3GPP TSG CN Plenary
Meeting #10, Bangkok, Thailand
6th – 8th December 2000

Tdoc NP-000683

3GPP TSG-CN-WG1, Meeting #14
20 - 24 November, 2000
Cardiff, Wales

Tdoc N1-001378 ~~N1-001189~~

Agenda Item: 8.2

WI / Topic: SIP call control protocol for the IM subsystem

Source: Lucent Technologies

Title: Proposed scope and contents for IP multimedia subsystem signalling flows

Effected Specifications / Releases: 24.228

Document for: Decision

Date: 8th November 2000

This document contains the output of the CN~~1~~ SIP ~~ad-hoc #1~~ drafting within the CN1 meeting held ~~October~~ November 2000 in ~~Sophia Antipolis~~Cardiff, and previously contained in N1-00118~~914~~9.

~~The only amendment made to this version are to incorporate the latest style and template changes.~~

3GPP TS 24.228 V0.1.0 (2000-11)

Technical Specification

3rd Generation Partnership Project;
Technical Specification Group Core Network;
Signalling flows for the IP multimedia call control based on SIP and SDP

(Release 5)

GLOBAL SYSTEM FOR
MOBILE COMMUNICATIONS

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Valbonne - FRANCE
Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

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Contents

Foreword

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

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

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the document.

Introduction

This clause is optional. If it exists, it is always the third unnumbered clause.

1 Scope

The present document gives examples of signalling flows for the the IP multimedia call control based on SIP and SDP.

These signalling flows demonstrate the interaction with the IP-connectivity network (GPRS), and with the protocol provided at the Cx interface.

These signalling flows provide detailed signalling flows, which expand on the overview information flows provided in 23.228.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

This specification may contain references to pre-Release-4 GSM specifications. These references shall be taken to refer to the Release 5 version where that version exists. Conversion from the pre-Release-4 number to the Release 4 (onwards) number is given in subclause 6.1 of 3GPP TR 41.001.

3 Definitions, symbols and abbreviations

3.1 Definitions

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

3.2 Symbols

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

3.3 Abbreviations

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

4 Methodology

Editor’s note: This clause is to give a general key to the interpretation of the signalling flows that are provided elsewhere in this document.

Editor’s note: Should the flows be broken down for individual networks, as shown in annex B of 23.228 (i.e. procedure blocks), or should the signalling flows be end-to-end, as shown in annex C of 23.228.

Editor’s note: The level of detail provided in the individual signalling flows should be as follows:

- A sequence diagram showing all methods involved in the transaction, and identifying all methods in the sequence with a referenceable identifier. This diagram may be a copy or a derivation of a diagram in 23.228, although there may be many more such diagrams. As such, the diagrams should use the same template and drawing package as the diagrams in 23.228. As such, as 23.228 evolves, so will this document.
- Using the referencerable identifier, an example of the contents of the method, showing all appropriate headers and body information (including SDP).
- Using the referencerable identifier, an example of the contents of the flows to the HSS, and the data stored in the HSS. Note that this may require some agreement with working group CN4.
- Using the referencerable identifier, an example of the data stored in the SIM. Note that this may require some agreement with TSG T.
- ……

Editor's note: The number of clauses at header level 1 is intended to be open ended, and header levels below this may be proken down as required. The lowest level clauses will always be x.x.1 – Normal operation, and x.x.2 – Exceptional operation. An emphasis is placed on the use of generic error handling where possible, particularly for the case of syntactic errors, therefore the "Exceptional operation subclause may well be empty, or used to show service level rejection of the request.

Editor's note: Some of the flows will have fixed contents, and some will be examples. Some will be constrained to a set of values. We need a mechanism of distinguishing these flow contents from each other.

Editor's note: It may be possibly to use the same flow contents to describe different message flows, in order to make the document more concise. This needs to be done after the various routeing and via headers have been inserted, and after the various naming conventions for the URLs have been rationalised.

5 Signalling flows for error handling

6 Elements of signalling flows for provision of IP-connectivity network

Editor's note: The intent of this subclause is to provide set of subflows, that can be referred to from other flows, that show the creation of appropriate PDP contexts, and the maintenance and tearing down of those PDP contexts.

Editor's note: The mechanism for communicating the CSCF address that the proxy forwards INVITEs to in the 200 OK message is for further study.

Editor's note: Input is required from SA3 on the authentication of the user and which network entity does this.

7 Signalling flows for REGISTER

Editor's note: In the absence of information, break as 23.228

Editor's note: The following issues, contributed in N1-001094 issue 2, needs to be reflected in flows for REGISTER, and for subsequent flows after REGISTER.

As a result of the registration procedures of TS 23.228 section 5.3, the UE and its Serving-CSCF have exchanged identity and routeing information, and have left behind a "trail of breadcrumbs" to enable future signalling messages sent by the UE to reach the S-CSCF (for call attempts from the UE), and signalling messages sent by the S-CSCF to reach the UE (for call attempts destined to the UE).

For signalling messages initiated by the UE, there are several ways to implement this "trail of breadcrumbs":

1. All of the message routeing information could be stored in the UE. This would likely take the form of a SIP ‘Route’ header, and would include information about the P-CSCF, any optional I-CSCF, and the S-CSCF. This ‘Route’ header would be included in all INVITE requests sent by the UE.

2. The mechanism of draft-dcsgroup-sip-state-02 could be extended to allow the CSCF to establish state information during registration, to be returned in all future INVITE requests.

3. All of the message routeing information could be stored in the P-CSCF, and added to the INVITE request sent by the UE

4. Each of the CSCFs could store a portion of the routeing information, the ‘next hop’ from each, so that the P-CSCF stores the name/address of the I-CSCF (or S-CSCF directly), and the I-CSCF stores the name/address of the S-CSCF.

Choice (1) and (2) minimize the storage requirements of the CSCFs. However, they cause additional information to be transferred over the air interface from the UE to P-CSCF.

Choices (3) and (4) minimize the storage requirements of the UE, and reduce the message size of the INVITE request. However, they require the P-CSCF to store information about all the UEs currently located in the area it serves. Note this includes all roaming mobiles, which is beyond the records normally stored about subscribers of the service.

For signalling messages regarding call attempts to the UE (i.e. mobile terminations), there are again several ways to implement the “trail of breadcrumbs”:

1. All of the message routeing information could be stored in the subscriber’s entry in his home network’s HSS. In addition to the S-CSCF name/address, routeing information from the S-CSCF to the UE could be included, such as a SIP ‘Route’ header. This is information that is written only at time of registration, and fetched only in handling of the initial INVITE request for a new call.

2. All of the message routeing information could be stored in the S-CSCF, and added to the INVITE request as part of the service control.

3. Each of the CSCFs could store a portion of the routeing information, the ‘next hop’ from each, so that the S-CSCF stores the name/address of the I-CSCF (or P-CSCF directly), and the I-CSCF stores the name/address of the P-CSCF.

Choice (1) has the advantage of storing the information in a place where there is already per-subscriber information, and adds no new storage requirements on the CSCFs.

Choices (2) and (3) seem to have no clear advantages.

7.1 Start of registration – User not registered

7.2 Continuation of registration – S-CSCF in home network

7.3 Continuation of registration – S-CSCF in visited network

7.4 Mobile initiated deregistration

7.5 Network initiated deregistration

8 Signalling flows for session initiation

Editor’s note: These have been listed by procedure block, as defined in 23.228.

Editor's note: The following issues, contributed in N1-001094 issue 3, needs to be reflected in flows for INVITE, and for subsequent flows after INVITE.

The requirement of caller-id-blocking (aka calling-line-identification-blocking, CLIB), in an IP environment requires that the IP address of the caller be blocked as well. If it was not, a mere 'traceroute' would provide the called party essentially all the information of caller-id. The SIP 'Via' and 'Record-Route' and 'Route' headers would also provide identity information about the caller, and should also be blocked.

Hiding of 'Via' headers is discussed in RFC2543 section 6.40.5, though that text is likely to be deleted in future versions of draft-ietf-sip-rfc2543bis. The mechanism should be retained for 3GPP, as a recommended extension to SIP.

Hiding of 'Route' and 'Record-Route' headers is discussed in draft-byerly-sip-hide-route-00. The mechanism should be adopted for 3GPP.

In both of these cases, there are generally two alternatives for hiding this information from the UE.

1. The information in the 'Via', 'Record-Route', or 'Route' headers could be removed from the SIP message and stored in the P-CSCF. When needed for a response or future request, they can be inserted by P-CSCF.

2. The information in the 'Via', 'Record-Route', and 'Route' headers can be encrypted by P-CSCF and the encrypted form be given to the UE. In responses or future requests, the P-CSCF will decrypt the values and restore them to their original values.

Choice (1) clearly increases the storage requirements of the P-CSCF, while choice (2) clearly increases the bandwidth requirements of the air interface.

Editor's note: The following issues, contributed in N1-001094 issue 4, needs to be reflected in flows for INVITE.

If the caller requested their caller-id to be blocked, but the network operator desires to offer the return-call service (*69), some mechanism is needed to hide the caller identity from the UE but still allow it to be addressed in a future call attempt. The PacketCable DCS specification used a 'private-URL' for this purpose, encrypting the destination information. The format of such a 'private-URL' was typically

sip:somelongstringofjibberishthatcanbedecryptedbytheCSCF@S-CSCF;private

There are actually two alternatives for dealing with this type of information

1. The information to be hidden from the user, e.g. caller identity, could be stored in the P-CSCF or S-CSCF. When needed for the subsequent call attempt, it can be inserted by the CSCF.

2. The design followed by DCS could be used, and the hidden information could be encrypted and stored in the UE

Choice (1) clearly increases the storage requirements of the P-CSCF or S-CSCF, while choice (2) clearly increases the bandwidth requirements of the air interface.

Editor's note: The following issues, contributed in N1-001094 issue 5, needs to be reflected in flows for INVITE and subsequent flows.

In developing mechanisms for call features in the PacketCable DCS group, there were several situations where hidden information was given to an endpoint for immediate use in establishing a new call. The DCS design was to keep the SIP proxy stateless, and this information (which included typically special billing arrangements for the new call to be established) was encrypted and given to the endpoint. The 'private-URL' always contained a timeout value, which limited its useable lifetime.

There are actually two alternatives for dealing with this type of information

1. The information to be hidden from the user, e.g. special billing information for a call, could be stored in the P-CSCF or S-CSCF. When needed for the subsequent call attempt, it can be inserted by the CSCF.

2. The design followed by DCS could be used, and the hidden information could be encrypted and stored in the UE

Choice (1) clearly increases the storage requirements of the P-CSCF or S-CSCF, while choice (2) clearly increases the bandwidth requirements of the air interface.

8.1 Origination sequence

8.1.1 (MO#1) Mobile origination, roaming, home control of services

8.1.2 (MO#2) Mobile origination, roaming, with visited network control of services

8.1.3 (MO#3) Mobile origination, located in home network

8.1.4 (PSTN-O) PSTN origination (where the S-CSCF is a MGCF)

8.2 Termination sequence

8.2.1 (MT#1) Mobile termination, roaming, home control of services

8.2.2 (MT#2) Mobile termination, roaming, with visited network control of services

8.2.3 (MT#3) Mobile termination, located in home network

8.2.4 (PSTN-T) PSTN termination (where the S-CSCF is a MGCF)

8.3 Serving-CSCF/MGCF-to-Serving-CSCF/MGCF sequences

8.3.1 (S-S#1) Call origination and termination are served by different network operators, with home control at termination.

8.3.2 (S-S#2) Call origination and termination are served by different network operators, termination is done by visited network control.

8.3.3 (S-S#3) Call origination and termination are served by the same operator, with home control at termination.

8.3.4 (S-S#4) Call origination and termination are served by the same operator, termination is done by visited network control.

9 Signalling flows for session termination

10 etc.

Annex A (informative): Documentation of preliminary material

Editor's note: This annex provides a temporary space for holding the contents of material that is to be achieving maturity, but is not yet regarded as stable.

When the material achieves stability, then it will move to the main body of the document.

Editor's note: Material in this area still needs final formatting, i.e. by the numbering and titling of all tables and figures, and the referencing of these from the text.

7 Signalling flows for REGISTER

Editor's note: Security Related Issues. A well established trust relationship is required between SIP servers of different networks. If the P-CSCF is allowed to alter the SIP REGISTER message, a mechanism is required to establish the trust –relationship between the P-CSCF and I-CSCF. The P-CSCF is required to have the authority to register a visiting mobile.

Editor's note: Security Related Issues The UE is required to send un-encrypted messages to the P-CSCF. The initial REGISTER message as shown in flow 3 can be cryptographically signed by the UE. In this case, the P-CSCF cannot replace the *Contact* field. Thus the above discussed solution for outbound proxy registration REQUIRES the UE to send messages with un-encrypted header fields to the P-CSCF. The P-CSCF can later encrypt these header fields before forwarding to the I-CSCF, if required. The un-encrypted header fields are listed as follows:

- To
- Via
- From
- Contact
- Expires
- Request URI

Editor's note: Call Flow/Protocol Related Issues. Passing S-CSCF selection information through SIP Register message from hI-CSCF to vI-CSCF According to the S2 call flows, the S-CSCF selection information is pull by the hI-CSCF , and forwarded to vI-CSCF via SIP message(REGISTER). The question is why can the vI-CSCF pull this information again from the HSS? If this information has to be forwarded via REGISTER message, how should it be carried? Call flow reference: Figure 2-2, flow 9, 10, and 13

Editor's note: Call Flow/Protocol Related Issues. Identifying Visited Network domain name from REGISTER message In S2 Visited Control Registration flows, hI-CSCF is required to forward the REGISTER message to vI-CSCF once the visited control decision is made by the HSS. In order to obtained the vI-CSCF's address, we need to construct a generic I-CSCF SIP Request URI using the Visited Network Domain Name derived from the REGITER message, and do a DNS look up. The issues is from which part of the message this information should be derived from? Should this information be passed using the message body? Can it be the Contact header (contains P-CSCF name/address), or the Via header (also contains the network address/hostname of P-CSCF), or even the use of the proposed Path extension? Call flow reference: Figure 2-2, flow 13

Editor's note: Call Flow/Protocol Related Issues. Passing forward rout information back to P-CSCF Currently, several solutions are being considered in CN1 WG. One solution is to use the SIP message body to carry this information, and the other is to use the proposed a generic extension to SIP (named Path header) to pass this information around during registration. This may require update to the call flows depends the outcome of CN1 WG decision.

Editor's note: Call Flow/Protocol Related Issues. Maintaining forward route when Firewall I_CSCF is used. When firewall I-CSCF is used, should both I-CSCF and S-CSCF to be maintained in the P-CSCF, or should only I-CSCF to be maintained in P-CSCF? One Solution is only I-CSCF to be maintained by P-CSCF, how does I-CSCF obtained the S-CSCF address? One possibility is to have the S-CSCF information saved in the HSS, and I-CSCF will query the HSS to obtain this information. Another solution is to use the proposed Path header to save both I-CSCF and S-CSCF name in P-CSCF. S-CSCF name should be encrypted by the I-CSCF in this case.

Editor's note: The format of the request URI in the REGISTER message is for further study. Should it be registrar.home_network.net or home_network.net or something else.

Editor's note: Is the formatting of a reregistration REGISTER message identical to the initial REGISTER message? Are any of the fields different, e.g. request-URI?

Editor's note: Current flows arbitrarily assign a timer value of 7200. Is this a recommended value, or can any value be chosen, and if so, what are the constraints?

7.2 Continuation of registration – S-CSCF in home network

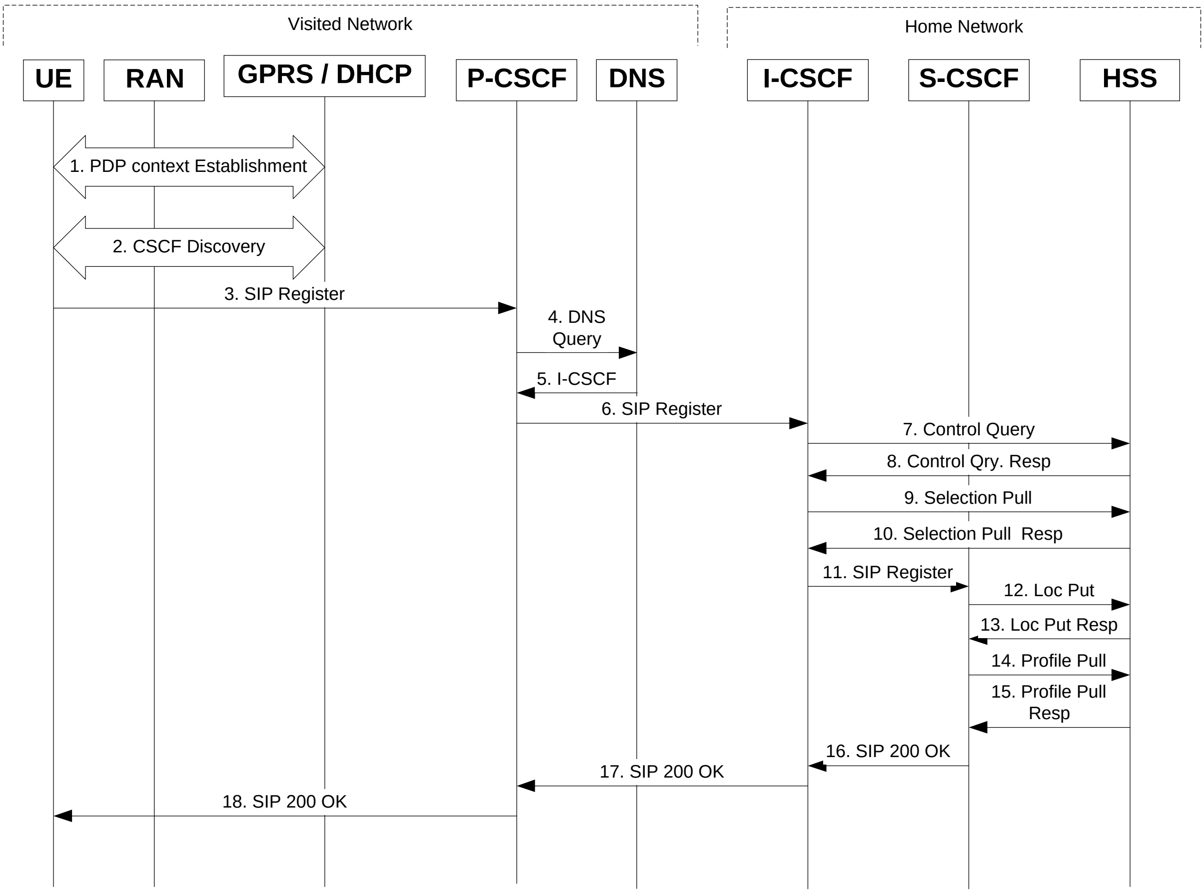


Figure 2-2: UE roaming with Visited control

1 **GPRS Attach / PDP Context Establishment (UE ⇔ GPRS)**

This signalling flow is shown to indicate the GRPS Attach and PDP Context Activation procedures that must be completed prior to application registration. When complete, the UE will have acquired an IP address (provided by the GGSN) which serves as the host address for the duration of the PDP context.

2 **CSCF Discovery (UE ⇔ GPRS/ DHCP)**

This signalling flow is the procedure to discover the Proxy CSCF using DHCP. When complete, the address of the proxy server (pcscf.home.net) is made known to the UE.

3 **SIP REGISTER (UE ⇨ P-CSCF)**

The purpose of this message is to register the user's SIP URI with a S-CSCF in the home network. This message is routed to the P-CSCF because it is the only SIP server known to the UE for the voice application. In the following SIP message, the Contact field contains the user's host address.

The P-CSCF will perform two actions, binding and forwarding. The binding is between the User's SIP address (user1@home_network.com) and the host (terminal) address ([5555::aaa:bbb:ccc:ddd]) which was acquired during PDP context activation process.

Table 7.xx

```
REGISTER sip:registrar.home_network.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Contact: <Sip:[5555::aaa:bbb:ccc:ddd]>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

4 **DNS Query (P-CSCF ⇨ DNS)**

Based on the user's URI, the P-CSCF determines that UE is registering from a visiting domain and performs a DNS query to locate the I-CSCF in the home network. The look up in the DNS is based on the address specified in the Request URI.

5 **DNS Response (DNS ⇨ P-CSCF)**

The DNS provides the P-CSCF with an address of the I-CSCF in the home network.

6 **SIP REGISTER (P-CSCF ⇨ I-CSCF)**

Since this P-CSCF is call stateful, it is required to be in the path for all Mobile Originated and Mobile Terminated requests for this user. To ensure this, the P-CSCF has to put itself into the path for future requests. One solution of achieving this is to have the P-CSCF as the contact point for this user at the home registrar.

To do this the P-CSCF creates a temporary SIP URI for the user called user1%40home_network.net@pcscf.visited_network.net. As part of its internal registration procedure the P-CSCF binds the temporary SIP URI to the user's SIP URI which was also bound to the IP address of the UE as shown in signalling flow 3. The P-CSCF then forwards the REGISTER message for user1@home_network.net, to the home registrar, using a contact address of user1%40home.net@pcscf.visited_network.net.

This signalling flow shows the SIP Register message being forward from the P-CSCF to the I-CSCF in the home domain.

Table 7.xx

```
REGISTER sip:registrar.home_network.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited_network.com
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Contact: <sip:user1%40home_network.net@pcscf.visited_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

7 **Control Query (I-CSCF ⇨ HSS)**

This signalling flow is initiated from the I-CSCF to the HSS for purpose of choosing the serving network. The HSS selects whether the serving network is in the home network or the visited network.

8 **Control Query Response (HSS ⇨ I-CSCF)**

Query Response is sent from the HSS to the I-CSCF with information required to select the serving system.

9 **Selection Pull (I-CSCF ⇨ HSS)**

The I-CSCF sends Selection Pull to the HSS to request the information related to the required S-CSCF capabilities which shall be input into the S-CSCF selection function.

10 **Selection Pull Response (HSS ⇨ I-CSCF)**

The HSS sends the Selection Pull Response to the I-CSCF with the capability information required for S-CSCF selection. The I-CSCF uses this information to select a suitable S-CSCF.

11 **SIP REGISTER (I-CSCF ⇨ S-CSCF)**

This signalling flow forwards the SIP Register message from the I-CSCF to the S-CSCF selected. The address in the request line is changed to the address of the S-CSCF.

Table 7.xx

```
REGISTER sip: scscf3.home_network.net SIP/2.0
Via: SIP/2.0/UDP icscf2.home_network.com
Via: SIP/2.0/UDP pcscf1.visited_network.com
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Contact: <sip:user1%40home.net@pcscf.visited_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

12 **Location Put (S-CSCF ⇨ HSS)**

The S-CSCF shall send Location Put to the HSS. The HSS stores the S-CSCF name for that subscriber.

13 **Location Response (HSS ⇨ S-CSCF)**

The HSS sends Location Put Response to the I-CSCF to acknowledge the sending of Location Put.

14 **Profile Pull (S-CSCF ⇨ HSS)**

On receipt of the Location Put Response signalling flow, the S-CSCF shall send the Location Pull signalling flow (subscriber identity) to the HSS in order to be able to download the subscriber profile to the S-CSCF.

15 **Profile Pull Response (HSS ⇨ S-CSCF)**

The HSS returns the signalling flow Profile Pull Resp (subscriber profile) to the S-CSCF. The S-CSCF shall store the subscriber profile for that indicated user.

16 **SIP 200 OK (S-CSCF ⇨ I-CSCF)**

The S_CSCF determines the contact name for the P-CSCF (S-CSCF or I-CSCF), and add this information to the 200 OK response. The S-CSCF sends acknowledgment to the I-CSCF indicating that Registration was successful. This message will traverse the path that the REGISTER message took as described in the Via list.

Table 7.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2.home_network.com
Via: SIP/2.0/UDP pcscf1.visited_network.com
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

17 **SIP 200 OK (I-CSCF ⇨ P-CSCF)**

The I-CSCF forwards acknowledgment from the S-CSCF to the P-CSCF indicating that Registration was successful. This message will traverse the path that the REGISTER message took as described in the Via list.

Table 7.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited_network.com
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

18 **SIP 200 OK (P-CSCF ⇨ UE)**

The P_CSCF stores the contact name in the 200 OK response (S-CSCF or I-CSCF) for the registered user for the duration of the registration, and removes the contact information from the 200 OK response. The P-CSCF then forwards acknowledgment from the I-CSCF to the UE indicating that Registration was successful.

Table 7.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

7.3 Continuation of registration – S-CSCF in visited network

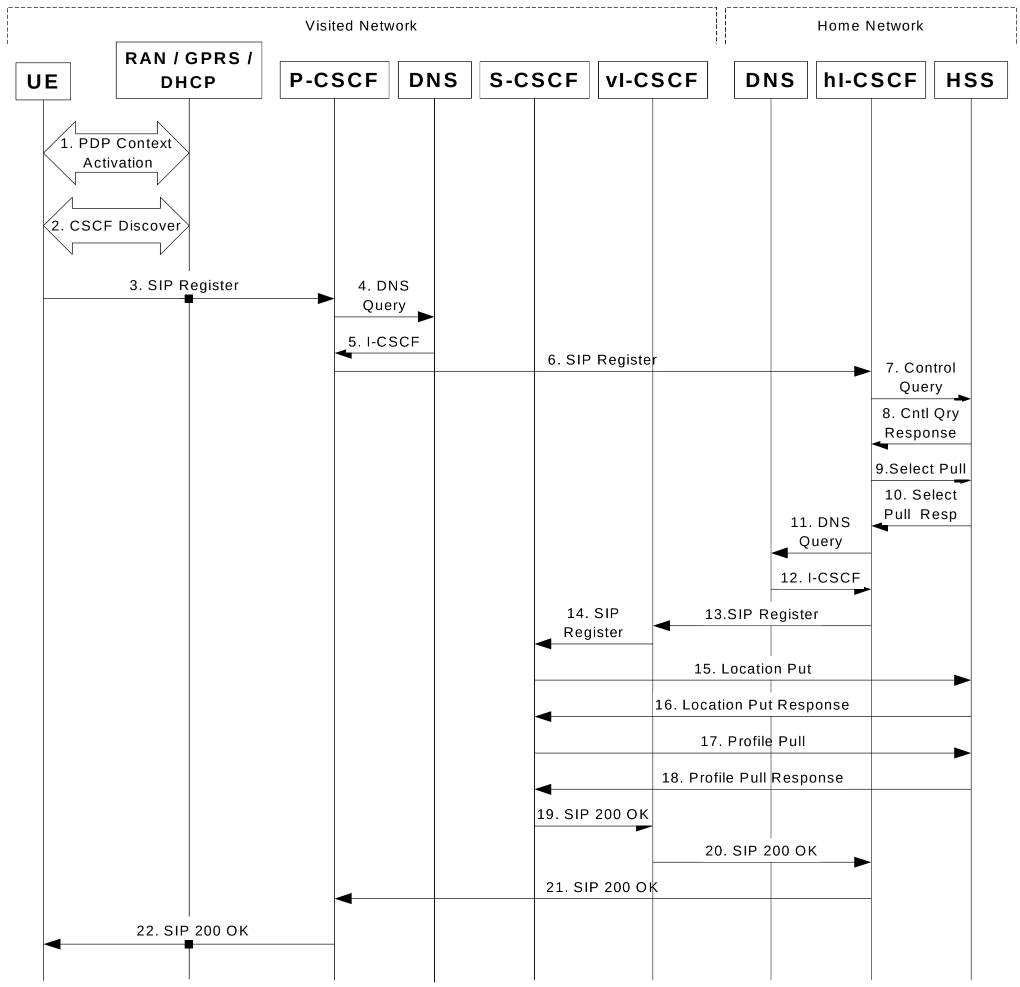


Figure 7.x

1 **GPRS Attach / PDP Context Establishment (UE ⇔ GPRS)**

This signalling flow is shown to indicate the GRPS Attach and PDP Context Activation procedures that must be completed prior to application registration. When complete, the UE will have acquired an IP address (provided by the GGSN) which serves as the host address for the duration of the PDP context.

2 **CSCF Discovery (UE ⇔ GPRS/ DHCP)**

This signalling flow is the procedure to discover the Proxy CSCF using DHCP. When complete, the address of the proxy server (pcscf.home.net) is made known to the UE.

3 **SIP REGISTER (UE ⇨ P-CSCF)**

The purpose of this message is to register the user's SIP URI with a S-CSCF in the home network. This message is routed to the P-CSCF because it is the only SIP server known to the UE for the voice application. In the following SIP message, the Contact field contains the user's host address.

The P-CSCF will perform two actions, binding and forwarding. The binding is between the User's SIP address (user1@home_network.com) and the host (terminal) address ([5555::aaa:bbb:ccc:ddd]) which was acquired during PDP context activation process.

Table 7.xx

```
REGISTER sip:registrar.home_network.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
```

```
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Contact: <Sip:[5555::aaa:bbb:ccc:ddd]>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

4 **DNS Query (P-CSCF ⇨ DNS)**

Based on the user's URI, the P-CSCF determines that UE is registering from a visiting domain and performs a DNS query to locate the I-CSCF in the home network. The look up in the DNS is based on the address specified in the Request URI.

5 **DNS Response (DNS ⇨ P-CSCF)**

The DNS provides the P-CSCF with an address of the I-CSCF in the home network (hI-CSCF).

6 **SIP REGISTER (P-CSCF ⇨ vI-CSCF)**

Since this P-CSCF is call stateful, it is required to be in the path for all Mobile Originated and Mobile Terminated requests for this user. To ensure this, the P-CSCF has to put itself into the path for future requests. One solution of achieving this is to have the P-CSCF as the contact point for this user at the home registrar.

To do this the P-CSCF creates a temporary SIP URI for the user called user1%40home_network.net@pcscf.visited_network.net. As part of its internal registration procedure the P-CSCF binds the temporary SIP URI to the user's SIP URI which was also bound to the IP address of the UE as shown in step 3. The P-CSCF then forwards the REGISTER message for user1@home_network.net, to the home registrar, using a contact address of user1%40home.net@pcscf.visited_network.net.

This signalling flow shows the SIP Register message being forward from the P-CSCF to the hI-CSCF in the home domain.

Table 7.xx

```
REGISTER sip:registrar.home_network.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited_network.com
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Contact: <sip:user1%40home_network.net@pcscf.visited_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

7 **Control Query (hI-CSCF ⇨ HSS)**

This signalling flow is initiated from the vI-CSCF to the HSS for purpose of choosing the serving network. The HSS selects whether the serving network is in the home network or the visited network.

8 **Control Query Response (HSS ⇨ hI-CSCF)**

Query Response is sent from the HSS to the I-CSCF with information required to select the serving system.

9 **Selection Pull (hI-CSCF ⇨ HSS)**

The hI-CSCF sends Selection Pull to the HSS to request the information related to the required S-CSCF capabilities which shall be input into the S-CSCF selection function.

10 **Selection Pull Response (HSS ⇨ hI-CSCF)**

The HSS sends the Selection Pull Response to the hI-CSCF with the capability information required for S-CSCF selection. The hI-CSCF uses this information to select a suitable S-CSCF.

Since this information is used to feed into the S-SCSCF selection function, in the case of visited, it need to be passed to I-CSCF in the visited network (vI-CSCF). It is assumed this information will be carried within the SIP REGISTER message from hI-CSCF to the vI-CSCF. How this is done is for further study Issues 1).

11 **DNS Query (hI-CSCF⇨ DNS)**

Based on the domain name in the CONTACT header of the REGISTER message, the hI-CSCF in the home network performs a DNS query to locate the vI-CSCF in the visited network.

12 **DNS Response (DNS⇨hI-CSCF)**

The DNS provides the P-CSCF with an address of the I-CSCF in the visited network (vI-CSCF).

13 **SIP REGISTER (hI-CSCF ⇨ vI-CSCF)**

This signalling flow forwards the SIP Register message from the hI-CSCF to the vI-CSCF selected. The Request URI has been changed to *registrar.visited_network.net* to indicate visited control. When vI-CSCF receives this request, it will use the selection information that is carried within the SIP REGISTER request to select the S-CSCF in the visited network.

Table 7.xx

```
REGISTER sip: registrar.visited_network.net SIP/2.0
Via: SIP/2.0/UDP icscf2.home_network.com
Via: SIP/2.0/UDP pcscf1.visited_network.com
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Contact: <sip:user1%40home.net@pcscf.visited_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

14 **SIP REGISTER (vI-CSCF ⇨ S-CSCF)**

This signalling flow forwards the SIP Register message from the vI-CSCF to the S-CSCF selected. The address in the request line is changed to the address of the S-CSCF.

Table 7.xx

```
REGISTER sip: scscf3.visited_network.net SIP/2.0
Via: SIP/2.0/UDP icscf2.visited_network.com
Via: SIP/2.0/UDP icscf2.home_network.com
Via: SIP/2.0/UDP pcscf1.visited_network.com
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Contact: <sip:user1%40home.net@pcscf.visited_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

15 **Location Put (S-CSCF ⇨ HSS)**

The S-CSCF shall send Location Put to the HSS. The HSS stores the S-CSCF name for that subscriber.

16 **Location Response (HSS ⇨ S-CSCF)**

The HSS sends Location Put Response to the vI-CSCF to acknowledge the sending of Location Put.

17 **Profile Pull (S-CSCF ⇨ HSS)**

On receipt of the Location Put Response signalling flow, the S-CSCF shall send the Location Pull signalling flow (subscriber identity) to the HSS in order to be able to download the subscriber profile to the S-CSCF.

18 **Profile Pull Response (HSS ⇨ S-CSCF)**

The HSS returns the signalling flow Profile Pull Resp (subscriber profile) to the S-CSCF. The S-CSCF shall store the subscriber profile for that indicated user.

19 **SIP 200 OK (S-CSCF ⇨ vI-CSCF)**

The S_CSCF determines the contact name for the P-CSCF (S-CSCF), and add this information to the 200OK response. The S-CSCF sends acknowledgment to the vI-CSCF indicating that Registration was successful. This message will traverse the path that the REGISTER message took as described in the Via list.

Table 7.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2.visited_network.com
Via: SIP/2.0/UDP icscf2.home_network.com
Via: SIP/2.0/UDP pcscf1.visited_network.com
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

20 **SIP 200 OK (vI-CSCF ⇨ hI-CSCF)**

The vI-CSCF sends acknowledgment to the hI-CSCF indicating that Registration was successful. This message will traverse the path that the REGISTER message took as described in the Via list.

Table 7.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2.home_network.com
Via: SIP/2.0/UDP pcscf1.visited_network.com
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

21 **SIP 200 OK (hI-CSCF ⇨ P-CSCF)**

The hI-CSCF forwards acknowledgment from the S-CSCF to the P-CSCF indicating that Registration was successful. This message will traverse the path that the REGISTER message took as described in the Via list.

Table 7.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited_network.com
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

22 **SIP 200 OK (P-CSCF ⇨ UE)**

The P_CSCF stores the contact name for the registered user for the duration of the registration, and removes the information from the 200 OK response. The P-CSCF then forwards acknowledgment from the hI-CSCF to the UE indicating that Registration was successful.

Table 7.xx

```
SIP/2.0 200 OK
```

```
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home_network.net>
To: <sip:user1@home_network.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

8

Editor's note: The contact field of a flow from the remote network should contain the information of the originating/terminating network endpoint. This could be the P-CSCF or the S-CSCF of the originating/terminating network and which requires further study.

Editor's note: Example naming conventions for different entities need to be defined and used consistently throughout the document.

8.1 Origination sequence

Editor's note: If an I-CSCF is to be used as a firewall I-CSCF then does it need to be statefull? According to the flows developed in 23.228, the I-CSCF (e.g.,look at Figure 1, messages 2b1 and 2b2) does not have a look up shown to find the address of the HSS. Does this imply statefulness of I-CSCFs?

Editor's note: Need to show procedures on how the MGCF determines the location of S-CSCF. If it is fielding calls from the PSTN and forwarding this to the S-CSCF then it is playing the role of an I-CSCF for PSTN Originated calls. This means that the MGCF will have to also contain I-CSCF functionality. One approach might be to have the MGCF always pass such Origination messages to an I-CSCF at all times and then the I-CSCF can address all issues of routeing the INVITE to the proper place.

Editor's note: For all UE to P-CSCF flows, the contents of the Contact header within the INVITE would appear to be redundant, but it is a mandatory header. Need to agree what the UE should populate this field with. Current contents is not the most appropriate. This value will be inserted by the P-CSCF.

Editor's note: Contents of the body (SDP) are not yet included in these flows. The presence, absence and content of the SDP needs to be addressed. Also need to identify which methods it appears in.

Editor's note: Flows 7-10 and 11-13 are missing the record-route and route header respectively. Need to add these headers with appropriate contents.

Editor's note: Contents of request-URI in INVITE flows other than that from the UE is for further study. Should it be as shown, or should it change to constrain the routeing of the method?

8.1.1 (MO#1) Mobile origination, roaming, home control of services

This origination procedure applies to roaming subscribers under home control. The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure. During registration, the home network decides to exercise home control of calls by this UE, and therefore allocates a S-CSCF in the home network. The home network advertises either the S-CSCF or an I-CSCF as the entry point from the visited network.

When registration is complete, P-CSCF knows the name/address of the next hop in the signalling path toward the serving-CSCF, either I-CSCF (if the home network wanted to hide their internal configuration) or S-CSCF (if there was no desire to hide the network configuration). I-CSCF, if it exists in the signalling path, knows the name/address of S-CSCF.

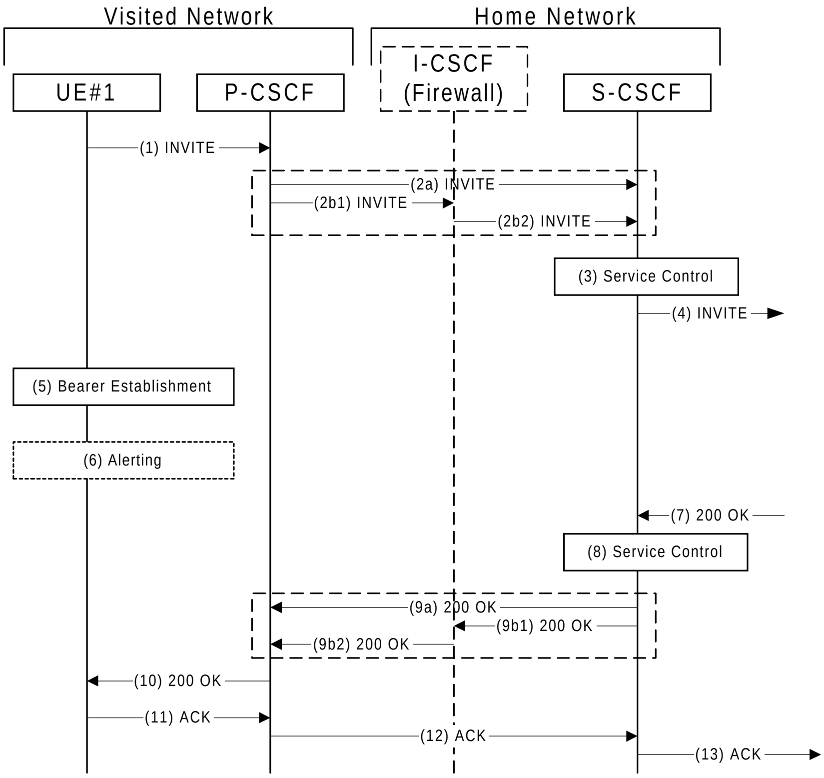


Figure 8.xx

Procedure MO#1 is as follows:

1. UE sends the SIP INVITE request to the P-CSCF determined via the CSCF discovery mechanism.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

SDP material in this Body section.
```

2. P-CSCF remembers (from the registration procedure) the next hop CSCF for this UE.

2a)

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

SDP material in this Body section.
```

This next hop is either the S-CSCF that is serving the visiting UE (choice (a)), or an I-CSCF within the home network that is performing the configuration hiding function for the home network operator (choice (b)).

(2a) If the home network operator does not desire to keep their network configuration hidden, the name/address of the S-CSCF was provided during registration, and the INVITE request is forwarded directly to the S-CSCF.

(2b) If the home network operator desires to keep their network configuration hidden, the name/address of an I-CSCF in the home network was provided during registration, and the INVITE request is forwarded through this I-CSCF to the S-CSCF.

(2b1) P-CSCF forwards the INVITE request to I-CSCF

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(2b2) I-CSCF forwards the INVITE request to S-CSCF

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf.home.com
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

3. S-CSCF performs any origination service control required by this subscriber.

4. S-CSCF forwards the request, as specified by the S-CSCF to S-CSCF procedures.

This SIP message assumes that message 2b2 is used to setup this call. In the case that message 2a is used the Via header containing the I-cscf .home.com is not included in the message.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
Via: SIP/2.0/UDP i-cscf.home.com
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

5. The originating UE and the terminating UE co-operatively establish the bearer path for the media flow.
6. The called UE may optionally perform alerting. If so, it signals this to the calling party.
7. When the called party answers, the called UE sends a SIP 200-OK final response, as specified by the termination procedures and the S-CSCF to S-CSCF procedures, to S-CSCF.

This SIP message assumes that message 2b2 is used to setup this call. . In the case that message 2a is used the Via header containing the I-cscf .home.com is not included in the message.

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf.home.com
Via: SIP/2.0/UDP i-cscf.home.com
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#2@home.com>
Content-Length: 0
```

8. S-CSCF performs whatever service control is appropriate for the completed call
9. S-CSCF sends a SIP 200-OK final response along the signalling path back to the call originator. Based on the choice made in (2) above, this response may either be sent directly from S-CSCF to P-CSCF (choice (a)), or be sent indirectly through I-CSCF firewall (choice (b)).

9a)

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Length: 0
```

9b1)

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP i-cscf.home.com
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Length: 0
```

9b2)

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Length: 0
```

10. P-CSCF sends a SIP 200-OK final response to the call originator

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Length: 0
```

11-13. UE responds to the final response with a SIP ACK message which is forwarded via the P-CSCF and S-CSCF.

11.

Table 8.xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

12 a)

Table 8.xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

12b1)

Table 8.xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

12b2)

Table 8.xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf.home.com
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

13.

Editor's note: Note here that the firewall I-CSCF is used. This is not shown in the flow from 23.228. It should also be noted here that message 12 should be broken down into (12a) and (12b1) and (12b2) as message 2 is broken up.

Table 8.xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
Via: SIP/2.0/UDP i-cscf.home.com
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

8.1.2 (MO#2) Mobile origination, roaming, with visited network control of services

This origination procedure applies to roaming subscribers, under visited network control.

The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure described in section 5.2.1. During registration, the home network decides to accept an offer of visited network control of calls by this UE, and therefore the visited network allocates a S-CSCF.

When registration is complete, P-CSCF knows the name/address of S-CSCF.

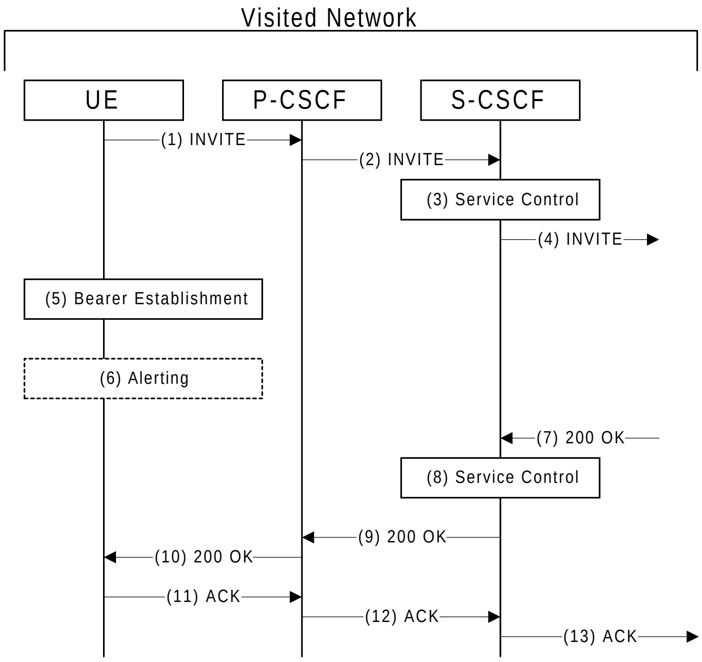


Figure xx

Procedure MO#2 is as follows:

1. UE sends the SIP INVITE request to the P-CSCF determined via the CSCF discovery mechanism.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

2. P-CSCF remembers (from the registration procedure) the next hop CSCF for this UE. This next hop is the S-CSCF that is serving the visiting UE.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

3. S-CSCF performs any origination service control required by this subscriber.

4. S-CSCF forwards the request, as specified by the S-CSCF to S-CSCF procedures.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.visited.com
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

5. The originating UE and the terminating UE cooperatively establish the bearer path for the media flow.
6. The called UE may optionally perform alerting. If so, it signals this to the calling party.
7. When the called party answers, the called UE sends a SIP 200-OK final response, as specified by the termination procedures and the S-CSCF to S-CSCF procedures, to S-CSCF.

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf.visited.com
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

8. S-CSCF performs whatever service control is appropriate for the completed call
9. S-CSCF sends a SIP 200-OK final response along the signalling path back to the call originator.

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Length: 0
```

10. P-CSCF sends a SIP 200-OK final response to the call originator

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Length: 0
```

11-13. UE responds to the final response with a SIP ACK message which is forwarded via the P-CSCF and the S-CSCF.

11.

Table 8.xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

12.

Table xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

13.

Table xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.visited.com
Via: SIP/2.0/UDP p-cscf.visited.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

8.1.3 (MO#3) Mobile origination, located in home network

This origination procedure applies to subscribers located in their home service area.

The UE is located in the home network, and determines the P-CSCF via the CSCF discovery procedure described in section 5.2.1. During registration, the home network allocates an S-CSCF in the home network.

When registration is complete, P-CSCF knows the name/address of S-CSCF.

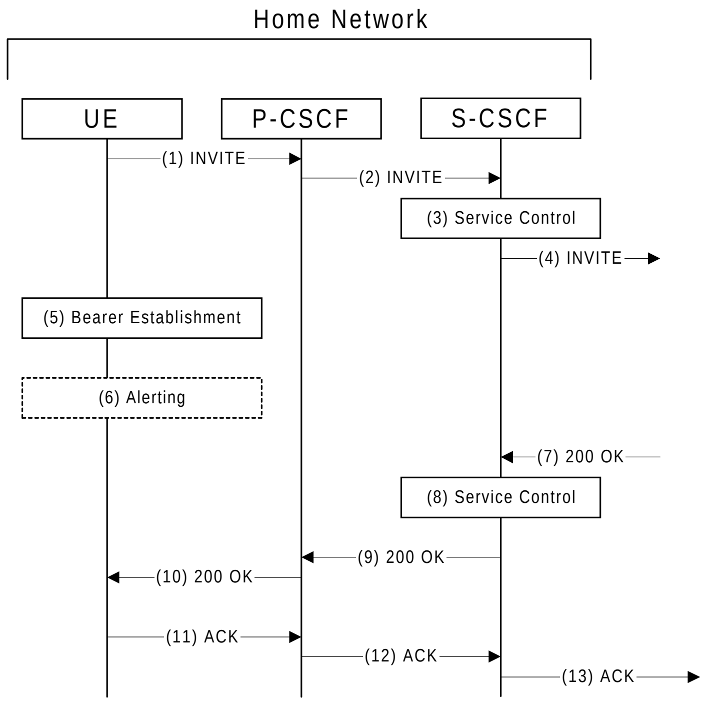


Figure xx

Procedure MO#4 is as follows:

1. UE#1 sends the SIP INVITE request to the P-CSCF determined via the CSCF discovery mechanism.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

2. P-CSCF remembers (from the registration procedure) the next hop CSCF for this UE. In this case it forwards the INVITE to the S-CSCF in the home network.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.home.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

3 S-CSCF performs any origination service control required by this subscriber.

4 S-CSCF forwards the request, as specified by the S-CSCF to S-CSCF procedures.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
Via: SIP/2.0/UDP p-cscf.home.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com> ----is this needed?
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

5. UE#1 establishes the bearer path for this session
6. UE#1 provides ringback in response to alerting at the destination
7. The SIP final response, 200-OK, is passed back to S-CSCF over the signalling path. This is typically generated by UE#2 when the subscriber has accepted the incoming call attempt.

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf.home.com
Via: SIP/2.0/UDP p-cscf.home.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Length: 0
```

8. S-CSCF performs any origination service control required by call completion.
9. S-CSCF passes the 200-OK response back to P-CSCF, following the path of the INVITE request of step (2) above.

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p-cscf.home.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Length: 0
```

10. P-CSCF passes the 200-OK response back to UE#1

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345600@UE#1.host
CSeq: 1 INVITE
Contact: <sip:UE#1@home.com>
Content-Length: 0
```

11-13. UE#1 responds to the SIP final response with a ACK message which is forwarded via the P-CSCF and the S-CSCF.

11.

Table xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

12.

Table xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.home.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

13.

Table xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
Via: SIP/2.0/UDP p-cscf.home.com
Via: SIP/2.0/UDP UE#1.host
From: <sip:UE#1@home.com>
To: <Called-Party-Identifier>
Call-ID: 12345601@UE#1.host
CSeq: 1 ACK
Content-Length: 0
```

8.1.4 (PSTN-O) PSTN origination (where the S-CSCF is a MGCF)

Editor's note: Flow 4 and flow need a contact contents.

The MGCF in the IM CN subsystem is a SIP endpoint that initiates requests on behalf of the PSTN and Media Gateway. The subsequent nodes consider the signalling as if it came from a S-CSCF. The MGCF incorporates the network security functionality of the S-CSCF. This MGCF does not invoke Sservice Control, as this may be carried out in the GSTN or at the terminating S-CSCF.This origination procedure can be used for any of the S-CSCF to S-CSCF procedures.

Due to routeing of calls within the PSTN, this origination procedure will only occur in the home network of the destination subscriber. However, due to the possibility of visited network control, the destination subscriber may be in a different operator's network. Further, due to cases of call fowarding and electronic surveillance, the destination of the call through the IM CN subsystem may actually be another PSTN termination.

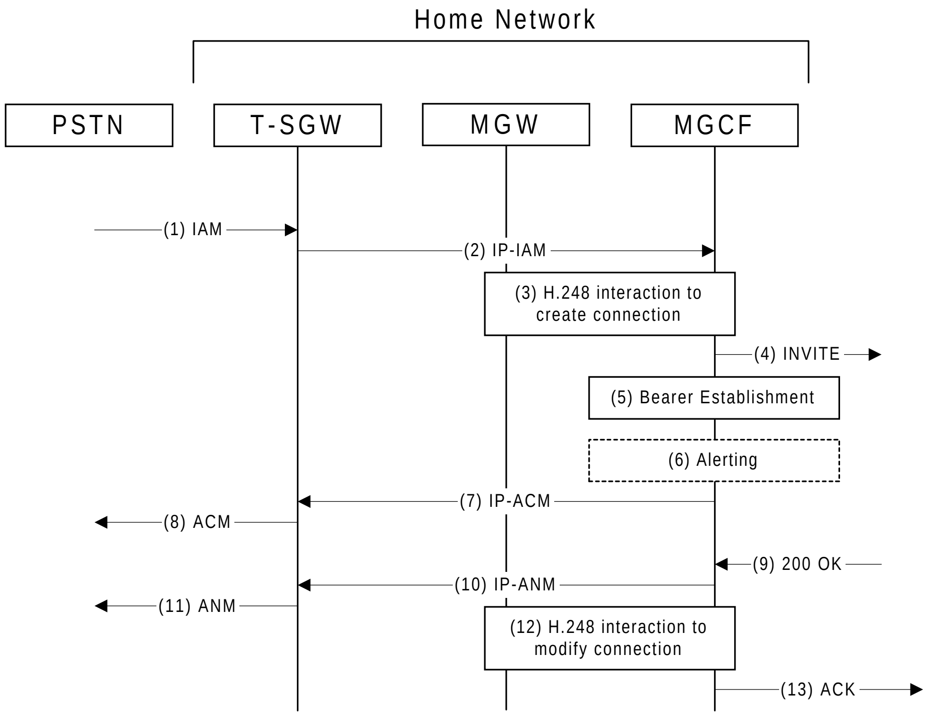


Figure xx

The PSTN Origination procedure is as follows:

1. The PSTN establishes a bearer path to the MGW, and signals to the T-SGW with a SS7 IAM message, giving the trunk identity and destination information
2. The T-SGW forwards the SS7 message, encapsulated in IP, to the MGCF.
3. The MGCF initiates a H.248 command, to seize the trunk and an IP port.
4. The MGCF initiates a SIP INVITE request, as per the proper S-CSCF to S-CSCF procedure.

Table xx

INVITE <Called-Party-Identifier> SIP/2.0 Via: SIP/2.0/UDP mgcf.home.com From: <Calling-Party-Identifier> To: <Called-Party-Identifier> Call-ID: 12345600@mgcf.home.com CSeq: 1 INVITE Content-Type: application/sdp Content-Length: xxx SDP material in this Body section.

5. The MGW and the terminating UE cooperatively establish the bearer path for the media flow
6. The called UE may optionally perform alerting. If so, it signals this to the calling party
7. If alerting is being performed, the MGCF forwards an IP-ACM message to T-SGW
8. If alerting is being performed, the T-SGW forwards a SS7 ACM message
9. When the called party answers, the terminating and S-CSCF to S-CSCF procedures result in a SIP 200 OK final response being sent to MGCF

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP mgcf.home.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@mgcf.home.com
CSeq: 1 INVITE
Content-Length: 0
```

10. MGCF forwards an IP-ANM message to T-SGW

11. T-SGW forwards an ANM message to the PSTN

12. MGCF initiates a H.248 command to alter the connection at MGW to make it bidirectional

13. MGCF acknowledges the SIP final response with a SIP ACK message

Table xx

```
ACK <Called-Party-Identifier> SIP/2.0
via: SIP/2.0/UDP mgcf.home.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@mgcf.home.com
CSeq: 1 ACK
Content-Length: 0
```

8.2 Termination sequence

Editor's note: The P-CSCF does the stripping of headers, especially the Via headers. In the case of Via headers from ACK messages, this headers can be stripped and discarded. For INVITES and its responses this header can be stripped and then appended to the response coming from the UE.

8.2.1 (MT#1) Mobile termination, roaming, home control of services

This termination procedure applies to roaming subscribers under home control. The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure. During registration, the home network decides to exercise home control of calls to/from this UE, and therefore allocates a S-CSCF in the home network. The home network advertises either the S-CSCF, or an I-CSCF firewall, as the entry point from the visited network.

When registration is complete, S-CSCF knows the name/address of its next hop in the signalling path, either I-CSCF or P-CSCF, I-CSCF (if it exists) knows the name/address of P-CSCF, and P-CSCF knows the name/address of the UE. The mechanism by which this information is stored is for further study.

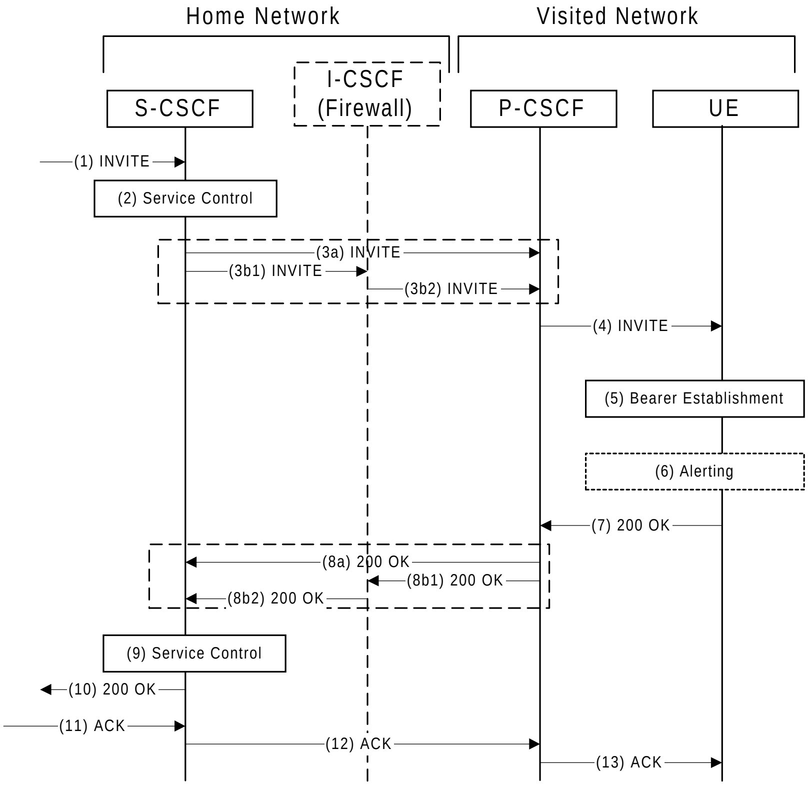


Figure 8.x

Procedure MT#1 is as follows:

1. The calling party sends the SIP INVITE request, via one of the origination procedures, and via one of the Inter-Serving procedures, to the Serving-CSCF for the terminating subscriber.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

2 S-CSCF performs any termination service control required by this subscriber

3 S-CSCF remembers (from the registration procedure) the next hop CSCF for this UE. It forwards the INVITE to the P-CSCF in the visited network, possibly through an I-CSCF.

3a)

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
```

```
Content-Length: xxx

 SDP material in this Body section.
```

3 This next hop is either the P-CSCF that is serving the visiting UE (choice (a)), or an I-CSCF within the home network that is performing the configuration hiding function for the home network operator (choice (b)).

(3a) If the home network operator does not desire to keep their network configuration hidden, the INVITE request is forwarded directly to the P-CSCF.

(3b) If the home network operator desires to keep their network configuration hidden, the INVITE request is forwarded through an I-CSCF to the P-CSCF.

(3b1) S-CSCF forwards the INVITE request to I-CSCF

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(3b2) I-CSCF forwards the INVITE request to P-CSCF

I-CSCF removes all previous Via headers and store them for subsequent addition to the response.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf-firewall.home.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

4. P-CSCF remembers (from the registration procedure) the UE address, and forwards the INVITE to the UE.

.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.visited.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

5. UE cooperatively with the call originator, establishes the bearer path for the media flow.

6. UE may alert the user and wait for an indication from the user before completing the call. If so, it indicates this to the calling party through the alerting procedure.

7. When the called party answers, the UE sends a SIP 200-OK final response to P-CSCF

Assuming 3a) coming in to P-CSCF AND that the P-CSCF DOES NOT remove the Via headers:

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p-cscf.visited.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

8. P-CSCF sends a SIP 200-OK final response along the signalling path back to the S-CSCF
   Based on the choice made in (3) above, this response may either be sent directly from P-CSCF to S-CSCF (choice (a)), or be sent indirectly through the I-CSCF firewall (choice (b)).

8a)

Assuming 3a) coming in to P-CSCF AND that the P-CSCF DOES NOT remove the Via headers:

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf.home.com
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

8b1)

Assuming 3b2) coming in to P-CSCF

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP i-cscf-firewall.home.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

8b2)

Assuming 3b2) coming in to P-CSCF

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf.home.com
Via: all the previous places that this message has traversed (since any firewall)
From: <Called-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
```

```
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

9. S-CSCF performs whatever service control is required for the call completion

10. S-CSCF forwards the SIP 200-OK final response along the signalling path back to the call originator.

Table 8.xx

```
SIP/2.0 200 OK
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

11-13. The calling party responds to the 200-OK final response with a SIP ACK message which is forwarded via the S-CSCF and the P-CSCF.

11.

Table 8.xx

```
ACK <Called-Party-Identifier> SIP/2.0
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@Calling-Party.host
CSeq: 1 ACK
Content-Length: 0
```

12.

Table 8.xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@Calling-Party.host
CSeq: 1 ACK
Content-Length: 0
```

13.

.

Table 8.xx

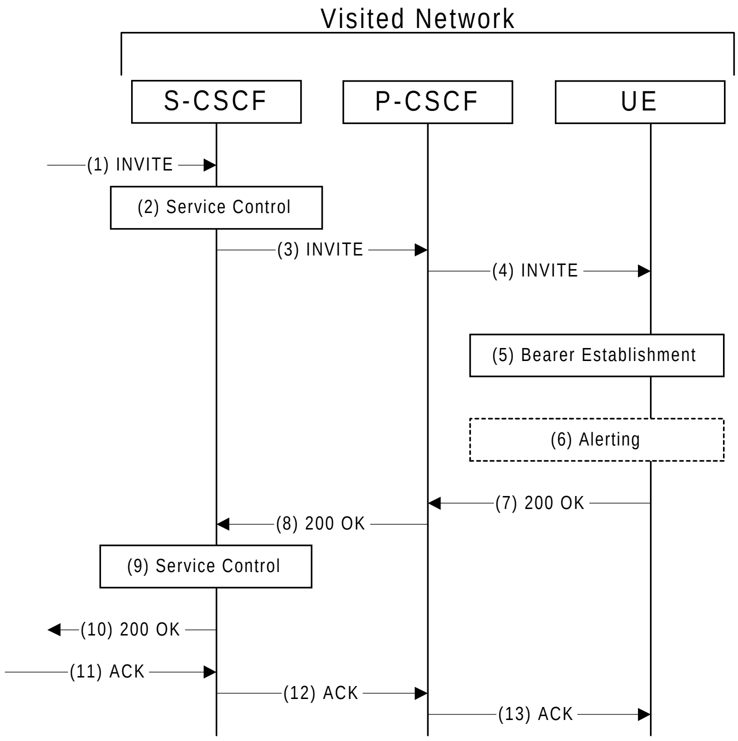
```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.visited.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@Calling-Party.host
CSeq: 1 ACK
Content-Length: 0
```

8.2.2 (MT#2) Mobile termination, roaming, with visited network control of services

This termination procedure applies to roaming subscribers, under visited network control.

The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure described in section 5.2.1. During registration, the home network decided to accept an offer of visited network control of calls by/to this UE, and therefore the visited network allocates the S-CSCF.

When registration is complete, S-CSCF knows the name/address of P-CSCF, and P-CSCF knows the name/address of the UE. The mechanism by which this information is stored is for further study.



Procedure MT#2 is as follows:

1. The caller sends the SIP INVITE request, via one of the origination procedures, and via one of the Inter-Serving CSCF procedures, to the Serving-CSCF for the terminating subscriber.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

2. S-CSCF performs any termination service control required by this subscriber

3. S-CSCF remembers (from the registration procedure) the next hop CSCF for this UE. It forwards the INVITE to the P-CSCF in the visited network.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.visited.com
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

4. P-CSCF remembers (from the registration procedure) the UE address, and forwards the INVITE to the UE.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.visited.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

5. UE, cooperatively with the call originator, establishes the bearer path for the media flow.

6. UE may alert the user and wait for an indication from the user before completing the call. If so, it indicates this to the calling party through the alerting procedure.

7. When the called party answers, UE sends a SIP 200-OK final response to P-CSCF.

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p-cscf.visited.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

8. P-CSCF sends a SIP 200-OK final response along the signalling path to S-CSCF.

Table 8.xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf.visited.com
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

9. S-CSCF performs whatever service control is required for the call completion

10. S-CSCF forwards the SIP 200-OK final response along the signalling path back to the call originator

Table 8.xx

```
SIP/2.0 200 OK
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

11-13. The calling party responds to the 200-OK final response with a SIP ACK message which is forwarded via the S-CSCF and the P-CSCF.

11.

Table 8.xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@Calling-Party.host
CSeq: 1 ACK
Contact: <Called-Party-Identifier>
Content-Length: 0
```

12.

Table 8.xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.visited.com
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@Calling-Party.host
CSeq: 1 ACK
Contact: <Called-Party-Identifier>
Content-Length: 0
```

13.

Table 8.xx

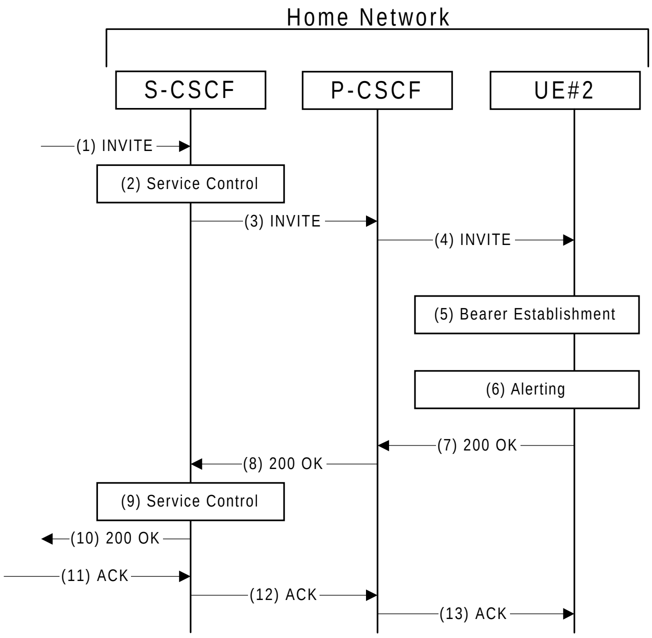
```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.visited.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@Calling-Party.host
CSeq: 1 ACK
Contact: <Called-Party-Identifier>
Content-Length: 0
```

8.2.3 (MT#3) Mobile termination, located in home network

This termination procedure applies to subscribers located in their home service area.

The UE is located in the home network, and determines the hP-CSCF via the CSCF discovery procedures described in section 5.2.1. During registration, the home network allocates a S-CSCF in the home network, hS-CSCF.

When registration is complete, hS-CSCF knows the name/address of hP-CSCF, and hP-CSCF knows the name/address of the UE.



Procedure MT#3 is as follows:

(1) UE#1 sends the SIP INVITE request, via one of the origination procedures, and via one of the Serving-Serving CSCF procedures, to the Serving-CSCF for the terminating subscriber.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(2) S-CSCF performs any termination service control required by this subscriber

(3) S-CSCF remembers (from the registration procedure) the next hop CSCF for this UE. It forwards the INVITE to the P-CSCF in the home network.

Table 8.xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(4) P-CSCF remembers (from the registration procedure) the UE address, and forwards the INVITE to the UE.Note that P-CSCF strips the Via headers.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.home.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(5) UE#2 establishes the bearer path for this call

(6) UE#2 alerts the user

(7) UE#2 generates the SIP final response, 200-OK, when the subscriber accepts the incoming call.

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP p-cscf.home.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

(8) P-CSCF forwards the 200-OK to S-CSCF, following the path of the INVITE request in step (3) above

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf.home.com
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

(9) S-CSCF performs any service control required on call completion.

(10) S-CSCF forwards the 200 OK final response, as per the appropriate S-CSCF to S-CSCF procedure.

Table xx

```
SIP/2.0 200 OK
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

11-13. The call originator responds to the 200-OK by sending the ACK message to UE#2 via the S-CSCF and the P-CSCF.

11.

Table xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@Calling-Party.host
CSeq: 1 ACK
Contact: <Called-Party-Identifier>
Content-Length: 0
```

12.

Table xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
Via: all the previous places that this message has traversed (since any firewall)
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@Calling-Party.host
CSeq: 1 ACK
Contact: <Called-Party-Identifier>
Content-Length: 0
```

13.

Table xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP p-cscf.home.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@Calling-Party.host
CSeq: 1 ACK
Contact: <Called-Party-Identifier>
Content-Length: 0
```

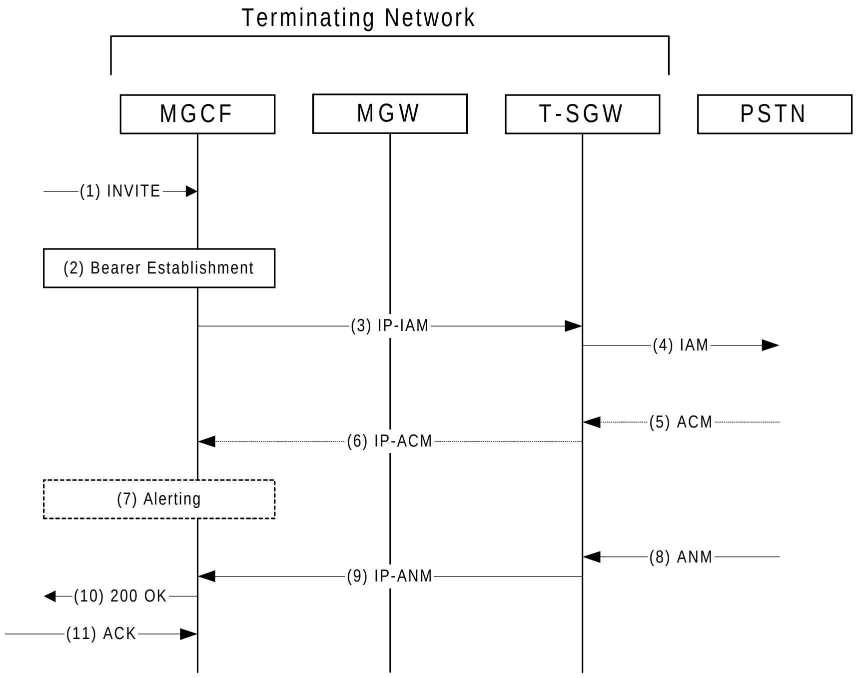
8.2.4 (PSTN-T) PSTN termination (where the S-CSCF is a MGCF)

The MGCF in the IM CN subsystem is a SIP endpoint that initiates and receives requests on behalf of the PSTN and Media Gateway (MGW).Other nodes consider the signalling as if it came from a S-CSCF. The MGCF incorporates the network security functionality of the S-CSCF.

PSTN termination may be done in the same operator's network as the S-CSCF of the call originator, e.g. the visited network for visited network control or the home network for home network control. Therefore, the location of the MGCF/MGW/T-SGW are given only as “Terminating Network” rather than “Home Network” or “Visited Network.”

Further, agreements between network operators may allow PSTN termination in a network other than the originator's visited network or home network. This may be done, for example, to avoid long distance or international tariffs.

This termination procedures can be used for any of the inter-serving procedures, in place of the S-CSCF.



The PSTN termination procedure is as follows:

1. MGCF receives an INVITE request, through one of the origination procedures and via one of the inter-serving procedures.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

2. MGCF initiates the establishment of the bearer path between the calling party and the MGW
3. MGCF sends an IP-IAM message to the T-SGW
4. T-SGW receives the IP-IAM and sends the SS7 IAM message into the PSTN.
5. The PSTN establishes the path to the destination. It may optionally alert the destination user before completing the call. If so, it responds with an SS7 ACM message
6. If the PSTN is alerting the destination user, T-SGW sends an IP-ACM message to MGCF
7. If the PSTN is alerting the destination user, MGCF and the calling party cooperatively perform the alerting function for the originating user.
8. When the called party answers, the PSTN sends an SS7 ANM message to T-SGW
9. T-SGW sends an IP-ANM message to MGCF
10. MGCF sends a SIP 200-OK final response along the signalling path back to the call originator

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf.home.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

11. The Calling party acknowledges the final response with a SIP ACK message

Table xx

```
ACK <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf.home.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345601@Calling-Party.host
CSeq: 1 ACK
Contact: <Called-Party-Identifier>
Content-Length: 0
```

8.3 Serving-CSCF/MGCF-to-Serving-CSCF/MGCF sequences

Editor's note: Contents of the flows require alignment with those for the originating and terminating flows.

8.3.1 (S-S#1) Call origination and termination are served by different network operators, with home control at termination

The Serving-CSCF handling call origination performs an analysis of the destination address, and determines that it belongs to a subscriber of a different operator. The request is therefore forwarded (optionally through an I-CSCF within the originating operator's network) to a well-known entry point in the destination operator's network, the I-CSCF. The I-CSCF queries the HSS for current location information, and finds the subscriber either located in the home service area, or roaming with home control. The I-CSCF therefore forwards the request to the S-CSCF serving the destination subscriber.

Origination sequences that share this common S-CSCF to S-CSCF procedure are:

MO#1 Mobile origination, roaming, home control of services. The "Originating Network" of S-S#1 is therefore a visited network, and S-CSCF#1 is located in the home network.

MO#2 Mobile origination, roaming, with visited control of services. The "Originating Network" of S-S#1 is therefore a visited network, and S-CSCF#1 is also located in the visited network.

MO#3 Mobile origination, located in home service area. The "Originating Network" of S-S#1 is therefore the home network, and S-CSCF#1 is also located in the home network.

PSTN-O PSTN origination. The "Originating Network" of S-S#1 is the home network. The elements labeled UE#1 and S-CSCF#1 are combined into the single MGCF of the PSTN-O procedure.

Termination sequences that share this common S-CSCF to S-CSCF procedure are:

MT#1 Mobile termination, roaming, home control of services. The "Terminating Network" of S-S#1 is a visited network.

MT#3 Mobile termination, located in home service area. The "Terminating Network" of S-S#1 is the home network.

PSTN-T PSTN termination. The "Terminating Network" of S-S#1 is the home network. The elements labeled S-CSCF#2 and UE#2 are combined into the single MGCF of the PSTN-T procedure.

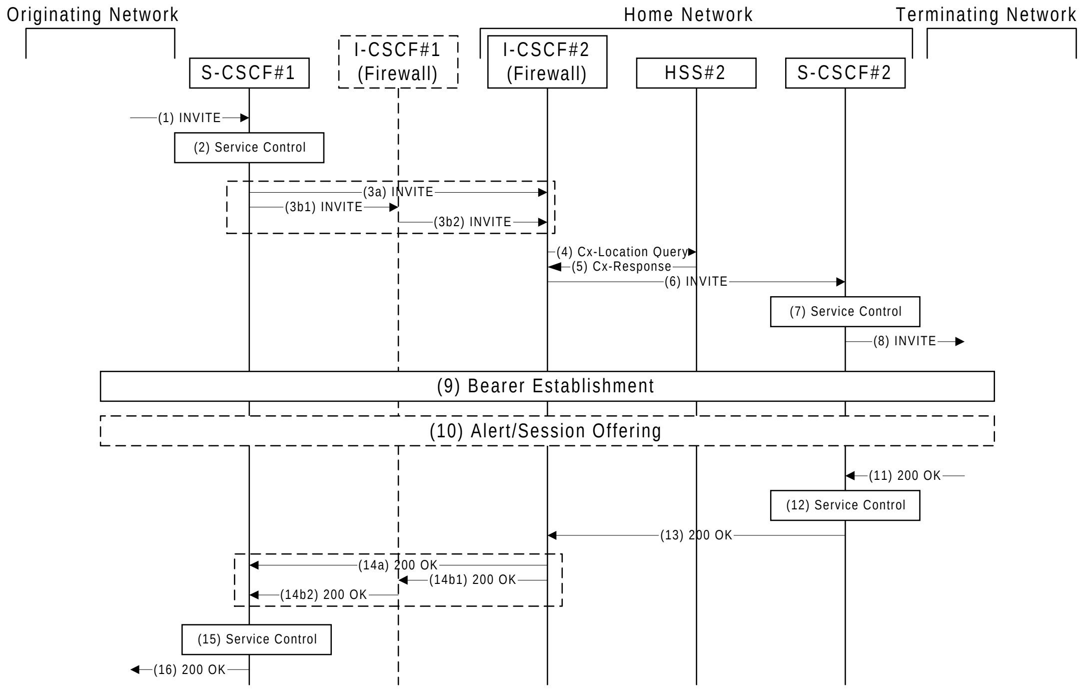


Figure xx

Editor's note: I-CSCF labelling of Firewall appears to be incorrect. Need to check against updated 23.228. This has an impact on the via headers.

Procedure S-S#1 is as follows:

1. The SIP INVITE request is sent from the UE to S-CSCF#1 by the procedures of the originating signalling flow.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

2. S-CSCF#1 performs whatever service control logic is appropriate for this call attempt.

3. S-CSCF#1 performs an analysis of the destination address, and determines the network operator to whom the subscriber belongs. For S-S#1, signalling flow (2) is an inter-operator message to the I-CSCF entry point for the terminating subscriber. If the originating operator desires to keep their internal configuration hidden, then S-CSCF#1 forwards the INVITE request through an I-CSCF (choice (b)); otherwise S-CSCF#1 forwards the INVITE request directly to I-CSCF#2, the well-known entry point into the terminating subscriber's network (choice (a)).

   (3a) If the originating network operator does not desire to keep their network configuration hidden, the INVITE request is sent directly to I-CSCF#2.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(3b) If the originating network operator desires to keep their network configuration hidden, the INVITE request is forwarded through an I-CSCF in the originating operator's network, I-CSCF#1.

(3b1) The INVITE request is sent from S-CSCF#1 to I-CSCF#1

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(3b2) I-CSCF#1 performs the configuration-hiding modifications to the request and forwards it to I-CSCF#2

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf#1.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

4 I-CSCF (at the border of the terminating subscriber's network) queries the HSS for current location information.

Editor's note: Not clear which field is mapped to the information in the Cx query. If the To: field is used, then if forwarding has occurred, then this may no longer carry the original information.

5 HSS responds with the address of the current Serving-CSCF for the terminating subscriber.

6 I-CSCF forwards the INVITE request to the S-CSCF that will handle the call termination.

This message assumes that we have received message 3b2 at I-CSCF#2: . In the case that message 3a is used the Via header containing the I-cscf #1-firewall.network#1.com is not included in the message and Via headers for all the previous places that this message has traversed since any I-cscf firewall are contained in the message.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf#1-firewall.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

Editor's note: Request URI should reflect information received from the HSS, and currently it does not.

7. S-CSCF#2 performs whatever service control logic is appropriate for this call attempt

8. The sequence continues with the signalling flows determined by the termination procedure.

This message assumes that we have received message 3b2 at I-CSCF#2: . In the case that message 3a is used the Via header containing the I-cscf #1-firewall.network#1.com is not included in the message and Via headers for all the previous places that this message has traversed since any I-cscf firewall are contained in the message.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf#1-firewall.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

9. Bearer path authorization and establishment, which will require additional SIP messages to carry the bearer information of the called party back to the calling party

10. The alerting phase, if required, which may require additional SIP messages to carry the indication from the called party back to the calling party

11. The SIP final response, 200-OK, is passed back to UE#1 over the signalling path. This is typically generated by UE#2 when the subscriber has accepted the incoming call attempt.

This message assumes that we have received message 3b2 at I-CSCF#2: . In the case that message 3a is used the Via header containing the I-cscf #1-firewall.network#1.com is not included in the message and Via headers for all the previous places that this message has traversed since any I-cscf firewall are contained in the message.

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf#1-firewall.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

12. S-CSCF#2 performs whatever service control logic is appropriate for this call completion

13. The 200-OK is passed to the I-CSCF#2.

This message assumes that we have received message 3b2 at I-CSCF#2: . In the case that message 3a is used the Via header containing the I-cscf #1-firewall.network#1.com is not included in the message and Via headers for all the previous places that this message has traversed since any I-cscf firewall are contained in the message.

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP i-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf#1-firewall.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

14. The 200-OK is passed to the S-CSCF#1 using the optional I-CSCF#1 if required.

This message assumes that we have received message 3b2 at I-CSCF#2: . In the case that message 3a is used the Via header containing the I-cscf #1-firewall.network#1.com is not included in the message and Via headers for all the previous places that this message has traversed since any I-cscf firewall are contained in the message.

14b1)

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP i-cscf#1-firewall.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

14b2)

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

15. S-CSCF#1 performs whatever service control logic is appropriate for this call completion

16. The 200-OK is returned to the originating network.

Table xx

```
SIP/2.0 200 OK
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
```

```
Contact: <Called-Party-Identifier>
Content-Length: 0
```

Editor's note: There are ACK messages that need to be shown for the completion of this picture.

8.3.2 (S-S#2) Call origination and termination are served by different network operators, termination is done by visited network control

The Serving-CSCF handling call origination performs an analysis of the destination address, and determines that it belongs to a subscriber of a different operator. The request is therefore forwarded (optionally through an I-CSCF within the originating operator's network) to a well-known entry point in the destination operator's network, the I-CSCF. The I-CSCF queries the HSS for current location information, and finds the subscriber is roaming with visited network control. The I-CSCF therefore forwards the request to the entry point provided by the visited network during registration. If the visited network did not want to hide their internal configuration, then this entry point was the S-CSCF serving the destination subscriber; otherwise it is an I-CSCF within the visited network.

Origination sequences that share this common S-CSCF to S-CSCF procedure are:

MO#1 Mobile origination, roaming, home control of services. The "Originating Network" of S-S#2 is therefore a visited network, and S-CSCF#1 is located in the home network.

MO#2 Mobile origination, roaming, with visited control of services. The "Originating Network" of S-S#2 is therefore a visited network, and S-CSCF#1 is also located in the visited network.

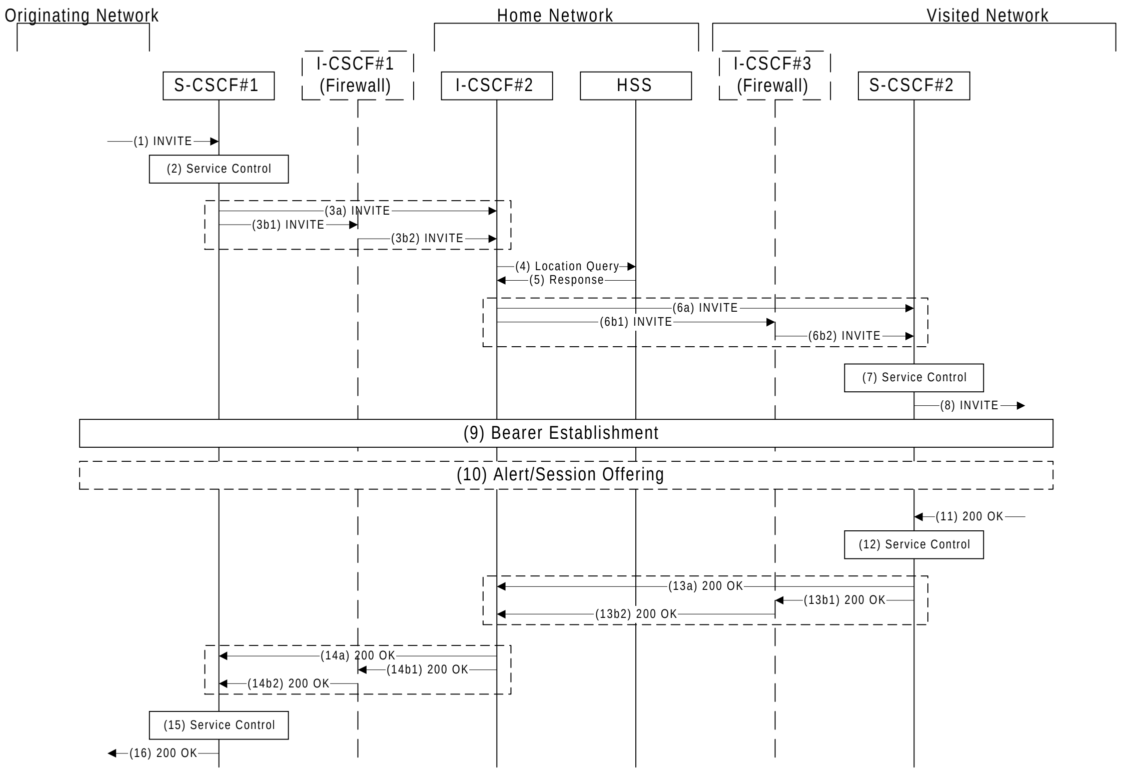
MO#3 Mobile origination, located in home service area. The "Originating Network" of S-S#2 is therefore the home network, and S-CSCF#1 is also located in the home network.

PSTN-O PSTN origination. The "Originating Network" of S-S#2 is the home network. The element S-CSCF#1 is the MGCF of the PSTN-O procedure.

Termination sequences that share this common S-CSCF to S-CSCF procedure are:

MT#2 Mobile termination, roaming, with visited control of services.

PSTN-T PSTN termination. The element labeled S-CSCF#2 is the MGCF of the PSTN-T procedure. This only occurs if there is an agreement between the network operators for termination of PSTN calls in the visited network.



Procedure S-S#2 is as follows:

1. The SIP INVITE request is sent from the UE to S-CSCF#1 by the procedures of the originating signalling flow.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

2. S-CSCF#1 performs whatever service control logic is appropriate for this call attempt

3. S-CSCF#1 performs an analysis of the destination address, and determines the network operator to whom the subscriber belongs. In this case it determines the destination subscriber belongs to a different network operator. Signalling flow (2) is an inter-operator message to the I-CSCF entry point for the terminating subscriber. If the originating operator desires to keep their internal configuration hidden, then S-CSCF#1 forwards the INVITE request through an I-CSCF (choice (b)); otherwise S-CSCF#1 forwards the INVITE request directly to I-CSCF#2, the well-known entry point into the terminating subscriber's network (choice (a)).

   (3a) If the originating network operator does not desire to keep their network configuration hidden, the INVITE request is sent directly to I-CSCF#2.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
```

```
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(3b) If the originating network operator desires to keep their network configuration hidden, the INVITE request is forwarded through I-CSCF#1 in the originating operator's network.

(3b1) The INVITE request is sent from S-CSCF#1 to I-CSCF#1

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(3b2) I-CSCF#1 performs the configuration-hiding modifications to the request and forwards it to I-CSCF#2

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf#1.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

4. I-CSCF#2 queries the HSS for current location information.

5. HSS responds with the address of the current Serving-CSCF for the terminating subscriber.

6. I-CSCF#2 forwards the INVITE request to the entry point whose address was provided during registration. This entry point is either the S-CSCF that is serving the visiting UE (choice (a)), or an I-CSCF within the visited network that is performing the configuration hiding function for the visited network operator (choice (b)).

   (6a) If the visited network operator does not desire to keep their network configuration hidden, the name/address of the S-CSCF was provided during registration, and the INVITE request is forwarded directly to S-CSCF#2.

This message assumes that 3b2 was received at I-CSCF#2. In the case that message 3a is used the Via header containing the I-cscf #1-firewall.network#1.com is not included in the message and Via headers for all the previous places that this message has traversed since any I-cscf firewall are contained in the message.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf#2.network#2.com
```

```
Via: SIP/2.0/UDP i-cscf#1-firewall.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(6b) If the visited network operator desires to keep their network configuration hidden, the name/address of an I-CSCF in the visited network was provided during registration, I-CSCF#3, and the INVITE request is forwarded through I-CSCF#3 to S-CSCF#2.

(6b1) I-CSCF#2 forwards the INVITE request to I-CSCF#3

This message assumes that 3b2 was received at I-CSCF#2. In the case that message 3a is used the Via header containing the I-cscf #1-firewall.network#1.com is not included in the message and Via headers for all the previous places that this message has traversed since any I-cscf firewall are contained in the message.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf#1firewall.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(6b2) I-CSCF#3 forwards the INVITE request to S-CSCF#2

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf#3-firewall.network#3.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

7. S-CSCF#2 performs whatever service control logic is appropriate for the call attempt.
8. The sequence continues with the signalling flows determined by the termination procedure.

This message assumes that 3b2 was received at I-CSCF#2 and that 6b2 was received at S-CSCF#2. In the case that message 6a is used the Via header containing the I-cscf #3-firewall.network#3.com is not included in the message and Via headers for all the previous places that this message has traversed since any I-cscf firewall are contained in the message.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf#3.network#3.com
Via: SIP/2.0/UDP i-cscf#3-firewall.network#3.com
```

```
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

9. The originating and terminating UE cooperatively establish the bearer path for the media flow.

10. The called UE may optionally perform alerting. If so, it signals this to the calling party.

11. When the called party answers, the termination procedure results in a SIP 200-OK final response being sent to S-CSCF#2

This message assumes that 3b2 was received at I-CSCF#2 and 6b2 at S-CSCF#3. . In the case that message 6a is used the Via header containing the I-cscf #3-firewall.network#3.com is not included in the message and Via headers for all the previous places that the original invite message traversed since any I-cscf firewall are contained in the message.

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf#3.network#3.com
Via: SIP/2.0/UDP i-cscf#3-firewall.network#3.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

12. S-CSCF#2 performs whatever service control logic is appropriate for the call completion.

13. S-CSCF#2 sends the SIP 200-OK final response along the signalling path back to the call originator. Based on the choice made in (5) above, this response may either be sent directly from S-CSCF#2 to I-CSCF#2 (choice (a)), or be sent indirectly through I-CSCF#3 (Firewall) (choice (b)).

This message assumes that 3b2 was received at I-CSCF#2 and 6b2 at S-CSCF#3.

13b1)

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP i-cscf#3-firewall.network#3.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

13b2)

This message assumes that 3b2 was received at I-CSCF#2 and 6b2 at S-CSCF#3

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP i-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf#1-firewall.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
```

```
Contact: <Called-Party-Identifier> ----is this needed?
Content-Length: 0
```

14. I-CSCF#2 sends the SIP 200-OK final response along the signalling path back to the call originator. Based on the choice made in (2) above, this response may either be sent directly from I-CSCF#2 to S-CSCF#1 (choice (a)), or be sent indirectly through I-CSCF#1 (Firewall) (choice (b)).

This message assumes that 3b2 was received at I-CSCF#2 and 6b2 at S-CSCF#3

14b1)

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP i-cscf#1-firewall.network#1.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

14b2)

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

15. S-CSCF#1 performs whatever service control logic is appropriate for the call completion.

16. S-CSCF#1 continues by the procedures of the originating signalling flow.

Table xx

```
SIP/2.0 200 OK
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

Editor’s note: That the figure from 23.228 does not show ACKs and these messages need to be added into this section.

8.3.3 (S-S#3) Call origination and termination are served by the same operator, with home control at termination.

The Serving-CSCF handling call origination performs an analysis of the destination address, and determines that it belongs to a subscriber of the same operator. The request is therefore forwarded to a local I-CSCF. The I-CSCF queries the HSS for current location information, and finds the subscriber either located in the home service area, or roaming with home control. The I-CSCF therefore forwards the request to the S-CSCF serving the destination subscriber.

Origination sequences that share this common S-CSCF to S-CSCF procedure are:

MO#1 Mobile origination, roaming, home control of services. The “Originating Network” of S-S#3 is therefore a visited network, and S-CSCF#1 is located in the home network.

MO#2 Mobile origination, roaming, with visited control of services. The “Originating Network” of S-S#3 is therefore a visited network, and S-CSCF#1 is also located in the visited network.

MO#3 Mobile origination, located in home service area. The “Originating Network” of S-S#3 is therefore the home network, and S-CSCF#1 is also located in the home network.

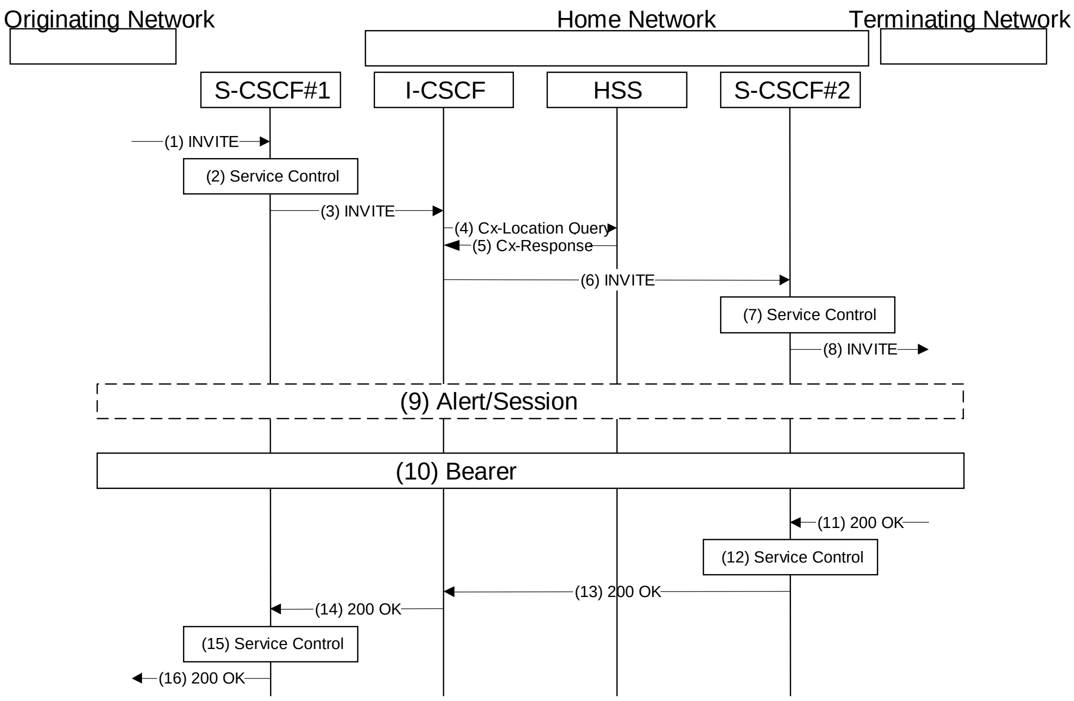
PSTN-O PSTN origination. The “Originating Network” of S-S#3 is the home network. The elements labeled UE#1 and hS-CSCF#1 are combined into the single MGCF of the PSTN-O procedure.

Termination sequences that share this common S-CSCF to S-CSCF procedure are:

MT#1 Mobile termination, roaming, home control of services. The “Terminating Network” of S-S#3 is a visited network.

MT#3 Mobile termination, located in home service area. The “Terminating Network” of S-S#3 is the home network.

PSTN-T PSTN termination. The “Terminating Network” of S-S#3 is the home network. The elements labeled S-CSCF#2 and UE#2 are combined into the single MGCF of the PSTN-T procedure.



Editor’s note: Boxes (9) and (10) appear to be the wrong way round. Will be corrected from the new version of 23.228.

Procedure S-S#3 is as follows:

1. The SIP INVITE request is sent from the UE to S-CSCF#1 by the procedures of the originating signalling flow.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx
```

```
 SDP material in this Body section.
```

2. S-CSCF#1 performs whatever service control logic is appropriate for this call attempt

3. S-CSCF#1 performs an analysis of the destination address, and determines the network operator to whom the subscriber belongs. Since it is local, the request is passed to a local I-CSCF.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

4. I-CSCF queries the HSS for current location information.

5. HSS responds with the address of the current Serving-CSCF for the terminating subscriber.

6. I-CSCF forwards the INVITE request to the S-CSCF that will handle the call termination.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf.network#2.com
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

7. S-CSCF#2 performs whatever service control logic is appropriate for this call attempt

8. The sequence continues with the signalling flows determined by the termination procedure.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf.network#2.com
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

9. Bearer path authorization and establishment, which will require additional SIP messages to carry the bearer information of the called party back to the calling party

10. The alerting phase, if required, which may require additional SIP messages to carry the indication from the called party back to the calling party

11. The SIP final response, 200-OK, is passed back to UE#1 over the signalling path. This is typically generated by UE#2 when the subscriber has accepted the incoming call attempt.

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf.network#2.com
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

12. S-CSCF#2 performs whatever service control logic is appropriate for this call completion

13. The 200-OK is passed to the I-CSCF

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP i-cscf.network#2.com
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

14. The 200-OK is passed to the S-CSCF#1

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

15. S-CSCF#1 performs whatever service control logic is appropriate for this call completion

16. The 200-OK is passed to the Originating Network

Table xx

```
SIP/2.0 200 OK
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

Editor's note: That the figure from 23.228 does not show ACKs and these messages need to be added into this section.

8.3.4 (S-S#4) Call origination and termination are served by the same operator, termination is done by visited network control

The Serving-CSCF handling call origination performs an analysis of the destination address, and determines that it belongs to a subscriber of the same operator. The request is therefore forwarded to a local I-CSCF. The I-CSCF queries the HSS for current location information, and finds the subscriber is roaming with visited network control. The I-CSCF therefore forwards the request to the entry point provided by the visited network during registration. If the visited network did not want to hide their internal configuration, then this entry point was the S-CSCF serving the destination subscriber; otherwise it is an I-CSCF within the visited network.

Origination sequences that share this common S-CSCF to S-CSCF procedure are:

MO#1 Mobile origination, roaming, home control of services. The "Originating Network" of S-S#4 is therefore a visited network, and S-CSCF#1 is located in the home network.

MO#2 Mobile origination, roaming, with visited control of services. The "Originating Network" of S-S#4 is therefore a visited network, and S-CSCF#1 is also located in the visited network.

MO#3 Mobile origination, located in home service area. The "Originating Network" of S-S#4 is therefore the home network, and S-CSCF#1 is also located in the home network.

PSTN-O PSTN origination. The "Originating Network" of S-S#4 is the home network. The element S-CSCF#1 is the MGCF of the PSTN-O procedure.

Termination sequences that share this common S-CSCF to S-CSCF procedure are:

MT#2 Mobile termination, roaming, with visited control of services.

PSTN-T PSTN termination. The element labeled S-CSCF#2 is the MGCF of the PSTN-T procedure. This only occurs if there is an agreement between the network operators for termination of PSTN calls in the visited network.

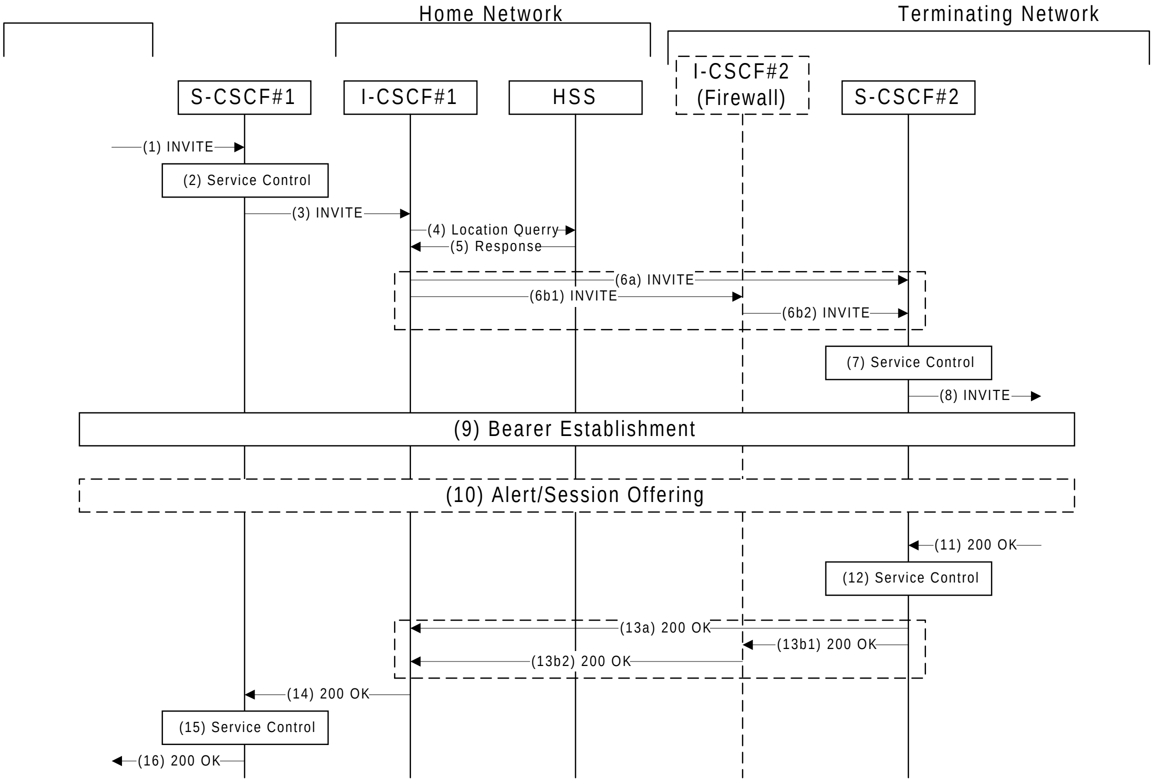


Figure xx

Procedure S-S#4 is as follows:

1. The SIP INVITE request is sent from the UE to S-CSCF#1 by the procedures of the originating signalling flow.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

2. S-CSCF#1 performs whatever service control logic is appropriate for this call attempt

3. S-CSCF#1 performs an analysis of the destination address, and determines the network operator to whom the subscriber belongs. Since it is local, the request is passed to a local I-CSCF, I-CSCF#1.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

4. I-CSCF#1 queries the HSS for current location information.

5. HSS responds with the address of the current Serving-CSCF for the terminating subscriber.

6. I-CSCF#1 forwards the INVITE request to the entry point whose address was provided during registration. This entry point is either the S-CSCF that is serving the visiting UE (choice (a)), or an I-CSCF within the visited network that is performing the configuration hiding function for the visited network operator (choice (b)).

   (6a) If the visited network operator does not desire to keep their network configuration hidden, the name/address of S-CSCF#2 was provided during registration, and the INVITE request is forwarded directly to S-CSCF#2.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf#1.network#1.com
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

   (6b) If the visited network operator desires to keep their network configuration hidden, the name/address of an I-CSCF in the visited network was provided during registration, and the INVITE request is forwarded through I-CSCF#2 (Firewall) to S-CSCF#2.

   (6b1) I-CSCF#1 forwards the INVITE request to I-CSCF#2 (Firewall)

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf#1.network#1.com
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

(6b2) I-CSCF#2 (Firewall) forwards the INVITE request to S-CSCF#2

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP i-cscf#2-firewall.network#2.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

7. S-CSCF#2 performs whatever service control is appropriate for this call attempt

8. S-CSCF#2 continues processing the call attempt using the terminating procedures.

This message assumes that message 6b2 is received at S-CSCF#2. In the case that message 6a is used the Via header containing the I-cscf #2-firewall.network#2.com is not included in the message and Via headers for all the previous places that this message has traversed since any I-cscf firewall are contained in the message.

Table xx

```
INVITE <Called-Party-Identifier> SIP/2.0
Via: SIP/2.0/UDP s-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf#2-firewall.network#2.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Calling-Party-Identifier>
Content-Type: application/sdp
Content-Length: xxx

 SDP material in this Body section.
```

9. The originating and terminating UE cooperatively establish the bearer path for the media flow.

10. The called UE may optionally perform alerting. If so, it signals this to the calling party.

11. When the called party answers, the termination procedure results in a SIP 200-OK final response being sent to S-CSCF#2

This message assumes that message 6b2 is received at S-CSCF#2. In the case that message 6a is used the Via header containing the I-cscf #2-firewall.network#2.com is not included in the message and Via headers for all the previous places that this message has traversed since any I-cscf firewall are contained in the message.

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf#2.network#2.com
Via: SIP/2.0/UDP i-cscf#2-firewall.network#2.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

12. S-CSCF#2 performs whatever service control is appropriate for this call completion

13. S-CSCF#2 sends the SIP 200-OK final response along the signalling path back to the call originator. Based on the choice made in (6) above, this response may either be sent directly from S-CSCF#2 to I-CSCF#1 (choice (a)), or be sent indirectly through I-CSCF#2 (Firewall) (choice (b)).

13 b1)

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP i-cscf#2-firewall.network#2.com
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

13b2)

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP i-cscf#1.network#1.com
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

14. I-CSCF#1 sends a SIP 200-OK final response along the signalling path back to S-CSCF#1

Table xx

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP s-cscf#1.network#1.com
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

15. S-CSCF#1 performs whatever service control is appropriate for this call completion

16. S-CSCF#1 continues by the procedures of the originating signalling flow.

Table xx

```
SIP/2.0 200 OK
… Via: all the previous places that this message has traversed.
From: <Calling-Party-Identifier>
To: <Called-Party-Identifier>
Call-ID: 12345600@Calling-Party.host
CSeq: 1 INVITE
Contact: <Called-Party-Identifier>
Content-Length: 0
```

Editor’s note: That the figure from 23.228 does not show ACKs and these messages need to be added into this section.

Annex B (informative):
Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
19/10/00		N1-001108			Version 0.0.1 Presented to CN1 SIP ad-hoc #1		
19/10/00		N1-001114			Version 0.0.2 Revisions identified by CN1 SIP ad-hoc #1		
08/11/00		N1-001189			Version 0.0.3 Changes to new 3GPP template and styles		
22/11/00		N1-00			Version 0.1.0 Inclusion of baseline material from CN #14 REGISTER and INVITE flows		