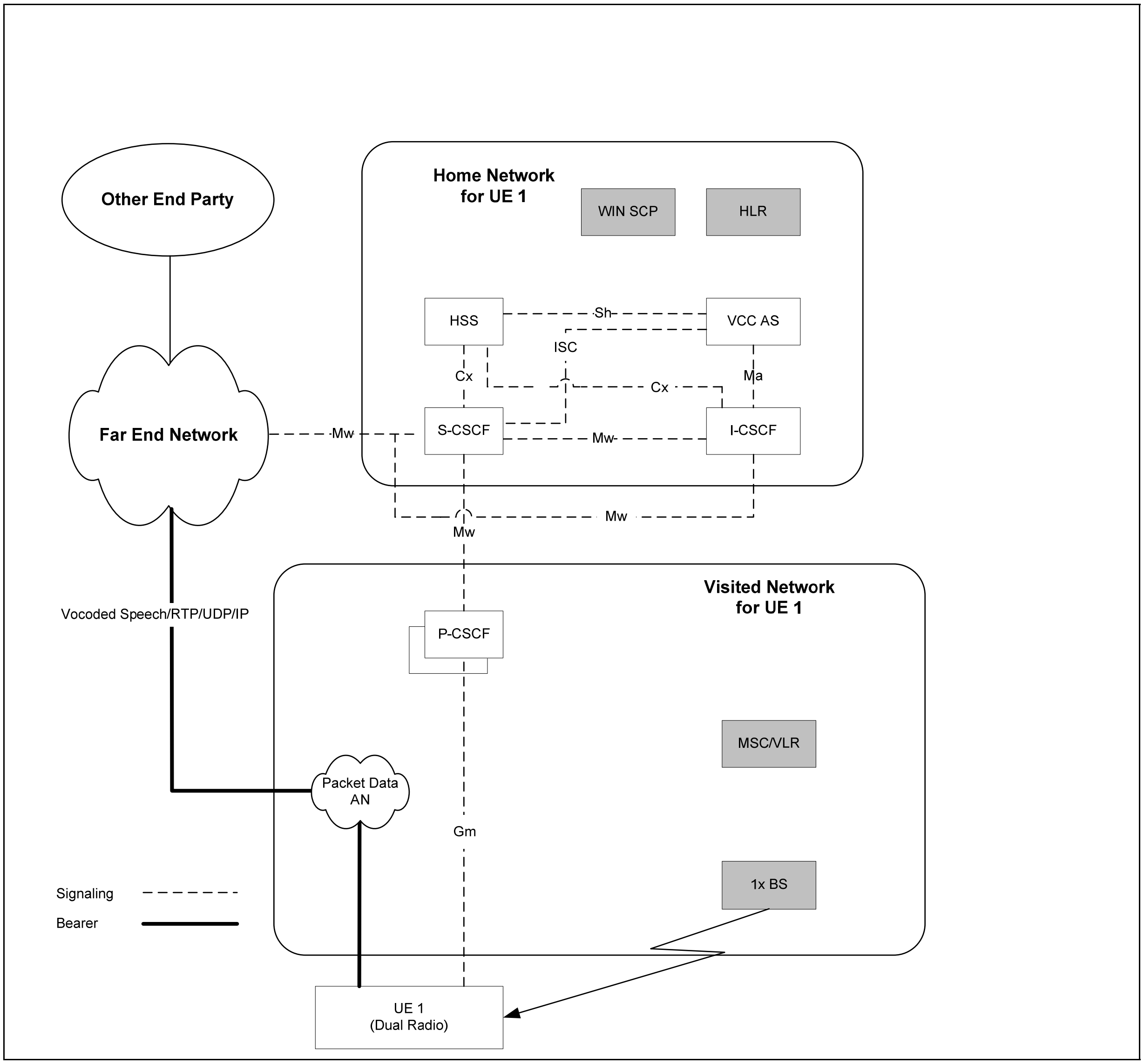


***Figure 18*** **Dual Radio VoIP-to-1x CS voice DT – Step 1: Before MGW is put into path**

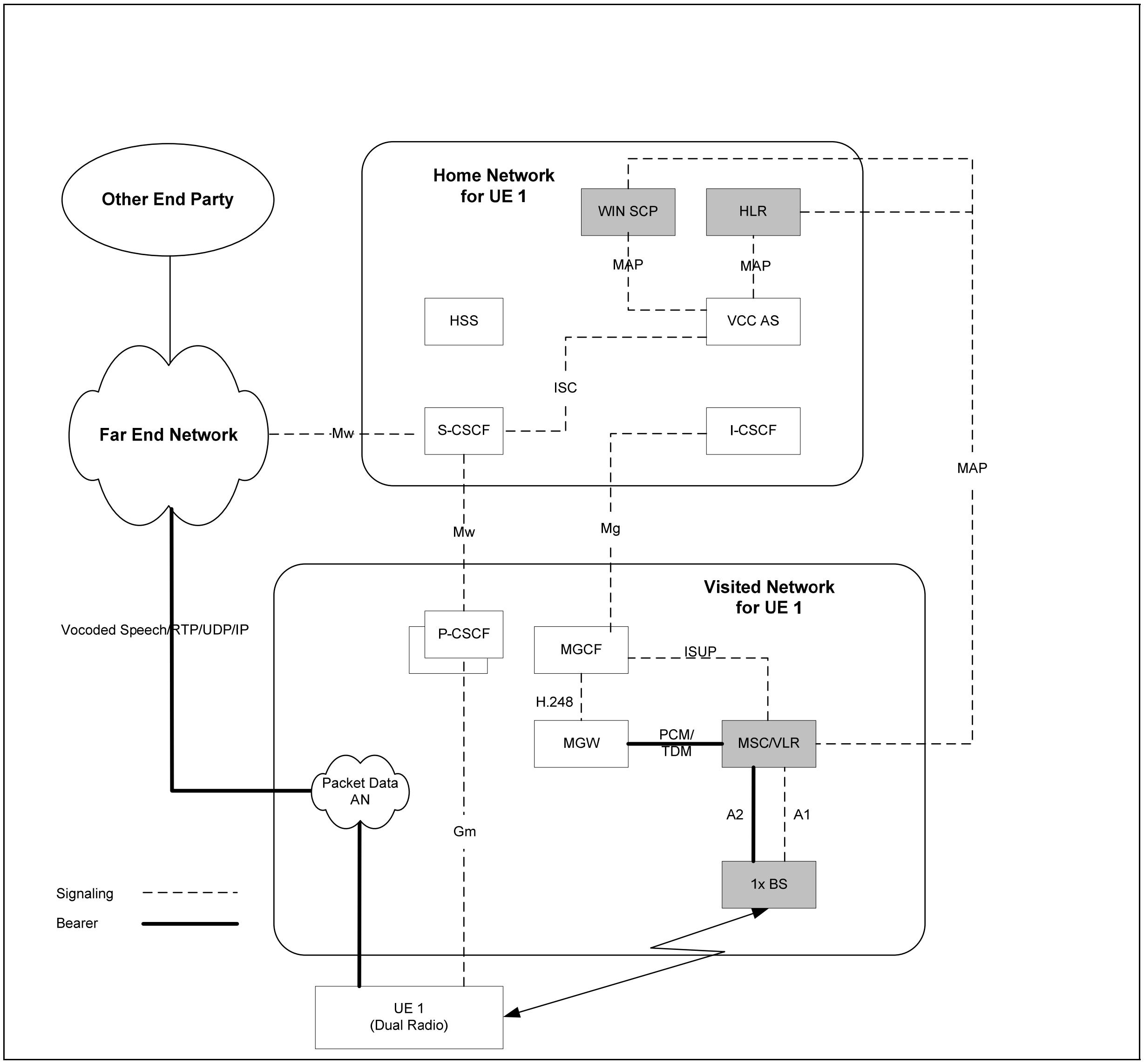


***Figure 19*** **Dual Radio VoIP-to-1x CS voice DT – Step 2: Before DT to 1x**

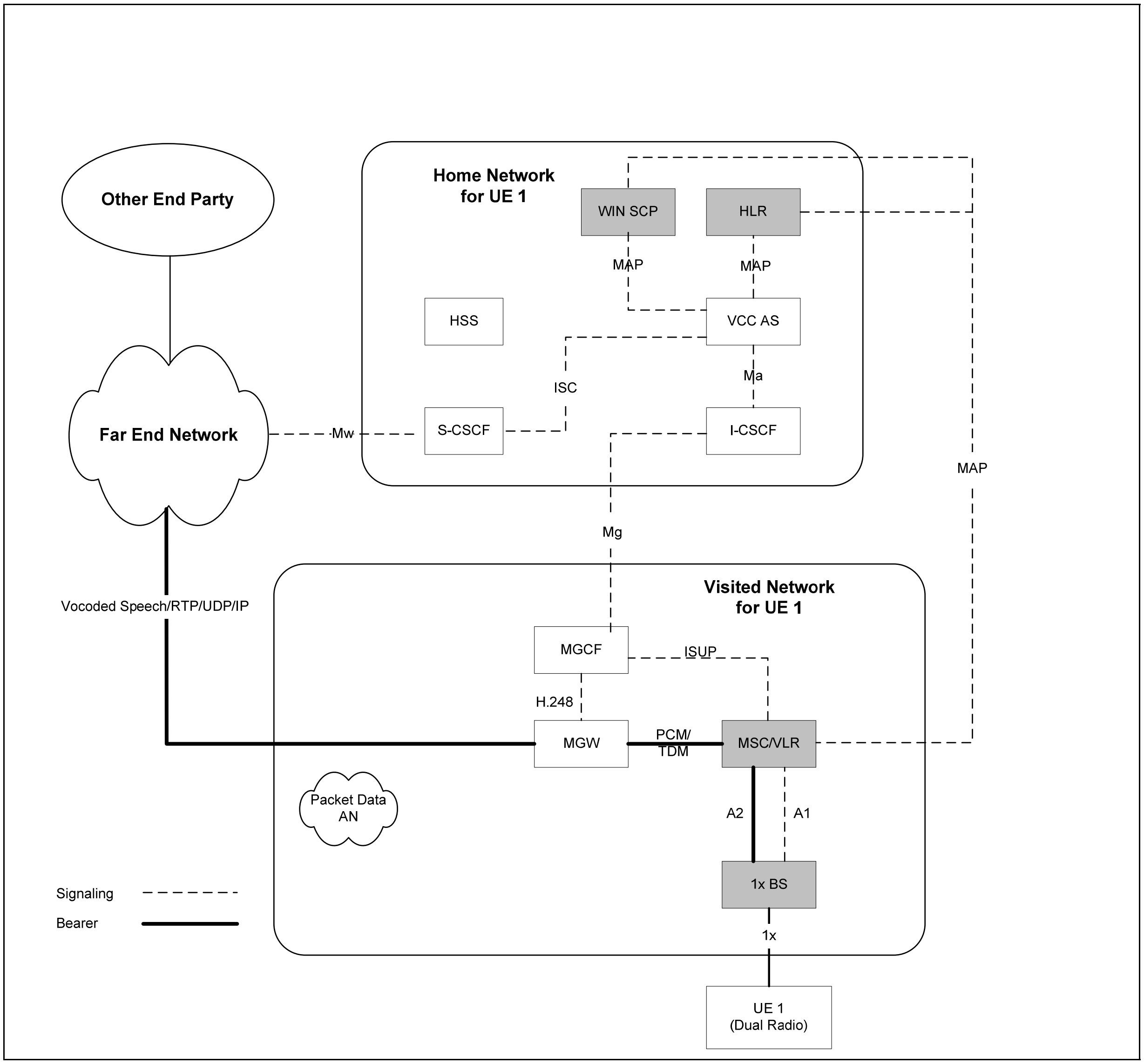


***Figure 20*** **Dual Radio VoIP-to-1x CS voice DT – Step 3: After DT to 1x performed**

### 4.6.1 Dual Radio VoIP-to-1x CS Voice DT

Figure 21 illustrates a detailed call flow for the Dual Radio VoIP-to-1x CS voice DT procedure.

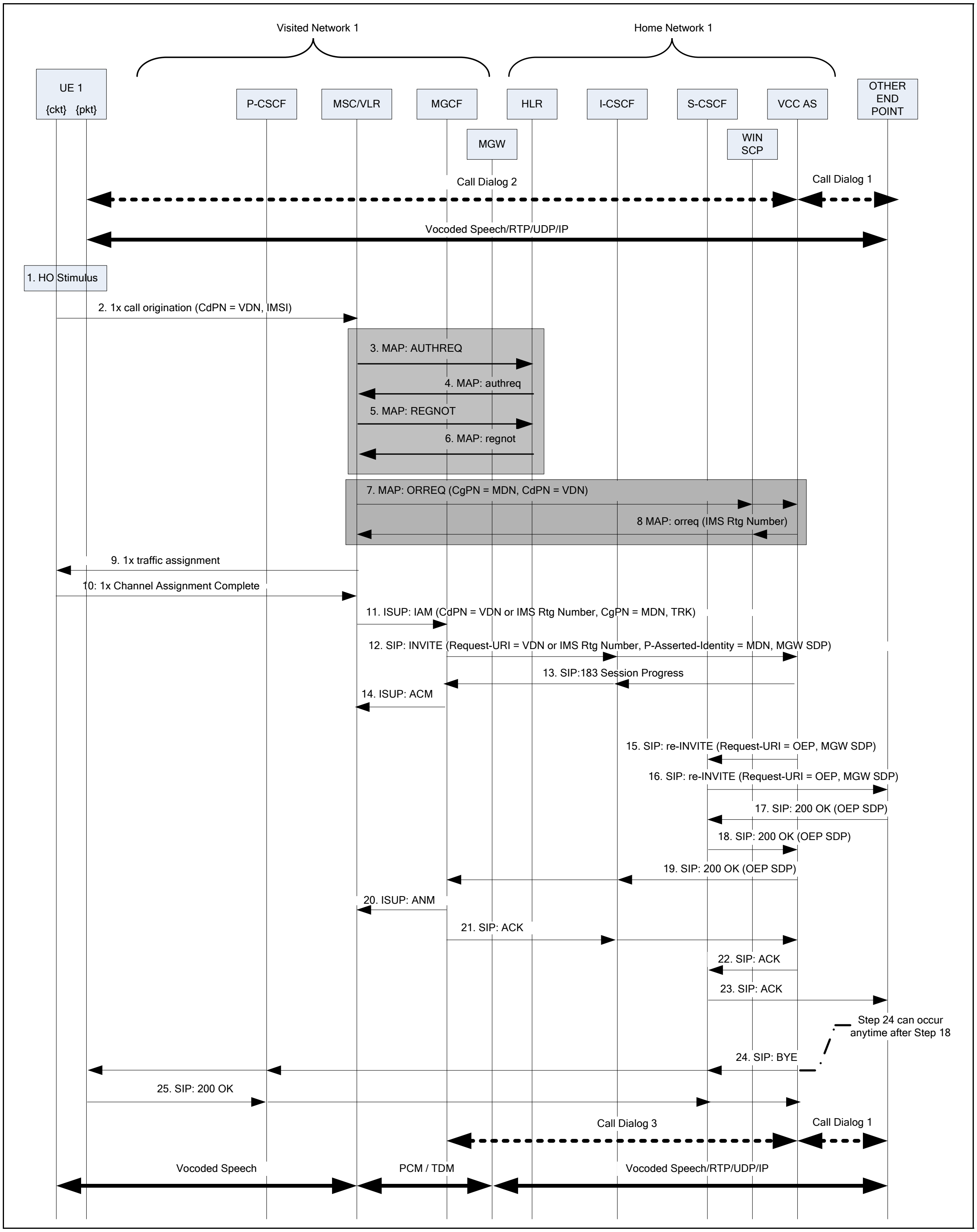


***Figure 21*** **Dual Radio VoIP-to-1x CS voice DT**

Pre-conditions:

It is assumed that initially there is an IMS VoIP call setup between the UE and the Other End Point (OEP). SIP call dialog 1 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the OEP. SIP call dialog 2 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the UE. The voice bearer path is illustrated by a heavy solid double arrow between the UE and the OEP.

1. UE 1 detects that a handoff from packet data VoIP to 1x CS is required. How UE 1 determines this is outside the scope of this document.

2. UE 1 sends a 1x *call origination* to the MSC/VLR (via the 1x BS) and includes the VDN. See [C.S0005] for the signaling details.

   NOTE 1: steps 3-6 are optional, depending on whether the UE 1 has previously been 1x CS registered and authenticated.

3. The Visited MSC/VLR may initiate a 1x registration procedure on behalf of UE 1. The Visited MSC sends an MAP *AUTHREQ* message to UE 1’s HLR to authenticate UE 1 prior to allowing registration and prior to allocating a 1x traffic channel to UE 1.

4. UE 1’s HLR responds by sending an MAP *authreq* message to the Visited MSC.

5. The Visited MSC sends an MAP *REGNOT* message to UE 1’s HLR.

6. UE 1’s HLR responds by sending an MAP *regnot* message to the Visited MSC.

   NOTE 2: Steps 7-8 are shown using the MAP *ORREQ* operation. Optionally, a post digit analysis trigger using the MAP *ANLYZD* operation may be used instead to obtain routing information for the DT.

   NOTE 3: If either origination triggers are not supported by the MSC/VLR or origination triggers are not armed for this subscriber, proceed to Step 9.

7. Once the visited MSC/VLR has obtained the service profile for the originating subscriber (i.e., by Step 3), the Visited MSC/VLR invokes a call origination trigger to obtain routing information. The Visited MSC/VLR sends a MAP *ORREQ* message to the WIN SCP (or to the HLR), containing the Calling Party Number (MDN) of UE 1 (derived from the IMSI) and the Called Party Number from the call origination. The WIN SCP (or HLR) sends the *ORREQ* message on to the VCC AS. Optionally, the Visited MSC/VLR may send the *ORREQ* message directly to a VCC AS that has an integrated WIN SCP function.

8. The VCC AS determines that this is a DT scenario based on the VDN in the Called Party Number (and the Calling Party Number) in the *ORREQ* message, and then allocates an IMS Routing Number, which is an E.164 temporary routing number associated with this DT. The VCC AS then sends back a MAP *orreq* message to WIN SCP (or HLR), which returns the *orreq* message to the MSC/VLR. Optionally, the VCC AS has an integrated WIN SCP function and sends the *orreq* message directly to the MSC/VLR.

9. Anytime after Step 2 and before Step 11, the Visited MSC sends a 1x *traffic assignment* (via the 1x BS) to UE 1. See [C.S0005] for the signaling details.

10. The 1x BS acquires UE 1’s reverse traffic channel and the voice path is established with the MSC/VLR. See [C.S0005] for the signaling details

11. The Called Party Number is either the dialed digit string in the 1x call origination message (Step 2) or, in the event that origination triggering resulted in the allocation of an IMRN, the IMRN from (Step 8). The translation of Called Party Number performed by the Visited MSC/VLR results in an ISUP *IAM* message being routed to either an MGCF in the serving network or to the PSTN, based upon operator policy. The Visited MSC includes the E.164 MDN of UE 1 in the Calling Party Number field of the *IAM*.

12. The Visited MGCF requests the MGW to create two terminations. The first termination is a TDM connection between the Visited MGW and the Visited MSC. The second termination is an RTP/UDP/IP ephemeral termination.

    The Visited MGCF sends a SIP *INVITE* message via the I-CSCF to the VCC AS containing the Request-URI, a P-Asserted-Identity, and an SDP offer. The Request-URI is based upon the *IAM* Called Party Number, the P-Asserted-Identity is based upon the *IAM* Calling Party Number, and the SDP offer is based upon the Visited MGW SDP information.

13. The VCC AS examines the P-Asserted-Identity header of the *INVITE* (see Step 12) to determine which subscriber is performing the VoIP-to-1x CS voice DT. Note that the VCC AS has already been put into the call flow signaling path during session setup. If a SIP dialog has been established between the user identified in the P-Asserted-Identity and a remote user the VCC AS sends a SIP *183 Session Progress* to the Visiting MGCF.

14. The Visiting MGCF [X.S0050] creates and returns an ISUP *ACM* message to the MSC/VLR.

15. If a SIP dialog has been established between the user identified in the P-Asserted-Identity (see Step 12) and a remote user, the VCC AS determines the OEP of the ongoing call. The VCC AS creates a SIP *re-INVITE* message with the Request-URI set to the OEP and an SDP offer based upon the Visited MGW SDP information. The VCC AS sends the *re-INVITE* to the S-CSCF.

16. The S-CSCF forwards the *re-INVITE* to the far end network.

17. The OEP (IMS user or PSTN MGCF/MGW) modifies its RTP bearer termination with the Visited MGW SDP and responds with a SIP *200 OK* message to the S-CSCF containing an SDP answer with the OEP SDP information.

18. The S-CSCF forwards the *200 OK* to the VCC AS.

19. The VCC AS sends a *200 OK* message via the I-CSCF to the Visited MGCF containing an SDP answer with the OEP SDP information.

20. The Visited MGCF requests modification of the Visited MGW ephemeral termination with the OEP SDP information and instructs the Visited MGW to reserve/commit Remote resources. The Visited MGCF sends an ISUP *ANM* message to the Visited MSC.

21. Anytime after the *200 OK* is received, the Visited MGCF sends a SIP *ACK* message via the I-CSCF to the VCC AS. This completes the establishment of SIP call dialog 3 between the MGCF and the VCC AS. The VCC AS records the location of UE 1 as present in the 1x CS domain.

22. Anytime after Step 21, the VCC AS sends a *ACK* message to the S-CSCF.

23. The S-CSCF forwards the *ACK* to the OEP.

24. Anytime after Step 21, the VCC AS sends a SIP *BYE* message to UE 1 via the S-CSCF to release SIP call dialog 2 between UE 1 and the VCC AS.

25. UE 1 responds to the VCC AS with a *200 OK*.

Post-conditions:

There is now voice call setup between UE 1 and the OEP via the 1x CS network. SIP call dialog 1 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the OEP. SIP call dialog 3 for this voice call is illustrated by a heavy dashed double arrow between the VCC AS and the MGCF. The voice bearer path is illustrated by heavy solid double arrows connecting the UE, MSC, MGW and the OEP.

## 4.7 Dual Radio Domain Transfer: 1x CS Voice to VoIP

This section of the specification illustrates signaling flows for 1x CS voice to-VoIP DT for a dual radio UE.

UE 1 may initiate a DT to IMS VoIP for a variety of reasons such as a degrading 1x CS radio environment or UE entered a VoIP capable packet data radio environment and IMS is preferred.

For 1x CS voice calls to be able to DT to IMS VoIP, the VCC AS must have been anchored into the call when initially setup (see Call Delivery to 1x CS Voice and 1x CS Call Origination scenarios for examples of this pre-condition). It is assumed that initially SIP call dialog 2 has been established between the VCC AS and the OEP, and SIP call dialog 1 has been established between the VCC AS and the MGCF.

Figure 22 Figure 23 and Figure 24 illustrate the three-step process involved during the 1x CS voice to-DUAL RADIO VoIP DT procedure.

In Step 1, bearer path between UE 1 and the MSC in the Visited Network for UE 1 is routed between the MSC and the far end via the MGW in the Visited Network for UE 1. Note that the VCC AS has already been put into the call flow signaling path during 1x CS voice call setup, e.g., as illustrated in Section 4.3.2.

In Step 2, UE 1 originates a VoIP call via IMS to the VCC AS. The VSS AS correlates this call with the original 1x CS voice call.

In Step 3, the bearer path between UE 1 and the far end is routed between the packet data AN in the Visited Network for UE 1 and the far end network. The 1x CS bearer via the MSC, MGW and far end network is removed and the DT is completed.

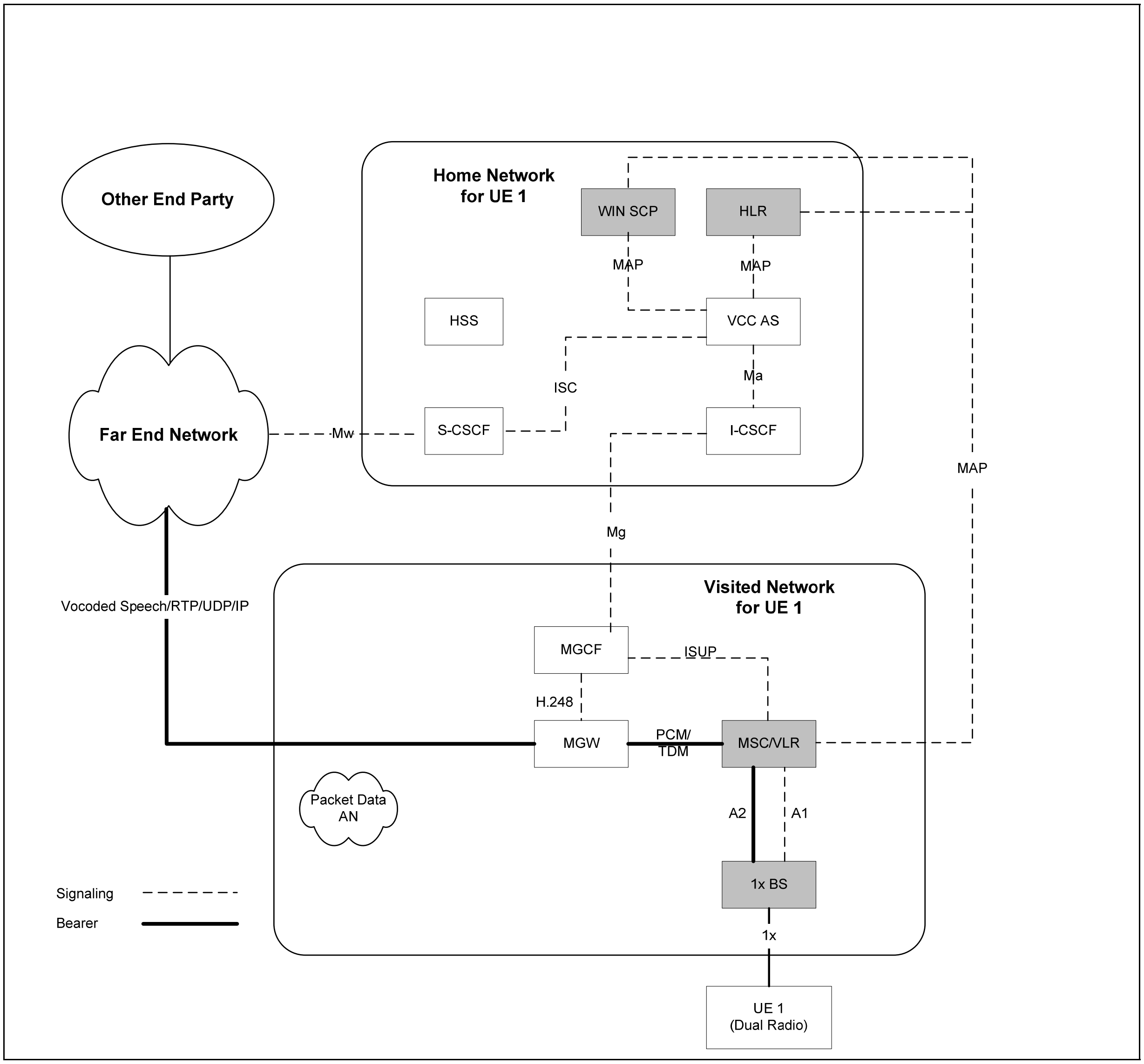


***Figure 22*** **Dual Radio 1x CS voice to VoIP DT – Step 1: Initial 1x CS voice call**

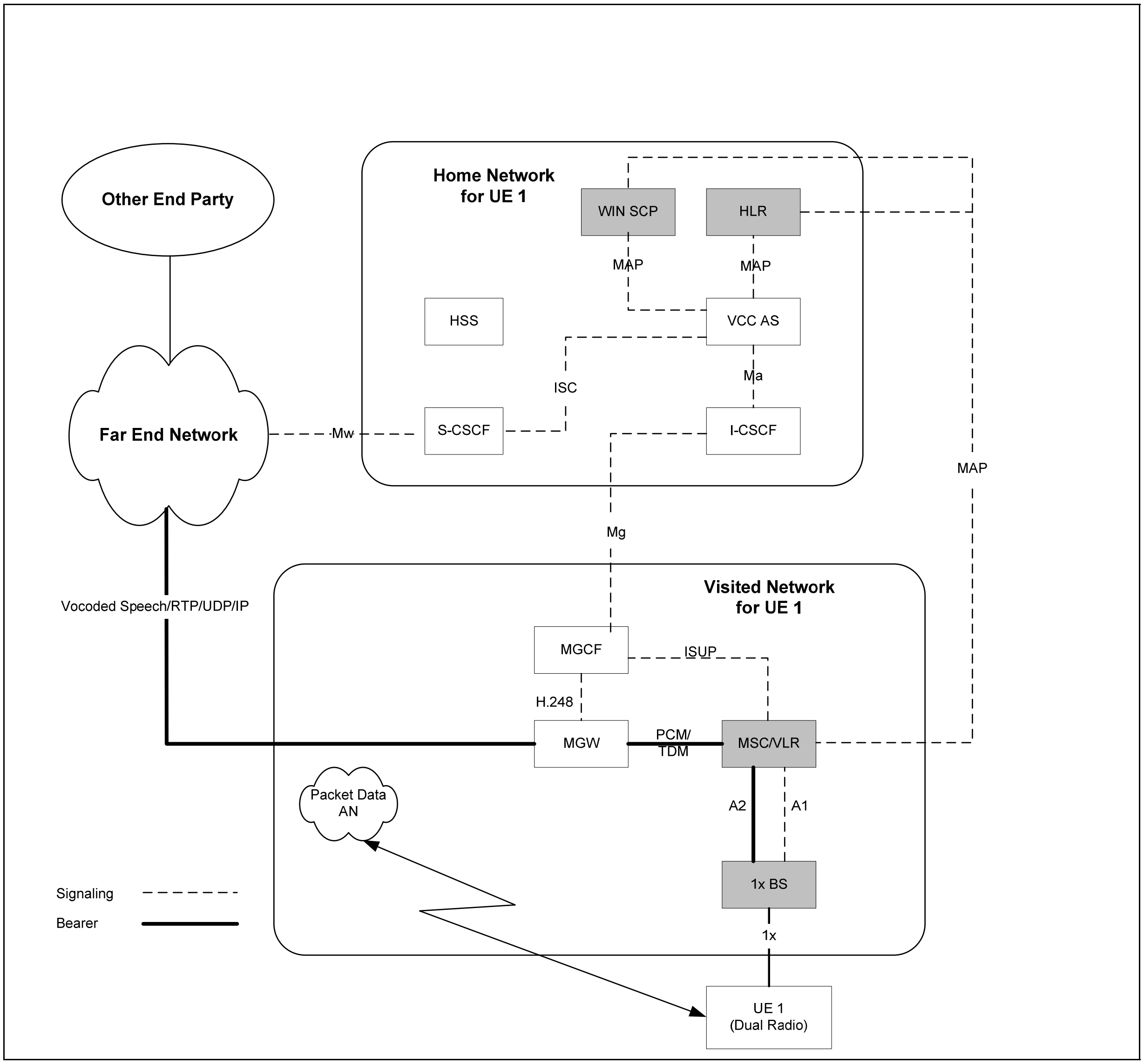


***Figure 23*** **Dual Radio 1x CS voice to VoIP DT – Step 2: Before DT to VoIP**

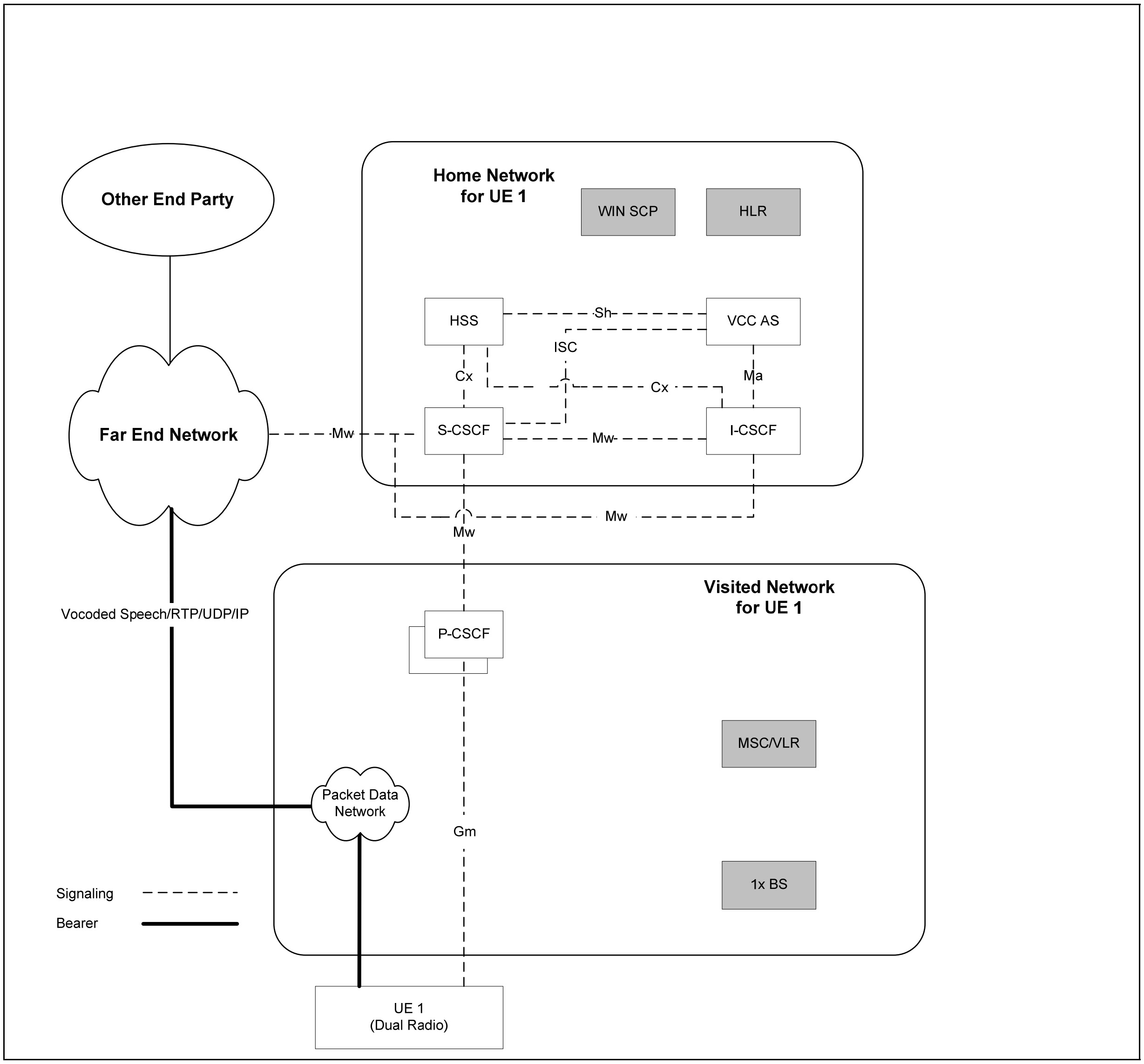


***Figure 24*** **Dual Radio 1x CS voice to VoIP DT – Step 3: After DT to VoIP**

### 4.7.1 Dual Radio 1x CS Voice to VoIP DT

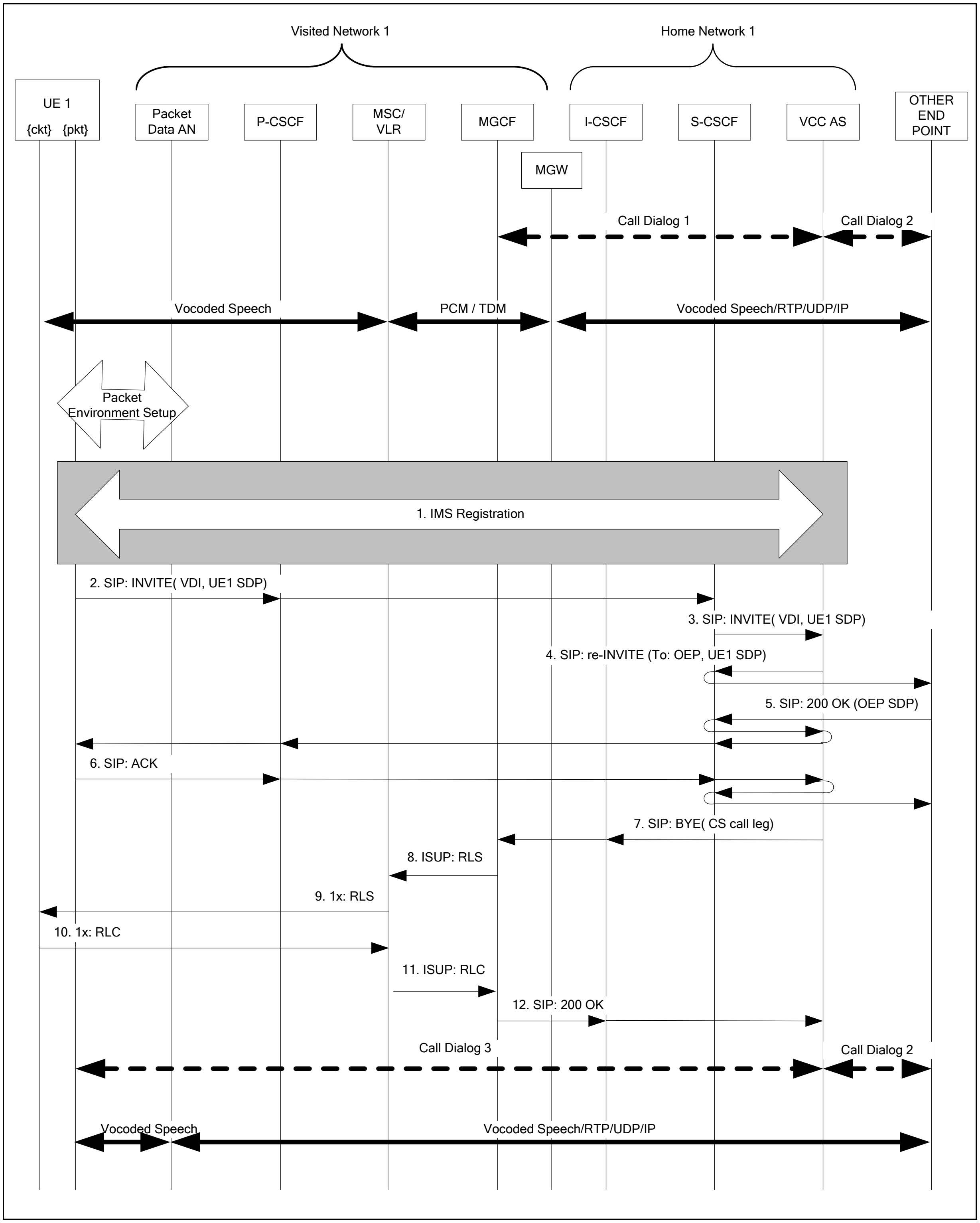


***Figure 25*** **Dual Radio Domain Transfer: 1x CS Voice to VoIP**

Precondition 1: A 1x CS Voice call is established. When the call was initially setup, the VCC AS was anchored into the call (see CS Call Delivery and/or CS Call Origination scenarios).

Precondition 2: Prior to the UE initiating a dual radio VoIP DT, the UE establishes the packet data environment.

1. Conditionally, if UE 1 is not IMS registered, IMS registration procedures per section 4.2.1 are executed.

2. UE 1 initiates a dual radio VoIP DT by initiating an IMS call to the VCC AS (i.e., VDI). The SIP *INVITE* is sent from the UE to the P-CSCF and then to the S-CSCF.

3. The S-CSCF forwards the *INVITE* to the VCC AS based on filter criteria. The combination of the known VDI along with the P-Asserted-Identity value identify the INVITE are the key indicators that DT treatment has been initiated.

4. The VCC AS sends a SIP *re-INVITE* message to the Other End Point (OEP) with the updated SDP for the UE 1 call leg via the S-CSCF. Note: this OEP may be an MGCF if the other end is an ISUP termination or an I-CSCF for a SIP termination.

5. The OEP acknowledges the *re-INVITE* with a SIP *200 OK* that is forwarded thru the IMS entities to UE 1.

6. The SIP *ACK* from UE 1 is forwarded thru the IMS entities to the OEP. This completes the establishment of call dialog 3.

7. On reception of the *ACK*, the VCC AS clears the original 1x CS Voice call leg (call dialog 1) by sending a SIP *BYE* to the MGCF via the I-CSCF. Note: The UE is not expected to clear the 1x CS call leg; the UE waits for the network to clear the leg to ensure the network is established prior to clearing the 1x CS call leg.

8. In steps 8-12, the 1x CS Voice call leg is cleared between the MSC and UE, and a SIP *200 OK* is sent back from the MGCF to the VCC AS.

## 4.8 Failure Scenarios

There is no mechanism for the UE to determine whether the call originated by it was anchored at the VCC AS or not, prior to initiating DT procedures. This section discusses the failure scenarios when a UE tries to initiate dual radio VoIP to 1x CS DT, or vice versa, for a call that is not anchored at the VCC AS.

### 4.8.1 Failure Scenario for Dual Radio Domain Transfer VoIP-1x CS Voice

Figure 26 provides an information flow when the UE tries to perform DT for a call originated by a dual radio UE in the IMS domain to 1x CS, but the original call-leg was not anchored at the VCC AS (due to operator policy etc.).

The flow is based on the precondition that the user is active in one or more IMS voice originating or terminating session(s) at the time of initiation of DT to CS. The call flows shown here are for dual radio VoIP-to-1x CS DT failure scenarios.

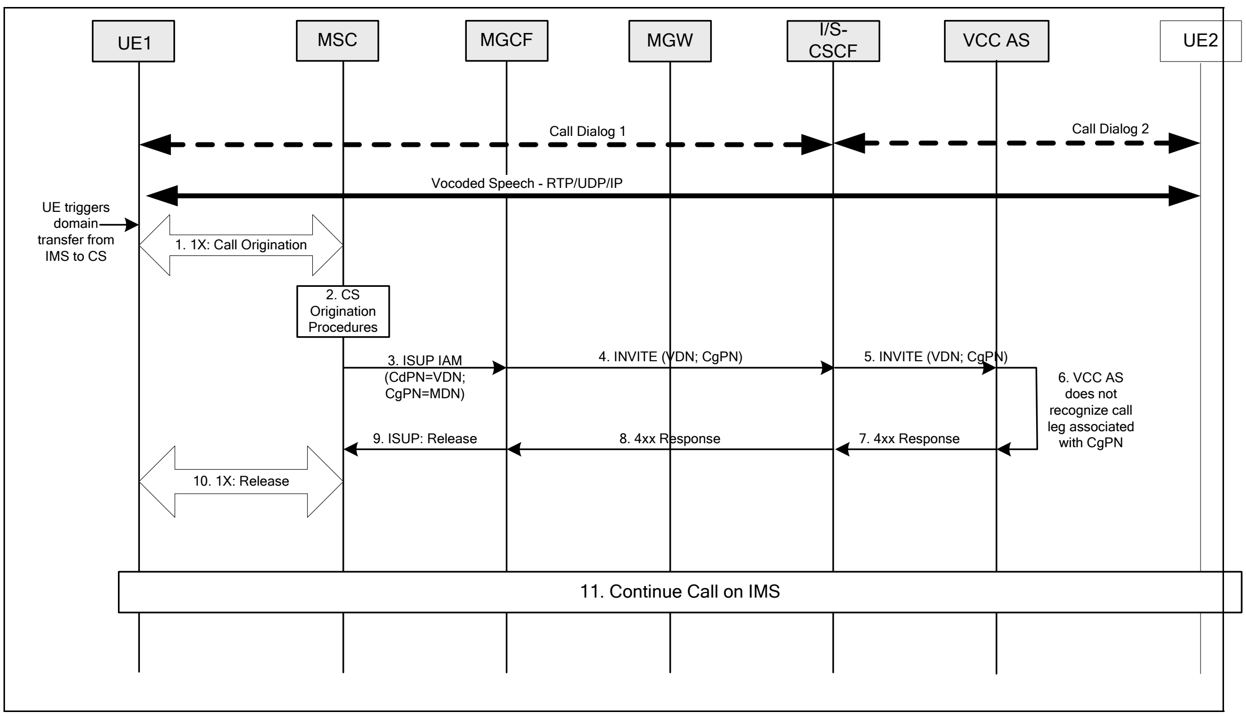


***Figure 26*** **Failure scenario for Dual Radio DT from VoIP to 1x CS**

1. In order to trigger a dual radio DT from VoIP to 1x CS, UE sends a 1x *Call Origination* to the Visited MSC as described in Section 4.6.1.

2-3. As described in Section 4.6.1, the Visited MSC performs the 1x registration and number translation procedures, and sends an ISUP *IAM* routed to a Visited MGCF in the serving network.

4-5. The Visited MGCF sends a SIP *INVITE* message via the I/S-CSCF to the VCC AS containing an SDP offer with the Visited MGW SDP information.

6. The VCC AS does not recognize the call-leg associated with the calling party number, since the call was not originally anchored at the VCC AS.

7-8. The VCC AS sends a SIP *4xx* error response to the Visited MGCF in response to the INVITE, routed through the I/S-CSCF.

9. The Visited MGCF converts the *4xx* response to ISUP *Release* message and forwards it to the Visited MSC.

10. The Visited MSC sends back a failure notification (1x *Release Order* message) to the UE 1 via the 1x BS. The UE 1 responds back with a 1x *Release* message to the Visited MSC.

11. At this point, UE 1 continues the call with the other end party in the IMS domain.

### 4.8.2 Failure Scenario for Dual Radio Domain Transfer: 1x CS - VoIP

Figure 27 provides an information flow when the UE tries to perform dual radio DT of a call originated in 1x to VoIP, but the original call-leg was not anchored at the VCC AS (due to operator policy, or no network support for VCC etc.).

The flow is based on the precondition that the user is active in the CS voice originating or terminating session(s) at the time of initiation of DT to CS.

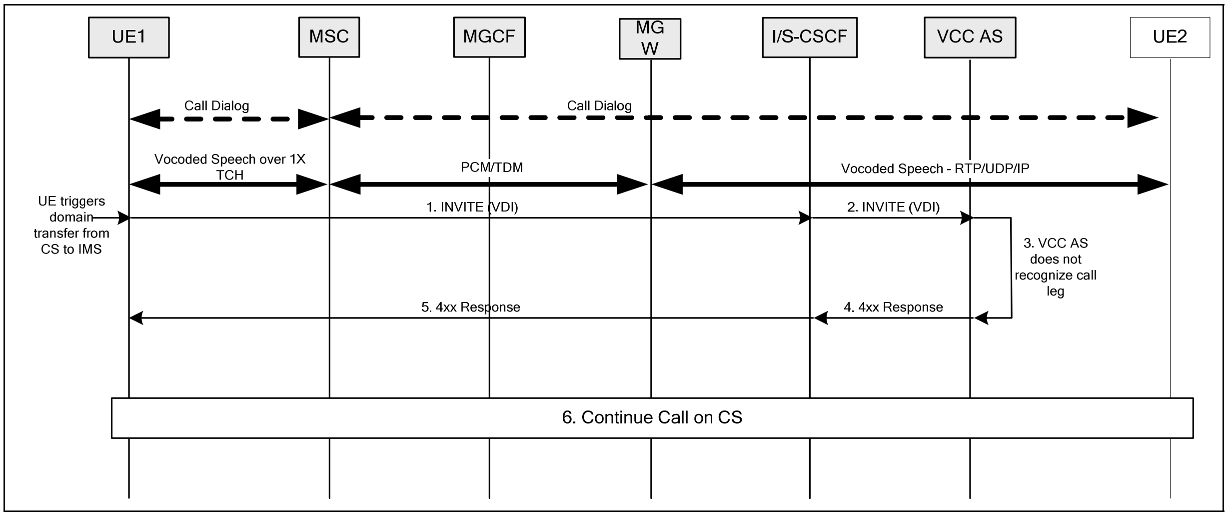


***Figure 27*** **Failure scenario for dual radio DT from 1x CS to VoIP**

1. UE 1 initiates a dual radio DT by initiating an IMS call to the VCC AS (i.e., VDI). The SIP *INVITE* is sent from the UE 1 to the I/S-CSCF.

2. The S-CSCF forwards the *INVITE* to the VCC AS based execution of iFC.

3. The VCC AS does not recognize the call-leg associated with the calling party number, since the call was not originally anchored.

4-5. The VCC AS sends a SIP *4xx*-error message in response to the *INVITE*, routed through the I/S-CSCF.

6. At this point, UE 1 continues the call in the CS domain.

### 4.8.3 Service Invocation during Call Delivery

The following call flow illustrates the scenario where the VCC AS is unable to deliver the incoming call to the UE, and hence returns a SIP *4xx* or SIP *3xx* response. It is assumed that the terminating IMS subscriber has subscribed to other services (e.g., voice mail) and that such services are performed by an AS. It is further assumed that the services AS is placed in the signaling path ahead of the VCC AS per the initial filter criteria (iFC). Upon any failure of the VCC AS to progress the SIP *INVITE*, then, the services AS will perform services as appropriate.

NOTE: This particular flow uses call forwarding just as an example. Other services (for example, voicemail routing) can be handled in a similar fashion.

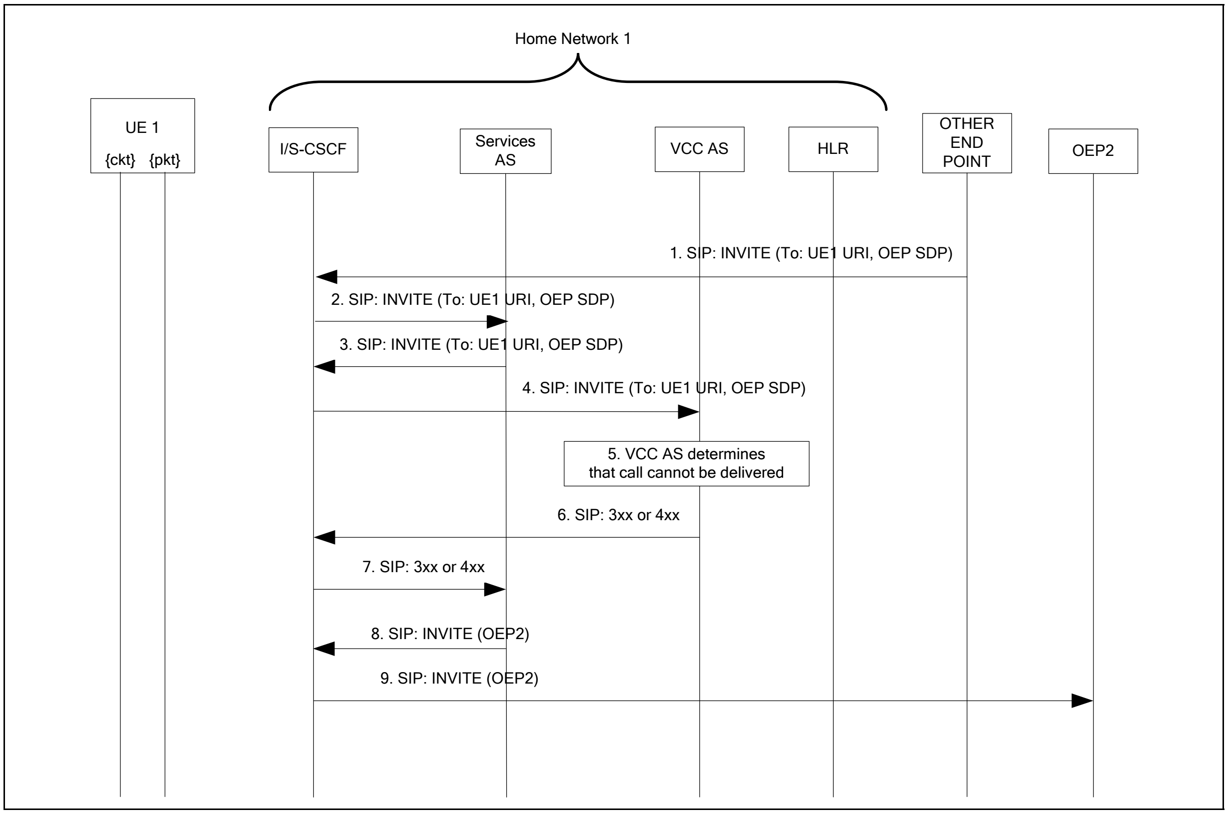


***Figure 28*** **Service Invocation during Call Delivery**

1. The Other End Point (OEP) sends a SIP *INVITE* to the I/S-CSCF in the IMS home network of UE 1.
2. Based on the filter criteria, the I/S-CSCF sends the *INVITE* to an AS providing IMS services (e.g., call forwarding).
3. The services AS sends the *INVITE* to the I/S-CSCF.
4. Based on the filter criteria, the I/S-CSCF sends the *INVITE* to the VCC AS.
5. The VCC AS determines that the call cannot be delivered to the CS; for example, the user may be unreachable in CS or CS call forwarding invoked and the VCC AS gets the information (e.g. forward-to number).
6. The VCC sends an appropriate (SIP *3xx* or SIP *4xx*) error response back to the S-CSCF.

7. The S-CSCF forwards the error response to the Services AS.
8. The Services AS decides to execute a call forwarding feature and forwards the call to another endpoint (OEP2).
9. The S-CSCF forwards the call to OEP2.

# 5 Stage 3: Procedures and Protocol

## 5.1 Overview of VCC between the CS domain and the MMD

### 5.1.1 General

VCC allows a UE employing two separate technologies, one the traditional CS domain accessed via cdma2000® 1x, and the other the MMD accessed via a number of access technologies, e.g. HRPD, E-UTRA or WLAN, to have calls flexibly delivered over either the CS domain or the MMD, and to transfer the calls from one technology to the other when access or other conditions alter.

Calls originated by VCC subscribers in both the MMD and in the CS domain are subject to anchoring in the MMD. Similarly calls terminated to VCC subscribers are subject to anchoring in the MMD. When anchoring occurs, such calls have a path to the VCC AS from either the CS domain or the MMD, so that the VCC AS can be used to execute a DT procedure of the voice session or the voice component of a multimedia session. If a call from a VCC subscriber is not anchored in the MMD, DT is not supported for that call.

For the above to occur, the following procedures are defined within this document:

- procedures for initializing a VCC AS for a specific subscriber before the VCC UE makes or receives calls are specified in Section 5.3.
- procedures for call origination are specified in Section 5.4.
- procedures for call termination are specified in Section 5.5;
- procedures for transfer of a call from the CS domain to the MMD are specified in Section 5.6; and
- procedures for transfer of a call from the MMD to the CS domain are specified in Section 5.7;

In this version of this document, VCC cannot be applied to emergency calls, and emergency calls are not therefore anchored.

### 5.1.2 Underlying Network Capabilities

VCC assumes the use of a number of underlying network capabilities:

1. deployment by the home network operator of VCC AS on the MMD;

---

cdma2000® is the trademark for the technical nomenclature for certain specifications and standards of the Organizational Partners (OPs) of 3GPP2. Geographically (and as of the date of publication), cdma2000® is a registered trademark of the Telecommunications Industry Association (TIA-USA) in the United States.

2. signalling within the CS domain (both within the home network and between the home network and any serving network) supported using ISUP ([ITU ISUP] or [ANSI ISUP]) and MAP [X.S0004-540];
3. provisioning of WIN capability (as specified in [WIN]) at the MSC (download of the subscriber's trigger profile) or the HLR (relay by the HLR of the ORREQ to the SCP);
4. interworking between CS domain and the MMD provided by an MGCF in accordance with [X.S0050]; and
5. capability of the IP-CAN to support VoIP.

### 5.1.3 URI and Address Assignments

In order to support VCC to a subscriber, the following URI and address assignments are assumed:

a. in this version of the document, the VCC UE will be configured with both a VDI and a VDN in order to initiate a DT;

b. the VCC UE will be configured to be reachable in both the MMD and the CS domain by one or more public telecommunication numbers which should be correlated between the CS domain and the MMD. A public telecommunication number can be a MDN used in the CS domain which is conveyed in international format to the VCC UE as a part of the implicit registration set associated with that VCC UE through MMD, or the VCC AS can be configured to provide a functional relationship between separate numbers providing each of these identities;

   NOTE: One way of correlating the public telecommunication numbers of the CS domain and the MMD is, to set them to the same values.

c. if WIN is supported, an IMRN may be assigned that can reach a VCC AS that can either support the VCC capabilities for that VCC UE, or otherwise locate the VCC AS supporting the VCC capabilities for that VCC UE. The IMRNs are dynamically allocated to route the call to MMD for anchoring purposes, or during DT from the MMD to the CS domain. The MMD is configured to treat the IMRN as a PSI;

d. the MDN will be subject to routing to the MMD in order for anchoring to be performed by the VCC AS. A TLDN is assigned to be able to route to a serving MSC (via an MGCF) such that the TLDN is not subject to the same routing back to the MMD and the VCC AS; and

e. not all calls are suitable for DT, and application of DT to other calls might be against subscriber or operator preferences.

## 5.2 Functional entities

### 5.2.1 Introduction

This clause associates the functional entities described for the MMD and for the CS domain, with the VCC roles described in the stage 2 architecture (see Section 4).

### 5.2.2 User Equipment (UE)

A UE compliant with this document shall implement the role of a VCC UE (see sections 5.3.2, 5.4.2, 5.5.2, 5.6.2 and 5.7.2).

### 5.2.3 Application Server (AS)

An AS compliant with this document shall implement the role of a VCC AS (see sections 5.3.3, 5.4.4, 5.5.4, 5.6.3 and 5.7.4).

### 5.2.4 Media Gateway Control Function (MGCF)

In order to support VCC for any call, the MGCF has to provide signaling interworking and control of the media between the CS domain and the MMD. The VCC AS can only be configured to operate where appropriate interworking is provided.

As the procedures for call termination in the CS domain may involve an MGCF provided by another network operator, the provision of appropriate interworking can extend to peering agreements between operators.

## 5.3 Roles for Registration in the MMD

### 5.3.1 Introduction

In addition to the procedures specified in this section, the VCC UE and the VCC AS shall support the procedures specified in [MMD Part-4] appropriate to the functional entity in which they are implemented. The VCC AS can be configured with any of various options for obtaining information from the MMD specified in [MMD Part-4], [MMD Part-10], [MMD Part-11] for example:

a. receipt of REGISTER request which causes a third-party REGISTER request to be sent to the VCC AS;

b. receipt of REGISTER request which causes a third-party REGISTER request to be sent to the VCC AS. The VCC AS then subscribes to the reg event package for that user to obtain information; or

c. receipt of REGISTER request which causes a third-party REGISTER request to be sent to the VCC AS. The VCC AS then uses the Sh interface to obtain information.

This document places no requirement on the use of all or any of these mechanisms.

### 5.3.2 VCC UE

When the VCC UE registers with the IMS subsystem, the VCC UE shall apply the procedures as specified in [MMD Part-4]. When constructing a REGISTER request, the UE shall include a “Timestamp” header [RFC 3261] in the SIP REGISTER request. The value of the “Timestamp” header shall be set to the time, in seconds since January 1, 1900 00:00 UTC, at which the UE generated the REGISTER message. The UE shall indicate its capabilities (for example, supported applications) using the procedures defined in [RFC 3840].

In this release of the specification, the UE shall use the “audio” feature tag defined in [RFC 3840] to indicate the support for real-time VoIP capability.

When the VCC UE enters an IP-CAN coverage area where the negotiated VoIP capabilities are different from the previous IP-CAN area, the VCC UE shall re-register with IMS to update its negotiated VoIP capabilities.

NOTE 1: Depending on operator policy, registration to the IMS subsystem can impact whether calls are delivered to the VCC UE using the MMD or using the CS domain.

NOTE 2: The VCC UE performs registration in the IMS subsystem independent of the UE's state in the CS domain.

If the VCC UE is both IMS registered and CS registered and the VCC UE detects that the IP-CAN connection is temporarily unavailable and still has CS network coverage, the VCC UE shall send an SMS, constructed as specified in section 5.3.2.1, on the CS network to notify the VCC AS that it is reachable only through the CS domain. The VCC UE shall follow procedures specified in section 5.3.2.2 for processing SMS acknowledgements. The "Destination Address" in the SMS message shall be set based on VDN. When the UE regains MMD coverage, the UE shall sends a SIP re-REGISTER message over IMS to indicate to the VCC AS that it has regained MMD coverage.

#### 5.3.2.1 Constructing SMS Message

The UE shall construct the SMS message as specified below.

- The UE shall set the SMS_MSG_TYPE of the SMS transport layer to '0000000' (SMS Point-to-Point).
- The UE shall set the contents of the SMS Point-to-Point message to the following:
  - set the Parameter ID '0000000' (Teleservice identifier) with the IDENTIFIER field set to'4242' (IMS Services Teleservice, IMSST).
  - set the Parameter ID '00000100' (Destination Address) to the VDN
  - include the Parameter ID '00000110' (Bearer Reply Option). The REPLY_SEQ field of the Bearer Reply Option shall be set to a value identifying the SMS message being constructed at the SMS transport layer.
  - set the Parameter ID '00001000' (Bearer Data) to value specified below.
- The UE shall set include the following sub parameters in Bearer Data:
  - include the sub parameter '00000000' (Message Identifier) with MESSAGE_TYPE set to '0010' (Submit) and the MESSAGE_ID field set to a value identifying the SMS message being constructed at the Teleservice Layer.
  - include the sub parameter '00000001' (User Data) with the MSG_ENCODING parameter set to '00000' (Octet). The UE shall set the CHARi field of the User Data as specified in section 5.8.1.1. The UE shall set the 'Domain-Status' field in VCC Message Type to 0x00 (Domain-Attachment-Message), and set the VCC Message Data to 0x00 (CS-only). The UE shall also set the 'Timestamp' field to the time, in seconds since January 1, 1900 00:00 UTC, at which the UE generated the IMSST message.
  - include the sub parameter '00001000' (Priority) with the PRIORITY field set to '10' (Urgent).
  - include the sub parameter '00001010' (Reply Option) with the USER_ACK_REQ field set to '0', DAK_REQ field set to '1', and READ_ACK_REQ field set to 0.

#### 5.3.2.2 Processing SMS Acknowledgments

Upon receiving an SMS Ack Message that corresponds to the previously sent SMS message for VCC, the UE shall:

- If the Cause Codes parameter indicates temporary failure, i.e. ERROR_CLASS is ‘10’, the UE may retry the VCC SMS message.
- Otherwise, no action is required.

Upon receiving an SMS Delivery Ack Message that corresponds to the previously sent SMS message, the UE shall:

- If the Delivery Ack Message does not contain the Message Status sub parameter, no further action is required.
- If the Delivery Ack Message contains the Message Status sub parameter:
  - If the ERROR_CLASS is ‘10’ (temporary condition), the UE may retry the VCC SMS message.
  - Otherwise, no action is required.

### 5.3.3 VCC AS

The VCC AS can obtain information from any received third-party REGISTER request, or any received reg event package, or the Sh interface, that it needs to implement the domain selection policy of the network operator.

If the third-party REGISTER request does not carry a P-Access-Network-Info header provided by the UE, and the VCC AS requires this knowledge for domain selection procedures, the mechanisms by which the VCC AS determines the domain are outside the scope of this document.

Upon receiving a third-party REGISTER from the S-CSCF, the VCC AS shall apply the procedures as specified in [MMD Part-4] with the following additions:

- The VCC AS shall store the S-CSCF information associated with the VCC UE;
- If the Expires header or the expires parameter in the Contact header has a value equal to zero, the VCC AS shall mark the VCC UE’s status as not reachable by IMS.
- The VCC AS shall check to see whether the time value specified in “Timestamp” field is later than the previously stored value of the Timestamp header. If the timestamp value is not later, the VCC AS shall not update the UE’s reachability status. Otherwise, the VCC AS shall store the value of the header and shall mark the VCC UE’s status as reachable via IMS.

When the VCC UE is reachable via IMS, the VCC AS may also store information on whether the VCC UE is VoIP capable based on information received in Section 5.3.2.

Upon receiving a SMS from the VCC UE with the Teleservice identifier set to IMSST, the VCC AS shall do the following:

- If the VCC Message Type is ‘Domain-Attachment-Status’ and the ‘Domain-Status’ field is set to 0x00 (CS-only), the VCC AS shall check to see whether the time value specified in “Timestamp” field is later than the previously stored Timestamp value. If the time in the SMS message is later, the VCC AS shall store the new Timestamp value and shall mark the UE’s status as reachable only over CS domain. Otherwise, the VCC AS shall ignore the SMS message.

### 5.3.4 S-CSCF

When the S-CSCF sends a third-party REGISTER message to the VCC AS, the S-CSCF shall follow the procedures as specified in [MMD Part-4]. In addition, if the "timestamp" header is sent by the UE in a REGISTER message, the S-CSCF shall transparently pass that along in the third-party REGISTER message to the VCC AS (as it would to all other AS in user's profile for the REGISTER message).

## 5.4 Roles for Call Origination

### 5.4.1 Introduction

This section specifies the procedures for call origination, both where the VCC UE is generating calls in the CS domain and where the VCC UE is generating calls using the MMD. Procedures are specified for the VCC UE and other VCC functional entities.

### 5.4.2 VCC UE

The VCC UE shall support origination of calls suitable for VCC via both the CS domain [C.S0005] and the MMD [MMD Part-4].

There are no VCC specific requirements for the origination of calls that may be subject to VCC.

NOTE 1: For INVITE requests initiated by the VCC UE in the MMD, the following are some of the response codes that can indicate failure of the VCC AS to process the request with its current characteristics:

- 484 (Address Incomplete);
- 488 (Not Acceptable Here) or 606 (Not Acceptable);
- 503 (Service Unavailable) or 603 (Decline).

### 5.4.3 MSC

There are no VCC specific procedures at the MSC.

NOTE: the MSC interacts with triggers for call origination.

### 5.4.4 VCC AS

#### 5.4.4.1 Distinction of Requests Sent to the VCC AS

The VCC AS needs to distinguish between the following initial SIP INVITE requests to provide specific functionality relating to call origination:

- SIP INVITE requests routed to the VCC AS over either the ISC interface or the Ma interface using the IMRN as a PSI, and therefore distinguished by the presence of the IMRN in the Request-URI header, and which are known by interaction with the MAP functionality [X.S0004-540] to relate to an originating request rather than a DT request or a termination request. In the procedures below, such requests are known as "SIP INVITE requests due to originating IMRN"; and

- SIP INVITE requests routed to the VCC AS over the ISC interface as a result of processing filter criteria at the S-CSCF according to the origination procedures defined in [MMD Part-4], are distinguished by the contents of the Request-URI. If the Request-URI contains a VDI, then it is for a DT request. However, absence of a VDI in the Request-URI indicates an origination request. In the procedures below, such requests are known as "SIP INVITE requests due to originating filter criteria".

Section 5.5.4.1, section 5.6.3.1 and section 5.7.4.1 detail other procedures for initial INVITE requests with different recognition conditions.

Other SIP initial requests for a dialog and requests for a SIP standalone transaction can be dealt with in any manner conformant with [MMD Part-4].

The VCC AS also processes MAP requests as defined in [X.S0004-540]. The VCC AS shall differentiate MAP requests that relate to an originating request, containing an ordinary called party number, covered in this section, and those relating to a DT request, which contain a VDN as the called party number.

#### 5.4.4.2 Call Origination in the MMD

When the VCC AS receives a SIP INVITE request due to originating filter criteria, the VCC AS shall:

1. check anchoring is possible for this session;

   NOTE 1: The conditions that prevent anchoring are a matter for implementation, but can include operator policy on a number of conditions, e.g. roaming of the VCC UE, the present IP-CAN used to access the IMS subsystem. In general, the number of calls presented for anchoring on behalf of the same user, and the media characteristics relating to these calls, will not prevent anchoring, as these issues are dealt with at DT.

2. if the session is not subject to anchoring, either:

   i. forward the SIP INVITE request by acting as a SIP proxy as specified in [MMD Part-4]. The VCC AS shall not Record-Route on such requests, and the request is not retargeted by changing the Request-URI; or

   ii. reject the SIP INVITE request. The following response codes are recommended:

      - 484 (Address Incomplete), if the Request-URI supplied is not resolvable in the home network (such a Request-URI may have been resolvable in the local network which may be visited by the VCC UE at this time);

      - 488 (Not Acceptable Here) or 606 (Not Acceptable), if some aspect of operator policy precluded anchoring; or

      - 503 (Service Unavailable) or 603 (Decline) for all other conditions;

   and no further VCC specific procedures are performed on this session;

   NOTE 2: Some checks may also form part of the initial filter criteria in the S-CSCF to determine if the SIP INVITE request is sent to the VCC AS in the first place.

3. if the session is subject to anchoring, operate as an application server providing 3rd party call control, and specifically as a routing B2BUA, as specified in [MMD Part-4] for this request and all future requests and responses in the same dialog with the following additions:

i. copy the Request-URI unchanged from the incoming SIP INVITE request to the outgoing SIP INVITE request;

ii. copy all other headers unchanged from the received SIP INVITE request to the outgoing SIP INVITE request; and

iii. copy the body unchanged from the received INVITE request to the outgoing SIP INVITE request.

NOTE 3: Call anchoring is performed before all originating services are executed and thus the VCC AS is invoked as the first AS in the originating initial Filter Criteria.

#### 5.4.4.3 Call Origination in the CS Domain – Procedures Towards the WIN SCP

When the VCC AS receives an MAP ORREQ or a MAP ANLYZD, containing a called party number that is not a VDN, the VCC AS shall:

1. check anchoring is possible for this call;

   NOTE 1: The conditions that prevent anchoring are a matter for implementation, but can include operator policy on a number of conditions, e.g. roaming of the VCC UE, lack of resources, such as available IMRNs, or a lack of translation rules if the called party number is not in international number format. In general, the number of calls presented for anchoring on behalf of the same user will not prevent anchoring, as these issues are dealt with at DT.

2. if the session is not subject to anchoring:

   i. set the AccessDeniedReason to the appropriate value and the AnnouncementList to the appropriate value and respond with an ORREQ or ANLYZD RETURN RESULT; or,

   ii. set the ActionCode to “continue processing” and respond with an ORREQ or ANLYZD RETURN RESULT.

3. if session is subject to anchoring, allocate an IMRN. The IMRN is such that when the VCC AS receives a SIP INVITE request it can derive by inspection that the request is due to an originating IMRN. How IMRNs are allocated may vary from one VCC AS to another and is not specified in this version of the specification. The VCC AS shall create a binding between the allocated IMRN and the ORREQ or ANLYZD Digits Parameter (i.e., the Called Party Number (CdPN) ).

4. if session is subject to anchoring, the VCC AS shall send an ORREQ or ANLYZD RETURN RESULT with the allocated IMRN in TerminationList.

#### 5.4.4.4 Call Origination in the CS Domain – Procedures Towards the MMD

When the VCC AS receives SIP INVITE request due to originating IMRN, the VCC AS shall:

1. operate as an application server providing 3rd party call control, and specifically as a routing B2BUA, as specified in [MMD Part-4] for this request and all future requests and responses in the same dialog;

2. set the Request-URI of the outgoing initial SIP INVITE request to a tel-URI which represents the original called party number of the call as initiated in the CS domain. This is mapped from the binding created in step 3 of section 5.4.4.3;

3. set the To header field of the outgoing initial SIP INVITE request to a tel-URI which represents the original called party number of the call as initiated in the CS domain. This is mapped from the binding created in step 3 of section 5.4.4.3;

4. insert a Route header pointing to the S-CSCF serving the VCC UE, or to the entry point of the VCC UE's network (e.g., I-CSCF) and append the orig parameter to the topmost Route header of the outgoing initial SIP INVITE request;

5. set the P-Asserted-Identity header of the outgoing INVITE request to a tel-URI which represents the calling party number of the call initiated in the CS domain. This is either available from information associated against the received IMRN or is the value as received in P-Asserted-Identity header of the incoming INVITE request.

   NOTE: It can happen that the P-Asserted-Identity header is not included in the incoming INVITE request.

The VCC AS should, in the outgoing requests and responses, include the same values as received in the incoming requests and responses in all other headers with the exception given in this section and in [MMD Part-4].

The VCC AS will handle the Privacy header in the outgoing INVITE request in the following way. The VCC AS shall either:

- if a Privacy header is received in the incoming INVITE request, include the Privacy header as received in the incoming INVITE request; or
- if a value is associated to IMRN and indicates that the presentation of the calling party number is restricted in the CS domain, include a Privacy header with the value set to "id".

6. release the IMRN and the ORREQ/ANLYZD Digits binding.

## 5.5 Roles for Call Termination

### 5.5.1 Introduction

This section specifies the procedures for call termination, both where the VCC UE is receiving calls in the CS domain and where the VCC UE is receiving calls using the MMD. Procedures are specified for the VCC UE and other VCC functional entities.

### 5.5.2 VCC UE

The VCC UE shall support termination of calls suitable for VCC both via the CS domain as specified in [C.S0005] and the MMD as specified [MMD Part-4].

### 5.5.3 MSC

There is no VCC specific procedure at the MSC.

NOTE: The MSC may interact with triggers for call termination.

### 5.5.4 VCC AS

#### 5.5.4.1 Distinction of Requests Sent to the VCC AS

The VCC AS needs to distinguish between the following initial SIP INVITE requests to provide specific functionality relating to call termination:

- SIP INVITE requests routed to the VCC AS over the ISC interface as a result of processing filter criteria at the S-CSCF according to the termination procedures as defined in [MMD Part-4] and therefore distinguished by the URI relating to this particular filter criteria appearing in the topmost entry in the Route header. In the procedures below such requests are known as "SIP INVITE requests due to terminating filter criteria"; and
- SIP INVITE requests routed to the VCC AS over the ISC interface or Ma interface using the IMRN as a PSI, and therefore distinguished by the presence of the IMRN in the Request-URI header, which is different from the VDN. In the procedures below such requests are known as "SIP INVITE requests due to terminating IMRN".

Section 5.4.4.1, section 5.6.3.1 and section 5.7.4.1 detail other procedures for initial INVITE requests with different recognition conditions.

Other SIP initial requests for a dialog and requests for a SIP standalone transaction can be dealt with in any manner conformant with [MMD Part-4].

The VCC AS also processes MAP requests related to call termination as defined in [X.S0004-540].

#### 5.5.4.2 Call Termination in the MMD

When the VCC AS receives a SIP INVITE request due to terminating filter criteria, the VCC AS shall:

1. check anchoring is possible for this session;

   NOTE 1: The conditions that prevent anchoring are a matter for implementation, but can include operator policy on a number of conditions, e.g. roaming of the VCC UE, or a lack of resources. In general, the number of calls presented for anchoring on behalf of the same user will not prevent anchoring, as these issues are dealt with at DT. If anchoring fails, the call is presented to the user without anchoring.

   NOTE 2: Such a check can also form part of the initial filter criteria in the S-CSCF to determine if the SIP INVITE request is sent to the VCC AS in the first place.

2. if the session is not subject to anchoring:

   i. if the preferred delivery domain for such unanchored calls is the MMD, forward the SIP INVITE request by acting as a SIP proxy as specified in [MMD Part-4]. The VCC AS shall not Record-Route on such requests. No further VCC specific procedures are performed on this session.

   ii. if the preferred delivery domain for such unanchored calls is the CS domain, the VCC AS shall send an MAP LOCREQ to the HLR of the terminating subscriber. Upon receiving the LOCREQ RETURN RESULT the VCC AS shall forward the SIP INVITE request by acting as a SIP proxy as specified in [MMD Part-4], after first retargeting the request by changing the Request-URI to the

TerminationList value (i.e., TLDN) in the LOCREQ RETURN RESULT . The VCC AS shall not Record-Route on such requests. No further VCC specific procedures are performed on this session.

NOTE 3: How the VCC AS determines the preferred delivery domain is outside the scope of this specification.

3. if the session is subject to anchoring, operate as an application server providing 3rd party call control, and specifically as a routeing B2BUA, as specified in [MMD Part-4] for this request and all future requests and responses in the same dialog;

4. if the session is subject to anchoring, perform domain selection based on:

   i. operator preferences;

   ii. callee capabilities of the terminating UE, as obtained during IMS registration;

   iii. access network information, as provided in the P-Access-Network-Info header of the terminating UE;

   iv. and session states;

5. if the MMD is selected, leave the Request-URI unchanged between the incoming SIP INVITE request and the outgoing SIP INVITE request;

6. if the CS domain is selected:

   i. send an MAP LOCREQ to the HLR of the terminating subscriber.

   ii. upon receiving the LOCREQ RETURN RESULT, the VCC AS

      1. shall set the Request-URI of the outgoing SIP INVITE request to a tel URI based upon the DestinationDigits [X.S0004-550] in the TerminationList in the LOCREQ RETURN RESULT; and

      2. should set the To header field of the outgoing SIP INVITE request to a tel URI based upon the DestinationDigits [X.S0004-550] in the TerminationList in the LOCREQ RETURN RESULT.

On completion of the above procedure, the call is anchored in the VCC AS.

NOTE 4: The VCC AS acting as a B2BUA - which performs the 3rd party call control - needs to be the last located application server to ensure that all application servers that need to remain in the path of a call after DT will do so.

If the call delivery attempt fails and if allowed by operator policy, the VCC AS may re-attempt the call termination to the CS Domain. If the re-attempted call delivery also fails, no further call attempts shall be made.

### 5.5.4.3 Call Termination in the CS Domain

The following section defines two options for implementing call termination in the CS domain.

#### 5.5.4.3.1 Call Termination in the CS Domain – Procedures Towards the WIN SCP

When the VCC AS receives an MAP ANLYZD request [X.S0004-540] with the TriggerType set to “ADVANCED_TERMINATION” the VCC AS shall:

1. check anchoring is possible for this call;

NOTE 1: The conditions that prevent anchoring are a matter for implementation, but can include operator policy on a number of conditions, e.g. roaming of the VCC UE, or a matter of lack of resources, e.g. available IMRNs. In general, the number of calls presented for anchoring on behalf of the same user will not prevent anchoring, as these issues are dealt with at DT. If anchoring fails, the call is presented to the user without anchoring.

2. if the session is not subject to anchoring:

   i. set the AccessDeniedReason to the appropriate value and the AnnouncementList to the appropriate value, and respond with an ANLYZD RETURN RESULT [X.S0004-540], or

   ii. set the ActionCode to "continue processing" and respond with an ANLYZD RETURN RESULT [X.S0004-540].

3. if the call is subject to anchoring, allocate an IMRN. How IMRNs are allocated may vary from one VCC AS to another and is not specified in this version of the specification. The VCC AS shall create a binding between the allocated IMRN and the ANLYZD Digits Parameter (i.e., the Called Party Number (CdPN) ).

4. if anchoring is possible for this call, the VCC AS shall send an ANLYZD RETURN RESULT with the allocated IMRN in TerminationList.

#### 5.5.4.3.2 Call Termination in the CS Domain – Procedures Towards the HLR

NOTE 1: HLR procedures for recognition of VCC subscribers and LOCREQ and ROUTEREQ handling are proprietary.

When the VCC AS receives an ROUTREQ request [X.S0004-540], the VCC AS shall:

1. check anchoring is possible for this call;

   NOTE 2: The conditions that prevent anchoring are a matter for implementation, but can include operator policy on a number of conditions, e.g. roaming of the VCC UE, or a matter of lack of resources, e.g. available IMRNs. In general, the number of calls presented for anchoring on behalf of the same user will not prevent anchoring, as these issues are dealt with at DT. If anchoring fails, the call is presented to the user without anchoring.

2. if the session is not subject to anchoring, the VCC AS shall:

   i. set the AccessDeniedReason to the appropriate value and the AnnouncementList to the appropriate value, and respond with an ROUTREQ RETURN RESULT [X.S0004-540].

3. if the call is subject to anchoring, allocate an IMRN. How IMRNs are allocated may vary from one VCC AS to another and is not specified in this version of the specification. The VCC AS shall create a binding between the allocated IMRN and the ROUTREQ Digits Parameter (i.e., the Called Party Number (CdPN) ).

4. if anchoring is possible for this call, the VCC AS shall send an ROUTREQ RETURN RESULT with the allocated IMRN in DestinationDigits.

### 5.5.4.4 Call Termination in the CS Domain – Procedures Towards MMD

When the VCC AS receives SIP INVITE request due to terminating IMRN, the VCC AS:

NOTE 1: All SIP INVITE requests directed to the VCC AS using an IMRN are assumed to be suitable for VCC anchoring, because any checks have been performed in conjunction with the MAP procedures.

1. shall operate as an application server providing 3rd party call control, and specifically as a routeing B2BUA, as specified in [MMD Part-4] for this request and all future requests and responses in the same dialog;

   NOTE 2: The SIP AS that implements the DT function of VCC AS acting as a B2BUA - which performs the 3rd party call control - needs to be the last located application server to ensure that all application servers that need to remain in the path of a call after DT will do so.

2. shall set the Request-URI of the outgoing initial SIP INVITE request to a tel-URI which represents the called party number of the original call as terminated in the CS domain. This is mapped from the binding created in section 5.5.4.3; and,

3. should set the To header field of the outgoing initial SIP INVITE request to a tel-URI which represents the called party number of the original call as terminated in the CS domain. This is mapped from the binding created in section 5.5.4.3;

4. shall release the IMRN and the ANLYZD Digits binding.

## 5.6 Roles for Domain Transfer of a Call from the CS Domain to the MMD

### 5.6.1 Introduction

This section specifies the procedures for DT of a call from the CS domain to the MMD when accessed via a dual radio UE. Procedures are specified for the VCC UE and other VCC functional entities.

### 5.6.2 VCC UE

If the VCC UE determines that an ongoing call in the CS domain needs to be supported over the MMD instead, e.g. based on radio conditions, then the VCC UE shall send a SIP INVITE request in accordance with [MMD Part-4]. For a UE originated call, the VCC UE shall send the SIP INVITE request only if it has entered the “Conversation Substate”. For a UE terminated call, the VCC UE shall send the SIP INVITE request only if it has entered the “Conversation Substate” and sent a “Connect” message to the BS. See [C.S0005] for the ongoing call in the CS domain. The VCC UE shall populate the SIP INVITE request as follows:

1. the Request-URI set to the VDI;

2. the To header field set to the VDI;

3. the P-Preferred-Identity header set to the a public telecommunication number of the calling party in accordance with Section 5.1.3; and

4. the SDP payload set for a single media line with media type “audio”, indicating all supported codecs for this media type, in accordance with [MMD Part-4]

If the VCC UE receives any 4xx – 6xx response to the SIP INVITE request, then DT has not occurred and the call will continue in the CS domain.

NOTE 1: If the VCC UE receivcs a 480 (Temporarily Unavailable) response to the SIP INVITE request, then this can indicate that the VCC AS was unable to correlate the request to a single anchored call in the CS domain.

NOTE 2: If the VCC UE receivcs a 488 (Not Acceptable Here) response or a 606 (Not Acceptable) response to the SIP INVITE request, then this can indicate that the remote terminal was not able to support the media characteristics of the SIP INVITE request, e.g. because the remote user is in the CS domain and the MGCF/MGW in the path does not support the specified interworking.

When the VCC UE receives a Release Order message from the network, the VCC UE shall comply with network initiated call release procedures as specified in [C.S0005].

### 5.6.3 VCC AS

#### 5.6.3.1 Distinction of Requests Sent to the VCC AS

The VCC AS needs to distinguish between the following initial SIP INVITE requests and other SIP INVITE requests to provide specific functionality relating to DT:

- SIP INVITE requests routed to the VCC AS over the ISC interface as a result of processing filter criteria at the S-CSCF according to the origination procedures (see [MMD Part-4] section 5.4.3.2), and therefore distinguished by the URI relating to this particular filter criteria appearing in the topmost entry in the Route header, but which contains a VDI belonging to the subscribed user as the Request-URI. In the procedures below such requests are known as "SIP INVITE requests due to VDI".

Sections 5.4.4.1, 5.5.4.1 and 5.7.4.1 detail other procedures for initial INVITE requests with different recognition conditions.

Other SIP initial requests for a dialog, and requests for a SIP standalone transaction can be dealt with in any manner conformant with [MMD Part-4].

#### 5.6.3.2 Domain transfer in the MMD

When the VCC AS receives a SIP INVITE request due to VDI, the VCC AS shall:

1. check whether existing call in CS domain is anchored to VCC AS. If the CS call is not anchored, the VCC AS shall return 480 (Temporarily Unavailable) response.

2. check the number of SIP Early Dialogs and SIP Dialogs of the user identified in the P-Asserted-Identity. If the number of SIP Early Dialogs and SIP Dialogs is greater than one, the VCC AS shall return 480 (Temporarily Unavailable) response.

3. check whether a SIP Dialog has been established between the user identified in the P-Asserted-Identity and a remote user. If the SIP Dialog has not been established, the VCC AS shall perform one of the following steps:

   i. return 183 (Session Progress) response and once the SIP dialog has been established the VCC AS proceeds with step 4, or,

   ii. Return 480 (Temporarily Unavailable) response to indicate that it cannot process the VCC request.

4. If the SIP dialog has been established, the VCC AS shall send a SIP reINVITE request towards the remote user using the existing established SIP Dialog. The VCC AS shall populate the SIP reINVITE request as follows:

i. set the Request-URI to the URI contained in the Contact header returned at the creation of the dialog with the remote user; and

ii. include a new SDP offer, setting the media characteristics as received in the SIP INVITE request due to VDI, by following the rules of [MMD Part-4].

Upon receiving the SIP ACK request from the UE, if the VCC AS not previously rejected the DT, the VCC AS shall initiate release of the old access leg by sending a SIP BYE request toward the MGCF.

### 5.6.4 MGCF

There are no VCC specific procedures at the MGCF.

## 5.7 Roles for Domain Transfer of a call from the MMD to the CS Domain

### 5.7.1 Introduction

This section specifies the procedures for DT of a call from the MMD to the CS domain. Procedures are specified for the VCC UE and other VCC functional entities.

### 5.7.2 VCC UE

If the VCC UE determines that an ongoing call in the MMD should be transferred to the CS domain, e.g. based on radio conditions, then if the ongoing call is via a single radio UE, the VCC UE shall send a 1x Origination message in accordance with [C.S0082] for an HRPD access network or [E-UTRA] for an E-UTRA access network. If the ongoing call is via a dual radio UE (e.g., WLAN) the VCC UE shall send a 1x Origination message in accordance with [C.S0005]. A VCC UE engaged in multiple sessions on the UE, may request DT according to operator policies. In the case that DT is allowed to be performed, the VCC UE shall initiate the release of all inactive sessions and all other sessions that are not the identified session of the DT (i.e., by sending a SIP BYE for the session), before initiating a DT. The VCC UE shall send the Origination message over the CS domain only if the ongoing call in the MMD has had the dialog accepted, i.e. a 200 (OK) response to the INVITE request has already been sent.

NOTE: The current media characteristics of the call in the MMD do not preclude DT, as the media characteristics are renegotiated as part of the DT.

The VCC UE shall populate the 1x Origination message as follows:

1. set the dialed digits to the VDN; and
2. other information elements populated for 1x Origination message as specified by [C.S0005], [C.S0082] or [E-UTRA] as applicable.

### 5.7.3 MSC

The MSC processes MS origination events as defined in [X.S0004-630]

NOTE: The MSC may interact with triggers for DT from MMD to the CS Domain.

## 5.7.4 VCC AS

### 5.7.4.1 Distinction of SIP INVITE Requests Sent to the VCC AS

The VCC AS needs to distinguish between the following initial SIP INVITE requests and other SIP INVITE requests to provide specific functionality relating to DT:

- SIP INVITE requests routed to the VCC AS over either the ISC interface or the Ma interface using the VDN or the IMRN as a PSI, and therefore distinguished by the presence of the VDN or the IMRN in the Request-URI header. The IMRN is known by interaction with the MAP functionality to relate to a DT request rather than an originating request or terminating request. In the procedures below, such requests are known as "SIP INVITE requests due to DT IMRN or VDN".

Sections 5.4.4.1, 5.5.4.1 and 5.6.3.1 detail other procedures for initial INVITE requests with different recognition conditions.

Other SIP initial requests for a dialog, and requests for a SIP standalone transaction can be dealt with in any manner conformant with 3GPP2 MMD [MMD Part-4].

The VCC AS also processes MAP requests as defined in [X.S0004-540].

NOTE: The functionality associated with these MAP requests depends on whether it contains an ordinary called party number indicating that it relates to an originating request, or it contains a VDN as the called party number indicating that it relates to a DT request.

### 5.7.4.2 Domain Transfer Procedures Towards the WIN SCP

When the VCC AS receives an MAP ORREQ or ANLYZD request, containing a called party number that is a VDN, the VCC AS shall:

1. check whether DT is possible;

   NOTE 1: The conditions that prevent DT are a matter for implementation, but in general for this check are a matter of lack of resources, e.g. available IMRNs. For other checks, the request will be continued so further checks can be performed at the VCC AS within the MMD.

2. if the call is not subject to DT, respond with a MAP ORREQ or ANLYZD RETURN RESULT indicating origination failure and no further VCC specific procedures are performed on this call;
3. if the call is subject to DT, allocate an IMRN. The IMRN is such that when the VCC AS receives a SIP INVITE request it can derive by inspection that the request is due to a DT IMRN. How IMRNs are allocated may vary from one VCC AS to another and is not specified in this version of the specification;
4. if the call is subject to DT, respond with a MAP ORREQ or ANLYZD RETURN RESULT message with the allocated IMRN in TerminationList parameter.

NOTE 2: The IMRN assigned for a DT request can be different from the one assigned for CS origination (different target PSI i.e. different sub-function of the VCC AS) and can be used as an indication of a DT request.

### 5.7.4.3 Domain Transfer in the MMD

When the VCC AS receives SIP INVITE request due to DT IMRN or VDN, the VCC AS shall associate the SIP INVITE request with an ongoing SIP dialog based on information associated with the received IMRN, if present, and P-Asserted-Identity header field and send a SIP reINVITE request towards the remote user using the existing established dialog.

By an ongoing SIP dialog, it is meant a dialog for which a SIP 2xx response to the initial SIP INVITE request has been sent or received. Multiple dialogs relating to the same VCC UE may have been anchored when the VCC AS receives a SIP INVITE due to DT IMRN or VDN. This may occur in the event that the UE does not succeed in releasing all inactive dialogs. If at least one SIP dialog exists for the user identified in the P-Asserted-Identity header field and a 2xx response has been sent for each dialog and the VCC AS is not able to identify one dialog for DT, then the VCC AS shall send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the DT. Otherwise, the identification of the associated dialog is subject to the following conditions:

- if only one SIP dialog exists for the user identified in the P-Asserted-Identity header field and a SIP 2xx response has been sent and there is active audio media, then continue the DT;
- if no SIP dialogs exist for the user identified in the P-Asserted-Identity header field where there is active audio media and a SIP 2xx response has been sent, then send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the DT;
- if more than one SIP dialog exists for the user identified in the P-Asserted-Identity header field and exactly one dialog exists where there is active audio media and a SIP 2xx response has been sent for that dialog, then:
  - if the remaining dialogs have inactive audio media, then the VCC AS may release the inactive dialogs and continue the DT procedures or the VCC AS may send a SIP 480 (Temporarily Unavailable) response to reject the SIP INVITE request relating to the DT.

Continuing the DT procedures, the VCC AS shall

1. populate the SIP reINVITE request as follows:
   i. set the Request-URI to the URI contained in the Contact header returned at the creation of the dialog with the remote user; and.
   ii. include a new SDP offer, setting the media characteristics as received in the SIP INVITE request due to DT IMRN or VDN, by following the rules of [MMD Part-4].
2. send a SIP 183 Session Progress response.
3. release the IMRN if allocated.

Upon receiving the SIP ACK request initiated from MGCF, the VCC AS shall initiate release of the old access leg by sending a SIP BYE request toward the S-CSCF for sending to the served VCC UE.

## 5.8 VCC Application Message Format

The VCC Application message shall have the following format:

| 8 bits | 8 bits | 8 bits | Variable |
|---|---|---|---|
| VCC Message Type | Identifier | Length | VCC Message Data |

***Table 1*** **VCC Application Message Format**

- *VCC Message Type:*

  The Message Type field is one octet, and identifies the type of SMS packet. When a packet is received with an invalid Message Type field, it is silently discarded.

  The following are the values for Message Type defined in this document:

  0x00 - Domain-Attachment-Status

  Others - Reserved

- *Identifier:*

  The Identifier field is one octet, and aids in matching replies to requests. The value of this field shall be different for every new request sent by the UE. For retransmissions, the Identifier field shall not change.

- *Length:*

  The Length field is one octet. It indicates the length of the packet, in octets, including the VCC Message Type, Identifier, Length, and VCC Message Data fields.

- *VCC Message Data:*

  The contents of this field are formatted according to the content of VCC Message Type.

### 5.8.1 Message Types

#### 5.8.1.1 Domain-Attachment-Status

The Domain-Attachment-Status message is sent by the UE to indicate to the VCC AS its attachment to a particular domain. This message indicates to the VCC AS whether the UE is attached to CS-only, IMS-only, or both domains. A summary of the Domain-Attachment-Status packet format is shown below. The fields are transmitted from left to right.

| 8 bits | 8 bits | 8 bits | 40 bits |
|---|---|---|---|
| VCC Message Type | Identifier | Length | VCC Message Data |

***Table 2*** **Message Type: Domain Attachment Status**

- *VCC Message Type*

  The Message Type is set to 0x00 for this message.

- *Identifier:*

  The Identifier field is set to a unique value for this message. The value of this field shall be different for every new request sent by the UE.

- *Length:*

The length field shall be set to 0x08.

- *VCC Message Data:*

The VCC Message Data shall have the following format.

| 8 bits | 32 bits |
|---|---|
| Domain Status | Timestamp |

***Table 3*** **VCC Message Data**

- *Domain-Status*:

The Domain-Status field can contain one of the following values:

0x00 CS-only

0x01 IMS-only

0x02 CS-IMS

NOTE: IMS-only and CS-IMS are not used in this specification. All other values are reserved.

- *Timestamp:*

The Timestamp field specifies the time at which the message was generated and is represented in seconds since January 1, 1900 00:00 UTC.