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3rd Generation Partnership Project;
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Signalling flows for the IP multimedia call control based on SIP and SDP

(Release 5)

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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:

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x the first digit:

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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 3GPP TS 23.228 [2].

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.

[1]   TR 21.905: "Vocabulary for 3GPP Specifications".

[2]   TS 23.228: "IP Multimedia (IM) Subsystem – Stage 2".

[3]   IETF 2543bis2: "SIP: Session Initiation Protocol" (ietf-sip-rfc2543bis-02.txt)

Editors Note: This reference to 2543 will need to be updated to reference the assigned RFC number, once the draft achieves RFC status within the IETF.

3 Definitions, symbols and abbreviations

3.1 Definitions

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

IM CN subsystem: (IP Multimedia CN subsystem) comprises of all CN elements for the provision of IP multimedia applications over IP multimedia sessions

IP multimedia session: an IP multimedia session is a set of multimedia senders and receivers and the data streams flowing from senders to receivers. IP multimedia sessions are supported by the IP multimedia CN Subsystem and are enabled by IP connectivity bearers (e.g. GPRS as a bearer). A user may invoke concurrent IP multimedia sessions.

Stateful Proxy: A logical entity that maintains state information at least for the duration of a SIP transaction. [3]

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:

BGCF	Breakout Gateway Control Function
CN	Core Network
CSCF	Call Session Control Function
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name System
GPRS	General Packet Radio Service
HSS	Home Subscriber Server
I-CSCF	Interrogating-CSCF
IM	IP Multimedia
IP	Internet Protocol
IPv6	IP version 6
MGCF	Media Gateway Control Function
P-CSCF	Proxy-CSCF
PDP	Packet Data Protocol
PSTN	Public Switched Telephone Network
QoS	Quality of Service
S-CSCF	Serving-CSCF
SDP	Session Description Protocol
SIP	Session Initiation Protocol
UE	User Equipment
URI	Uniform Resource Identifier
USIM	User Service Identity Module

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: 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.

4.1 Key required to interpret signalling flows

This sub-clause lists the keys required to interpret all Signalling Flows in this Specification.

4.1.1 Key required to interpret signalling flow contents tables

Contents Tables are used to describe the contents of SIP Methods. The following key (rules) have been applied to each contents table to improve readability, reduce errors and increase maintainability:-

a) Where a header field in the SIP method contents tables show only the header name, the contents are identical to the received request/response.

b) The **(…)** sequence of characters is used to indicate that the Content-length field needs to be filled in, with the appropriate value i.e. the number of bytes in the payload.

c) Repeated headers within a method are listed on a single line, with a comma used as delimiter. This convention is not mandatory but used in this specification for improved readability.

d) Headers are listed within a table in the following order:-

   1) hop by hop e.g. Via:

   2) end to end e.g. To:

   3) entity headers e.g. Content-length:

   This convention is not mandatory but used in this specification for improved readability.

5 <not currently used>

Editor's Note: The contents of this clause have been moved towards the end of the specification – currently Clauses 100 and 101. This is as a result of many requests to locate the Error Handling flows after the normal flows. The primary reason for this is to enable easy referencing to earlier material i.e. to minimize the need for forward references.

It has been decided to minimize confusion and not to re-order the complete specification as a result of this clause move. Hence should an additional clause be required in the future "5" is available.

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 subclause 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 session attempts from the UE), and signalling messages sent by the S-CSCF to reach the UE (for session 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 session 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 session.

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.

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: Session Flow/Protocol Related Issues. Passing S-CSCF selection information through SIP Register message from hI-CSCF to vI-CSCF According to the S2 session 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? Session flow reference: Figure 2-2, flow 9, 10, and 13

Editor's note: Session 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? Session flow reference: Figure 2-2, flow 13

Editor's note: Session 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.home1.net or home1.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?

7.1   Registration signalling: user not registered

This shows the registration signalling flow for the scenario when the user is not registered. For the purpose of this registration flow, the subscriber is considered to be roaming. In this signalling flow, the home network has network configuration hiding active.

Figure 7.1-1: UE roaming

1. **GPRS Attach / PDP Context Establishment (UE to GPRS)**

    This signalling flow is shown to indicate the GPRS 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 to GPRS/ DHCP)**

    This signalling flow is the procedure to discover the Proxy CSCF in the visited IM CN subsystem.

    The UE should be able to obtain the IP address of the P-CSCF during the PDP Context Activation procedure. At the PDP Context Activation time, the IP address of the P-CSCF shall be conveyed to the UE in the Activate PDP Context Accept message. The UEs that do not support DHCP shall use this P-CSCF discovery procedure.

    Additionally, the UEs that incorporate the DNS and DHCP client software that supports DHCP extensions (specified in draft-ietf-sip-dhcp-03.txt) may employ the DHCP mechanism to discover the P-CSCF in the visited IM CN subsystem. When allocating an IP address to the UE, the DHCP server may provide the UE with the fully-qualified domain name (FQDN) of the P-CSCF and the address of the DNS server in the visited IM CN

subsystem. Subsequently, the DNS client in the UE will utilise the provided FQDN and perform an address-record lookup (i.e. type A DNS access) to obtain the IP address of the P-CSCF in the visited IM CN subsystem.

NOTE: A UE may be roaming within the home network.

Editor's Note: IANA Considerations - Currently the IANA has not assigned an "DHCP option number" for the *SIP Servers DHCP Option* defined in the draft-ietf-sip-dhcp-03.txt. Therefore, the DHCP alternative can not be currently implemented.

Editor's Note: Second approach needs further study on the interactions with the restrictions on the Signalling PDP Context, TS 23.228 subclause 4.2.6.

3. **SIP REGISTER request (UE to P-CSCF) – see example in Table 7.1-3**

The purpose of this request is to register the user's SIP URI with a S-CSCF in the home network. This request is routed to the P-CSCF because it is the only SIP server known to the UE. In the following SIP request, 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@home1.net) and the host (terminal) address ([5555::aaa:bbb:ccc:ddd]) which was acquired during PDP context activation process.

Table 7.1-3 SIP REGISTER request (UE to P-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home1.net>
To: <sip:user1@home1.net>
Contact: <Sip:[5555::aaa:bbb:ccc:ddd]>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Allow-Events: org.3gpp.nwinitdereg
Content-Length: 0
```

Request-URI: The Request-URI (the URI that follows the method name, "REGISTER", in the first line) indicates the destination domain of this REGISTER request. The rules for routeing a SIP request describe how to use DNS to resolve this domain name ("home1.net") into an address or entry point into the home operator's network (the I-CSCF). This information is stored in the USIM.

Editor's note: One proposal is that the above URI forms an integral part of a global private identifier.

Via: IPv6 PDP address of the SIP session allocated during the PDP Context Activation process.

From: This indicates the SIP identity of the user (stored in the USIM) originating the REGISTER request. In SIP, this can be a third-party.

Editor's note: One proposal is: "This is a natural place for the private identifier or NAI for the subscriber. Forming a SIP URL from the NAI is a simple matter of prepending "sip:". For example, if the subscriber's NAI is 19725835472@operator.com, then the From: header would be sip:19725835472@operator.com." Alternatively it could be the SIP-URL of the party registering.

To: This indicates the public identifier being registered. This is the identity by which other parties know this subscriber. It is obtained from the USIM.

Editor's note: One proposed additional text: "In this case, this is the global SIP URL for the subscriber."

Contact: This indicates the point-of-presence for the subscriber – the IP address of the UE. This is the temporary point of contact for the subscriber that is being registered. Subsequent requests destined for this subscriber will be sent to this address. This information is stored in the P-CSCF.

Editor's note: It is for further study whether this information is stored in the HSS and the S-CSCF for the subscriber in order to support multiple registrations.

Call Id: Call Identifier for this Registration generated as per [3]

Cseq: Cseq for this Registration generated as per [3]

Allow-Events: Allow-Events is used to implicitly subscribe to the event "org.3gpp.nwinitdereg". The P-CSCF will remove this field from the REGISTER request before sending it on to the home network.

Editor's note: At this point, all that is missing from the REGISTER request is the visited operator's domain name. This piece of information is inserted into the Contact: header by the P-CSCF in the visited network upon receiving the REGISTER request from the UE. The P-CSCF takes the Contact: header indicating the IP address of the UE and replaces it with a new Contact: header indicating the address of the P-CSCF (including the visited operator's domain name). The IP address of the UE is stored in the P-CSCF for routing incoming requests to the appropriate UE.

For example, the UE's Contact: address ("sip:[5555::aaa:bbb:ccc:ddd]") is replaced by a Contact: address for the proxy ("sip:sean.olson%40home1.net@pcscf1.visited1.net"). This new Contact: address contains a specially encoded version of the subscriber's identity (To: header), plus the address of the P-CSCF.

If the proposed private identifier is not included in the From header, a mechanism is also needed to transfer this proposed private identifier.

Upon receiving this request the P-CSCF will set it's SIP registration timer for this UE to the Expires time in this request.

4. **DNS-Q**

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.

The P-CSCF sends the REGISTRATION request - after local processing - to the address indicated in the Request-URI. When forwarding the REGISTRATION request the P-CSCF needs to specify the protocol, port number and IP address of the I-CSCF server in the home network to which to send the REGISTRATION request. The P-CSCF tries to find this information by querying the DNS. Since the Request-URI does not specify the transport protocol the, P-CSCF selects the UDP.

Table 7.1-4a DNS Query (P-CSCF to DNS)

```
OPCODE=SQUERY
QNAME=_sip.udp.registrar.home_network.net, QCLASS=IN, QTYPE=SRV
```

The DNS records are retrieved according to RFC2782.

Table 7.1-4b DNS Query Response (DNS to P-CSCF)

```
OPCODE=SQUERY, RESPONSE, AA
QNAME=_sip.udp.registrar.home_network.net, QCLASS=IN, QTYPE=SRV

_sip._udp.registrar.home_network.net    0 IN SRV 1 10 5060 icscf2.home_network.com
                                         0 IN SRV 1  0 5060 icscf7.home_network.com

icscf2.home_network.com                  0 IN AAAA    5555::aba:dab:aaa:daa
icscf7.home_network.com                  0 IN AAAA    5555::a1a:b2b:c3c:d4d
```

In the Answer field of the query-response each I-CSCF is identified by its host domain name. The returned SRV Resource Records (RRs) are merged and ordered, and the selection technique (employing the Priority and Weight parameters returned in the RRs) as specified in RFC2782 is used to select the I-CSCF (i.e., the icscf2.home_network.com). Since the Additional Data field of the query-response also contains the IP address of the selected I-CSCF (i.e., 5555::aba:dab:aaa:daa), a new query to the DNS is not required.

Once the IP address of the I-CSCF is obtained, the P-CSCF forwards the REGISTRATION request to this IP address (i.e., 5555::aba:dab:aaa:daa) using the UDP protocol and port number 5060.

5. **SIP REGISTER request (P-CSCF to I-CSCF) – see example in Table 7.1-5**

Since this P-CSCF is a stateful proxy, 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%40home1.net@pcscf1.visited1.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 request for user1@home1.net, to the home registrar, using a contact address of user1%40home.net@pcscf1.visited1.net.

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

Table 7.1-5 SIP REGISTER request (P-CSCF to I-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:pcscf1.visited1.net>
Proxy-require: path
Require: path
From:
To:
Contact: <sip:user1%40home1.net@pcscf1.visited1.net>
Call-ID:
CSeq:
Expires:
Content-Length:
```

Path: This is the address of the P-CSCF and is included to inform the S-CSCF where to route terminating sessions.

Require, Proxy-Require: These headers are included to ensure that the recipient correctly handles the Path header. If the recipient does not support the path header, a response will be received with a status code of 420 and an Unsupported header indicating "path". Such a response indicates a misconfiguration of the routing tables and the request has been routed outside the IM CN subsystem.

6. **Cx-Query**

The I-CSCF requests information related to the required S-CSCF capabilities from the HSS. The HSS provides the I-CSCF with either the S-CSCF address for the subscriber (if the subscriber is currently registered) or the S-CSCF required capabilities (if the subscriber is not currently registered.) Since the subscriber is not registered in this case, the HSS returns the S-CSCF required capabilities and the I-CSCF uses this information to select a suitable S-CSCF.

7. **SIP REGISTER request (I-CSCF to S-CSCF) – see example in Table 7.1-7**

I-CSCF adds a proper I-CSCF name to the Path header.

This signalling flow forwards the SIP REGISTER from the I-CSCF to the S-CSCF selected. The Request-URI is changed to the address of the S-CSCF.

Table 7.1-7 SIP REGISTER request (I-CSCF to S-CSCF)

```
REGISTER sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
     [5555::aaa:bbb:ccc:ddd]
Path: <sip:icscf2.home1.net>, <sip:pcscf1.visited1.net>
Proxy-require:
Require:
From:
To:
Contact:
Call-ID:
CSeq:
Expires:
Content-Length:
```

Path: The S-CSCF stores the contents of the Path headers and uses these addresses for routing mobile terminated sessions.

Upon receiving this request the S-CSCF will set it's SIP registration timer for this UE to the Expires time in this request.

8. **Cx-Location**

The S-CSCF shall send its location information to the HSS. The HSS stores the S-CSCF name for that subscriber. The HSS sends a response to the S-CSCF to acknowledge the sending of location information.

9. **Cx-Profile**

The S-CSCF shall send the subscriber's identity to the HSS in order to be able to download the subscriber profile to the S-CSCF. The HSS returns the subscriber's profile to the S-CSCF. The S-CSCF shall store the subscriber profile for that indicated user.

10. **SIP 200 OK response (S-CSCF to I-CSCF) – see example in Table 7.1-10**

The S-CSCF sends acknowledgment to the I-CSCF indicating that Registration was successful. This response will traverse the path that the REGISTER request took as described in the Via list.

Table 7.1-10 SIP 200 OK response (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]
Path: <sip:scscf1.home1.net>, <sip:icscf2.home1.net>, <sip:pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

Path: The S-CSCF inserts its own name to the front of the list.

11. **SIP 200 OK response (I-CSCF to P-CSCF) – see example in Table 7.1-11**

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

Table 7.1-11 SIP 200 OK response (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:token(scscf1.home1.net)>, <sip:icscf2.home1.net>, <sip:pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

12. **SIP 200 OK response (P-CSCF to UE) – see example in Table 7.1-12**

The P-CSCF removes its address from the Path header, reverses the order of the fields, saves the resulting Path header and associates it with the UE. The P-CSCF then removes the Path header from the 200 OK response. The P-CSCF then forwards acknowledgment from the I-CSCF to the UE indicating that Registration was successful.

Table 7.1-12 SIP 200 OK response (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

13. **SIP NOTIFY (P-CSCF to UE) – see example in Table 7.1-13**

    The P-CSCF sends a first NOTIFY request to the UE in order to confirm the successful (implicit) subscription to the event "org.3gpp.nwinitdereg". A new Call-ID (here: ab01cd23de45@pcscf1.visited1.net) is generated.

    Editors Note: The choice of Call-ID as done here is rather unstable; to be clarified in IETF first

Table 7.1-13 SIP NOTIFY (P-CSCF to UE)

```
NOTIFY sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net
From: <sip:pcscf1.visited1.net>
To: <sip:sip:user1@home1.net>
Call-ID: ab01cd23de45@pcscf1.visited1.net
CSeq: 1 NOTIFY
Expires: 7200
Event: org.3gpp.nwinitdereg
Content-Length: 0
```

14. **SIP 200 OK (UE to P-CSCF) – see example in Table 7.1-14**

    The UE stores the Call-ID, CSeq, etc. and generates a 200 OK response to the NOFIFY

Table 7.1-14 SIP 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net
From:
To:
Call-ID:
CSeq:
Content-Length:
```

7.2 Registration signalling: re-registration – user currently registered

For the purpose of this re-registration flow, the subscriber is considered to be roaming. In this signalling flow, the home network has network configuration hiding active.
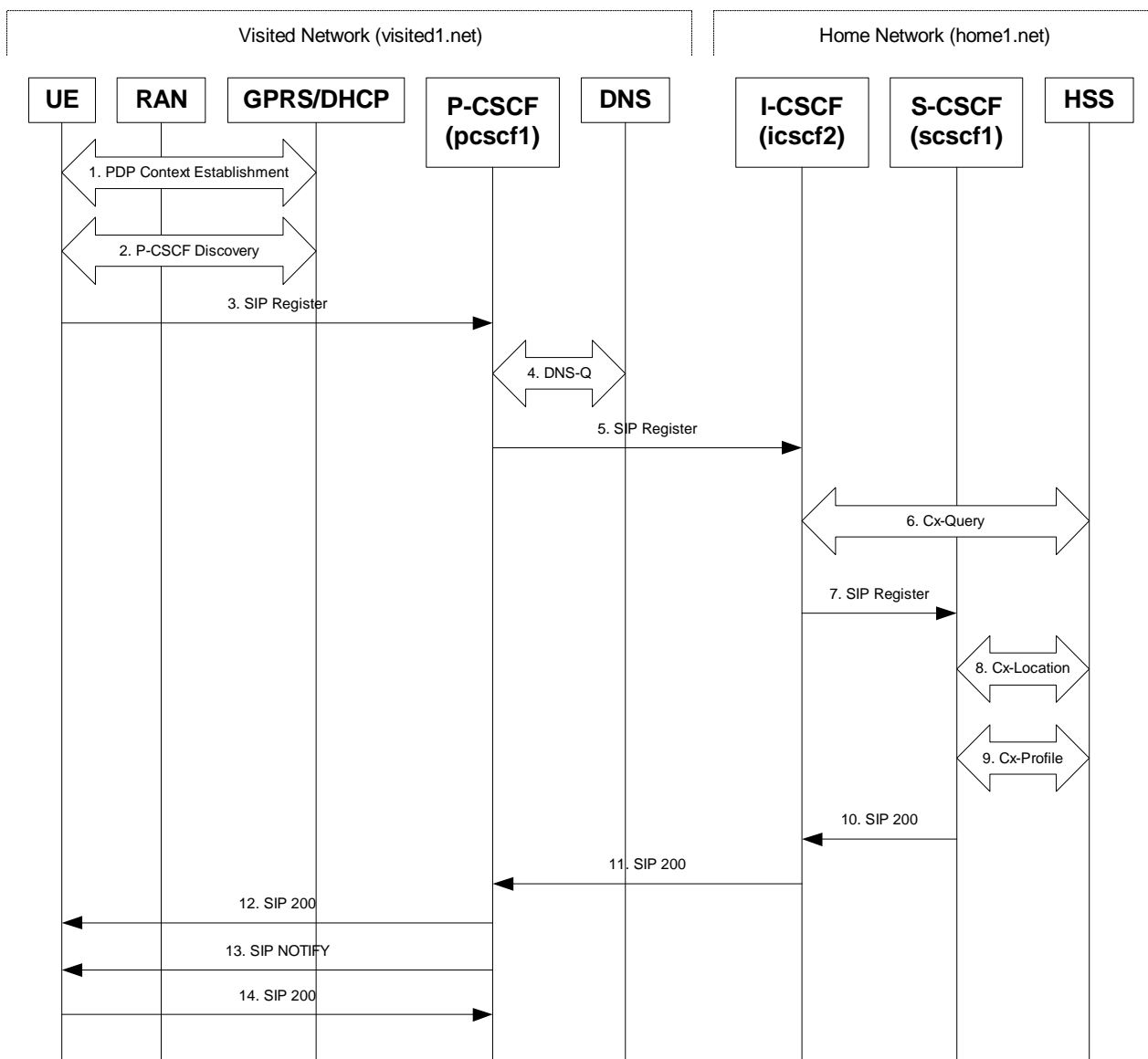
This flow assumes :

1. That the same PDP Context allocated during the initial registration scenario is still used for re-registration. For the case when the UE does not still have an active PDP context then PDP context procedures from subclause 7.1 must first be completed.

2. The DHCP procedure employed for P-CSCF discovery is not needed.

3. The S-CSCF selection procedure invoked by the I-CSCF is not needed.

Periodic application level re-registration is initiated by the UE either in response to the expiration of the existing registration information or in response to a change in the registration status of the UE. Re-registration follows the same path as described in subclause 7.1.

Figure 7.2-1: UE roaming

1. **SIP REGISTER request (UE to P-CSCF) – see example in Table 7.2-1**

   The registration expires in the UE. The UE re-registers by sending a new REGISTER request. This request is sent to the same P-CSCF with which the UE initially registered. The P-CSCF maintains the same binding between the User's SIP public address (user_public1@home1.net) and the host (terminal) address ([5555::aaa:bbb:ccc:ddd]) which it established during the original registration.

Table 7.2-1 SIP REGISTER (UE to P-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user_private1@home1.net>
To: <sip:user_public1@home1.net>
Contact: <Sip:[5555::aaa:bbb:ccc:ddd]>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 7 REGISTER
Expires: 7200
Allow-Events: org.3gpp.nwinitdereg
Content-Length: 0
```

   The header field usage is the same as for the initial registration scenario:

From: This indicates the private identity of the user (stored in the USIM) originating the REGISTER request.

Editor's note: The requirement for and impact on third party registration with this usage of the From header needs further study.

To: This indicates the target of the REGISTER request. The target is the public identity that is being registered. This is the identity by which other parties know this subscriber.

Contact: This indicates the point-of-presence for the subscriber – the IP address of the UE. This is the temporary identifier for the subscriber that is being registered. Subsequent requests destined for this subscriber will be sent to this address. This information is stored in the P-CSCF.

Editor's note: It is for further study whether this information is stored in the HSS and the S-CSCF for the subscriber in order to support multiple registrations.

Request-URI: The Request-URI (the URI that follows the method name, "REGISTER", in the first line) indicates the destination domain of this REGISTER request. The rules for routeing a SIP request describe how to use DNS to resolve this domain name ("home1.net") into an address or entry point into the home operator's network (the I-CSCF). This information is stored in the USIM.

Editor's note: At this point, all that is missing from the REGISTER request is the visited operator's domain name. This piece of information is inserted into the Contact: header by the P-CSCF in the visited network upon receiving the REGISTER request from the UE. The P-CSCF takes the Contact: header indicating the IP address of the UE and replaces it with a new Contact: header indicating the address of the P-CSCF (including the visited operator's domain name). The IP address of the UE is stored in the P-CSCF for routing incoming requests to the appropriate UE.

For example, the UE's Contact: address ("sip:[5555::aaa:bbb:ccc:ddd]") is replaced by a Contact: address for the proxy ("sip:sean.olson%40home1.net@pcscf1.visited1.net"). This new Contact: address contains a specially encoded version of the subscriber's identity (To: header), plus the address of the P-CSCF.

Upon receiving this request the P-CSCF will detect that it already has a registration record for this UE and will reset it's SIP registration timer for this UE to the Expires time in this request.

2. **DNS-Q**

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. The DNS provides the P-CSCF with an address of the I-CSCF in the home network. The P-CSCF must not use the I-CSCF address cached as a result of the previous registration.

3. **SIP REGISTER request (P-CSCF to I-CSCF) – see example in Table 7.2-3**

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

Table 7.2-3 SIP REGISTER request (P-CSCF to I-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:pcscf1.visited1.net>
Proxy-require: path
Require: path
From:
To:
Contact: <sip:user1%40home1.net@pcscf1.visited1.net>
Call-ID: CSeq:
Expires:
Content-Length:
```

Path: This is the address of the P-CSCF and is included to inform the S-CSCF where to route terminating sessions.

Require, Proxy-Require: These headers are included to ensure that the recipient correctly handles the Path header. If the recipient does not support the path header, a response will be received with a status code of 420 and an Unsupported header indicating "path". Such a response indicates a misconfiguration of the routing tables and the request has been routed outside the IM CN subsystem.

4. **Cx-Query**

The I-CSCF requests information related to the required S-CSCF capabilities from the HSS. The HSS shall determine that the user is currently registered, and send an indication of current S-CSCF to the I-CSCF. Hence, the S-CSCF selection procedure is not needed.

5. **SIP REGISTER request (I-CSCF to S-CSCF) – see example in Table 7.2-5**

This signalling flow forwards the SIP REGISTER request from the I-CSCF to the S-CSCF selected. The Request-URI is changed to the address of the S-CSCF.

I-CSCF adds a proper I-CSCF name to the Path header.

Table 7.2-5 SIP REGISTER request (I-CSCF to S-CSCF)

```
REGISTER sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Path: <sip:icscf2.home1.net>, <sip: pcscf1.visited1.net
Proxy-require:
Require:
From:
To:
Contact:
Call-ID:
CSeq:
Expires:
Content-Length:
```

Path: The S-CSCF stores the contents of the Path headers and uses these addresses for routing mobile terminated sessions.

Upon receiving this request the S-CSCF will detect that it already has a registration record for this UE and will reset it's SIP registration timer for this UE to the Expires time in this request.

6. **Cx-Location**

The S-CSCF shall send its location information to the HSS. The HSS stores the S-CSCF name for that subscriber. The HSS sends a response to the S-CSCF to acknowledge the sending of location information.

Editor's Note: Since the S-CSCF has detected this as a re-registration it may omit this step as an optimisation.

7. **Cx-Profile**

The S-CSCF shall send the subscriber's identity to the HSS in order to be able to download the subscriber profile to the S-CSCF. The HSS returns the subscriber's profile to the S-CSCF. The S-CSCF shall store the subscriber profile for that indicated user.

Editor's Note: Since the S-CSCF has detected this as a re-registration it may omit this step as an optimisation.

8. **SIP 200 OK response (S-CSCF to I-CSCF) – see example in Table 7.2-8**

The S-CSCF sends acknowledgment to the I-CSCF indicating that Registration was successful. This response will traverse the path that the REGISTER request took as described in the Via list.

Table 7.2-8 SIP 200 OK response (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Path: <sip: scscf2.home1.net>, <sip: icscf2.home1.net>, <sip: pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

Path: The S-CSCF inserts its own name to the front of the list.

9. **SIP 200 OK response (I-CSCF to P-CSCF) – see example in Table 7.2-9**

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

Table 7.2-9 SIP 200 OK response (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:token(scscf2.home1.net)>, <sip: icscf2.home1.net>, <sip: pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

10. **SIP 200 OK response (P-CSCF to UE) – see example in Table 7.2-10**

The P-CSCF removes its address from the Path header, reverses the order of the fields, saves the resulting Path header and associates it with the UE. The P-CSCF then removes the Path header from the 200 OK response. The P-CSCF then forwards acknowledgment from the I-CSCF to the UE indicating that Registration was successful.

Table 7.2-10 SIP 200 OK response (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Expires: Content-Length:
```

11. **SIP NOTIFY (P-CSCF to UE) – see example in Table 7.2-11**

The P-CSCF sends a first NOTIFY request to the UE in order to confirm the successful (implicit) subscription to the event "org.3gpp.nwinitdereg". A new Call-ID (here: uv45wx67yz89@pcscf1.visited1.net) is generated.
Note: This Call-Id has also to be different from the last implicit subscription

Editors Note: The choice of Call-ID as done here is rather unstable; to be clarified in IETF first

Table 7.2-11 SIP NOTIFY (P-CSCF to UE)

```
NOTIFY sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net
From: <sip:pcscf1.visited1.net>
To: <sip:sip:user1@home1.net>
Call-ID: uv45wx67yz89@pcscf1.visited1.net
CSeq: 1 NOTIFY
Expires: 7200
Event: org.3gpp.nwinitdereg
Content-Length: 0
```

12. **SIP 200 OK (UE to P-CSCF) – see example in Table 7.2-12**

The UE stores the Call-ID, CSeq, etc. and generates a 200 OK response to the NOFIFY

Table 7.2-12 SIP 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net
From:
To:
Call-ID:
CSeq:
Content-Length:
```

7.3   Registration signalling: user not registered

This shows the registration signalling flow for the scenario when the user is not registered. For the purpose of this registration flow, the subscriber is considered to be roaming. In this signalling flow, the home network does not have network configuration hiding active.

The example below assumes that the UE does not subscribe to the org.3gpp.nwinitdereg or any other event during Registration.

Figure 7.1-1: UE roaming

1. **GPRS Attach / PDP Context Establishment (UE to GPRS)**

   This signalling flow is shown to indicate the GPRS 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 to GPRS/ DHCP)**

   This signalling flow is the procedure to discover the Proxy CSCF in the visited IM CN subsystem.

   The UE should be able to obtain the IP address of the P-CSCF during the PDP Context Activation procedure. At the PDP Context Activation time, the IP address of the P-CSCF shall be conveyed to the UE in the Activate PDP Context Accept message. The UEs that do not support DHCP shall use this P-CSCF discovery procedure.

   Additionally, the UEs that incorporate the DNS and DHCP client software that supports DHCP extensions (specified in draft-ietf-sip-dhcp-03.txt) may employ the DHCP mechanism to discover the P-CSCF in the visited IM CN subsystem. When allocating an IP address to the UE, the DHCP server may provide the UE with the

fully-qualified domain name (FQDN) of the P-CSCF and the address of the DNS server in the visited IM CN subsystem. Subsequently, the DNS client in the UE will utilise the provided FQDN and perform an address-record lookup (i.e. type A DNS access) to obtain the IP address of the P-CSCF in the visited IM CN subsystem.

NOTE: A UE may be roaming within the home network.

Editor's Note: IANA Considerations - Currently the IANA has not assigned an "DHCP option number" for the *SIP Servers DHCP Option* defined in the draft-ietf-sip-dhcp-03.txt. Therefore, the DHCP alternative can not be currently implemented.

Editor's Note: Second approach needs further study on the interactions with the restrictions on the Signalling PDP Context, TS 23.228 subclause 4.2.6.

3. **SIP REGISTER request (UE to P-CSCF) – see example in Table 7.1-3**

The purpose of this request is to register the user's SIP URI with a S-CSCF in the home network. This request is routed to the P-CSCF because it is the only SIP server known to the UE. In the following SIP request, 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@home1.net) and the host (terminal) address ([5555::aaa:bbb:ccc:ddd]) which was acquired during PDP context activation process.

Table 7.1-3 SIP REGISTER request (UE to P-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home1.net>
To: <sip:user1@home1.net>
Contact: <Sip:[5555::aaa:bbb:ccc:ddd]>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Content-Length: 0
```

Request-URI: The Request-URI (the URI that follows the method name, "REGISTER", in the first line) indicates the destination domain of this REGISTER request. The rules for routeing a SIP request describe how to use DNS to resolve this domain name ("home1.net") into an address or entry point into the home operator's network (the I-CSCF). This information is stored in the USIM.

Editor's note: One proposal is that the above URI forms an integral part of a global private identifier.

Via: IPv6 PDP address of the SIP session allocated during the PDP Context Activation process.

From: This indicates the SIP identity of the user (stored in the USIM) originating the REGISTER request. In SIP, this can be a third-party.

Editor's note: One proposal is: "This is a natural place for the private identifier or NAI for the subscriber. Forming a SIP URL from the NAI is a simple matter of prepending "sip:". For example, if the subscriber's NAI is 19725835472@operator.com, then the From: header would be sip:19725835472@operator.com." Alternatively it could be the SIP-URL of the party registering.

To: This indicates the public identifier being registered. This is the identity by which other parties know this subscriber. It is obtained from the USIM.

Editor's note: One proposed additional text: "In this case, this is the global SIP URL for the subscriber."

Contact: This indicates the point-of-presence for the subscriber – the IP address of the UE. This is the temporary point of contact for the subscriber that is being registered. Subsequent requests destined for this subscriber will be sent to this address. This information is stored in the P-CSCF.

Editor's note: It is for further study whether this information is stored in the HSS and the S-CSCF for the subscriber in order to support multiple registrations.

Call Id: Call Identifier for this Registration generated as per [3]

> **Cseq:** Cseq for this Registration generated as per [3]

Editor's note: At this point, all that is missing from the REGISTER request is the visited operator's domain name. This piece of information is inserted into the Contact: header by the P-CSCF in the visited network upon receiving the REGISTER request from the UE. The P-CSCF takes the Contact: header indicating the IP address of the UE and replaces it with a new Contact: header indicating the address of the P-CSCF (including the visited operator's domain name). The IP address of the UE is stored in the P-CSCF for routing incoming requests to the appropriate UE.

For example, the UE's Contact: address ("sip:[5555::aaa:bbb:ccc:ddd]") is replaced by a Contact: address for the proxy ("sip:sean.olson%40home1.net@pcscf1.visited1.net"). This new Contact: address contains a specially encoded version of the subscriber's identity (To: header), plus the address of the P-CSCF.

If the proposed private identifier is not included in the From header, a mechanism is also needed to transfer this proposed private identifier.

Upon receiving this request the P-CSCF will set it's SIP registration timer for this UE to the Expires time in this request.

4. **DNS-Q**

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.

The P-CSCF sends the REGISTRATION request - after local processing - to the address indicated in the Request-URI. When forwarding the REGISTRATION request the P-CSCF needs to specify the protocol, port number and IP address of the I-CSCF server in the home network to which to send the REGISTRATION request. The P-CSCF tries to find this information by querying the DNS. Since the Request-URI does not specify the transport protocol the, P-CSCF selects the UDP.

Table 7.1-4a DNS Query (P-CSCF to DNS)

```
OPCODE=SQUERY
QNAME=_sip.udp.registrar.home_network.net, QCLASS=IN, QTYPE=SRV
```

The DNS records are retrieved according to RFC2782.

Table 7.1-4b DNS Query Response (DNS to P-CSCF)

```
OPCODE=SQUERY, RESPONSE, AA
QNAME=_sip.udp.registrar.home_network.net, QCLASS=IN, QTYPE=SRV

_sip._udp.registrar.home_network.net   0 IN SRV 1 10 5060 icscf2.home_network.com
                                       0 IN SRV 1  0 5060 icscf7.home_network.com

icscf2.home_network.com                0 IN AAAA    5555::aba:dab:aaa:daa
icscf7.home_network.com                0 IN AAAA    5555::a1a:b2b:c3c:d4d
```

In the Answer field of the query-response each I-CSCF is identified by its host domain name. The returned SRV Resource Records (RRs) are merged and ordered, and the selection technique (employing the Priority and Weight parameters returned in the RRs) as specified in RFC2782 is used to select the I-CSCF (i.e., the icscf2.home_network.com). Since the Additional Data field of the query-response also contains the IP address of the selected I-CSCF (i.e., 5555::aba:dab:aaa:daa), a new query to the DNS is not required.

Once the IP address of the I-CSCF is obtained, the P-CSCF forwards the REGISTRATION request to this IP address (i.e., 5555::aba:dab:aaa:daa) using the UDP protocol and port number 5060.

5. **SIP REGISTER request (P-CSCF to I-CSCF) – see example in Table 7.1-5**

Since this P-CSCF is a stateful proxy, 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%40home1.net@pcscf1.visited1.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 request for user1@home1.net, to the home registrar, using a contact address of user1%40home.net@pcscf1.visited1.net.

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

Table 7.1-5 SIP REGISTER request (P-CSCF to I-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:pcscf1.visited1.net>
Proxy-require: path
Require: path
From:
To:
Contact: <sip:user1%40home1.net@pcscf1.visited1.net>
Call-ID:
CSeq:
Expires:
Content-Length:
```

Path: This is the address of the P-CSCF and is included to inform the S-CSCF where to route terminating sessions.

Require, Proxy-Require: These headers are included to ensure that the recipient correctly handles the Path header. If the recipient does not support the path header, a response will be received with a status code of 420 and an Unsupported header indicating "path". Such a response indicates a misconfiguration of the routing tables and the request has been routed outside the IM CN subsystem.

6. **Cx-Query**

   The I-CSCF requests information related to the required S-CSCF capabilities from the HSS. The HSS provides the I-CSCF with either the S-CSCF address for the subscriber (if the subscriber is currently registered) or the S-CSCF required capabilities (if the subscriber is not currently registered.) Since the subscriber is not registered in this case, the HSS returns the S-CSCF required capabilities and the I-CSCF uses this information to select a suitable S-CSCF.

7. **SIP REGISTER request (I-CSCF to S-CSCF) – see example in Table 7.1-7**

   I-CSCF does not modifythe Path header.

   This signalling flow forwards the SIP REGISTER from the I-CSCF to the S-CSCF selected. The Request-URI is changed to the address of the S-CSCF.

Table 7.1-7 SIP REGISTER request (I-CSCF to S-CSCF)

```
REGISTER sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Path: <sip:pcscf1.visited1.net>
Proxy-require:
Require:
From:
To:
Contact:
Call-ID:
CSeq:
Expires:
Content-Length:
```

Path: The S-CSCF stores the contents of the Path headers and uses these addresses for routing mobile terminated sessions.

Upon receiving this request the S-CSCF will set it's SIP registration timer for this UE to the Expires time in this request.

8. **Cx-Location**

   The S-CSCF shall send its location information to the HSS. The HSS stores the S-CSCF name for that subscriber. The HSS sends a response to the S-CSCF to acknowledge the sending of location information.

9. **Cx-Profile**

   The S-CSCF shall send the subscriber's identity to the HSS in order to be able to download the subscriber profile to the S-CSCF. The HSS returns the subscriber's profile to the S-CSCF. The S-CSCF shall store the subscriber profile for that indicated user.

10. **SIP 200 OK response (S-CSCF to I-CSCF) – see example in Table 7.1-10**

    The S-CSCF sends acknowledgment to the I-CSCF indicating that Registration was successful. This response will traverse the path that the REGISTER request took as described in the Via list.

Table 7.1-10 SIP 200 OK response (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]
Path: <sip:scscf1.home1.net>, <sip:pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

   Path: The S-CSCF inserts its own name to the front of the list.

11. **SIP 200 OK response (I-CSCF to P-CSCF) – see example in Table 7.1-11**

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

Table 7.1-11 SIP 200 OK response (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:scscf1.home1.net, <sip:pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

12. **SIP 200 OK response (P-CSCF to UE) – see example in Table 7.1-12**

    The P-CSCF removes its address from the Path header, reverses the order of the fields, saves the resulting Path header and associates it with the UE. The P-CSCF then removes the Path header from the 200 OK response. The P-CSCF then forwards acknowledgment from the I-CSCF to the UE indicating that Registration was successful.

Table 7.1-12 SIP 200 OK response (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

7.4 Registration signalling: mobile initiated deregistration

For the purposes of this deregistration flow, the subscriber is considered to be roaming. In this flow, the home network has configuration hiding active.

This flow assumes:

1. That the same PDP Context allocated during the initial registration scenario is still used for deregistration. For the case when the UE does not still have an active PDP context then PDP context procedures from subclause 7.1 must first be completed.

Editor's Note: If the same PDP-Context is not available, is it guaranteed that the UE will get back the same IP address at this point? If this is not possible, would there be a problem with the binding in the P-CSCF (user_public1@home1.net and [5555::aaa:bbb:ccc:ddd])?

2. The procedure employed for P-CSCF discovery is not needed.

3. The S-CSCF selection procedure invoked by the I-CSCF is not needed.

Editor's note: The roaming case needs to be studied in a separate flow. The specific case to be studied is where the UE roams into a new roaming area. The old PDP-context is probably lost, before this Mobile Initiated Deregistration procedure can be started.
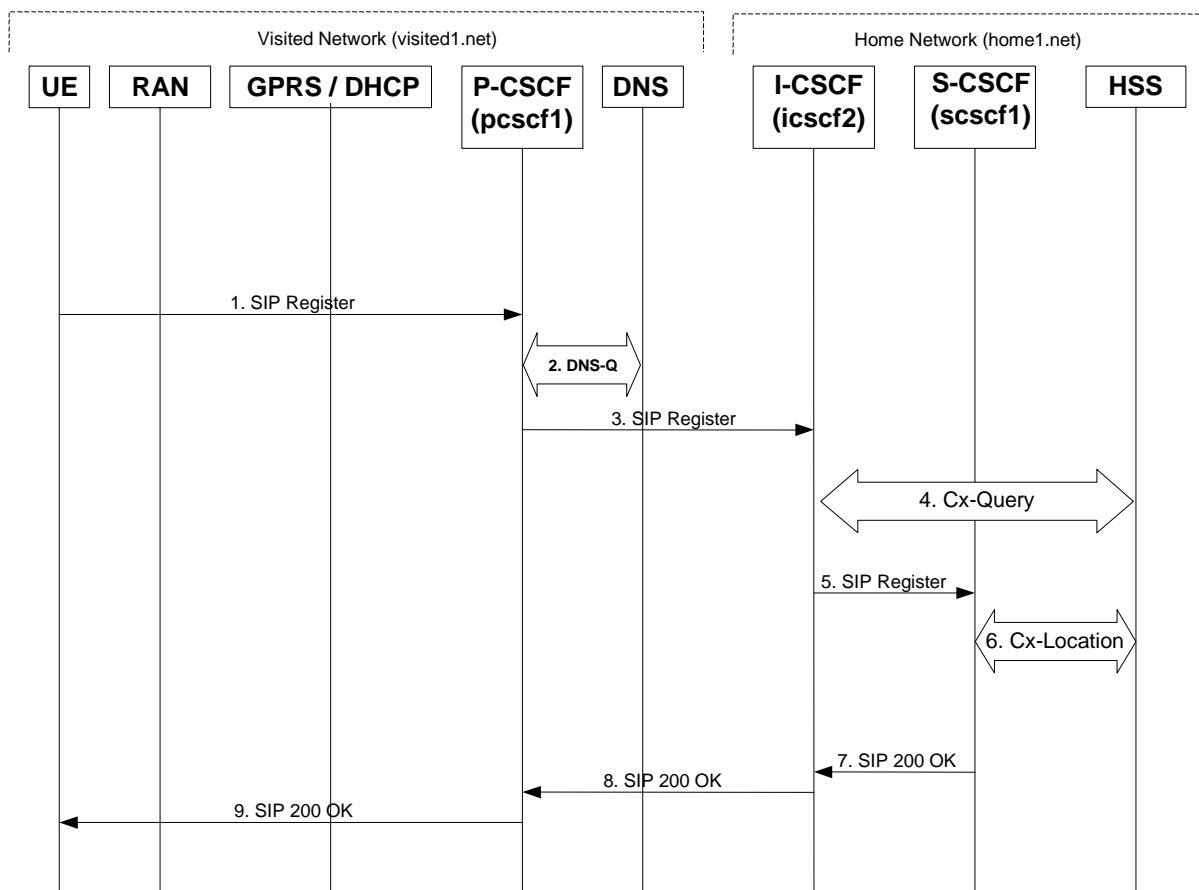
Steps 1 through 4 are identical as for "Re-Registration – User Currently Registered" – see subclause 7.2.

5. **SIP REGISTER (I-CSCF to S-CSCF) – see example in Table 7.3-5**

   I-CSCF adds a proper I-CSCF name to the Path header.

   This signalling flow forwards the SIP REGISTER request from the I-CSCF to the S-CSCF selected. The Request-URI is changed to the address of the S-CSCF.

Table 7.3-5 SIP REGISTER request (I-CSCF to S-CSCF)

```
REGISTER sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Path: <sip:icscf2.home1.net> <sip:pcscf1.visited1.net>
Proxy-require: path
Require: path
From: <sip:user_private1@home1.net>
To: <sip:user_public1@home1.net>
Contact: <sip:user1%40home1.net@pcscf.visited1.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 3 REGISTER
Expires: 0
Content-Length: 0
```

Expires: The 0 value indicates the registration is being cancelled.

Upon receiving this request the S-CSCF will reset the SIP registration timer for this UE to 0.

6. **Cx-Location**

   The S-CSCF shall notify the HSS to clear its location information for that subscriber. The HSS deletes the S-CSCF name for that subscriber. The HSS sends a response to the S-CSCF to acknowledge the clearing of location information.

7. **SIP 200 OK (S-CSCF to I-CSCF) – see example in Table 7.3-7**

The S-CSCF sends acknowledgment to the I-CSCF indicating that deregistration was successful. This request will traverse the path that the REGISTER request took as described in the Via list. The S-CSCF clears its information for that subscriber.

Table 7.3-7 SIP 200 OK response (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Path: <sip:scscf1.home1.net>, <sip:icscf2.home1.net>, <sip:pcscf1.visited1.net>
From: <sip:user_private1@home1.net>
To: <sip:user_public1@home1.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 3 REGISTER
Expires: 0
Content-Length: 0
```

Path: The S-CSCF inserts its own name to the front of the list.

8. **SIP 200 OK (I-CSCF to P-CSCF) – see example in Table 7.3-8**

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

Table 7.3-8 SIP 200 OK response (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:token(scscf1.home1.net)>, <sip:icscf2.home1.net>, <sip:pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

9. **SIP 200 OK (P-CSCF to UE) – see example in Table 7.3-9**

The P-CSCF forwards the acknowledgment from the I-CSCF to the UE indicating that deregistration was successful. The P-CSCF clears its information for that subscriber after sending the acknowledgment to the UE.

Table 7.3-9 SIP 200 OK response (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

8 Signalling flows for session initiation

8.1 Introduction

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 subclause 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 session 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 session 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 session 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 session 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 session, could be stored in the P-CSCF or S-CSCF. When needed for the subsequent session 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 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: The use of the term "Firewall" is FFS and may be changed at a future time. This is dependant on a corresponding change in 23.228.

Editor's Note: Need to show ENUM interactions at the S-CSCF.

This clause breaks down the session flows into a number of individual procedures, following the same principles as 3GPP TS 23.228 [2] subclause 5.4.9.

For the purposes of this document, a further breakdown has been necessary, and therefore a number of flows have been given an (a) or (b) suffix, so that the session flows for configuration independence may be distinguished from those without, e.g.

- (MO#1a) Mobile origination, roaming, without I-CSCF providing configuration independence

- (MO#1b) Mobile origination, roaming, with I-CSCF in home network providing configuration independence

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 session 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

3. 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 session attempt, it can be inserted by the CSCF.

4. 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 session 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

3. 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 session attempt, it can be inserted by the CSCF.

4. 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.

8.2 Origination procedures

8.2.1 Introduction

This clause presents the detailed application level flows to define the Procedures for session originations.

The session origination procedures specify the signalling path between the UE initiating a session attempt and the Serving-CSCF that is assigned to perform the session origination service. This signalling path is determined at the time of UE registration, and remains fixed for the life of the registration.

A UE always has a proxy (P-CSCF) associated with it. This P-CSCF is located in the same network as the UE, performs resource authorization, and may have additional functions in handling of emergency sessions. The P-CSCF is determined by the CSCF discovery process.

As a result of the registration procedure, the P-CSCF determines the next hop toward the Serving-CSCF. This next hop may be directly to the S-CSCF (MO#1a for the roaming case, MO#2 for the home case), or to an I-CSCF who forwards the request to the S-CSCF (MO#1b). These next-hop addresses could be IPv6 addresses, or could be names that are translated via DNS to an IPv6 address.

Sessions originated in the PSTN to a mobile destination are a special case of the Origination procedures and three possibilities to route such sessions are detailed. In the first one, all sessions originated in the PSTN are routed towards the IM CN subsystem. The MGCF uses H.248/MEGACO to control a Media Gateway, and communicates with the SS7 network via the TSGW. The MGCF initiates the SIP request, and subsequent nodes consider the signalling as if it came from a S-CSCF. In the second one, all sessions originated in the PSTN are routed towards the CS domain. The entry point of the network is then a G-MSC. In the third one, the operator can choose to handle simultaneously the first two routing possibilities and a way to handle this flexibility is detailed.

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: 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 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?

8.2.2 (MO#1a) Mobile origination, roaming, without I-CSCF in home network providing configuration independence

This origination procedure applies to roaming subscribers when the home network operator does not desire to keep its internal configuration hidden from the visited network. The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure. During registration, the home network allocates a S-CSCF. The home network provides the S-CSCF name/address as the entry point from the visited network.

When registration is complete, P-CSCF knows the name/address of the S-CSCF.
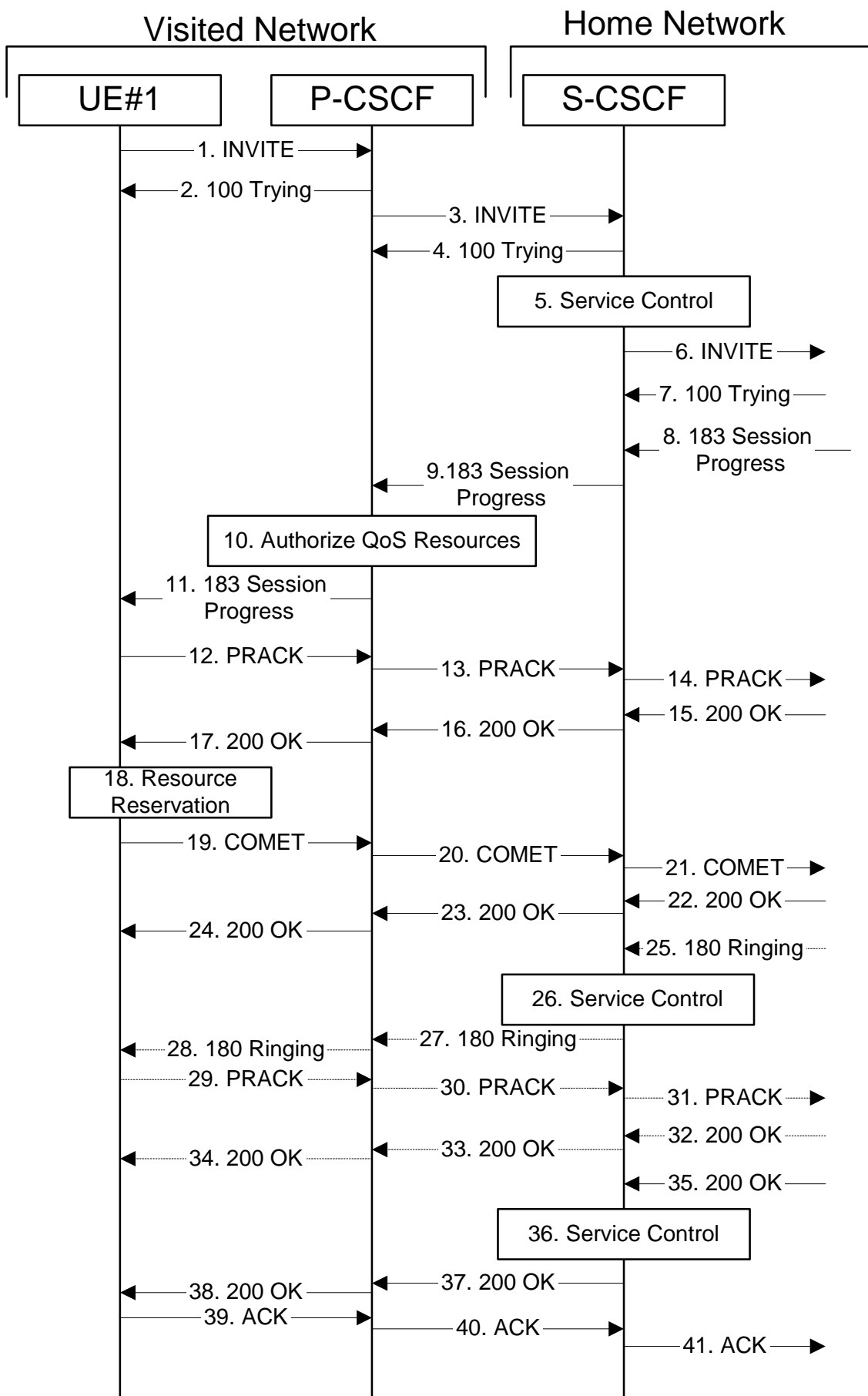
Figure 8.2.2 – MO#1a

Procedure MO#1a is as follows:

1. **INVITE (UE to P-CSCF) – see example in Table 8.2.2-1**

   UE#1 determines the complete set of codecs that it is capable of supporting for this session. It builds a SDP containing bandwidth requirements and characteristics of each, and assigns local port numbers for each possible media flow. Multiple media flows may be offered, and for each media flow (m= line in SDP), there may be multiple codec choices offered.

   For this example, assume UE#1 is capable of sending two simultaneous video streams, either H261 or MPV format, and two simultaneous audio streams, either AMR, G726-32, PCMU, or G728.

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

   Editor's Note: Certain fields in the SDP carry no information. In particular the "o=", "s=" fields and "t=". These are, however, mandatory fields within SDP. Does 3GPP wish to define a non-standard version of SDP that removes these, and if so, how does this interwork with outside SIP networks that use standard SDP.

   Editor's Note: Need to insure the codec negotiation procedures are compatible with the procedures brought into release 4 for CS domain services (BICC).

Table 8.2.2-1: INVITE (UE to P-CSCF)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 98 99
a=qos:mandatory sendrecv
a=rtpmap:98 H261
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 98 99
a=rtpmap:98 H261
a=rtpmap:99:MPV
a=qos:mandatory sendrecv
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

   Request-URI: contains the keyed number from the user.

   Via: contains the IP address or FQDN of the originating UE.

Remote-Party-ID: contains the public identity of the UE. The Display name is optional.

Proxy-Require: The sip privacy draft specifies that the usage of the Remote-Party-Id MUST be accompanied by a Proxy-Require header specifying "privacy" in all INVITE requests.

From:, To:, Call-ID: follow the recommendations of draft-ietf-sip-privacy-01, even though anonymity is not being requested for this session.

Cseq: is a random starting number.

Contact: is the IP address or FQDN of the originating UE.

SDP The SDP contains the full set of codecs supported by UE#1

2. **100 Trying (P-CSCF to UE) – see example in Table 8.2.2-2**

P-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.2.2-2: 100 Trying (P-CSCF to UE)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **INVITE (P-CSCF to S-CSCF) – see example in Table 8.2.2-3**

P-CSCF remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. This next hop is the S-CSCF within the home network.

P-CSCF rewrites the Contact header, with a locally defined value that identifies the UE. P-CSCF adds itself to the Record-Route header, and adds a Via header.

P-CSCF#1 examines the media parameters, and removes any choices that the network operator decides based on local policy, not to allow on the network.

For this example, assume the network operator disallows H261 video encoding.

Editor's Note: Need to describe the procedures for a P-CSCF/S-CSCF to follow when it doesn't understand a media line in the SDP. It is clear that it should not remove such lines, as that destroys the ability to create new services; however the CSCF needs sufficient information about the unknown media stream to perform the authorization.

The INVITE request is forwarded to the S-CSCF.

Table 8.2.2-3: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv

a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Request-URI: is the first component in the remembered Path header from Registration.

Route: contains the remaining elements from the Path header from Registration, with the initial Request-URI (received from the UE) appended as the final component.

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the video media streams no longer list code 98 (H261).

Editor's Note: Modified text for this step is contained in Annex A.

4. **100 Trying (S-CSCF to P-CSCF) – see example in Table 8.2.2-4**

   S-CSCF responds to the INVITE request (3) with a 100 Trying provisional response.

Table 8.2.2-4: 100 Trying (S-CSCF to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

5. **Service Control**

   S-CSCF validates the service profile, and performs any origination service control required for this subscriber.

   S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

For this example, assume the subscriber is not allowed video.

Editor's Note: Modified text for this step is contained in Annex A.

6. **INVITE (MO#1 to S-S) – see example in Table 8.2.2-6**

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

Editor's Note: Need for additional headers to transport e.g. Billing-Correlation-Identifier is FFS.

Table 8.2.2-6: INVITE request (MO#1a to S-S)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
a=qos:mandatory sendrecv

a=rtpmap:99:MPV
m=video 0 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

SDP   The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the video media streams show a port number zero, which removes them from the negotiation.

Request-URI: In the case where the Route header of the incoming INVITE request to S-CSCF contains a TEL-URL, it has to be translated to a globally routable SIP-URL before applying it as Request-URI of the outgoing INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: Need to verify that it is the Route header of the incoming INVITE request which may contain the TEL-URL, and not the Request-URI as indicated in the text of Tdoc N1-010353.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

7. **100 Trying (S-S to MO#1a) – see example in Table 8.2.2-7**

S-CSCF receives a 100 Trying provisional response, as specified by the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.2-7: 100 Trying (S-S to MO#1a)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

8. **183 Session Progress (S-S to MO#1a) – see example in Table 8.2.2-8**

   The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response (to 6), per the S-CSCF to S-CSCF procedures.

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.2-8: 183 Session Progress (S-S to MO#1a)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

9. **183 Session Progress (S-CSCF to P-CSCF) – see example in Table 8.2.2-9**

   S-CSCF forwards the 183 Session Progress response to P-CSCF.

Table 8.2.2-9: 183 Session Progress (S-SCSF to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=a=
a=
a=
a=
m=
```

10. **Authorize QoS Resources**

    P-CSCF authorizes the resources necessary for this session.

11. **183 Session Progress (P-CSCF to UE) – see example in Table 8.2.2-11**

    P-CSCF forwards the 183 Session Progress response to the originating endpoint.

Table 8.2.2-11: 183 Session Progress (P-CSCF to UE)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Media-Authorization: 00200001001001017063663312e78797a2e6e6574000c02013942563330373200
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token1@pcscf1.visited.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=a=
a=
a=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf.home.net, sip:scscf2.home.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
```

Contact: a locally unique token to identify the saved routing information.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf1.xyz.net" with credentials "9BV3072". "00" at the end of the authorixation token is required to pad to a multiple of 4 bytes.

Editor's Note: Modified text for this step is contained in Annex A.

12. **PRACK (UE to P-CSCF) – see example in Table 8.2.2-12**

UE#1 determines which media flows should be used for this session, and which codecs should be used for each of those media flows. If there was any change in media flows, or if there was more than one choice of codec for a media flow, then UE#1 must include an SDP in the PRACK message sent to UE#2.

For this example, assume UE#1 chooses AMR as the codec to use for the single audio stream.

UE includes this information in the PRACK request to P-CSCF.

Editor's Note: The use of three-message codec negotiation (one round-trip to determine common capabilities, then originator picks the ones to use) is allowed by RFC2543, but will apparently not be supported by 2543bis. This inconsistency needs to be resolved.

Table 8.2.2-12: PRACK (UE to P-CSCF)

```
PRACK sip:token1@pcscf1.visited.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

Request-URI: takes the value of the Contact header of the received 183 Session Progress response.

Via:, Contact: take the value of either the IP address of RQDN of the originating UE.

From:, To:, Call-ID: copied from the 183 Session Progress response so that they include any tag parameter.

Cseq: takes a higher value than that in the previous request.

The final selection of the media stream from the set of those supported by the terminating endpoint, given in the received 183 Session Progress response (14), is made by the originating UE and included in the SDP.

13. **PRACK (P-CSCF to S-CSCF) – see example in Table 8.2.2-13**

   P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

   P-CSCF forwards the PRACK request to S-CSCF.

Table 8.2.2-13: PRACK (P-CSCF to S-CSCF)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

   Request-URI: taken from the first component of the saved Route header.

   Route: as saved from the previous response (with first element moved to Request-URI).

   Contact: re-written by P-CSCF with a locally defined value that identifies the UE.

14. **PRACK (MO#1a to S-S) – see example in Table 8.2.2-14**

   S-CSCF forwards the PRACK request to the terminating endpoint, as per the S-CSCF to S-CSCF procedure.

Table 8.2.2-14: PRACK (MO#1a to S-S)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Request-URI: the first component of the Route header. This will vary according to which S-CSCF to S-CSCF flow is used.

15. **200 OK (S-S to MO#1a) – see example in Table 8.2.2-15**

The destination endpoint responds to the PRACK request (14) with a 200 OK response, per the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.2-15: 200 OK (S-S to MO#1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

16. **200 OK (S-CSCF to P-CSCF) – see example in Table 8.2.2-16**

S-CSCF forwards the 200 OK response to P-CSCF.

Table 8.2.2-16: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

17. **200 OK (P-CSCF to UE) – see example in Table 8.2.2-17**

P-CSCF forwards the 200 OK response to UE.

Table 8.2.2-17: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf1.visited.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

18. **Resource Reservation**

After determining the final media streams in step #11, UE initiates the reservation procedures for the resources needed for this session.

19. **COMET (UE to P-CSCF) – see example in Table 8.2.2-19**

When the resource reservation is completed, UE sends the COMET request to the terminating endpoint, via the signalling path established by the INVITE request. The request is sent first to P-CSCF.

Table 8.2.2-19: COMET (UE to P-CSCF)

```
COMET sip:token2@pcscf1.visited.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

Request-URI: takes the value of the Contact header of the received 183 Session Progress response.

Via:, Contact: take the value of either the IP address or FQDN of the originating UE.

From:, To:, Call-ID: copied from the 183 Session Progress response so that they include any tag parameters.

Cseq: takes a higher value than that in the previous request.

The SDP indicates that the resource reservation was successful.

20. **COMET (P-CSCF to S-CSCF) – see example in Table 8.2.2-20**

P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF forwards the COMET request to S-CSCF.

Table 8.2.2-20: COMET (P-CSCF to S-CSCF)

```
COMET sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Request-URI: taken from the first component of the saved Route header.

Route: saved from the 183 Session Progress response (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

21. **COMET (MO#1a to S-S) – see example in Table 8.2.2-21**

S-CSCF forwards the COMET request to the terminating endpoint, as per the S-CSCF to S-CSCF procedure.

Table 8.2.2-21: COMET (MO#1a to S-S)

```
COMET sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Request-URI: the first component of the Route header. This will vary according to which S-CSCF to S-CSCF flow is used.

22. **200 OK (S-S to MO#1a) – see example in Table 8.2.2-22**

The destination endpoint responds to the COMET request (21) with a 200 OK, per the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.2-22: 200 OK (S-S to MO#1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

23. **200 OK (S-CSCF to P-CSCF) – see example in Table 8.2.2-23**

S-CSCF forwards the 200 OK response to P-CSCF.

Table 8.2.2-23: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

24. **200 OK (P-CSCF to UE) – see example in Table 8.2.2-24**

P-CSCF forwards the 200 OK response to UE.

Table 8.2.2-24: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

25. **180 Ringing (S-S to MO#1a) – see example in Table 8.2.2-25**

The called UE may optionally perform alerting. If so, it signals this to the calling party by a 180 Ringing provisional response to (6). This response is sent to S-CSCF per the S-CSCF to S-CSCF procedure.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.2-25: 180 Ringing (S-S to MO#1a)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
RSeq: 9022
Content-length: 0
```

26. **Service Control**

The S-CSCF validates the service profile and performs any service control required for this subscriber

27. **180 Ringing (S-CSCF to P-CSCF) – see example in Table 8.2.2-27**

S-CSCF forwards the 180 Ringing response to P-CSCF.

Table 8.2.2-27: 180 Ringing (S-CSCF to P-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

28. **180 Ringing (P-CSCF to UE) – see example in Table 8.1.1-28**

P-CSCF removes the Record-Route and Contact headers, calculates the propoer Route header to add to future requests, and saves that information without passing it to UE. A Contact header is added with a locally unique token to identify the saved routing information.

P-CSCF forwards the 180 Ringing response to UE.

Table 8.2.2-28: 180 Ringing (P-CSCF to UE)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token4@pcscf1.visited.net
RSeq:
Content-length:
```

Editor's Note: Additional QoS interactions to handle one-way media at this point (e.g. for PSTN ringback and announcements) is for further study.

29. **PRACK (UE to P-CSCF) – see example in Table 8.2.2-29**

UE indicates to the originating subscriber that the destination is ringing. It responds to the 180 Ringing provisional response (28) with a PRACK request.

Table 8.2.2-29: PRACK (UE to P-CSCF)

```
PRACK sip:token4@pcscf1.visited.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 9022 127 INVITE
Content-length: 0
```

Request-URI: takes the value of the Contact header of the 180 Ringing response.

Via:, Contact: take the value of either the IP address or FQDN of the UE.

From:, To:, Call-ID: copied from the 180 Ringing response so that they include any revised tag parameters.

Cseq: takes a higher value than in the previous request.

30. **PRACK (P-CSCF to S-CSCF) – see example in Table 8.2.2-30**

   P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

   P-CSCF forwards the PRACK request to S-CSCF.

Table 8.2.2-30: PRACK (P-CSCF to S-CSCF)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Rack:
Content-length:
```

31. **PRACK (MO#1a to S-S) – see example in Table 8.2.2-31**

   S-CSCF forwards the PRACK request to the terminating endpoint, as per the S-CSCF to S-CSCF procedure.

Table 8.2.2-31: PRACK (MO#1a to S-S)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

32. **200 OK (S-S to MO#1a) – see example in Table 8.2.2-32**

   The destination endpoint responds to the PRACK request (31) with a 200 OK response.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.2-32: 200 OK (S-S to MO#1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

33. **200 OK (S-CSCF to P-CSCF) – see example in Table 8.2.2-33**

    S-CSCF forwards the 200 OK response to P-CSCF.

Table 8.2.2-33: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

34. **200 OK (P-CSCF to UE) – see example in Table 8.2.2-34**

    P-CSCF forwards the 200 OK response to UE.

Table 8.2.2-34: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token5@pcscf1.visited.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

35. **200 OK (S-S to MO#1a) – see example in Table 8.2.2-35**

    When the called party answers, the terminating endpoint sends a 200 OK final response to the INVITE request (6), as specified by the termination procedures and the S-CSCF to S-CSCF procedures, to S-CSCF.

Table 8.2.2-35: 200 OK (S-S to MO#1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

36. **Service Control**

S-CSCF performs whatever service control is appropriate for the completed session

Editor's Note: Modified text for this step is contained in Annex A.

37. **200 OK (S-CSCF to P-CSCF) – see example in Table 8.2.2-37**

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

Table 8.2.2-37: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

38. **200 OK (P-CSCF to UE) – see example in Table 8.2.2-38**

P-CSCF indicates the resources reserved for this session should now be committed, and forwards the 200 OK final response to the session originator. Ue can start the media flow(s) for this session

Table 8.2.2-38: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token6@pcscf1.visited.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

39. **ACK (UE to P-CSCF) – see example in Table 8.2.2-39**

UE starts the media flow for this session, and responds to the 200 OK (37) with an ACK request sent to P-CSCF.

Table 8.2.2-39: ACK (UE to P-CSCF)

```
ACK sip:token6@pcscf1.visited.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 ACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

Cseq: is required to be the same value as Cseq contained in original INVITE request [3]

40. **ACK (P-CSCF to S-CSCF) – see example in Table 8.2.2-40**

P-CSCF forwards the ACK request to S-CSCF.

Table 8.2.2-40: ACK (P-CSCF to S-CSCF)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaaa%3abbbb%3accc%3addd%5d@pcscf1.visited.net
Content-length:
```

> **Request-URI:** the first component of the saved Route header.
>
> **Route:** is as saved from the 200 OK response (with first element moved to Request-URI).
>
> **Contact:** a locally defined value that identifies the UE.

41. **ACK (MO#1a to S-S) – see example in Table 8.2.2-41**

> S-CSCF forwards the ACK request to the terminating endpoint, per the S-CSCF to S-CSCF procedure.

Table 8.2.2-41: ACK (MO#1a to S-S)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Length:
```

8.2.3 (MO#1b) Mobile origination, roaming, with I-CSCF in home network providing configuration independence

This origination procedure applies to roaming subscribers when the home network operator desires to keep its internal configuration hidden from the visited network. The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure. During registration, the home network allocates an S-CSCF. The home network advertises an I-CSCF as the entry point from the visited network, who forwards requests to the S-CSCF.

When registration is complete, P-CSCF knows the name/address of the next hop in the signalling path toward the serving-CSCF, the I-CSCF. I-CSCF receives information in the request, from which it determines the name/address of the proper S-CSCF.
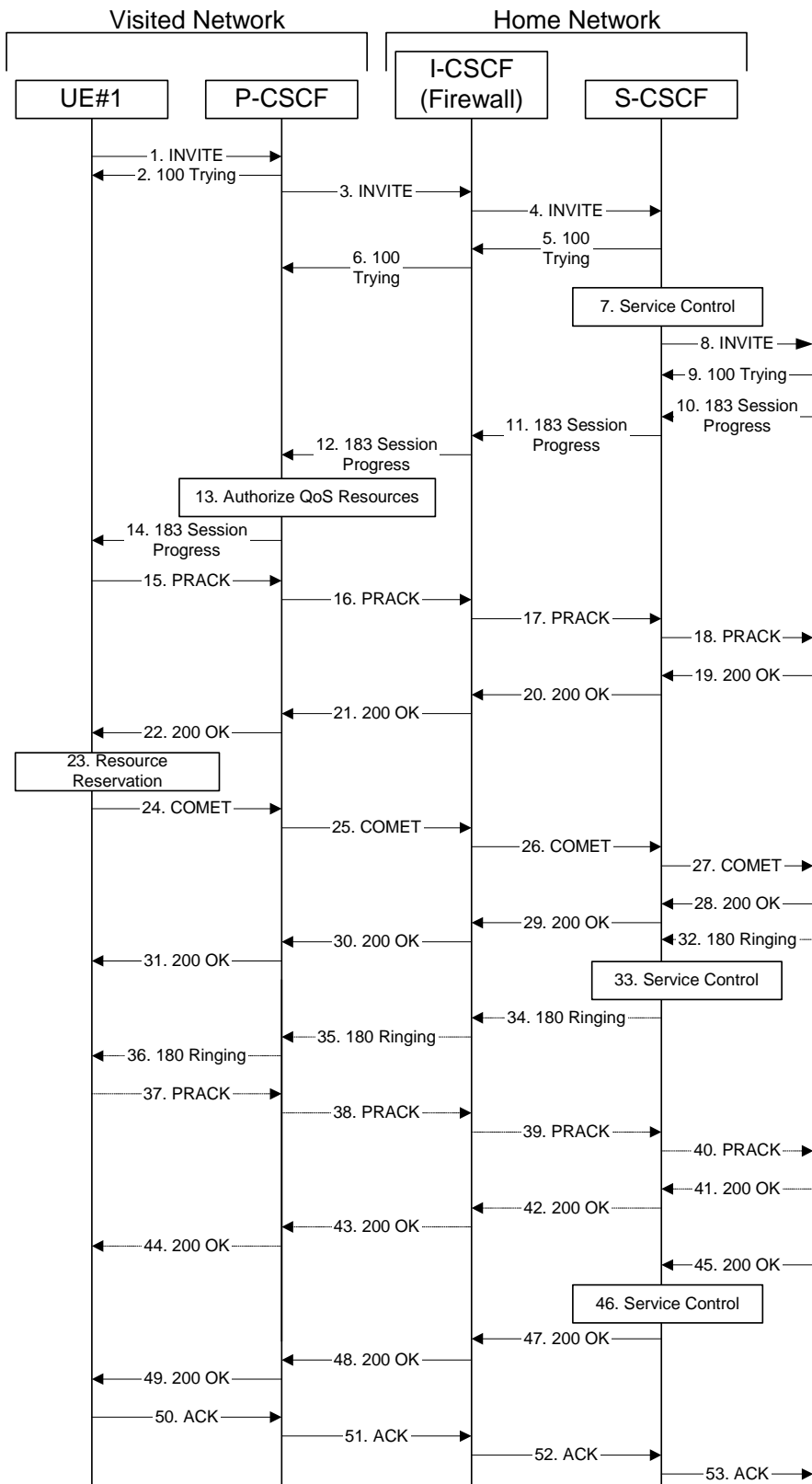
Figure 8.2.3 – MO#1b

Procedure MO#1b is as follows:

1. **INVITE (UE to P-CSCF) – see example in Table 8.2.3-1**

UE#1 determines the complete set of codecs that it is capable of supporting for this session. It builds a SDP containing bandwidth requirements and characteristics of each, and assigns local port numbers for each possible media flow. Multiple media flows may be offered, and for each media flow (m= line in SDP), there may be multiple codec choices offered.

For this example, assume UE#1 is capable of sending two simultaneous video streams, either H261 or MPV format, and two simultaneous audio streams, either AMR, G726-32, PCMU, or G728.

UE sends the INVITE request, containing an initial SDP, to the P-CSCF determined via the CSCF discovery mechanism. An example is contained in table 8.2.3-1.

Editor's Note: Certain fields in the SDP carry no information. In particular the "o=", "s=" fields and "t=". These are, however, mandatory fields within SDP. Does 3GPP wish to define a non-standard version of SDP that removes these, and if so, how does this interwork with outside SIP networks that use standard SDP.

Table 8.2.3-1: INVITE (UE to P-CSCF)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111;time=36123E5B;seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 98 99
a=qos:mandatory sendrecv
a=rtpmap:98 H261
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 98 99
a=qos:mandatory sendrecv
a=rtpmap:98 H261
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

Request-URI: contains the keyed number from the user.

Via: contains the IP address or FQDN of the originating UE.

Remote-Party-ID: contains the public identity of the UE. The Display name is optional.

Proxy-Require: The sip privacy draft specifies that the usage of the Remote-Party-Id MUST be accompanied by a Proxy-Require header specifying "privacy" in all INVITE requests.

From:, To:, Call-ID: follow the recommendations of draft-ietf-sip-privacy-01, even though anonymity is not being requested for this session.

Cseq: is a random starting number.

Contact: is the IP address or FQDN of the originating UE.

> **SDP** The SDP contains the full set of codecs supported by UE#1

2. **100 Trying (P-CSCF to UE) – see example in Table 8.2.3-2**

P-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.2.3-2: 100 Trying (P-CSCF to UE)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **INVITE (P-CSCF to I-CSCF) – see example in Table 8.2.3-3**

P-CSCF#1 examines the media parameters, and removes any choices that the network operator decides based on local policy, not to allow on the network.

For this example, assume the network operator disallows H261 video encoding.

The INVITE request is forwarded through this I-CSCF to the S-CSCF.

Table 8.2.3-3: INVITE (P-CSCF to I-CSCF)

```
INVITE sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(scscf.home.net), sip:+1-212-555-2222@home.net;user=phone
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecva=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

> **Route:** P-CSCF remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. The initial Request-URI is appended to the Route header.

> **Contact:** a locally defined value that identifies the UE.

> **Request-URI:** the first component in the remembered Path header from Registration.

SDP   The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the video media streams no longer list code 98 (H261).

Editor's Note: Modified text for this step is contained in Annex A.

4. **INVITE (I-CSCF to S-CSCF) – see example in Table 8.2.3-4**

   I-CSCF adds itself to the Record-Route header, and adds a Via header.

   I-CSCF determines the routing information contained in the request, and forwards the request to S-CSCF that is serving the UE.

Table 8.2.3-4: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:icscf.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
m=
a=
a=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

5. **100 Trying (S-CSCF to I-CSCF) – see example in Table 8.2.3-5**

   S-CSCF responds to the INVITE request (4) with a 100 Trying provisional response.

Table 8.2.3-5: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **100 Trying (I-CSCF to P-CSCF) – see example in Table 8.2.3-6**

   I-CSCF forwards the 100 Trying provisional response to P-CSCF.

Table 8.2.3-6: 100 Trying (I-CSCF to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

7. **Service Control**

S-CSCF validates the service profile, and performs any origination service control required for this subscriber.

S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

For this example, assume the subscriber is not allowed video.

Editor's Note: Modified text for this step is contained in Annex A.

8. **INVITE (MO#1b to S-S) – see example in Table 8.2.3-8**

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

Editor's Note: Need for additional headers to transport e.g. Billing-Correlation-Identifier is FFS.

Table 8.2.3-8: INVITE (MO#1b to S-S)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:icscf.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the video media streams show a port number zero, which removes them from the negotiation.

Request-URI: In the case where the Route header of the incoming INVITE request to S-CSCF contains a TEL-URL, it has to be translated to a globally routable SIP-URL before applying it as Request-URI of the outgoing INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: Need to verify that it is the Route header of the incoming INVITE request which may contain the TEL-URL, and not the Request-URI as indicated in the text of Tdoc N1-010353.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

9. **100 Trying (S-S to MO#1b) – see example in Table 8.2.3-9**

S-CSCF receives a 100 Trying provisional response, as specified by the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.3-9: 100 Trying (S-S to MO#1b)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

10. **183 Session Progress (S-S to MO#1b) – see example in Table 8.2.3-10**

The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response (to (8)), per the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.3-10: 183 Session Progress response (S-S to MO#1b)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net, sip:icscf.home.net
Remote-Party-ID: John Smith <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

Editor's Note: Future contributions should address the alternative of P-CSCF not modifying the Contact header and instead inserting itself in the Record-Route.

11. **183 Session Progress (S-CSCF to I-CSCF) – see example in Table 8.2.3-11**

S-CSCF forwards the 183 Session Progress response to I-CSCF.

Table 8.2.3-11: 183 Session Progress (S-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

12. **183 Session Progress (I-CSCF to P-CSCF) – see example in Table 8.2.3-12**

I-CSCF forwards the 183 Session Progress response to P-CSCF

Table 8.2.3-12: 183 Session Progress (I-CSCF to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home.net, sip:scscf2.home.net), sip:icscf.home.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Record-Route: header entries to the left of I-CSCF's entry are reversed and translated.

13. **Authorize QoS Resources**

P-CSCF authorizes the resources necessary for this session

14. **183 Session Progress (P-CSCF to UE) – see example in Table 8.2.3-14**

P-CSCF forwards the 183 Session Progress response to the originating endpoint

Table 8.2.3-14: 183 Session Progress (P-CSCF to UE)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Media-Authorization: 0020000100100101706366312e78797a2e6e6574000c02013942563330373200
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token1@pcscf1.visited.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:icscf.home.net,
    sip:Token(sip:scscf.home.net, sip:scscf2.home.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
```

Contact: a locally unique token to identify the saved routing information.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf1.xyz.net" with credentials "9BV3072". "00" at the end of the authorixation token is required to pad to a multiple of 4 bytes.

Editor's Note: Modified text for this step is contained in Annex A.

15. **PRACK (UE to P-CSCF) – see example in Table 8.2.3-15**

UE decides the final set of media streams for this session, and includes this information in the PRACK request to P-CSCF.

Table 8.2.3-15: PRACK (UE to P-CSCF)

```
PRACK sip:token1@pcscf1.visited.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

Request-URI: takes the value of the Contact header of the received 183 Session Progress response.

Via:, Contact: take the value of either the IP address of RQDN of the originating UE.

From:, To:, Call-ID: copied from the 183 Session Progress response so that they include any tag parameter.

Cseq: takes a higher value than that in the previous request.

The final selection of the media stream from the set of those supported by the terminating endpoint, given in the received 183 Session Progress response (14), is made by the originating UE and included in the SDP.

16. **PRACK (P-CSCF to I-CSCF) – see example in Table 8.2.3-16**

P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF forwards the PRACK request to I-CSCF.

Table 8.2.3-16: PRACK (P-CSCF to I-CSCF)

```
PRACK sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(sip:scscf.home.net, sip:scscf2.home.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Request-URI: taken from the first component of the saved Route header.

Route: saved from the previous response (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

17. **PRACK (I-CSCF to S-CSCF) – see example in Table 8.2.3-17**

I-CSCF determines the routing information, and forwards the PRACK request to S-CSCF.

Table 8.2.3-17: PRACK (I-CSCF to S-CSCF)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:icscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

18. **PRACK (MO#1b to S-S) – see example in Table 8.2.3-18**

S-CSCF forwards the PRACK request to the terminating endpoint, as per the S-CSCF to S-CSCF procedure.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.3-18: PRACK (MO#1b to S-S)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net, sip:iscsf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Request-URI: the first component of the Route header. This will vary according to which S-CSCF to S-CSCF flow is used.

19. **200 OK (S-S to MO#1b) – see example in Table 8.2.3-19**

The destination endpoint responds to the PRACK request (18) with a 200 OK response, per the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.3-19: 200 OK (S-S to MO#1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net, sip:icscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

20. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.2.3-20**

S-CSCF forwards the 200 OK response to I-CSCF

Table 8.2.3-20: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net, sip:icscf.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

21. **200 OK (I-CSCF to P-CSCF) – see example in Table 8.2.3-21**

    I-CSCF forwards the 200 OK response to P-CSCF.

Table 8.2.3-21: 200 OK (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home.net, sip:scscf2.home.net), sip:icscf.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

Record-Route: entries to the left of I-CSCF's entry are reversed and translated.

22. **200 OK (P-CSCF to UE) – see example in Table 8.2.3-22**

    P-CSCF forwards the 200 OK response to UE.

Table 8.2.3-22: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf1.visited.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

Editor's Note: The need to generate a new token for Record-Route and Contact information for a non-INVITE response is FFS.

23. **Resource Reservation**

    After determining the final media streams in step #14, UE initiates the reservation procedures for the resources needed for this session.

24. **COMET (UE to P-CSCF) – see example in Table 8.2.3-24**

    When the resource reservation is completed, UE sends the COMET request to the terminating endpoint, via the signalling path established by the INVITE request. The request is sent first to P-CSCF.

Table 8.2.3-24: COMET (UE to P-CSCF)

```
COMET sip:token2@pcscf1.visited.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

Request-URI: takes the value of the Contact header of the received 183 Session Progress response.

Via, Contact: take the value of either the IP address or FQDN of the originating UE.

From:, To:, Call-ID: copied from the 183 Session Progress response so that they include any tag parameters.

Cseq: takes a higher value than that in the previous request.

   The SDP indicates that the resource reservation was successful.

25. **COMET (P-CSCF to I-CSCF) – see example in Table 8.2.3-25**

   P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

   P-CSCF forwards the COMET request to I-CSCF.

Table 8.2.3-25: COMET (P-CSCF to I-CSCF)

```
COMET sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(sip:scscf.home.net, sip:scscf2.home.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Request-URI: taken from the first component of the saved Route header.

> **Route:** saved from the 183 Session Progress response (with first element moved to Request-URI).

> **Contact:** a locally defined value that identifies the UE.

26. **COMET (I-CSCF to S-CSCF) – see example in Table 8.2.3-26**

> I-CSCF determines the routing information, and forwards the request to S-CSCF.

Table 8.2.3-26: COMET (I-CSCF to S-CSCF)

```
COMET sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:icscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

27. **COMET (MO#1b to S-S) – see example in Table 8.2.3-27**

> S-CSCF forwards the COMET request to the terminating endpoint, as per the S-CSCF to S-CSCF procedure.

Table 8.2.3-27 COMET (MO#1b to S-S)

```
COMET sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net, sip:icscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

> **Request-URI:** the first component of the Route header. This will vary according to which S-CSCF to S-CSCF flow is used.

28. **200 OK (S-S to MO#1b) – see example in Table 8.2.3-28**

    The destination endpoint responds to the COMET request (27) with a 200 OK, per the S-CSCF to S-CSCF procedures.

    NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.3-28: 200 OK (S-S to MO#1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net, sip:icscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

29. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.2.3-29**

    S-CSCF forwards the 200 OK response to I-CSCF.

Table 8.2.3-29 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

30. **200 OK (I-CSCF to P-CSCF) – see example in Table 8.2.3-30**

    I-CSCF forwards the 200 OK response to P-CSCF.

Table 8.2.3-30: 200 OK (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home.net, sip:scscf2.home.net),sip:icscf.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

Record-Route: entries to the left of I-CSCF's entry are reversed and translated.

31. **200 OK (P-CSCF to UE) – see example in Table 8.2.3-31**

    P-CSCF forwards the 200 OK response to UE.

Table 8.2.3-31: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

32. **180 Ringing (S-S to MO#1b) – see example in Table 8.2.3-32**

The called UE may optionally perform alerting. If so, it signals this to the calling party by a 180 Ringing provisional response to (8). This response is sent to S-CSCF per the S-CSCF to S-CSCF procedure.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.3-32: 180 Ringing (S-S to MO#1b)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net, sip:icscf.home.net
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
RSeq: 9022
Content-length: 0
```

33. **Service Control**

The S-CSCF validates the service profile and performs any service control required for this subscriber.

34. **180 Ringing (S-CSCF to I-CSCF) – see example in Table 8.1.2-34**

S-CSCF forwards the 180 Ringing response to I-CSCF.

Table 8.2.3-34: 180 Ringing (S-CSCF to I-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

35. **180 Ringing (I-CSCF to P-CSCF) – see example in Table 8.2.3-35**

I-CSCF forwards the 180 Ringing response to P-CSCF.

Table 8.2.3-35: 180 Ringing (I-CSCF to P-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home.net, sip:scscf2.home.net), sip:icscf.home.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

Record-Route: entries to the left of I-CSCF's entry are reversed and translated.

36. **180 Ringing (P-CSCF to UE) – see example in Table 8.2.3-36**

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

P-CSCF forwards the 180 Ringing response to UE.

Table 8.2.3-36: 180 Ringing (P-CSCF to UE)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token4@pcscf1.visited.net
RSeq:
Content-length:
```

Contact: a locally unique token to identify the saved routing information

Editor's Note: Additional QoS interactions to handle one-way media at this point (e.g. for PSTN ringback and announcements) is for further study.

37. **PRACK (UE to P-CSCF) – see example in Table 8.2.3-37**

UE indicates to the originating subscriber that the destination is ringing. It acknowledges the 180 Ringing provisional response (36) with a PRACK request.

Table 8.2.3-37: PRACK (UE to P-CSCF)

```
PRACK sip:token4@pcscf1.visited.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 9022 127 INVITE
Content-length: 0
```

Request-URI: takes the value of the Contact header of the 180 Ringing response.

Via:, Contact: take the value of either the IP address or FQDN of the UE.

From:, To:, Call-ID: copied from the 180 Ringing response so that they include any revised tag parameters.

Cseq: takes a higher value than in the previous request.

38. **PRACK (P-CSCF to I-CSCF) – see example in Table 8.2.3-38**

P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF forwards the PRACK request to I-CSCF.

Table 8.2.3-38: PRACK (P-CSCF to I-CSCF)

```
PRACK sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(sip:scscf.home.net, sip:scscf2.home.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Rack:
Content-length:
```

39. **PRACK (I-CSCF to S-CSCF) – see example in Table 8.2.3-39**

I-CSCF forwards the PRACK request to S-CSCF.

Table 8.2.3-39: PRACK (I-CSCF to S-CSCF)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:icscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

40. **PRACK (MO#1b to S-S) – see example in Table 8.2.3-40**

S-CSCF forwards the PRACK request to the terminating endpoint, as per the S-CSCF to S-CSCF procedure.

Table 8.2.3-40: PRACK (MO#1b to S-S)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net, sip:icscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

41. **200 OK (S-S to MO#1b) – see example in Table 8.2.3-41**

The destination endpoint responds to the PRACK request (40) with a 200 OK response.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.3-41: 200 OK (S-S to MO#1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net, sip:icscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

42. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.2.3-42**

   S-CSCF forwards the 200 OK response to I-CSCF.

Table 8.2.3-42: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

43. **200 OK (I-CSCF to P-CSCF) – see example in Table 8.2.3-43**

   I-CSCF forwards the 200 OK response to P-CSCF.

Table 8.2.3-43: 200 OK (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home.net, sip:scscf2.home.net),sip:icscf.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

Record-Route: entries to the left of I-CSCF's entry are reversed and translated.

44. **200 OK (P-CSCF to UE) – see example in Table 8.2.3-44**

   P-CSCF forwards the 200 OK response to UE.

Table 8.2.3-44: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token5@pcscf1.visited.net
Content-Length:
```

   P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

45. **200 OK (S-S to MO#1b) – see example in Table 8.2.3-45**

When the called party answers, the terminating endpoint sends a 200 OK final response to the INVITE request (8), as specified by the termination procedures and the S-CSCF to S-CSCF procedures, to S-CSCF.

Table 8.2.3-45: 200 OK (S-S to MO#1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net, sip:icscf.home.net
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbbb%5d@pcscf2.home.net
Content-Length: 0
```

46. **Service Control**

S-CSCF performs whatever service control is appropriate for the completed session

Editor's Note: Modified text for this step is contained in Annex A.

47. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.2.3-47**

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

Table 8.2.3-47: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net, sip:icscf.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

48. **200 OK (I-CSCF to P-CSCF) – see example in Table 8.2.3-48**

I-CSCF sends the 200 OK final response to P-CSCF.

Table 8.2.3-48: 200 OK (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home.net, sip:scscf2.home.net), sip:icscf.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

Record-Route: entries to the left of I-CSCF's entry are reversed and translated.

49. **200 OK (P-CSCF to UE) – see example in Table 8.2.3-49**

P-CSCF indicates the resources reserved for this session should now be committed, and forwards the 200 OK final response to the session originator. Ue can start the media flow(s) for this session.

Table 8.2.3-49: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token6@pcscf1.visited.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

50. **ACK (UE to P-CSCF) – see example in Table 8.2.3-50**

UE starts the media flow for this session, and responds to the 200 OK (49) with an ACK request sent to P-CSCF.

Table 8.2.3-50: ACK (UE to P-CSCF)

```
ACK sip:token6@pcscf1.visited.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 ACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

Cseq: is required to be the same value as Cseq contained in original INVITE request [3]

51. **ACK (P-CSCF to I-CSCF) – see example in Table 8.2.3-51**

P-CSCF forwards the ACK request to I-CSCF.

Table 8.2.3-51: ACK (P-CSCF to I-CSCF)

```
ACK sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(sip:scscf.home.net, sip:scscf2.home.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200 OK response (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

52. **ACK (I-CSCF to S-CSCF) – see example in Table 8.2.3-52**

I-CSCF determines the routing information, and forwards the ACK request to S-CSCF.

Table 8.2.3-52: ACK (I-CSCF to S-CSCF)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:icscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Length:
```

53. **ACK (MO#1b to S-S) – see example in Table 8.2.3-53**

S-CSCF forwards the ACK request to the terminating endpoint, per the S-CSCF to S-CSCF procedure.

Table 8.2.3-53: ACK (MO#1b to S-S)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net, sip:icscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Length:
```

8.2.4 (MO#2) 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. During registration, the home network allocates an S-CSCF in the home network.

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

Home Network
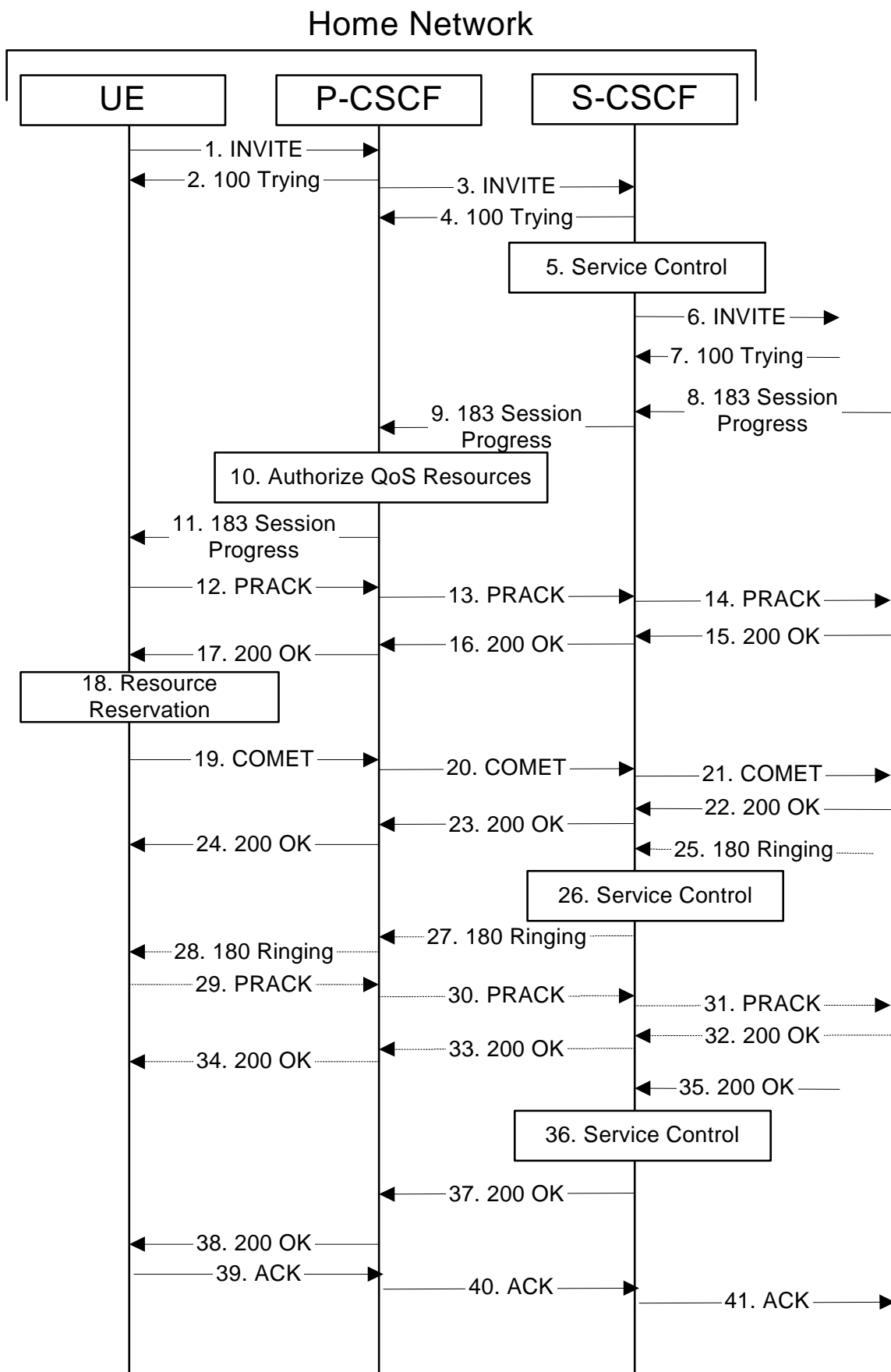
Figure 8.1.3 – MO#2

Procedure MO#2 is as follows:

1. **INVITE (UE to P-CSCF) – see example in Table 8.2.4-1**

   UE#1 determines the complete set of codecs that it is capable of supporting for this session. It builds a SDP containing bandwidth requirements and characteristics of each, and assigns local port numbers for each possible media flow. Multiple media flows may be offered, and for each media flow (m= line in SDP), there may be multiple codec choices offered.

   For this example, assume UE#1 is capable of sending two simultaneous video streams, either H261 or MPV format, and two simultaneous audio streams, either AMR, G726-32, PCMU, or G728.

   UE sends the INVITE request, containing an initial SDP, to the P-CSCF determined via the CSCF discovery mechanism.

   Editor's Note: Certain fields in the SDP carry no information. In particular the "o=", "s=" fields and "t=". These are, however, mandatory fields within SDP. Does 3GPP wish to define a non-standard version of SDP that removes these, and if so, how does this interwork with outside SIP networks that use standard SDP.

Table 8.2.4-1: INVITE (UE to P-CSCF)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 98 99
a=qos:mandatory sendrecv
a=rtpmap:98 H261
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 98 99
a=qos:mandatory sendrecv
a=rtpmap:98 H261
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

Request-URI: contains the keyed number from the user.

Via: contains the IP address or FQDN of the originating UE.

Remote-Party-ID: contains the public identity of the UE. The Display name is optional.

Proxy-Require: The sip privacy draft specifies that the usage of the Remote-Party-Id MUST be accompanied by a Proxy-Require header specifying "privacy" in all INVITE requests.

From:, To:, Call-ID: follow the recommendations of draft-ietf-sip-privacy-01, even though anonymity is not being requested for this session.

Cseq: a random starting number.

Contact: the IP address or FQDN of the originating UE.

SDP   The SDP contains the full set of codecs supported by UE#1

2. **100 Trying (P-CSCF to UE) – see example in Table 8.2.4-2**

    P-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.2.4-2: 100 Trying (P-CSCF to UE)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **INVITE (P-CSCF to S-CSCF) – see example in Table 8.2.4-3**

    P-CSCF remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. This next hop is the S-CSCF within the home network.

    P-CSCF rewrites the Contact header, with a locally defined value that identifies the UE. P-CSCF adds itself to the Record-Route header, and adds a Via header.

    P-CSCF#1 examines the media parameters, and removes any choices that the network operator decides based on local policy, not to allow on the network.

    For this example, assume the network operator disallows H261 video encoding.

    The INVITE request is forwarded to the S-CSCF.

Table 8.2.4-3: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Request-URI: the first component in the remembered Path header from Registration.

Route: contains the remaining elements from the Path header from Registration, with the initial Request-URI (received from the UE) appended as the final component.

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the video media streams no longer list code 98 (H261).

Editor's Note: Modified text for this step is contained in Annex A.

4. **100 Trying (S-CSCF to P-CSCF) – see example in Table 8.2.4-4**

   S-CSCF responds to the INVITE request (3) with a 100 Trying provisional response.

Table 8.2.4-4: 100 Trying (S-CSCF to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

5. **Service Control**

   S-CSCF validates the service profile, and performs any origination service control required for this subscriber.

   S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

For this example, assume the subscriber is not allowed video.

Editor's Note: Modified text for this step is contained in Annex A.

6. **INVITE (MO#2 to S-S) – see example in Table 8.2.4-6**

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

Editor's Note: Need for additional headers to transport e.g. Billing-Correlation-Identifier is FFS.

Table 8.2.4-6: INVITE (MO#2 to S-S)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

SDP   The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the video media streams show a port number zero, which removes them from the negotiation.

Request-URI: In the case where the Route header of the incoming INVITE request to S-CSCF contains a TEL-URL, it has to be translated to a globally routable SIP-URL before applying it as Request-URI of the outgoing INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: Need to verify that it is the Route header of the incoming INVITE request which may contain the TEL-URL, and not the Request-URI as indicated in the text of Tdoc N1-010353.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

7. **100 Trying (S-S to MO#2) – see example in Table 8.2.4-7**

   S-CSCF receives a 100 Trying provisional response, as specified by the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.4-7: 100 Trying (S-S to MO#2)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

8. **183 Session Progress (S-S to MO#2) – see example in Table 8.2.4-8**

The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response (to (6)), per the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.4-8: 183 Session Progress (S-S to MO#2)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

9. **183 Session Progress (S-CSCF to P-CSCF) – see example in Table 8.2.4-9**

S-CSCF forwards the 183 Session Progress response to P-CSCF.

Table 8.2.4-9: 183 Session Progress (S-CSCF to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
```

10. **Authorize QoS Resources**

   P-CSCF authorizes the resources necessary for this session

11. **183 Session Progress (P-CSCF to UE) – see example in Table 8.2.4-11**

   P-CSCF forwards the 183 Session Progress response to the originating endpoint.

Table 8.2.4-11: 183 Session Progress (P-CSCF to UE)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Media-Authorization: 0020000100100101706366312e78797a2e6e6574000c02013942563330373200
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token1@pcscf1.home.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf.home.net, sip:scscf2.home.net,
       sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
```

Contact: a locally unique token to identify the saved routing information.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf1.xyz.net" with credentials "9BV3072". "00" at the end of the authorixation token is required to pad to a multiple of 4 bytes.

Editor's Note: Modified text for this step is contained in Annex A.

12. **PRACK (UE to P-CSCF) – see example in Table 8.2.4-12**

UE#1 determines which media flows should be used for this session, and which codecs should be used for each of those media flows. If there was any change in media flows, or if there was more than one choice of codec for a media flow, then UE#1 must include an SDP in the PRACK message sent to UE#2.

For this example, assume UE#1 chooses AMR as the codec to use for the single audio stream.

UE includes this information in the PRACK request to P-CSCF.

Table 8.2.4-12: PRACK (UE to P-CSCF)

```
PRACK sip:token1@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

Request-URI: takes the value of the Contact header of the received 183 Session Progress response.

Via:, Contact: take the value of either the IP address of RQDN of the originating UE.

From:, To:, Call-ID: copied from the 183 Session Progress response so that they include any tag parameter.

Cseq: takes a higher value than that in the previous request.

The final selection of the media stream from the set of those supported by the terminating endpoint, given in the received 183 Session Progress response (14), is made by the originating UE and included in the SDP.

13. **PRACK (P-CSCF to S-CSCF) – see example in Table 8.2.4-13**

P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF forwards the PRACK request to S-CSCF.

Table 8.2.4-13: PRACK (P-CSCF to S-CSCF)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Request-URI: the first component of the saved Route header.

Route: saved from the previous response (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

14. **PRACK (MO#2 to S-S) – see example in Table 8.2.4-14**

S-CSCF forwards the PRACK request to the terminating endpoint, as per the S-CSCF to S-CSCF procedure.

Table 8.2.4-14: PRACK (MO#2 to S-S)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Request-URI: the first component of the Route header. This will vary according to which S-CSCF to S-CSCF flow is used.

15. **200 OK (S-S to MO#2) – see example in Table 8.2.4-15**

The destination endpoint responds to the PRACK request (14) with a 200 OK response, per the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.4-15: 200 OK (S-S to MO#2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

16. **200 OK (S-CSCF to P-CSCF) – see example in Table 8.2.4-16**

S-CSCF forwards the 200 OK response to P-CSCF.

Table 8.2.4-16: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

17. **200 OK (P-CSCF to UE) – see example in Table 8.2.4-17**

P-CSCF forwards the 200 OK response to UE.

Table 8.2.4-17: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf1.home.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

18. **Resource Reservation**

After determining the final media streams in step #11, UE initiates the reservation procedures for the resources needed for this session.

19. **COMET (UE to P-CSCF) – see example in Table 8.2.4-19**

When the resource reservation is completed, UE sends the COMET request to the terminating endpoint, via the signalling path established by the INVITE request. The request is sent first to P-CSCF.

Table 8.2.4-19: COMET (UE to P-CSCF)

```
COMET sip:token2@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

Request-URI: takes the value of the Contact header of the received 183 Session Progress response.

Via:, Contact: take the value of either the IP address or FQDN of the originating UE.

From:, To:, Call-ID: copied from the 183 Session Progress response so that they include any tag parameters.

CSeq: takes a higher value than that in the previous request.

The SDP indicates that the resource reservation was successful.

20. **COMET (P-CSCF to S-CSCF) – see example in Table 8.2.4-20**

P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF forwards the COMET request to S-CSCF.

Table 8.2.4-20: COMET (P-CSCF to S-CSCF)

```
COMET sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Request-URI: the first component of the saved Route header.

Route: saved from the 183 Session Progress response (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

21. **COMET (MO#2 to S-S) – see example in Table 8.2.4-21**

S-CSCF forwards the COMET request to the terminating endpoint, as per the S-CSCF to S-CSCF procedure.

Table 8.2.4-21: COMET (MO#2 to S-S)

```
COMET sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Request-URI: the first component of the Route header. This will vary according to which S-CSCF to S-CSCF flow is used.

22. **200 OK (S-S to MO#2) – see example in Table 8.2.4-22**

The destination endpoint responds to the COMET request (21) with a 200 OK, per the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.4-22: 200 OK (S-S to MO#2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

23. **200 OK (S-CSCF to P-CSCF) – see example in Table 8.2.4-23**

S-CSCF forwards the 200 OK response to P-CSCF.

Table 8.2.4-23: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

24. **200 OK (P-CSCF to UE) – see example in Table 8.2.4-24**

    P-CSCF forwards the 200 OK response to UE.

Table 8.2.4-24: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.home.net
Content-Length:
```

    P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

    Contact: a locally unique token to identify the saved routing information

25. **180 Ringing (S-S to MO#2) – see example in Table 8.2.4-25**

    The called UE may optionally perform alerting. If so, it signals this to the calling party by a 180 Ringing provisional response to (6). This response is sent to S-CSCF per the S-CSCF to S-CSCF procedure.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.4-25: 180 Ringing (S-S to MO#2)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
RSeq: 9022
Content-length: 0
```

26. **Service Control**

    The S-CSCF validates the service profile and performs any service control required for this subscriber.

27. **180 Ringing (S-CSCF to P-CSCF) – see example in Table 8.2.4-27**

    S-CSCF forwards the 180 Ringing response to P-CSCF.

Table 8.2.4-27: 180 Ringing (S-CSCF to P-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

28. **180 Ringing (P-CSCF to UE) – see example in Table 8.2.4-28**

   P-CSCF removes the Record-Route and Contact headers, calculates the propoer Route header to add to future requests, and saves that information without passing it to UE. A Contact header is added with a locally unique token to identify the saved routing information.

   P-CSCF forwards the 180 Ringing response to UE.

Table 8.2.4-28: 180 Ringing (P-CSCF to UE)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token4@pcscf1.home.net
RSeq:
Content-length:
```

Editor's Note: Additional QoS interactions to handle one-way media at this point (e.g. for PSTN ringback and announcements) is for further study.

29. **PRACK (UE to P-CSCF) – see example in Table 8.2.4-29**

   UE indicates to the originating subscriber that the destination is ringing. It acknowledges the 180 Ringing provisional response (27) with a PRACK request.

Table 8.2.4-29: PRACK (UE to P-CSCF)

```
PRACK sip:token4@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 9022 127 INVITE
Content-length: 0
```

   Request-URI: takes the value of the Contact header of the 180 Ringing response.

   Via:, Contact: take the value of either the IP address or FQDN of the UE.

   From:, To:, Call-ID: copied from the 180 Ringing response so that they include any revised tag parameters.

   Cseq: takes a higher value than in the previous request.

30. **PRACK (P-CSCF to S-CSCF) – see example in Table 8.2.4-30**

   P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF forwards the PRACK request to S-CSCF.

Table 8.2.4-30: PRACK (P-CSCF to S-CSCF)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack:
Content-length:
```

31. **PRACK (MO#2 to S-S) – see example in Table 8.2.4-31**

   S-CSCF forwards the PRACK request to the terminating endpoint, as per the S-CSCF to S-CSCF procedure.

Table 8.2.4-31: PRACK (MO#2 to S-S)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

32. **200 OK (S-S to MO#2) – see example in Table 8.2.4-32**

   The destination endpoint responds to the PRACK request (31) with a 200 OK response.

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.4-32: 200 OK (S-S to MO#2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

33. **200 OK (S-CSCF to P-CSCF) – see example in Table 8.2.4-33**

   S-CSCF forwards the 200 OK response to P-CSCF.

Table 8.2.4-33: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

34. **200 OK (P-CSCF to UE) – see example in Table 8.2.4-34**

P-CSCF forwards the 200 OK response to UE.

Table 8.2.4-34: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token5@pcscf1.home.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

35. **200 OK (S-S to MO#2) – see example in Table 8.2.4-35**

When the called party answers, the terminating endpoint sends a 200 OK final response to the INVITE request (6), as specified by the termination procedures and the S-CSCF to S-CSCF procedures, to S-CSCF.

Table 8.2.4-35: 200 OK (S-S to MO#2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

36. **Service Control**

S-CSCF performs whatever service control is appropriate for the completed session

Editor's Note: Modified text for this step is contained in Annex A.

37. **200 OK (S-CSCF to P-CSCF) – see example in Table 8.2.4-37**

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

Table 8.2.4-37: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

38. **200 OK (P-CSCF to UE) – see example in Table 8.2.4-38**

P-CSCF indicates the resources reserved for this session should now be committed, and forwards the 200 OK final response to the session originator. Ue can start media flow(s) for this session.

Table 8.2.4-38: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token6@pcscf1.home.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

39. **ACK (UE to P-CSCF) – see example in Table 8.2.4-39**

UE starts the media flow for this session, and responds to the 200 OK (38) with an ACK request sent to P-CSCF.

Table 8.2.4-39: ACK (UE to P-CSCF)

```
ACK sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 ACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

Cseq: is required to be the same value as Cseq is original INVITE request [3]

40. **ACK (P-CSCF to S-CSCF) – see example in Table 8.2.4-40**

P-CSCF forwards the ACK request to S-CSCF.

Table 8.2.4-40: ACK (P-CSCF to S-CSCF)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200 OK response (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

41. **ACK (MO#2 to S-S) – see example in Table 8.2.4-41**

S-CSCF forwards the ACK request to the terminating endpoint, per the S-CSCF to S-CSCF procedure.

Table 8.2.4-41: ACK (MO#2 to S-S)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Length:
```

8.2.5   (PSTN-O) PSTN origination

The MGCF in the IM 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 Service 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-S procedures.

Due to routing of sessions within the PSTN, this origination procedure will only occur in the home network of the destination subscriber. However, the destination subscriber may be roaming in a different operator's network. Further, due to cases of session forwarding and electronic surveillance, the destination of the session through the IM subsystem may actually be another PSTN termination.

8.2.5.1   PSTN originated sessions routed towards IM CN subsystem (through MGCF)

This clause presents only the case of PSTN originated sessions routed towards the IP Multimedia subsystem reaching first a MGCF.

Home Network
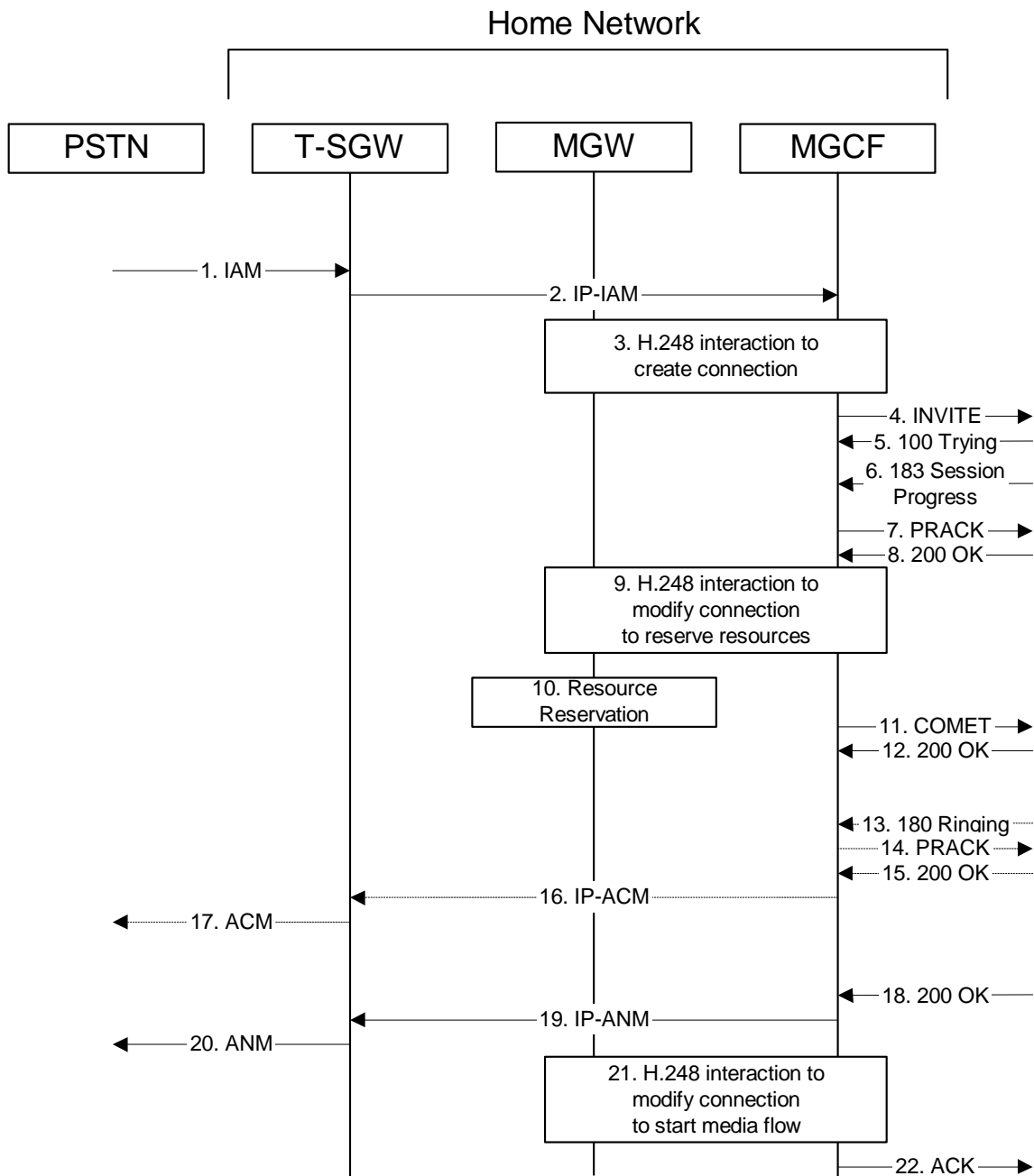
Figure 8.2.5.1 – PSTN-O

The PSTN Origination procedure is as follows:

1. **SS7 IAM**

   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. **IP-IAM**

   The T-SGW forwards the SS7 message, encapsulated in IP, to the MGCF.

3. **H.248 Interaction**

   The MGCF initiates a H.248 command, to seize the trunk and an IP port.

4. **INVITE (PSTN-O to S-S) – see example in Table 8.2.5.1-4**

The MGCF initiates an INVITE request, containing an initial SDP, as per the proper S-CSCF to S-CSCF procedure.

Table 8.2.5.1-4: INVITE (PSTN-O to S-S)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP mgcf21.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require: privacy
Anonymity: Off
From: <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:017/4@mgcf21.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

Request-URI: contains the keyed number from the user, as obtained from SS7 signalling.

Via: contains the IP address or FQDN of the originating MGCF.

Remote-Party-ID: contains the identity of the originator. The Display name is optional.

Proxy-Require: The sip privacy draft specifies that the usage of the Remote-Party-Id MUST be accompanied by a Proxy-Require header specifying "privacy" in all INVITE requests.

From:, To:, Call-ID: follow the recommendations of draft-ietf-sip-privacy-01, even though anonymity is not being requested for this session.

Cseq: a random starting number.

Contact: is the line identification and IP address or FQDN of the originating MGCF.

5. **100 Trying (S-S to PSTN-O) – see example in Table 8.2.5.1-5**

MGCF receives a 100 Trying provisional response, as specified by the S-CSCF to S-CSCF procedures.

Table 8.2.5.1-5: 100 Trying (S-S to PSTN-O)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP mgcf21.home.net
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

6. **183 Session Progress (S-S to PSTN-O) – see example in Table 8.2.5.1-6**

The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response, per the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.2.5.1-6: 183 Session Progress (S-S to PSTN-O)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP mgcf21.home.net
Record-Route: sip:pcscf2.home.net, sip:scscf2.home.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Proxy-Require:
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
```

7. **PRACK (PSTN-O to S-S) – see example in Table 8.2.5.1-7**

   MGCF decides the final set of media streams for this session, and includes this information in the PRACK request, send to the destination per the S-CSCF to S-CSCF procedures.

Table 8.2.5.1-7: PRACK (PSTN-O to S-S)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP mgcf21.home.net
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From: <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:017/4@mgcf21.home.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
```

Request-URI: takes the value of the Contact header of the received 183 Session Progress response.

Via: takes the value of either the IP address or FQDN of the originating MGCF.

From:, To:, Call-ID: copied from the 183 Session Progress response so that they include any tag parameter.

Contact: is the line identification and IP address or FQDN of the originating MGCF.

Cseq: takes a higher value than that in the previous request.

The final selection of the media stream from the set of those supported by the terminating endpoint, given in the received 183 Session Progress response (14), is made by the originating MGCF and included in the SDP.

8. **200 OK (S-S to PSTN-O) – see example in Table 8.2.5.1-8**

The destination responds to the PRACK request (7) with a 200 OK response.

Table 8.2.5.1-8: 200OK (S-S to PSTN-O)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP mgcf21.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

9. **H.248 Interaction**

MGCF initiates a H.248 command to modify the connection parameters and instruct the MGW to reserve the resources needed for the session.

10. **Reserve Resources**

MGW reserves the resources needed for the session

11. **COMET (PSTN-O to S-S) – see example in Table 8.2.5.1-11**

When the resource reservation is completed, MGCF sends the COMET request to the terminating endpoint, per the S-S procedures.

Table 8.2.5.1-11: COMET (PSTN-O to S-S)

```
COMET sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP mgcf21.home.net
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:mgcf21.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
```

Request-URI: the first component of the Route header, built from the Record-Route response of the received 183 Session Progress response.

Via: contains the IP address or FQDN of the originating MGCF.

From:, To:, Call-ID: copied from the 183 Session Progress response so that they include any tag parameters.

Contact: is the line identification and IP address or FQDN of the originating MGCF.

Cseq: takes a higher value than that in the previous request.

The SDP indicates that the resource reservation was successful.

12. **200 OK (S-S to PSTN-O) – see example in Table 8.2.5.1-12**

The destination endpoint responds to the COMET request (11) with a 200 OK response.

Table 8.2.5.1-12: 200 OK (S-S to PSTN-O)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP mgcf21.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

13. **180 Ringing (S-S to PSTN-O) – see example in Table 8.2.5.1-13**

The destination endpoint may optionally perform alerting. If so, it signals this to the calling party by a 180 Ringing provisional response. This response is sent to MGCF per the S-CSCF to S-CSCF procedure.

Table 8.2.5.1-13: 180 Ringing (S-S to PSTN-O)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP mgcf21.home.net
Record-Route: sip:scscf2.home.net
Require: 100rel
From:
To:
Call-ID:
CSeq: 127 INVITE
RSeq: 9022
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

14. **PRACK (PSTN-O to S-S) – see example in Table 8.2.5.1-14**

MGCF acknowledges the 180 Ringing provisional response (13) with a PRACK request.

Table 8.2.5.1-14: PRACK (PSTN-O to S-S)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP mgcf21.home.net
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq: 130 PRACK
Contact: sip:mgcf21.home.net
Rack: 9022 127 INVITE
Content-length: 0
```

15. **200 OK (S-S to PSTN-O) – see example in Table 8.2.5.1-15**

The destination endpoint responds to the PRACK request (14) with a 200 OK response.

Table 8.2.5.1-15: 200 OK (S-S to PSTN-O)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP mgcf21.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

16. **IP-ACM**

    If alerting is being performed, the MGCF forwards an IP-ACM message to T-SGW

17. **ACM**

    If alerting is being performed, the T-SGW forwards a SS7 ACM message

18. **200 OK (S-S to PSTN-O) – see example in Table 8.2.5.1-18**

    When the called party answers, the terminating and S-S procedures result in a 200 OK final response being sent to MGCF

Table 8.2.5.1-18: 200 OK (S-S to PSTN-O)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP mgcf21.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Length: 0
```

19. **IP-ANM**

    MGCF forwards an IP-ANM message to T-SGW

20. **ANM**

    T-SGW forwards an ANM message to the PSTN

21. **H.248 Interaction**

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

22. **ACK (PSTN-O to S-S) – see example in Table 8.2.5.1-22**

    MGCF acknowledges the 200 OK final response (18) with an ACK request.

Table 8.2.5.1-22: ACK (PSTN-O to S-S)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP mgcf21.home.net
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq: 127 ACK
Contact: sip:mgcf21.home.net
Content-length: 0
```

Cseq: is required to be the same value as Cseq is original INVITE request [3]

8.2.5.2 PSTN originated sessions routed towards CS domain (through G-MSC)

This clause presents only the case of PSTN originated calls routed towards the CS domain reaching first a G-MSC.

Editor's Note: This clause needs further study.

8.2.5.3 PSTN originated sessions routed either towards IM CN subsystem or towards CS domain

This clause presents the case where the operator has chosen to route some of its PSTN originated calls towards its CS domain and some of them towards the IM CN subsystem.

Editor's Note: This clause needs further study.

8.3 Termination procedures

8.3.1 Introduction

This clause presents the detailed application level flows to define the Procedures for session terminations.

The session termination procedures specify the signalling path between the Serving-CSCF assigned to perform the session termination service and the UE. This signalling path is determined at the time of UE registration, and remains fixed for the life of the registration. This signalling path is the reverse of the session initiation signalling path of Clause 8.2. Therefore there is a one-to-one correspondence between the origination procedures of subclause 8.2 and the termination procedures of this clause.

A UE always has a proxy (P-CSCF) associated with it. This P-CSCF is located in the same network as the UE, and performs resource authorization for the sessions to the UE. The P-CSCF is determined by the CSCF discovery process, described in Subclause 5.2.1 (Local CSCF Discovery).

As a result of the registration procedure, the P-CSCF knows the address of the UE. The assigned S-CSCF, in the home network, knows the name/address of the P-CSCF. If the network operator owning the S-CSCF wants to keep their configuration private, the S-CSCF will have chosen an Interrogating-CSCF, I-CSCF, who will perform the firewall functions and pass messages to the P-CSCF (procedure MT#1b).

Sessions destined to the PSTN are a special case of the Termination procedures. Two of the S-CSCF to S-CSCF procedures deal specifically with PSTN termination, and route the session signalling through a BGCF that allocates a MGCF. The MGCF uses H.248/MEGACO to control a Media Gateway, and communicates with the SS7 network via the T-SGW. The MGCF receives and processes SIP requests, and subsequent nodes consider the signalling as if it came from a S-CSCF.

8.3.2 (MT#1a) Mobile termination, roaming, without I-CSCF in home network providing configuration independence

This termination procedure applies to roaming subscribers when the home network operator does not desire to keep its internal configuration hidden from the visited network. The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure. During registration, the home 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.
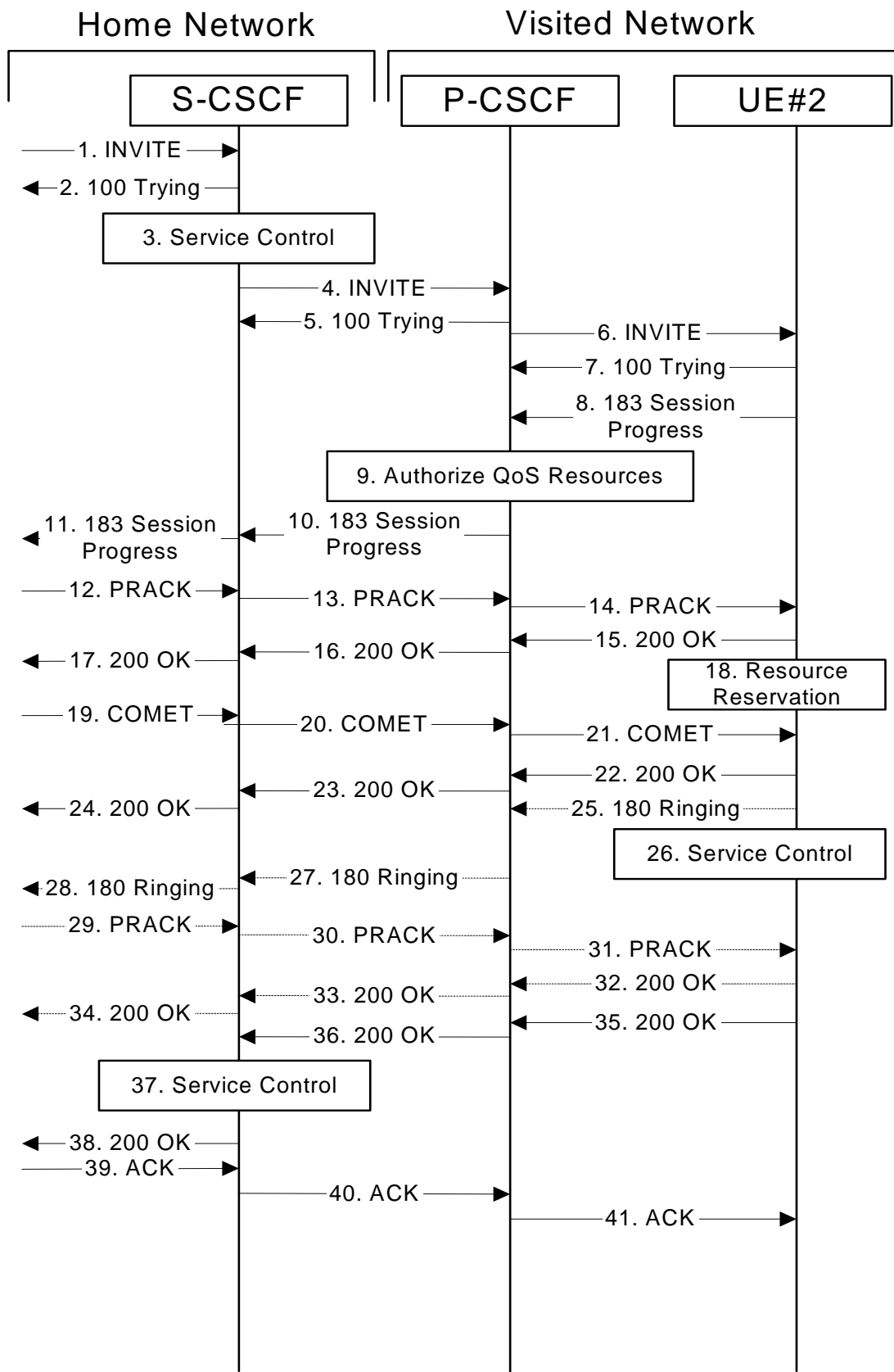
Figure 8.3.2 – MT#1a

Procedure MT#1a is as follows:

1. **INVITE (S-S to MT#1a) – see example in Table 8.3.2-1**

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

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.2-1: INVITE (S-S to MT#1a)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 0 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

SDP   The SDP contains the complete set of supported codecs from the session originator, as restricted by the originating network operator. The "m=" lines for the video media streams show a port number zero, which removes them from the negotiation.

2. **100 Trying (MT#1a to S-S) – see example in Table 8.3.2-2**

   S-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.3.2-2: 100 Trying (MT#1a to S-S)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

   S-CSCF validates the service profile, and performs any termination service control required for this subscriber

   S-CSCF examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

   For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

   Editor's Note: Modified text for this step is contained in Annex A.

4. **INVITE (S-CSCF to P-CSCF) – see example in Table 83.2-4**

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

Table 8.3.2-4: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

    Route: built from the registration information, (pcscf and UE contact name), followed by the initial Request-URI from the incoming INVITE request. The first component of the Route header, pcscf, is moved to the Request-URI of the request.

Via:, Record-Route: S-CSCF adds itself

SDP   The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the second audio stream shows a port number zero, which removes it from the negotiation.

5. **100 Trying (P-CSCF to S-CSCF) – see example in Table 8.3.2-5**

   P-CSCF responds to the INVITE request (4) with a 100 Trying provisional response.

Table 8.3.2-5: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **INVITE (P-CSCF to UE) – see example in Table 8.3.2-6**

   P-CSCF examines the media parameters, and removes any that the network operator decides, based on local policy, not to allow on the network.

   For this example, assume the network operator does not allow 64 kb/s audio, so the PCMU codec is removed.

   P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.2-6: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.visited.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf.home.net, sip:scscf0.home.net,
    sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the first audio stream no longer contains codec "0" (PCMU), which removes it from the negotiation.

Editor's Note: Modified text for this step is contained in Annex A.

7. **100 Trying (UE to P-CSCF) – see example in Table 8.3.2-7**

   UE may optionally send a 100 Trying provisional response to P-CSCF.

Table 8.3.2-7: 100 Trying (UE to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

8. **183 Session Progress (UE to P-CSCF) – see example in Table 8.3.2-8**

   UE#2 determines the complete set of codecs that it is capable of supporting for this session. It determines the intersection with those appearing in the SDP in the INVITE request. For each media flow that is not supported, UE#2 inserts a SDP entry for media (m= line) with port=0. For each media flow that is supported, UE#2 inserts a SDP entry with an assigned port and with the codecs in common with those in the SDP from UE#1.

   For this example, assume UE#2 supports both AMR and G726, but not G728 (code 15)

   UE responds with a 183 Session Progress response containing SDP back to the originator. This SDP may represent one or more media for a multi-media session. This response is sent to P-CSCF.

Table 8.3.2-8: 183 Session Progress (UE to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

Remote-Party-ID: identifies the answering subscriber. It contains the public identifier URL, and the name of the answering party.

To: A tag is added to the To header.

Contact: header identifies the IP address or FQDN of the UE.

SDP The SDP contains the subset of codecs supported by UE. It requests a confirmation of the QoS preconditions for establishing the session

9. **Authorize QoS Resources**

P-CSCF authorizes the resources necessary for this session.

10. **183 Session Progress (P-CSCF to S-CSCF) – see example in Table 8.3.2-10**

P-CSCF forwards the 183 Session Progress response to S-CSCF.

Table 8.3.2-10: 183 Session Progress (P-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.visited.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
```

Via:, Record-Route: P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

11. **183 Session Progress (MT#1a to S-S) – see example in Table 8.3.2-11**

S-CSCF forwards the 183 Session Progress response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.2-11: 183 Session Progress (MT#1a to S-S)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
```

12. **PRACK (S-S to MT#1a) – see example in Table 8.3.2-12**

The originating endpoint sends a PRACK request containing the final SDP to be used in this session, via the S-CSCF to S-CSCF procedure, to S-CSCF.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.2-12: PRACK (S-S to MT#1a)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf0.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

13. **PRACK (S-CSCF to P-CSCF) – see example in Table 8.3.2-13**

S-CSCF forwards the PRACK request to P-CSCF.

Table 8.3.2-13: PRACK (S-CSCF to P-CSCF)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

14. **PRACK (P-CSCF to UE) – see example in Table 8.3.2-14**

    P-CSCF forwards the PRACK request to UE.

Table 8.3.2-14: PRACK (P-CSCF to UE)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token2
From:
To:
Call-ID:
Cseq:
Contact: token2@pcscf.visited.net
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

15. **200 OK (UE to P-CSCF) – see example in Table 8.3.2-15**

UE acknowledges the PRACK request (14) with a 200 OK response.

Table 8.3.2-15: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.visited.net;branch=token2
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

16. **200 OK (P-CSCF to S-CSCF) – see example in Table 8.3.2-16**

P-CSCF forwards the 200 OK response to S-CSCF.

Table 8.3.2-16: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

17. **200 OK (MT#1a to S-S) – see example in Table 8.3.2-17**

S-CSCF forwards the 200 OK response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.2-17: 200 OK (MT#1a to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

18. **Resource Reservation**

UE initiates the reservation procedures for the resources needed for this session.

19. **COMET (S-S to MT#1a) – see example in Table 8.3.2-19**

When the originating endpoint has completed its resource reservation, it sends the COMET request to S-CSCF, via the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.2-19: COMET (S-S to MT#1a)

```
COMET sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf0.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

20. **OMET (S-CSCF to P-CSCF) – see example in Table 8.3.2-20**

    S-CSCF forwards the COMET request to P-CSCF.

Table 8.3.2-20: COMET (S-CSCF to P-CSCF)

```
COMET sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

21. **COMET (P-CSCF to UE) – see example in Table 8.3.2-21**

    P-CSCF forwards the COMET request to UE.

Table 8.3.2-21: COMET (P-CSCF to UE)

```
COMET sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf.visited.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

22. **200 OK (UE to P-CSCF) – see example in Table 8.3.2-22**

UE acknowledges the COMET request (21) with a 200 OK response.

Table 8.3.2-22: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.visited.net;branch=token3
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

23. **200 OK (P-CSCF to S-CSCF) – see example in Table 8.3.2-23**

P-CSCF forwards the 200 OK response to S-CSCF.

Table 8.3.2-23: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

24. **200 OK (MT#1a to S-S) – see example in Table 8.3.2-24**

S-CSCF forwards the 200 OK response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.2-24: 200 OK (MT#1a to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

25. **180 Ringing (UE to P-CSCF) – see example in Table 8.3.2-25**

Before proceeding with session establishment, the UE waits for two events. First, the resource reservation initiated in step #18 must complete successfully. Second, the resource reservation initiated by the originating endpoint must complete successfully (which is indicated by message #21 received by UE). The UE may now immediately accept the session (and proceed with step #35), or alert the destination subscriber of an incoming session attempt; if the latter it indicates this to the calling party by a 180 Ringing provisional response sent to P-CSCF.

Table 8.3.2-25: 180 Ringing (UE to P-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9022
Content-length: 0
```

26. **180 Ringing (P-CSCF to S-CSCF) – see example in Table 8.3.2-26**

P-CSCF forwards the 180 Ringing response to S-CSCF.

Table 8.3.2-26: 180 Ringing (P-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

27. **Service Control**

The S-CSCF validates the service profile and performs any service control required for this subscriber.

28. **180 Ringing (MT#1a to S-S) – see example in Table 8.3.2-28**

S-CSCF forwards the 180 Ringing response to the originating endpoint, per the S-CSCF to S-CSCF procedure.

Table 8.3.2-28: 180 Ringing (MT#1a to S-S)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

29. **PRACK (S-S to MT#1a) – see example in Table 8.3.2-29**

The originator acknowledges the 180 Ringing response (28) with a PRACK request.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.2-29: PRACK (S-S to MT#1a)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf0.home.net
From:
To:
Call-ID:
Cseq: 130 PRACK
Contact: sip%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Rack: 9022 127 INVITE
Content-length: 0
```

30. **PRACK (S-CSCF to P-CSCF) – see example in Table 8.3.2-30**

S-CSCF forwards the PRACK request to P-CSCF.

Table 8.3.2-30: PRACK (S-CSCF to P-CSCF)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

31. **PRACK (P-CSCF to UE) – see example in Table 8.3.2-31**

P-CSCF forwards the PRACK request to UE.

Table 8.3.2-31: PRACK (P-CSCF to UE)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token4
From:
To:
Call-ID:
Cseq:
Contact: token4@pcscf.visited.net
Rack:
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

32. **200 OK (UE to P-CSCF) – see example in Table 8.3.2-32**

UE acknowledges the PRACK request (31) with a 200 OK response.

Table 8.3.2-32: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.visited.net;branch=token4
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

33. **200 OK (P-CSCF to S-CSCF) – see example in Table 8.3.2-33**

P-CSCF forwards the 200 OK response to S-CSCF.

Table 8.3.2-33: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

34. **200 OK (MT#1a to S-S) – see example in Table 8.3.2-34**

S-CSCF forwards the 200 OK response to the session originator, per the S-CSCF to S-CSCF procedures.

Table 8.3.2-34: 200 OK (MT#1a to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

35. **200 OK (UE to P-CSCF) – see example in Table 8.3.2-35**

When the called party answers the UE sends a 200 OK final response to the INVITE request (6) to P-CSCF, and starts the media flow(s) for this session.

Table 8.3.2-35: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
From:
To:
Call-ID:
CSeq: 127 INVITE
Content-length: 0
```

36. **200 OK (P-CSCF to S-CSCF) – see example in Table 8.3.2-36**

P-CSCF indicates the resources reserved for this session should now be committed, and sends the 200 OK final response to S-CSCF.

Table 8.3.2-36: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

37. **Service Control**

S-CSCF performs whatever service control is required for the session completion

38. **200 OK (MT#1a to S-S) – see example in Table 8.3.2-38**

S-CSCF forwards the 200 OK final response along the signalling path back to the session originator, as per the S-CSCF to S-CSCF procedure.

Table 8.3.2-38: 200 OK (MT#1a to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

39. **ACK (S-S to MT#1a) – see example in Table 8.3.2-39**

The calling party responds to the 200 OK final response (38) with an ACK request which is sent to S-CSCF via the S-CSCF to S-CSCF procedure.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.2-39: ACK (S-S to MT#1a)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf0.home.net
From:
To:
Call-ID:
Cseq: 127 ACK
Contact: sip%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-length: 0
```

40. **ACK (S-CSCF to P-CSCF) – see example in Table 8.3.2-40**

     S-CSCF forwards the ACK request to P-CSCF.

Table 8.3.2-40: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

41. **ACK (P-CSCF to UE) – see example in Table 8.3.2-41**

     P-CSCF forwards the ACK request to UE.

Table 8.3.2-41: ACK (P-CSCF to UE)

```
ACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token5
From:
To:
Call-ID:
Cseq:
Contact: token5@pcscf.visited.net
Content-length:
```

     P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

8.3.3   (MT#1b) Mobile termination, roaming, with I-CSCF in home network providing configuration independence

This termination procedure applies to roaming subscribers when the home network operator desires to keep its internal configuration hidden from the visited network. The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure. During registration, the home network allocates a S-CSCF. The home network advertises an I-CSCF as the entry point from the visited network, who protects the S-CSCF identity and forwards requests to the P-CSCF.

When registration is complete, S-CSCF knows the name/address of its next hop in the signalling path toward the UE, the I-CSCF. I-CSCF receives information in the request, which it translates and obtains the name/address of P-CSCF, and P-CSCF knows the name/address of the UE.
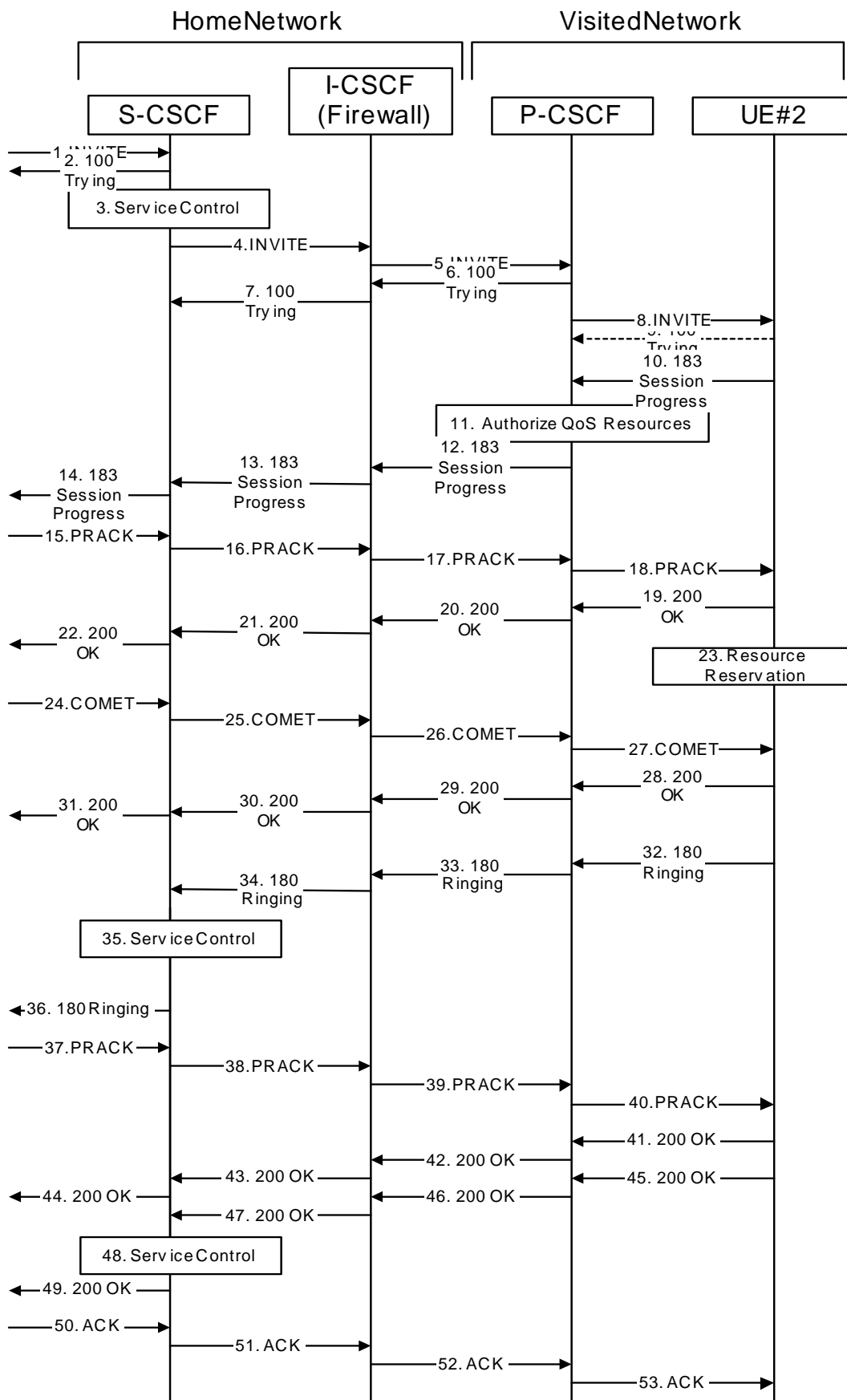
Figure 8.3.3 – MT#1b

Procedure MT#1b is as follows:

1. **INVITE (S-S to MT#1b) – see example in Table 8.3.3-1**

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

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.3-1: INVITE (S-S to MT#1b)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

   SDP The SDP contains the complete set of supported codecs from the session originator, as restricted by the originating network operator. The "m=" lines for the video media streams show a port number zero, which removes them from the negotiation.

2. **100 Trying (MT#1b to S-S) – see example in Table 8.3.3-2**

   S-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.3.3-2: 100 Trying (MT#1b to S-S)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

   S-CSCF validates the service profile, and performs any termination service control required for this subscriber

   S-CSCF examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

Editor's Note: Modified text for this step is contained in Annex A.

4. **INVITE (S-CSCF to I-CSCF) – see example in Table 8.3.3-4**

S-CSCF remembers (from the registration procedure) the next hop CSCF for this UE. It forwards the INVITE to the I-CSCF to perform the firewall functions.

Table 8.3.3-4: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net, sip:+1-212-555-
    2222@home.net;user=phone
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

Route: built from the registration information, icscf, pcscf (and UE contact name), followed by the initial Request-URI from the incoming INVITE request. The first component of the Route header, icscf, is moved to the Request-URI of the request.

Via:, Record-Route: S-CSCF adds itself in the Record-Route and Via headers.

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the second audio stream shows a port number zero, which removes it from the negotiation.

5. **INVITE (I-CSCF to P-CSCF) – see example in Table 8.3.3-5**

I-CSCF translates the Via headers in the request, and forwards the INVITE request to P-CSCF.

Table 8.3.3-5: INVITE (I-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
m=
a=
a=
a=
a=
m=
```

Via: translated to preserve configuration independence of the home network.

Record-Route: translated to preserve configuration independence of the home network.

6. **100 Trying (P-CSCF to I-CSCF) – see example in Table 8.3.3-6**

   P-CSCF responds to the INVITE request (5) with a 100 Trying provisional response.

Table 8.3.3-6: 100 Trying (P-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

7. **100 Trying (I-CSCF to S-CSCF) – see example in Table 8.3.3-7**

   I-CSCF determines the Via header, and forwards the 100 Trying provisional response to S-CSCF.

Table 8.3.3-7: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

8. **INVITE (P-CSCF to UE) – see example in Table 8.3.3-8**

P-CSCF examines the media parameters, and removes any that the network operator decides, based on local policy, not to allow on the network.

For this example, assume the network operator does not allow 64 kb/s audio, so the PCMU codec is removed.

P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.3-8: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.visited.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:icscf.home.net,
    sip:Token(sip:scscf.home.net, sip:scscf0.home.net),
    sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

SDP   The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the first audio stream no longer contains codec "0" (PCMU), which removes it from the negotiation.

Editor's Note: Modified text for this step is contained in Annex A.

9. **100 Trying (UE to P-CSCF) – see example in Table 8.3.3-9**

UE may optionally send a 100 Trying provisional response to P-CSCF.

Table 8.3.3-9: 100 Trying (UE to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

10. **183 Session Progress (UE to P-CSCF) – see example in Table 8.3.3-10**

UE#2 determines the complete set of codecs that it is capable of supporting for this session. It determines the intersection with those appearing in the SDP in the INVITE request. For each media flow that is not supported, UE#2 inserts a SDP entry for media (m= line) with port=0. For each media flow that is supported, UE#2 inserts a SDP entry with an assigned port and with the codecs in common with those in the SDP from UE#1.

For this example, assume UE#2 supports both AMR and G726, but not G728 (code 15)

UE responds with a 183 Session Progress response containing SDP back to the originator. This SDP may represent one or more media for a multi-media session.This response is sent to P-CSCF.

Table 8.3.3-10: 183 Session Progress (UE to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

Remote-Party-ID: identifies the answering subscriber. It contains the public identifier URL, and the name of the answering party.

To: A tag is added to the To header.

Contact: identifies the IP address or FQDN of the UE.

SDP   The SDP contains the subset of codecs supported by UE. It requests a confirmation of the QoS preconditions for establishing the session

11. **Authorize QoS Resources**

P-CSCF authorizes the resources necessary for this session.

12. **183 Session Progress (P-CSCF to I-CSCF) – see example in Table 8.3.3-12**

P-CSCF forwards the 183 Session Progress response to I-CSCF.

Table 8.3.3-12: 183 Session Progress (P-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaaa%3abbbb%5d@pcscf.visited.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

13. **183-Session-Progress (I-CSCF to S-CSCF) – see example in Table 8.3.3-13**

    I-CSCF determines the Via and Record-Route headers, and forwards the response to S-CSCF.

Table 8.3.3-13: 183 Session Progress (I-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf.home.net, sip:scscf.home.net, sip:scscf0.home.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
```

14. **183 Session Progress (MT#1b to S-S) – see example in Table 8.3.3-14**

S-CSCF forwards the 183 Session Progress response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.3-14: 183 Session Progress (MT#1b to S-S)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
```

15. **PRACK (S-S to MT#1b) – see example in Table 8.3.3-15**

The originating endpoint sends a PRACK request containing the final SDP to be used in this session, via the S-CSCF to S-CSCF procedure, to S-CSCF.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.3-15: PRACK (S-S to MT#1b)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.visited1.net, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf0.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

16. **PRACK (S-CSCF to I-CSCF) – see example in Table 8.3.3-16**

S-CSCF forwards the PRACK request to I-CSCF.

Table 8.3.3-16: PRACK (S-CSCF to I-CSCF)

```
PRACK sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
   SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

17. **PRACK (I-CSCF to P-CSCF) – see example in Table 8.3.3-17**

I-CSCF translates the Via headers in the PRACK request, and forwards the request to P-CSCF.

Table 8.3.3-17: PRACK (I-CSCF to P-CSCF)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Via: translated to preserve configuration independence of the home network.

Record-Route: translated to preserve configuration independence of the home network.

18. **PRACK (P-CSCF to UE) – see example in Table 8.3.3-18**

P-CSCF forwards the PRACK request to UE.

Table 8.3.3-18: PRACK (P-CSCF to UE)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token2
From:
To:
Call-ID:
Cseq:
Contact: token2@pcscf.visited.net
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

19. **200 OK (UE to P-CSCF) – see example in Table 8.3.3-19**

UE acknowledges the PRACK request (18) with a 200 OK response.

Table 8.3.3-19: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.visited.net;branch=token2
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

20. **200 OK (P-CSCF to I-CSCF) – see example in Table 8.3.3-20**

P-CSCF forwards the 200 OK response to I-CSCF.

Table 8.3.3-20: 200 OK (P-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
From:
To:
Call-ID:
CSeq:
Content-length:
```

Via:, Record-Route: P-CSCF restores the Via and Record-Route headers from saved values, based on the token value in the branch parameter of its Via.

21. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.3.3-21**

I-CSCF determines the Via and Record-Route headers, and forwards the 200 OK response to S-CSCF.

Table 8.3.3-21: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf.home.net, sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

22. **200 OK (MT#1b to S-S) – see example in Table 8.3.3-22**

S-CSCF forwards the 200 OK response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.3-22: 200 OK (MT#1b to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

23. **Resource Reservation**

UE initiates the reservation procedures for the resources needed for this session.

24. **COMET (S-S to MT#1b) – see example in Table 8.3.3-24**

When the originating endpoint has completed its resource reservation, it sends the COMET request to S-CSCF, via the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.3-24: COMET (S-S to MT#1b)

```
COMET sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf0.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

25. **COMET (S-CSCF to I-CSCF) – see example in Table 8.3.3-25**

S-CSCF forwards the COMET request to I-CSCF.

Table 8.3.3-25: COMET (S-CSCF to I-CSCF)

```
COMET sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

26. **COMET (I-CSCF to P-CSCF) – see example in Table 8.3.3-26**

   I-CSCF translates the Via headers in the COMET request, and forwards the request to P-CSCF.

Table 8.3.3-26: COMET (I-CSCF to P-CSCF)

```
COMET sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

   Via: translated to preserve configuration independence of the home network.

   Record-Route: translated to preserve configuration independence of the home network.

27. **COMET (P-CSCF to UE) – see example in Table 8.3.3-27**

   P-CSCF forwards the COMET request to UE.

Table 8.3.3-27: COMET (P-CSCF to UE)

```
COMET sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf.visited.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

28. **200 OK (UE to P-CSCF) – see example in Table 8.3.3-28**

UE acknowledges the COMET request (27) with a 200 OK response.

Table 8.3.3-28: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.visited.net;branch=token3
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

29. **200 OK (P-CSCF to I-CSCF) – see example in Table 8.3.3-29**

P-CSCF forwards the 200 OK response to I-CSCF.

Table 8.3.3-29: 200 OK (P-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF restores the Via and Record-Route headers from saved values, based on the token value in the branch parameter of its Via.

30. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.3.3-30**

I-CSCF determines the Via and Record-Route headers, and forwards the 200 OK to S-CSCF

Table 8.3.3-30: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf.home.net, sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

31. **200 OK (MT#1b to S-S) – see example in Table 8.3.3-31**

    S-CSCF forwards the 200 OK response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.3-31: 200 OK (MT#1b to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

32. **180 Ringing (UE to P-CSCF) – see example in Table 8.3.3-32**

    Before proceeding with session establishment, the UE waits for two events. First, the resource reservation initiated in step #23 must complete successfully. Second, the resource reservation initiated by the originating endpoint must complete successfully (which is indicated by message #27 received by UE). The UE may now immediately accept the session (and proceed with step #45), or alert the destination subscriber of an incoming session attempt; if the latter it indicates this to the calling party by a 180 Ringing provisional response sent to P-CSCF.

Table 8.3.3-32: 180 Ringing (UE to P-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9022
Content-length: 0
```

33. **180 Ringing (P-CSCF to I-CSCF) – see example in Table 8.3.3-33**

    P-CSCF forwards the 180 Ringing response to I-CSCF.

Table 8.3.3-33: 180 Ringing (P-CSCF to I-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.visited.net
RSeq:
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

34. 1**80 Ringing (I-CSCF to S-CSCF) – see example in Table 8.3.3-34**

I-CSCF determines the Via and Record-Route headers, and forwards the 180 Ringing response to S-CSCF.

Table 8.3.3-34: 180 Ringing (I-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf.home.net, sip:scscf.home.net, sip:scscf0.home.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

35. **Service Control**

The S-CSCF validates the service profile and performs any service control required for this subscriber.

36. **180 Ringing (MT#1b to S-S) – see example in Table 8.3.3-36**

S-CSCF forwards the 180 Ringing response to the originating endpoint, per the S-CSCF to S-CSCF procedure.

Table 8.3.3-36: 180 Ringing (MT#1b to S-S)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

37. **PRACK (S-S to MT#1b) – see example in Table 8.3.3-37**

The originator acknowledges the 180 Ringing response (36) with a PRACK request.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.3-37: PRACK (S-S to MT#1b)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf0.home.net
From:
To:
Call-ID:
Cseq: 130 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Rack: 9022 127 INVITE
Content-length: 0
```

38. **PRACK (S-CSCF to I-CSCF) – see example in Table 8.3.3-38**

    S-CSCF forwards the PRACK request to I-CSCF.

Table 8.3.3-38: PRACK (S-CSCF to I-CSCF)

```
PRACK sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

39. **PRACK (I-CSCF to P-CSCF) – see example in Table 8.3.3-39**

    I-CSCF translates the Via headers in the PRACK request, and forwards the request to P-CSCF.

Table 8.3.3-39: PRACK (I-CSCF to P-CSCF)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

Via: translated to preserve configuration independence of the home network.

Record-Route: translated to preserve configuration independence of the home network.

40. **PRACK (P-CSCF to UE) – see example in Table 8.3.3-40**

    P-CSCF forwards the PRACK request to UE.

Table 8.3.3-40: PRACK (P-CSCF to UE)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token4
From:
To:
Call-ID:
Cseq:
Contact: token4@pcscf.visited.net
Rack:
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

41. **200 OK (UE to P-CSCF) – see example in Table 8.3.3-41**

UE acknowledges the PRACK request (40) with a 200 OK response.

Table 8.3.3-41: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.visited.net;branch=token4
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

42. **200 OK (P-CSCF to I-CSCF) – see example in Table 8.3.3-42**

P-CSCF forwards the 200 OK to I-CSCF.

Table 8.3.3-42: 200 OK (P-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
   scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF restores the Via and Record-Route headers from saved values, based on the token value in the branch parameter of its Via.

43. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.3.3-43**

I-CSCF determines the Via and Record-Route headers, and forwards the 200 OK response to S-CSCF.

Table 8.3.3-43: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf.home.net, sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

44. **200 OK (MT#1b to S-S) – see example in Table 8.3.3-44**

S-CSCF forwards the 200 OK to the session originator, per the S-CSCF to S-CSCF procedures.

Table 8.3.3-44: 200 OK (MT#1b to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

45. **200 OK (UE to P-CSCF) – see example in Table 8.3.3-45**

When the called party answers, the UE sends a 200 OK final response to the INVITE request (8) to P-CSCF, and starts the media flow(s) for this session.

Table 8.3.3-45: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
From:
To:
Call-ID:
CSeq: 127 INVITE
Content-length: 0
```

46. **200 OK (P-CSCF to I-CSCF) – see example in Table 8.3.3-46**

P-CSCF indicates the resources reserved for this session should now be committed, and sends the 200 OK final response to I-CSCF.

Table 8.3.3-46: 200 OK (P-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF restores the Via and Record-Route headers from saved values, based on the token value in the branch parameter of its Via.

47. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.3.3-47**

I-CSCF determines the Via and Record-Route headers, and forwards the 200 OK response to S-CSCF.

Table 8.3.3-47: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf.home.net, sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

48. **Service Control**

S-CSCF performs whatever service control is required for the session completion

49. **200 OK (MT#1b to S-S) – see example in Table 8.3.3-49**

S-CSCF forwards the 200 OK final response along the signalling path back to the session originator, as per the S-CSCF to S-CSCF procedure.

Table 8.3.3-49: 200 OK (MT#1b to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

50. **ACK (S-S to MT#1b) – see example in Table 8.3.3-50**

The calling party responds to the 200 OK final response (49) with an ACK request which is sent to S-CSCF via the S-CSCF to S-CSCF procedure.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.3-50: ACK (S-S to MT#1b)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf0.home.net
From:
To:
Call-ID:
Cseq: 127 ACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-length: 0
```

51. **ACK (S-CSCF to I-CSCF) – see example in Table 8.3.3-51**

S-CSCF forwards the ACK request to I-CSCF.

Table 8.3.3-51: ACK (S-CSCF to I-CSCF)

```
ACK sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

52. **ACK (I-CSCF to P-CSCF) – see example in Table 8.3.3-52**

I-CSCF forwards the ACK request to P-CSCF.

Table 8.3.3-52: ACK (I-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

Via: translated to preserve configuration independence of the home network.

Record-Route: translated to preserve configuration independence of the home network.

53. **ACK (P-CSCF to UE) – see example in Table 8.3.3-53**

P-CSCF forwards the ACK request to UE.

Table 8.3.3-53: ACK (P-CSCF to UE)

```
ACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token5
From:
To:
Call-ID:
Cseq:
Contact: token5@pcscf.visited.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

8.3.4 (MT#2) 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 P-CSCF via the CSCF discovery procedure. During registration, the home network allocates a S-CSCF in the home network, 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.
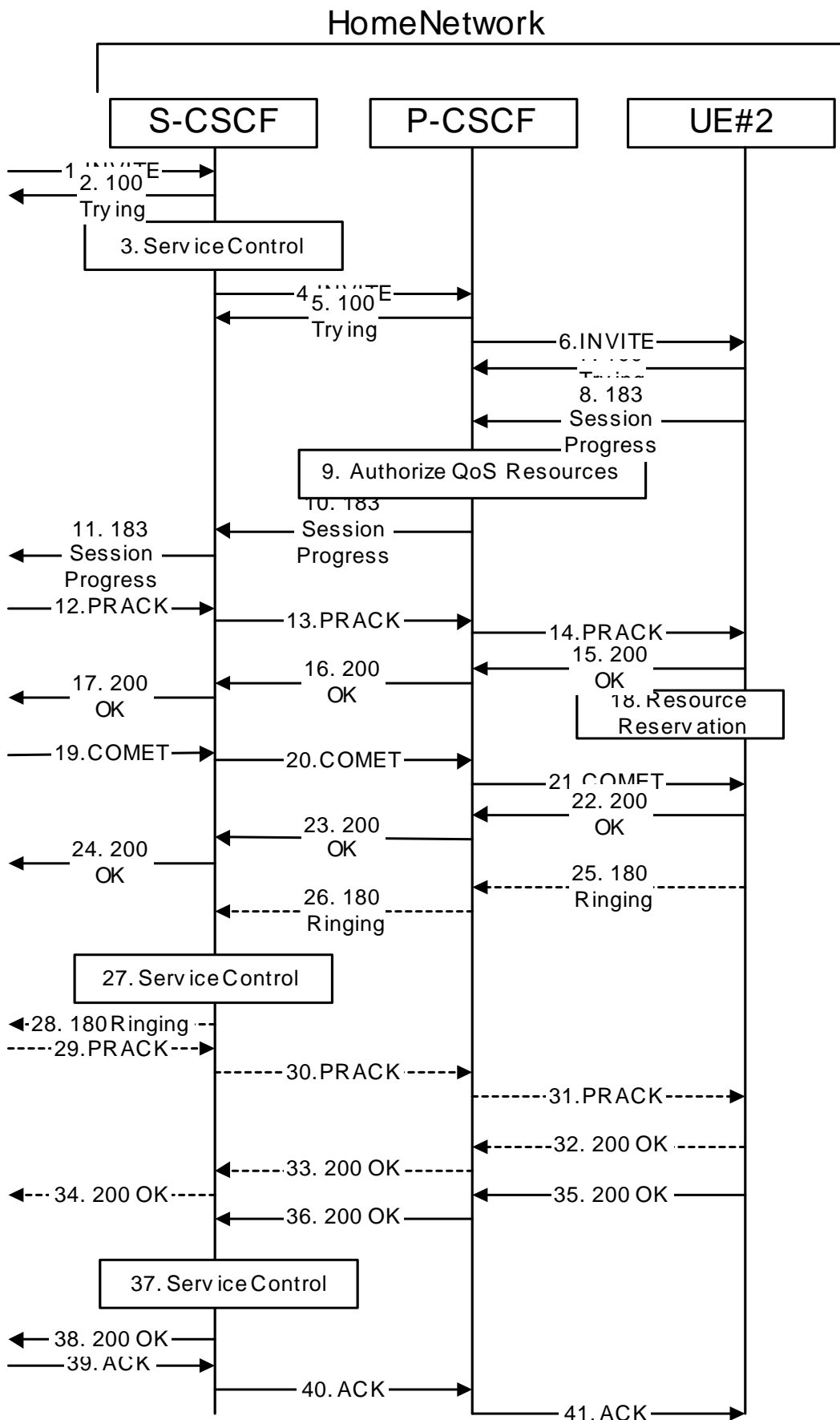
Figure 8.2.3 – MT#2

Procedure MT#2 is as follows:

1. **INVITE (S-S to MT#2) – see example in Table 8.3.4-1**

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

    NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.4-1: INVITE (S-S to MT#2)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

   SDP The SDP contains the complete set of supported codecs from the session originator, as restricted by the originating network operator. The "m=" lines for the video media streams show a port number zero, which removes them from the negotiation.

2. **100 Trying (MT#2 to S-S) – see example in Table 8.3.4-2**

    S-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.3.4-2: 100 Trying (MT#2 to S-S)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

   S-CSCF validates the service profile, and performs any termination service control required for this subscriber

   S-CSCF examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

   For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

   Editor's Note: Modified text for this step is contained in Annex A.

4. **INVITE (S-CSCF to P-CSCF) – see example in Table 8.3.4-4**

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

Table 8.3.4-4: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

Route: built from the registration information, (pcscf and UE contact name), followed by the initial Request-URI from the incoming INVITE request. The first component of the Route header, pcscf, is moved to the Request-URI of the request.

Via:, Record-Route: S-CSCF adds itself in the Record-Route and Via headers.

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the second audio stream shows a port number zero, which removes it from the negotiation.

5. **100 Trying (P-CSCF to S-CSCF) – see example in Table 8.3.4-5**

P-CSCF responds to the INVITE request (4) with a 100 Trying provisional response.

Table 8.3.4-5: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **INVITE (P-CSCF to UE) – see example in Table 8.3.4-6**

   P-CSCF examines the media parameters, and removes any that the network operator decides, based on local policy, not to allow on the network.

   For this example, assume the network operator does not allow 64 kb/s audio, so the PCMU codec is removed.

   P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.4-6: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.home.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf.home.net, sip:scscf0.home.net,
    sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

> **SDP** The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the first audio stream no longer contains codec "0" (PCMU), which removes it from the negotiation.

Editor's Note: Modified text for this step is contained in Annex A.

7. **100 Trying (UE to P-CSCF) – see example in Table 8.3.4-7**

UE may optionally send a 100 Trying provisional response to P-CSCF.

Table 8.3.4-7: 100 Trying (UE to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

8. **183 Session Progress (UE to P-CSCF) – see example in Table 8.3.4-8**

UE#2 determines the complete set of codecs that it is capable of supporting for this session. It determines the intersection with those appearing in the SDP in the INVITE request. For each media flow that is not supported, UE#2 inserts a SDP entry for media (m= line) with port=0. For each media flow that is supported, UE#2 inserts a SDP entry with an assigned port and with the codecs in common with those in the SDP from UE#1.

For this example, assume UE#2 supports both AMR and G726, but not G728 (code 15)

UE responds with a 183 Session Progress response containing SDP back to the originator. This SDP may represent one or more media for a multi-media session. This response is sent to P-CSCF.

Table 8.3.4-8: 183 Session Progress (UE to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

> **Remote-Party-ID:** identifies the answering subscriber. It contains the public identifier URL, and the name of the answering party.

> **To:** A tag is added to the To header.

Contact: identifies the IP address or FQDN of the UE.

SDP The SDP contains the subset of codecs supported by UE. It requests a confirmation of the QoS preconditions for establishing the session

9. **Authorize QoS Resources**

   P-CSCF authorizes the resources necessary for this session.

10. **183 Session Progress (P-CSCF to S-CSCF) – see example in Table 8.3.4-10**

   P-CSCF forwards the 183 Session Progress response to S-CSCF.

Table 8.3.4-10: 183 Session Progress (P-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.home.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
```

   P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

   Contact: a locally defined value that identifies the UE.

11. **183 Session Progress (MT#2 to S-S) – see example in Table 8.3.4-11**

   S-CSCF forwards the 183 Session Progress response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.4-11: 183 Session Progress (MT#2 to S-S)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
```

12. **PRACK (S-S to MT#2) – see example in Table 8.3.4-12**

   The originating endpoint sends a PRACK request containing the final SDP to be used in this session, via the S-CSCF to S-CSCF procedure, to S-CSCF.

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.4-12: PRACK (S-S to MT#2)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net
Record-Route: sip:scscf.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

13. **PRACK (S-CSCF to P-CSCF) – see example in Table 8.3.4-13**

S-CSCF forwards the PRACK request to P-CSCF.

Table 8.3.4-13: PRACK (S-CSCF to P-CSCF)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

14. **PRACK (P-CSCF to UE) – see example in Table 8.3.4-14**

P-CSCF forwards the PRACK request to UE.

Table 8.3.4-14: PRACK (P-CSCF to UE)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net;branch=token2
From:
To:
Call-ID:
Cseq:
Contact: token2@pcscf.home.net
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

15. **200 OK (UE to P-CSCF) – see example in Table 8.3.4-15**

UE acknowledges the PRACK request (14) with a 200 OK response.

Table 8.3.4-15: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home.net;branch=token2
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

16. **200 OK (P-CSCF to S-CSCF) – see example in Table 8.3.4-16**

P-CSCF forwards the 200 OK response to S-CSCF.

Table 8.3.4-16: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

17. **200 OK (MT#2 to S-S) – see example in Table 8.3.4-17**

S-CSCF forwards the 200 OK response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.4-17: 200 OK (MT#2 to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

18. **Resource Reservation**

UE initiates the reservation procedures for the resources needed for this session.

19. **COMET (S-S to MT#2) – see example in Table 8.3.4-19**

When the originating endpoint has completed its resource reservation, it sends the COMET request to S-CSCF, via the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.4-19: COMET (S-S to MT#2)

```
COMET sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net
Record-Route: sip:scscf0.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m-video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

20. **COMET (S-CSCF to P-CSCF) – see example in Table 8.3.4-20**

    S-CSCF forwards the COMET request to P-CSCF.

Table 8.3.4-20: COMET (S-CSCF to P-CSCF)

```
COMET sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

21. **COMET (P-CSCF to UE) – see example in Table 8.3.4-21**

    P-CSCF forwards the COMET request to UE.

Table 8.3.4-21: COMET (P-CSCF to UE)

```
COMET sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

22. **200 OK (UE to P-CSCF) – see example in Table 8.3.4-22**

UE acknowledges the COMET request (21) with a 200 OK response.

Table 8.3.4-22: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

23. **200 OK (P-CSCF to S-CSCF) – see example in Table 8.3.4-23**

P-CSCF forwards the 200 OK response to S-CSCF.

Table 8.3.4-23: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

24. **200 OK (MT#2 to S-S) – see example in Table 8.3.4-24**

S-CSCF forwards the 200 OK response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.4-24: 200 OK (MT#2 to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

25. **180 Ringing (UE to P-CSCF) – see example in Table 8.3.4-25**

Before proceeding with session establishment, the UE waits for two events. First, the resource reservation initiated in step #18 must complete successfully. Second, the resource reservation initiated by the originating endpoint must complete successfully (which is indicated by message #21 received by UE). The UE may now immediately accept the session (and proceed with step #35), or alert the destination subscriber of an incoming session attempt; if the latter it indicates this to the calling party by a 180 Ringing provisional response sent to P-CSCF.

Table 8.3.4-25: 180 Ringing (UE to P-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9022
Content-length: 0
```

26. **180 Ringing (P-CSCF to S-CSCF) – see example in Table 8.3.4-26**

P-CSCF forwards the 180 Ringing response to S-CSCF.

Table 8.3.4-26: 180 Ringing (P-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

27. **Service Control**

The S-CSCF validates the service profile and performs any service control required for this subscriber.

28. **180 Ringing (MT#2 to S-S) – see example in Table 8.3.4-28**

S-CSCF forwards the 180 Ringing response to the originating endpoint, per the S-CSCF to S-CSCF procedure.

Table 8.3.4-28: 180 Ringing (MT#2 to S-S)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

29. **PRACK (S-S to MT#2) – see example in Table 8.3.4-29**

The originator acknowledges the 180 Ringing response (28) with a PRACK request.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.4-29: PRACK (S-S to MT#2)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net
Record-Route: sip:scscf0.home.net
From:
To:
Call-ID:
Cseq: 130 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Rack: 9022 127 INVITE
Content-length: 0
```

30. **PRACK (S-CSCF to P-CSCF) – see example in Table 8.3.4-30**

S-CSCF forwards the PRACK request to P-CSCF.

Table 8.3.4-30: PRACK (S-CSCF to P-CSCF)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

31. **PRACK (P-CSCF to UE) – see example in Table 8.3.4-31**

P-CSCF forwards the PRACK request to UE.

Table 8.3.4-31: PRACK (P-CSCF to UE)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net;branch=token4
From:
To:
Call-ID:
Cseq:
Contact: token4@pcscf.home.net
Rack:
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

32. **200 OK (UE to P-CSCF) – see example in Table 8.3.4-32**

UE acknowledges the PRACK request (31) with a 200 OK response.

Table 8.3.4-32: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home.net;branch=token4
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

33. **200 OK (P-CSCF to S-CSCF) – see example in Table 8.3.4-33**

P-CSCF forwards the 200 OK response to S-CSCF.

Table 8.3.4-33: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

34. **200 OK (MT#2 to S-S) – see example in Table 8.3.4-34**

S-CSCF forwards the 200 OK response to the session originator, per the S-CSCF to S-CSCF procedures.

Table 8.3.4-34: 200 OK (MT#2 to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

35. **200 OK (UE to P-CSCF) – see example in Table 8.3.4-35**

When the called party answers, the UE sends a 200 OK final response to the INVITE request (6) to P-CSCF, and starts the media flow(s) for this session.

Table 8.3.4-35: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
From:
To:
Call-ID:
CSeq: 127 INVITE
Content-length: 0
```

36. **200 OK (P-CSCF to S-CSCF) – see example in Table 8.3.4-36**

   P-CSCF indicates the resources reserved for this session should now be committed, and sends the 200 OK final response to S-CSCF.

Table 8.3.4-36: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

37. **Service Control**

   S-CSCF performs whatever service control is required for the session completion

38. **200 OK (MT#2 to S-S) – see example in Table 8.3.4-38**

   S-CSCF forwards the 200 OK final response along the signalling path back to the session originator, as per the S-CSCF to S-CSCF procedure.

Table 8.3.4-38: 200 OK (MT#2 to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

39. **ACK (S-S to MT#2) – see example in Table 8.3.4-39**

   The calling party responds to the 200 OK final response (38) with an ACK request which is sent to S-CSCF via the S-CSCF to S-CSCF procedure.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.4-39: ACK (S-S to MT#2)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net
Record-Route: sip:scscf0.home.net
From:
To:
Call-ID:
Cseq: 127 ACK
Contact: sip%5b5555%3a%3aaaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-length: 0
```

40. **ACK (S-CSCF to P-CSCF) – see example in Table 8.3.4-40**

   S-CSCF forwards the ACK request to P-CSCF.

Table 8.3.4-40: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

41. **ACK (P-CSCF to UE) – see example in Table 8.3.4-41**

   P-CSCF forwards the ACK request to UE.

Table 8.3.4-41: ACK (P-CSCF to UE)

```
ACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net;branch=token5
From:
To:
Call-ID:
Cseq:
Contact: token5@pcscf.home.net
Content-length:
```

   P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

   Contact: a locally unique token to identify the saved routing information.

   Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

8.3.5 (PSTN-T) PSTN termination

The MGCF in the IM 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.

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

This termination procedure can be used in either S-S#3 or S-S#4.
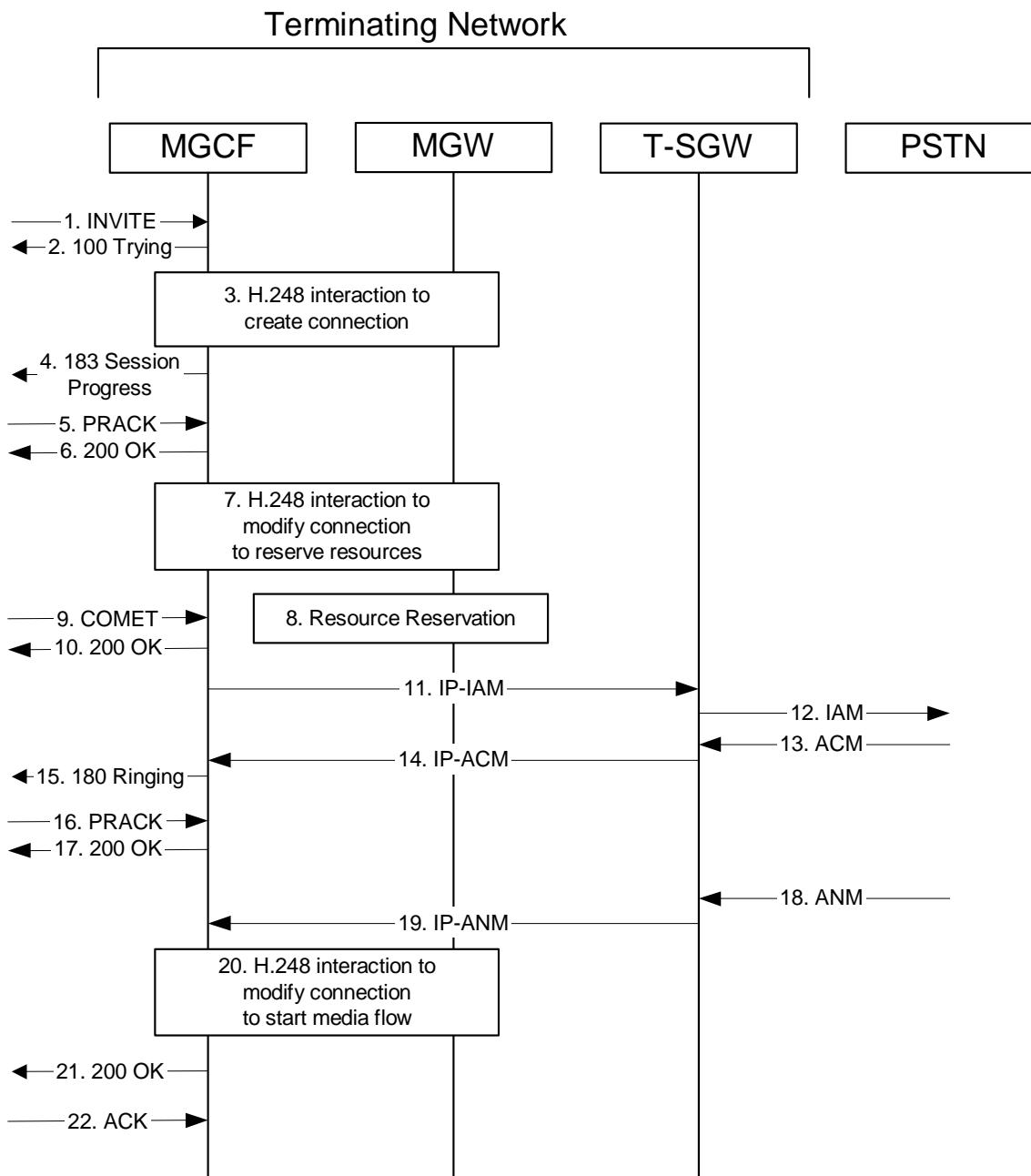
Figure 8.3.5 – PSTN-T

The PSTN termination procedure is as follows:

1. **INVITE (S-S to PSTN-T) – see example in Table 8.3.5-1**

   MGCF receives an INVITE request, through one of the origination procedures and via one of the S-CSCF to S-CSCF procedures.

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#3 and MO#2 are assumed.

Table 8.3.5-1: INVITE (S-S to PSTN-T)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf.home.net, sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

SDP The SDP contains the complete set of supported codecs from the session originator, as restricted by the originating network operator. The "m=" lines for the video media streams show a port number zero, which removes them from the negotiation.

2. **100 Trying (PSTN-T to S-S) – see example in Table 8.3.5-2**

MGCF may respond to the INVITE request with a 100 Trying provisional response.

Table 8.3.5-2: 100 Trying (PSTN-T to S-S)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

3. **H.248 Interaction to Create Connection**

MGCF initiates a H.248 interaction to pick an outgoing channel and determine media capabilities of the MGW.

4. **183 Session Progress (PSTN-T to S-S) – see example in Table 8.3.5-4**

MGCF determines the subset of the media flows proposed by the originating endpoint that it supports, and responds with a 183 Session Progress response back to the originator. This response is sent via the S-CSCF to S-CSCF procedure.

Table 8.3.5-4: 183 Session Progress (PSTN-T to S-S)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf.home.net, sip:scscf0.home.net
Require: 100rel
Remote-Party-ID: <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity: Off
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:017/4@mgcf4.home.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m-video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

5. **PRACK (S-S to PSTN-T) – see example in Table 8.3.5-5**

   The originating endpoint sends a PRACK request containing the final SDP to be used in this session, via the S-CSCF to S-CSCF procedure, to MGCF.

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#3 and MO#2 are assumed.

Table 8.3.5-5: PRACK (S-S to PSTN-T)

```
PRACK sip:mgcf.home.net SIP/2.0
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf.home.net, sip:scscf0.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m-video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

6. **200 OK (PSTN-T to S-S) – see example in Table 8.3.5-6**

MGCF acknowledges the PRACK request (5) with a 200 OK response.

Table 8.3.5-6: 200 OK (PSTN-T to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

7. **H.248 Interaction to Modify Connection**

   MGCF initiates a H.248 interaction to modify the connection established in step #3 and instruct MGW to reserve the resources necessary for the media streams.

8. **Resource Reservation**

   MGW reserved the resources necessary for the media streams.

9. **COMET (S-S to PSTN-T) – see example in Table 8.3.5-9**

   When the originating endpoint has completed its resource reservation, it sends the COMET request to MGCF, via the S-CSCF to S-CSCF procedures.

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#3 and MO#2 are assumed.

Table 8.3.5-9: COMET (S-S to PSTN-T)

```
COMET sip:mgcf.home.net SIP/2.0
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

10. **200 OK (PSTN-T to S-S) – see example in Table 8.3.5-10**

    MGCF acknowledges the COMET request (9) with a 200 OK response.

Table 8.3.5-10: 200 OK (PSTN-T to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

11. **IP-IAM**

   MGCF sends an IP-IAM message to the T-SGW

12. SS7 IAM

   T-SGW receives the IP-IAM and sends the SS7 IAM message into the PSTN.

13. **SS7 ACM**

   The PSTN establishes the path to the destination. In the present case the PSTN responds with an SS7 ACM message containing a "subscriber free" indication, implying that the called party is being alerted.

   Editor's Note: Alternatively the SS7 ACM may also be sent without any "subscriber free" indication meaning that the address is complete but not implying that the party is being alerted. This could be shown in a separate flow.

14. **IP-ACM**

   T-SGW sends an IP-ACM message to MGCF

15. **180 Ringing (PSTN-T to S-S) – see example in Table 8.3.5-15**

   If the PSTN is alerting the destination user, MGCF indicates this to the calling party by a 180 Ringing provisional response. This response is sent via the S-CSCF to S-CSCF procedures.

Table 8.3.5-15: 180 Ringing (PSTN-T to S-S)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf.home.net, sip:scscf0.home.net
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:017/4@mgcf4.home.net
RSeq: 9022
Content-length: 0
```

   The 180 Ringing is used when the SS7 ACM has indicated that the called party is being alerted.

   Editor's Note: The 183 Session Progress is used when there is no such indication from the SS7 ACM (no information about the called party being free or alerted). This could be shown in a separate flow.

   Editor's Note: Additional QoS interactions to handle one-way media at this point (e.g. for PSTN ringback and announcements) is for further study. This may depend on the SDP information included in the 180/183 messages.

16. **PRACK (S-S to PSTN-T) – see example in Table 8.3.5-16**

   The originator acknowledges the 180 Ringing provisional response (15) with a PRACK request.

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#3 and MO#2 are assumed.

Table 8.3.5-16: PRACK (S-S to PSTN-T)

```
PRACK sip:mgcf.home.net SIP/2.0
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq: 130 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Rack: 9022 127 INVITE
Content-length: 0
```

17. **200 OK (PSTN-T to S-S) – see example in Table 8.3.5-17**

     MGCF acknowledges the PRACK request (16) with a 200 OK response.

Table 8.3.5-17: 200 OK (PSTN-T to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

18. **SS7 ANM**

     When the called party answers, the PSTN sends an SS7 ANM message to T-SGW

19. **IP-ANM**

     T-SGW sends an IP-ANM message to MGCF

20. **H.248 Interaction to Modify Connection**

     MGCF initiates a H.248 interaction to make the connection in the MGW bi-directional.

21. **200 OK (PSTN-T to S-S) – see example in Table 8.3.5-21**

     MGCF sends a 200 OK final response along the signalling path back to the session originator.

Table 8.3.5-21: 200 OK (PSTN-T to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq: 127 INVITE
Content-length: 0
```

22. **ACK (S-S to PSTN-T) – see example in Table 8.3.5-22**

     The Calling party acknowledges the final response (21) with an ACK request.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#3 and MO#2 are assumed.

Table 8.3.5-22: ACK (S-S to PSTN-T)

```
ACK sip:mgcf.home.net SIP/2.0
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf.home.net, sip:scscf0.home.net
From:
To:
Call-ID:
Cseq: 127 ACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-length: 0
```

8.3.6 (MT#1c) Mobile termination, roaming, without I-CSCF in home network providing configuration independence, reject the invite request

This termination procedure applies to roaming subscribers when the home network operator does not desire to keep its internal configuration hidden from the visited network. The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure. During registration, the home 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.

Figure 8.3.6 – MT#1c

Procedure MT#1c is as follows:

1. **NVITE (S-S to MT#1a) – see example in Table 8.3.6-1**

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

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.6-1: INVITE (S-S to MT#1c)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (MT#1c to S-S) – see example in Table 8.3.6-2**

   S-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.3.6-2: 100 Trying (MT#1c to S-S)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

   S-CSCF validates the service profile, and performs any termination service control required for this subscriber

   S-CSCF examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

4. **INVITE (S-CSCF to P-CSCF) – see example in Table 8.3.6-4**

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

Table 8.3.6-4: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Route: built from the registration information, (pcscf and UE contact name), followed by the initial Request-URI from the incoming INVITE request. The first component of the Route header, pcscf, is moved to the Request-URI of the request.

Via:, Record-Route: S-CSCF adds itself

5. **100 Trying (P-CSCF to S-CSCF) – see example in Table 8.3.6-5**

    P-CSCF responds to the INVITE request (4) with a 100 Trying provisional response.

Table 8.3.6-5: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **INVITE (P-CSCF to UE) – see example in Table 8.3.6-6**

    P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.6-6: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.visited.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf.home.net, sip:scscf0.home.net,
 sip:%5b5555%3a%3aaaaa%3abbbb%3accc%3addd%5d@pcscf0.home.net
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

7. **486 Busy Here (UE to P-CSCF) – see example in Table 8.3.6-7**

UE is contacted successfully but it is currently not willing or able to take additional sessions. The response MAY indicate a better time later to call in the Retry-After header.

Table 8.3.6-7: 486 Busy Here (UE to P-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP pcscf.visited.net;
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:[5555::eee:fff:aaa:bbb]
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

Retry -After: a time period in seconds indicating how long the caller can try again

8. **ACK (P-CSCF to UE) – see example in Table 8.3.6-8**

Upon receive the 486 reponse from the UE, P-CSCF sends ACK back to the UE.

Table 8.3.6-8: ACK (P-CSCF to UE)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

9. **486 Busy Here (P-CSCF to S-CSCF) – see example in Table 8.3.6-9**

   P-CSCF forwards the 486 reponse to the S-CSCF.

Table 8.3.6-9: 486 Busy Here (P-CSCF to S-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=4321
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.visited.net
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

10. **ACK (S-CSCF to P-CSCF) – see example in Table 8.3.6-10**

    S-CSCF sends ACK to the P-CSCF.

Table 8.3.6-10: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

11. **Service Control**

    The S-CSCF validates the service profile and performs any service control required for this subscriber.

12. **486 Busy Here (MT#1c to S-S) – see example in Table 8.3.6-12**

    S-CSCF forwards the 486 Busy Here response to the originator, per the S-CSCF to S-CSCF procedure. Also indicates the voice mail address of the callee

Table 8.3.6-12: 486 Busy Here (MT#1c to S-S)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=9999
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

13. **ACK (S-S to MT#1c) – see example in Table 8.3.6-13**

    The S-CSCF of calling party responds to the 486 Busy Here response with an ACK request that is sent to S-CSCF via the S-CSCF to S-CSCF procedure.

Table 8.3.6-13: ACK (S-S to MT#1c)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.3.7 (MT#1d) Mobile termination, roaming, with I-CSCF in home network providing configuration independence, reject the invite request

This termination procedure applies to roaming subscribers when the home network operator does not desire to keep its internal configuration hidden from the visited network. The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure. During registration, the home 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.
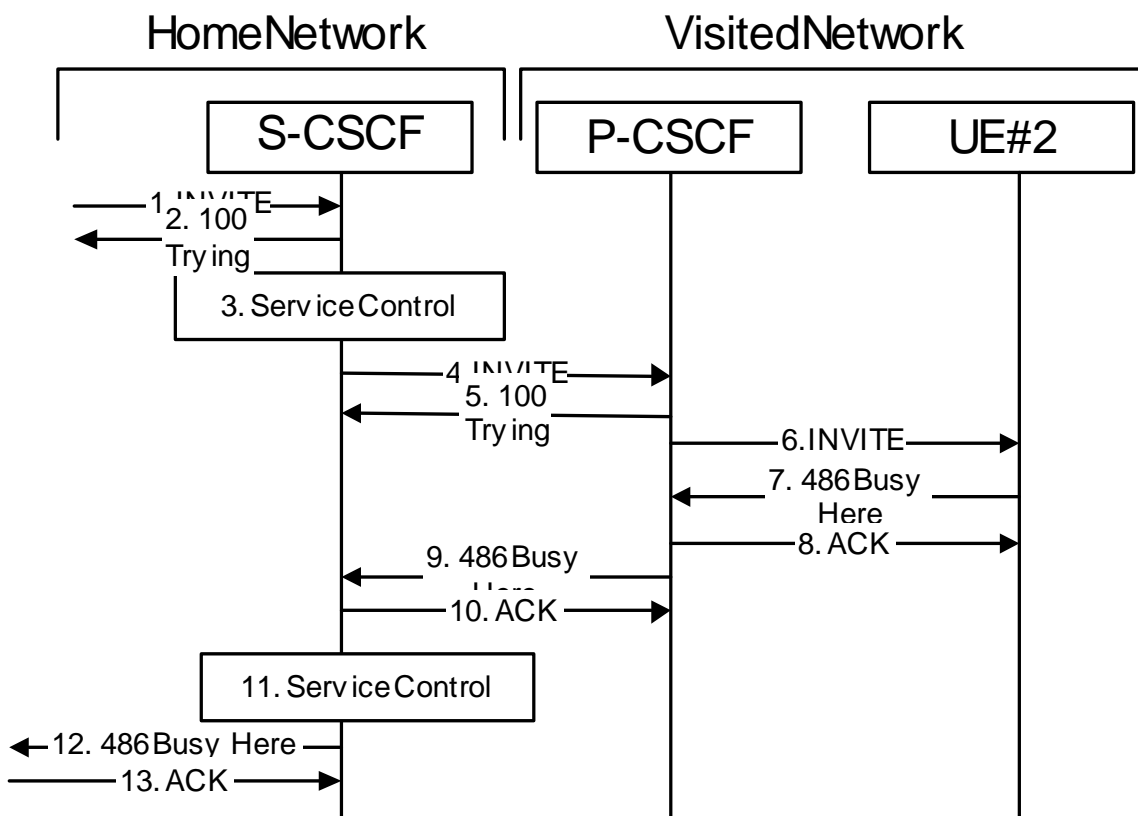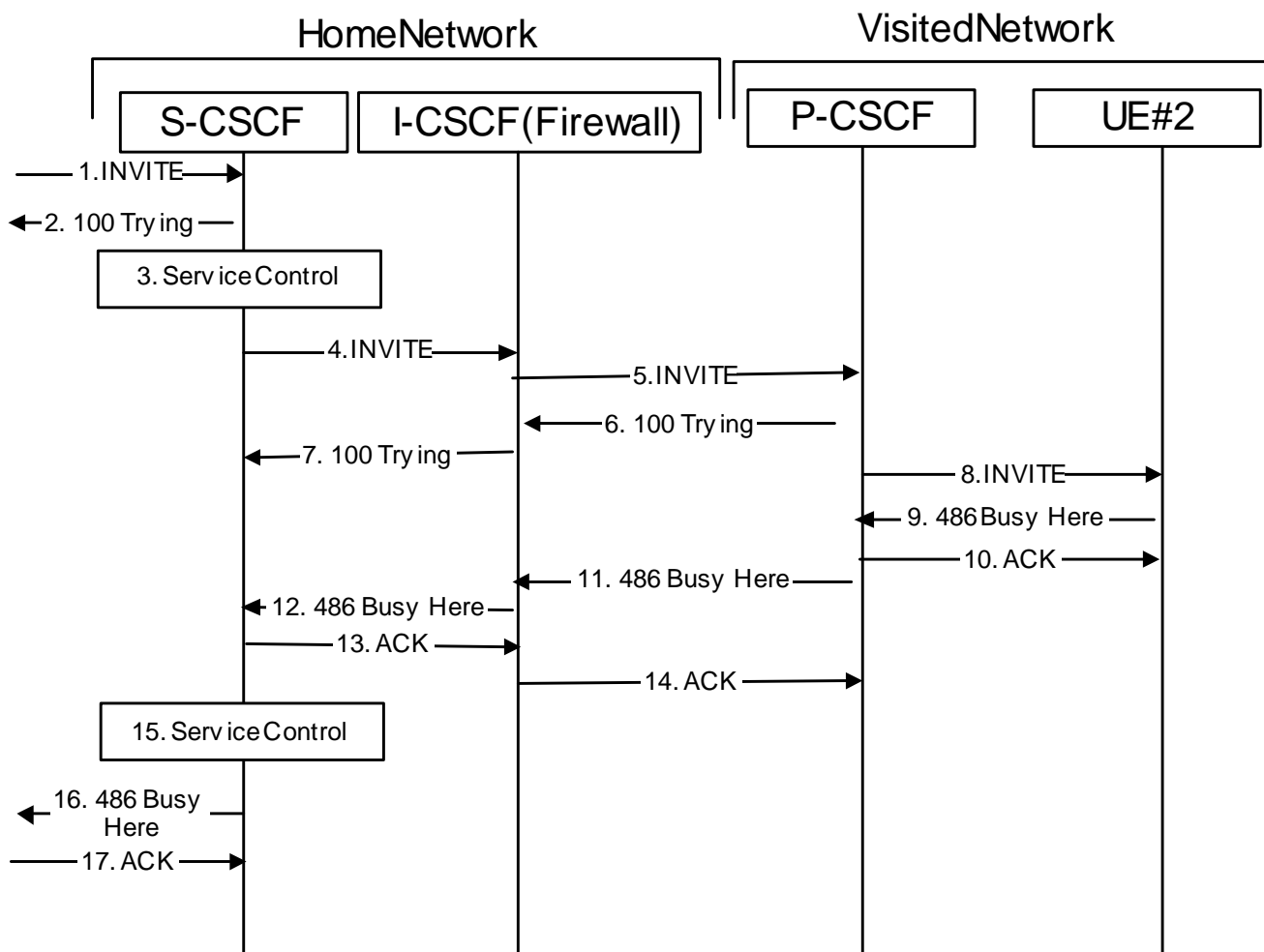

Figure 8.3.7 – MT#1d

Procedure MT#1d is as follows:

1. **INVITE (S-S to MT#1d) – see example in Table 8.3.7-1**

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

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.7-1: INVITE (S-S to MT#1d)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (MT#1d to S-S) – see example in Table 8.3.7-2**

   S-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.3.7-2: 100 Trying (MT#1d to S-S)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

   S-CSCF validates the service profile, and performs any termination service control required for this subscriber

   S-CSCF examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

4. **INVITE (S-CSCF to I-CSCF) – see example in Table 8.3.7-4**

   S-CSCF remembers (from the registration procedure) the next hop CSCF for this UE. It forwards the INVITE to the I-CSCF to perform the firewall functions.

Table 8.3.7-4: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net, sip:+1-212-555-
    2222@home.net;user=phone
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Route: built from the registration information, icscf, pcscf (and UE contact name), followed by the initial Request-URI from the incoming INVITE request. The first component of the Route header, icscf, is moved to the Request-URI of the request.

Via:, Record-Route: S-CSCF adds itself in the Record-Route and Via headers.

5. **INVITE (I-CSCF to P-CSCF) – see example in Table 8.3.7-5**

I-CSCF translates the Via headers in the request, and forwards the INVITE request to P-CSCF.

Table 8.3.7-5: INVITE (I-CSCF to P-CSCF)

```
INVITE sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:icscf.home.net, sip:Token(sip:scscf.home.net, sip:scscf0.home.net)
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Via: translated to preserve configuration independence of the home network.

Record-Route: translated to preserve configuration independence of the home network.

6. **100 Trying (P-CSCF to I-CSCF) – see example in Table 8.3.7-6**

   P-CSCF responds to the INVITE request (5) with a 100 Trying provisional response.

Table 8.3.7-6: 100 Trying (P-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

7. **100 Trying (I-CSCF to S-CSCF) – see example in Table 8.3.7-7**

   I-CSCF determines the Via header, and forwards the 100 Trying provisional response to S-CSCF.

Table 8.3.7-7: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

8. **INVITE (P-CSCF to UE) – see example in Table 8.3.7-8**

   P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.7-8: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.visited.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:icscf.home.net,
```

```
         sip:Token(sip:scscf.home.net, sip:scscf0.home.net),
         sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

9. **486 Busy Here (UE to P-CSCF) – see example in Table 8.3.7-9**

UE is contacted successfully but it is currently not willing or able to take additional sessions. The response MAY indicate a better time to call in the Retry-After header.

Table 8.3.7-9: 486 Busy Here (UE to P-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
Content-length:0
Retry-After:3600
```

Retry-After: Indicates how long the caller can try again

10. **ACK (P-CSCF to UE) – see example in Table 8.3.7-10**

Upon receive the 486 reponse from the UE, P-CSCF sends ACK back to the UE.

Table 8.3.7-10: ACK (P-CSCF to UE)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

11. **486 Busy Here (P-CSCF to I-CSCF) – see example in Table 8.3.7-11**

P-CSCF forwards the 486 reponse to the I-CSCF.

Table 8.3.7-11: 486 Busy Here (P-CSCF to I-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=4321
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Content-length:0
Retry-After:3600
```

12. **486 Busy Here (I-CSCF to S-CSCF) – see example in Table 8.3.7-12**

I-CSCF forwards the 486 reponse to the S-CSCF.

Table 8.3.7-12: 486 Busy Here (I-CSCF to S-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:0
Retry-After:3600
```

13. **ACK (S-CSCF to I-CSCF) – see example in Table 8.3.7-13**

The S-CSCF copies the Requet-URL and Route headers from the original INVITE request to ACK and send it to the P-CSCF via I-CSCF.

Table 8.3.7-13: ACK (S-CSCF to I-CSCF)

```
ACK: sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net, sip:+1-212-555-
    2222@home.net;user=phone
From:
To:
Call-ID:
CSeq:
Content-length:0
```

14. **Service Control**

The S-CSCF validates the service profile and performs any service control required for this subscriber.

15. **ACK (I-CSCF to P-CSCF) – see example in Table 8.3.7-15**

I-CSCF forwards the ACK to the P-CSCF, P-CSCF checks the ACK and makes sure this is for a 4xx response, so P-CSCF will not forward it further down.

Table 8.3.7-15: ACK (I-CSCF to P-CSCF)

```
ACK: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home.net)
Route: sip:+1-212-555-2222@home.net;user=phone
From:
To:
Call-ID:
CSeq:
Content-length:0
```

16. **486 Busy Here (MT#1d to S-S) – see example in Table 8.3.7-16**

S-CSCF forwards the 486 response to the originator, per the S-CSCF to S-CSCF procedure

Table 8.3.7-16: 486 Busy Here (MT#1d to S-S)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=9999
Call-ID:
CSeq:
Contact:sip[voice mailbox]
Content-length:0
Retry-After:3600
```

17. **ACK (S-S to MT#1d) – see example in Table 8.3.7-17**

S-CSCF0 sends the ACK to the S-CSCF

Table 8.3.7-17: ACK (S-S to MT#1d)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.3.8 (MT#2a) Mobile termination, located in home network, reject the INVITE request

This termination 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. During registration, the home network allocates a S-CSCF in the home network, 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.

Figure 8.3.8 – MT#2a

Procedure MT#2a is as follows:

1. **INVITE (S-S to MT#2a) – see example in Table 8.3.8-1**

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

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.8-1: INVITE (S-S to MT#2A)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (MT#2a to S-S) – see example in Table 8.3.8-2**

   S-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.3.8-2: 100 Trying (MT#2A to S-S)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

   S-CSCF validates the service profile, and performs any termination service control required for this subscriber.

   S-CSCF examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

4. **INVITE (S-CSCF to P-CSCF) – see example in Table 8.3.8-4**

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

Table 8.3.8-4: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Route: built from the registration information, (pcscf and UE contact name), followed by the initial Request-URI from the incoming INVITE request. The first component of the Route header, pcscf, is moved to the Request-URI of the request.

Via:, Record-Route: S-CSCF adds itself in the Record-Route and Via headers.

5. **100 Trying (P-CSCF to S-CSCF) – see example in Table 8.3.8-5**

    P-CSCF responds to the INVITE request (4) with a 100 Trying provisional response.

Table 8.3.8-5: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **INVITE (P-CSCF to UE) – see example in Table 8.3.8-6**

    P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.8-6: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf.home.net, sip:scscf0.home.net,
    sip:%5b5555%3a%3aaaaa%3abbbb%3accc%3addd%5d@pcscf0.home.net
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

7. **486 Busy Here (UE to P-CSCF) – see example in Table 8.3.8-7**

UE is contacted successfully but it is currently not willing or able to take additional sessions. The response MAY indicate a better time to call in the Retry-After header.

Table 8.3.8-7: 486 Busy Here (UE to P-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:[5555::eee:fff:aaa:bbb]
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

8. **ACK (P-CSCF to UE) – see example in Table 8.3.8-8**

Upon receive the 486 response from the UE, P-CSCF sends ACK back to the UE.

Table 8.3.8-8: ACK (P-CSCF to UE)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

9. **486 Busy Here (P-CSCF to S-CSCF) – see example in Table 8.3.8-9**

   P-CSCF forwards the 486 response to the S-CSCF.

Table 8.3.8-9: 486 Busy Here (P-CSCF to S-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=4321
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.home.net
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

10. **ACK (S-CSCF to P-CSCF) – see example in Table 8.3.8-10**

    S-CSCF sends ACK to the P-CSCF.

Table 8.3.8-10: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

11. **Service Control**

    The S-CSCF validates the service profile and performs any service control required for this subscriber.

12. **486 Busy Here (MT#2a to S-S) – see example in Table 8.3.8-12**

    S-CSCF forwards the 486 BusyHere response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.8-12: 486 Busy Here (MT#2a to S-S)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=9999
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

13. **ACK (S-S to MT#2a) – see example in Table 8.3.8-13**

    The S-CSCF of calling party responds to the 486 Busy Here response with an ACK request that is sent to S-CSCF via the S-CSCF to S-CSCF procedure.

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.3.9 (MT#1e) Mobile termination, roaming, without I-CSCF in home network providing configuration independence, service is refused by S-CSCF when receiving INVITE request

This termination procedure applies to roaming subscribers when the home network operator does not desire to keep its internal configuration hidden from the visited network. The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure. During registration, the home network allocates the S-CSCF.

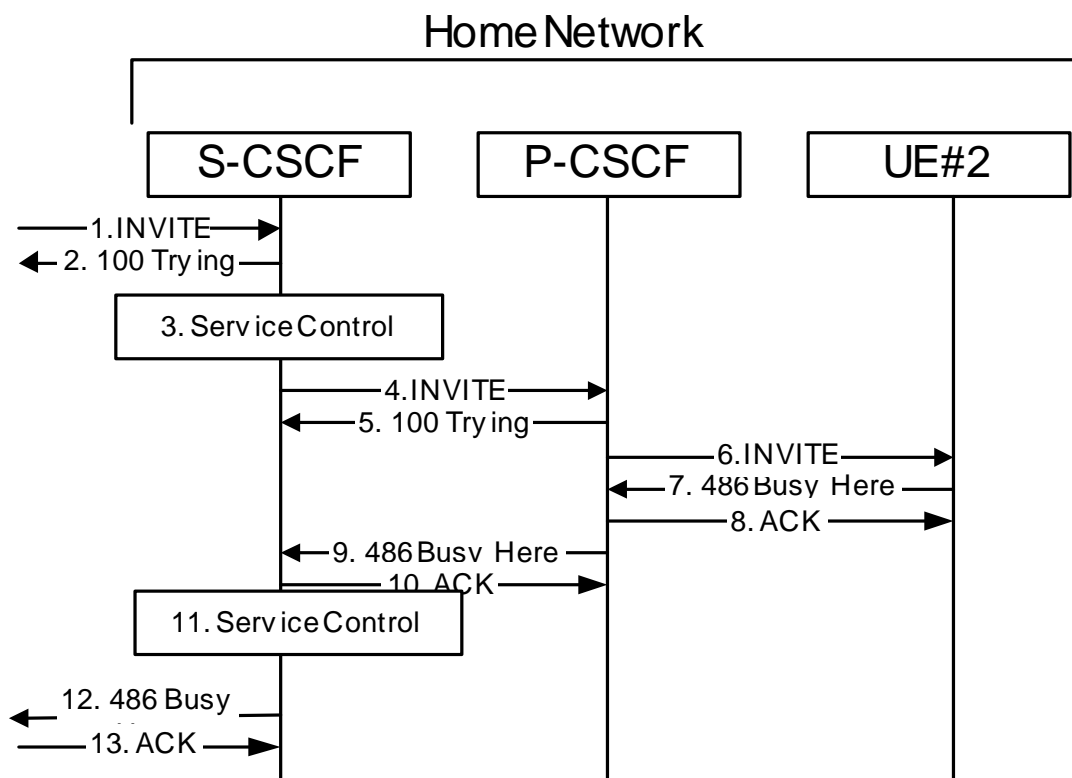When registration is complete, S-CSCF knows the name/address of P-CSCF, and P-CSCF knows the name/address of the UE.

Figure 8.3.9-1: Mobile termination, roaming, without I-CSCF in home network providing configuration independence, service is refused by S-CSCF when receiving INVITE request

1. **INVITE (S-S to MT#1e) - see example in Table 8.3.9-1**

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

Table 8.3.9-1: INVITE (S-S to MT#1e)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (MT#1e to S-S) - see example in Table 8.3.9-2**

   S-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.3.9-2: 100 Trying (MT#1e to S-S)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

   The S-CSCF validates the service profile and performs any service control required for this subscriber.

4. **403 Forbidden (MT#1e to S-S) - see example in Table 8.3.9-4**

   S-CSCF forwards the 403 Forbidden response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.9-4: 403 Forbidden (MT#1e to S-S)

```
SIP/2.0 403 Forbidden
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Call-ID:
CSeq:
Content-length:0
```

5. **ACK (S-S to MT#1e) - see example in Table 8.3.9-5**

   The S-CSCF of calling party responds to the 403 Forbidden response with an ACK request that is sent to S-CSCF via the S-CSCF to S-CSCF procedure.

Table 8.3.9-5: ACK (S-S to MT#1e)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Call-ID:
CSeq:
Content-length:0
```

8.4 Serving CSCF (MGCF) to Serving-CSCF (MGCF) procedures

8.4.1 Introduction

This clause presents the detailed application level flows to define the procedures for Serving-CSCF to Serving-CSCF.

This clause contains four session flow procedures, showing variations on the signalling path between the Serving-CSCF (or MGCF) that handles session origination, and the Serving-CSCF (or MGCF) that handles session termination. This signalling path depends on:

- whether the originator and destination are served by the same network operator,

    - agreements between operators for optimum PSTN gateway location.

Between separate operators, there are additional sub-cases covering the optional network configuration hiding – hiding required by both operators, neither operator, or just one operator.

The Serving-CSCF handling session origination performs an analysis of the destination address, and determines whether it is a PSTN destination, a subscriber of the same network operator or a subscriber of a different operator.

If the analysis of the destination address determined that it belongs to a subscriber of a different operator, the request is 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. The I-CSCF then forwards the request to the S-CSCF. This is session flow procedure S-S#1.

If the analysis of the destination address determines that it belongs to a subscriber of the same operator, the S-CSCF forwards the request to a local I-CSCF, who queries the HSS for current location information. The I-CSCF then forwards the request to the S-CSCF. This is session flow procedure S-S#2.

If the analysis of the destination address determines that it is a PSTN destination, the S-CSCF forwards the request to a local BGCF. Based on further analysis of the destination address, and on agreements between operators for PSTN termination, the BGCF will either select a local MGCF to perform the termination (procedure S-S#3) or will forward the request to a BGCF in another operator's network who will select the MGCF to perform the termination (procedures S-S#4).

8.4.2 (S-S#1a) Different network operators performing origination and termination, without configuration hiding by either operator

The Serving-CSCF handling session origination (S-CSCF#1) performs an analysis of the destination address, and determines that it belongs to a subscriber of a different operator. The originating network operator does not desire to keep their configuration hidden, so forwards the request to a well-known entry point in the destination operator's network, I-CSCF. I-CSCF queries the HSS for current location information, and finds the S-CSCF assigned to the subscriber (S-CSCF#2), and forwards the request to S-CSCF#2. The terminating network operator does not desire to keep their configuration hidden, so the I-CSCF does not insert itself into the signaling path for future exchanges.

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

MO#1a Mobile origination, roaming, without firewall. The "Originating Network" of S-S#1a is therefore a visited network.

MO#1b   Mobile origination, roaming, with firewall in home network. The "Originating Network" of S-S#1a is therefore a visited network.

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

PSTN-O PSTN origination. The "Originating Network" of S-S#1a is the home network. The element labeled 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#1a   Mobile termination, roaming, without firewall. The "Terminating Network" of S-S#1a is a visited network.

MT#1b   Mobile termination, roaming, with firewall in home network. The "Terminating Network" of S-S#1a is a visited network.

MT#2   Mobile termination, located in home service area. The "Terminating Network" of S-S#1a is the home network.

Figure 8.4.2 – S-S#1a

Procedure S-S#1a is as follows:

1. **INVITE (MO to S-S#1a) – see example in Table 8.4.2-1**

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

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-1: INVITE (MO to S-S#1a)

```
INVITE sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv

a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (S-S#1a to MO) – see example in Table 8.4.2-2**

   S-CSCF#1 responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.4.2-2: 100 Trying (S-S#1a to MO)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

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

   S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

   For this example, assume the subscriber is not allowed video.

4. **INVITE (S-CSCF to I-CSCF) – see example in Table 8.4.2-4**

   S-CSCF#1 performs an analysis of the destination address, and determines the network operator to whom the destination subscriber belongs. Since the originating operator does not desire to keep their internal configuration hidden, S-CSCF#1 forwards the INVITE request directly to to I-CSCF in the destination network.

Table 8.4.2-4: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP sip:scscf.home1.net SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home1.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Request-URI: In the case where the Route header of the incoming INVITE request to S-CSCF contains a TEL-URL, it has to be translated to a globally routable SIP-URL before applying it as Request-URI of the outgoing INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: Need to verify that it is the Route header of the incoming INVITE request which may contain the TEL-URL, and not the Request-URI as indicated in the text of Tdoc N1-010353.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

5. **100 Trying (I-CSCF to S-CSCF) – see example in Table 8.4.2-5**

   I-CSCF responds to the INVITE request (4) by sending a 100 Trying provisional response to S-CSCF#1.

Table 8.4.2-5: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **Location Query**

   I-CSCF (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

7. Location Response

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

8. **INVITE (I-CSCF to S-CSCF) – see example in Table 8.4.2-8**

I-CSCF forwards the INVITE request to the S-CSCF (S-CSCF#2) that will handle the session termination.

Table 8.4.2-8: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home1.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which the INVITE is routed to S-CSCF, and the mechanism by which S-CSCF retrieves the initial Request URI (which identifies the real destination of the session) is FFS.

NOTE: The I-CSCF does not add itself to the Record-Route header, as it has no need to remain in the signalling path once the session is established.

9. **100 Trying (S-CSCF to I-CSCF) – see example in Table 8.4.2-9**

S-CSCF#2 responds to the INVITE request (8) with a 100 Trying provisional response.

Table 8.4.2-9: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

10. **Service Control**

S-CSCF#2 performs whatever service control logic is appropriate for this session attempt

S-CSCF#2 examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

11. **INVITE (S-S#1a to MT) – see example in Table 8.4.2-11**

S-CSCF#2 forwards the INVITE request, as determined by the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-11: INVITE (S-S#1a to MT)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

12. **100 Trying (MT to S-S#1a) – see example in Table 8.4.2-12**

S-CSCF#2 receives a 100 Trying provisional response to the INVITE request (11), as specified by the termination procedures.

Table 8.4.2-12: 100 Trying (MT to S-S#1a)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

13. **183 Session Progress (MT to S-S#1a) – see example in Table 8.4.2-13**

The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response to the INVITE request (11), as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-13: 183 Session Progress (MT to S-S#1a)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

14. **183 Session Progress (S-CSCF to I-CSCF) – see example in Table 8.4.2-14**

    S-CSCF#2 forwards the 183 Session Progress provisional response to I-CSCF

Table 8.4.2-14: 183 Session Progress (S-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

15. **183 Session Progress (I-CSCF to S-CSCF) – see example in Table 8.4.2-15**

    I-CSCF forwards the 183 Session Progress provisional response to S-CSCF#1.

Table 8.4.2-15: 183 Session Progress (I-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

16. **183 Session Progress (S-S#1a to MO) – see example in Table 8.4.2-16**

    S-CSCF#1 forwards the 183 Session Progress to the originator, as per the originating procedure.

    NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-16: 183 Session Progress (S-S#1a to MO)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

17. **PRACK (MO to S-S#1a) – see example in Table 8.4.2-17**

The originator decides the final set of media streams, and includes this information in the PRACK request sent to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-17: PRACK (MO to S-S#1a)

```
PRACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

18. **PRACK (S-CSCF to S-CSCF) – see example in Table 8.4.2-18**

S-CSCF#1 forwards the PRACK request to S-CSCF#2.

Table 8.4.2-18: PRACK (S-CSCF to S-CSCF)

```
PRACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

19. **PRACK (S-S#1a to MT) – see example in Table 8.4.2-19**

S-CSCF#2 forwards the PRACK request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-19: PRACK (S-S#1a to MT)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

20. **200 OK (MT to S-S#1a) – see example in Table 8.4.2-20**

   The terminating endpoint responds to the PRACK request (19) with a 200 OK response.

   NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-20: 200 OK (MT to S-S#1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

21. **200 OK (S-CSCF to S-CSCF) – see example in Table 8.4.2-21**

   S-CSCF#2 forwards the 200 OK response to S-CSCF#1.

Table 8.4.2-21: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

22. **200 OK (S-S#1a to MO) – see example in Table 8.4.2-22**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-22: 200 OK (S-S#1a to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

23. **COMET (MO to S-S#1a) – see example in Table 8.4.2-23**

When the originating endpoint has completed the resource reservation procedures, it sends the COMET request to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-23: COMET (MO to S-S#1a)

```
COMET sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

24. **COMET (S-CSCF to S-CSCF) – see example in Table 8.4.2-24**

S-CSCF#1 forwards the COMET request to S-CSCF#2.

Table 8.4.2-24: COMET (S-CSCF to S-CSCF)

```
COMET sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

25. **COMET (S-S#1a to MT) – see example in Table 8.4.2-25**

S-CSCF#2 forwards the COMET request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-25: COMET (S-S#1a to MT)

```
COMET sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

26. **200 OK (MT to S-S#1a) – see example in Table 8.4.2-26**

The terminating endpoint responds to the COMET request (25) with a 200 OK response.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-26: 200 OK (MT to S-S#1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

27. **200 OK (S-CSCF to S-CSCF) – see example in Table 8.4.2-27**

S-CSCF#2 forwards the 200 OK response to S-CSCF#1.

Table 8.4.2-27: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

28. **200 OK (S-S#1a to MO) – see example in Table 8.4.2-28**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-28: 200 OK (S-S#1a to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

29. **180 Ringing (MT to S-S#1a) – see example in Table 8.4.2-29**

The terminating endpoint may optionally send a 180 Ringing provisional response indicating alerting is in progress. This response is sent by the termination procedure to S-CSCF#2.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-29: 180 Ringing (MT to S-S#1a)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaaa%3abbbb%5d@pcscf.home2.net
RSeq: 9022
Content-length: 0
```

30. **Service Control**

    The S-CSCF validates the service profile and performs any service control required for this subscriber.

31. **180 Ringing (S-CSCF to I-CSCF) – see example in Table 8.4.2-31**

    S-CSCF#2 forwards the 180 Ringing response to I-CSCF.

Table 8.4.2-31: 180 Ringing (S-CSCF to I-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

32. **180 Ringing (I-CSCF to S-CSCF) – see example in Table 8.4.2-32**

    I-CSCF forwards the 180 Ringing response to S-CSCF#1.

Table 8.4.2-32: 180 Ringing (I-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

33. **Service Control**

    The S-CSCF validates the service profile and performs any service control required for this subscriber.

34. **180 Ringing (S-S#1a to MO) – see example in Table 8.4.2-34**

    S-CSCF#1 forwards the 180 Ringing response to the originator, per the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-34: 180 Ringing (S-S#1a to MO)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

35. **PRACK (MO to S-S#1a) – see example in Table 8.4.2-35**

The originator acknowledges the 180 Ringing provisional response (34) with a PRACK request.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-35: PRACK (MO to S-S#1a)

```
PRACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Rack: 9022 127 INVITE
Content-length: 0
```

36. **PRACK (S-CSCF to S-CSCF) – see example in Table 8.4.2-36**

S-CSCF#1 forwards the PRACK request to S-CSCF#2.

Table 8.4.2-36: PRACK (S-CSCF to S-CSCF)

```
PRACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

37. **PRACK (S-S#1a to MT) – see example in Table 8.4.2-37**

S-CSCF#2 forwards the PRACK request to the terminating endpoint.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-37: PRACK (S-S#1a to MT)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

38. **200 OK (MT to S-S#1a) – see example in Table 8.4.2-38**

The terminating endpoint responds to the PRACK request (37) with a 200 OK response.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-38: 200 OK (MT to S-S#1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

39. **200 OK (S-CSCF to S-CSCF) – see example in Table 8.4.2-39**

S-CSCF#2 forwards the 200 OK response to S-CSCF#1.

Table 8.4.2-39: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

40. **200 OK (S-S#1a to MO) – see example in Table 8.4.2-40**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-40: 200 OK (S-S#1a to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

41. **200 OK (MT to S-S#1a) – see example in Table 8.4.2-41**

The final response to the INVITE request (11), 200 OK, is sent by the terminating endpoint over the signalling path. This is typically generated when the subscriber has accepted the incoming session attempt. The response is sent to S-CSCF#2 per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-41: 200 OK (MT to S-S#1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

42. **Service Control**

S-CSCF#2 performs whatever service control logic is appropriate for this session completion

43. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.2-43**

The 200 OK response is forwarded to the I-CSCF.

Table 8.4.2-43: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

44. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.2-44**

The 200 OK response is forwarded to S-CSCF#1.

Table 8.4.2-44: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

45. **Service Control**

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

46. **200 OK (S-S#1a to MO) – see example in Table 8.4.2-46**

The 200 OK response is returned to the originating endpoint, by the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-46: 200 OK (S-S#1a to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

47. **ACK (MO to S-S#1a) – see example in Table 8.4.2-47**

The originating endpoint sends the final acknowledgement to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-47: ACK (MO to S-S#1a)

```
ACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 ACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-length: 0
```

48. **ACK (S-CSCF to S-CSCF) – see example in Table 8.4.2-48**

S-CSCF#1 forwards the ACK request to S-CSCF#2.

Table 8.4.2-48: ACK (S-CSCF to S-CSCF)

```
ACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

49. **ACK (S-S#1a to MT) – see example in Table 8.4.2-49**

> S-CSCF#2 forwards the ACK request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.2-49: ACK (S-S#1a to MT)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

8.4.3 (S-S#1b) Different network operators performing origination and termination, with configuration hiding by both network operators

The Serving-CSCF handling session origination (S-CSCF#1) performs an analysis of the destination address, and determines that it belongs to a subscriber of a different operator. The originating network operator desires to keep their configuration hidden, so forwards the request through an I-CSCF (I-CSCF#1) to a well-known entry point in the destination operator's network, I-CSCF#2. I-CSCF#2 queries the HSS for current location information, and finds the S-CSCF assigned to the subscriber (S-CSCF#2), and forwards the request to S-CSCF#2. The terminating network operator also desires to keep their configuration hidden, so I-CSCF#2 inserts itself into the signaling path for future exchanges.

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

MO#1a Mobile origination, roaming, without firewall. The "Originating Network" of S-S#1b is therefore a visited network.

MO#1b Mobile origination, roaming, with firewall in home network. The "Originating Network" of S-S#1b is therefore a visited network.

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

PSTN-O PSTN origination. The "Originating Network" of S-S#1b is the home network. The element labeled 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#1aMobile termination, roaming, without firewall. The "Terminating Network" of S-S#1b is a visited network.

MT#1b Mobile termination, roaming, with firewall in home network. The "Terminating Network" of S-S#1b is a visited network.

MT#2   Mobile termination, located in home service area. The "Terminating Network" of S-S#1b is the home
network.

Figure 8.4.3 – S-S#1b

Procedure S-S#1b is as follows:

1. **INVITE (MO to S-S#1b) – see example in Table 8.4.3-1**

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

    NOTE: There are a number of different origination flows, and the table represents a typical example of what one
    of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-1: INVITE (MO to S-S#1b)

```
INVITE sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (S-S#1b to MO) – see example in Table 8.4.3-2**

   S-CSCF#1 responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.4.3-2: 100 Trying (S-S#1b to MO)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

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

   S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

   For this example, assume the subscriber is not allowed video.

4. **INVITE (S-CSCF to I-CSCF) – see example in Table 8.4.3-4**

   S-CSCF#1 performs an analysis of the destination address, and determines the network operator to whom the destination subscriber belongs. Since the originating operator desires to keep their internal configuration hidden, S-CSCF#1 forwards the INVITE request to I-CSCF#1.

Table 8.4.3-4: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:icscf.home1.net SIP/2.0
Via: SIP/2.0/UDP sip:scscf.home1.net SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home1.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Request-URI: set to the I-CSCF that will perform the translation needed to maintain configuration independence.

Route: updated to cause I-CSCF to forward the request to the proper terminating network operator. In the case of a TEL-URL, it has to be translated to a globally routable SIP-URL before sending the INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

5. **INVITE (I-CSCF to I-CSCF) – see example in Table 8.4.3-5**

I-CSCF#1 forwards the INVITE request to I-CSCF#2.

Table 8.4.3-5: INVITE (I-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

6. **100 Trying (I-CSCF to I-CSCF) – see example in Table 8.4.3-6**

   I-CSCF#2 respond to the INVITE request (5) with a 100 Trying provisional response.

Table 8.4.3-6: 100 Trying (I-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

7. **100 Trying (I-CSCF to S-CSCF) – see example in Table 8.4.3-7**

   I-CSCF#1 determines the Via header, and forwards the 100 Trying provisional response to S-CSCF#1.

Table 8.4.3-7: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

8. **Location Query**

I-CSCF#2 (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

9. **Location Response**

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

10. **INVITE (I-CSCF to S-CSCF) – see example in Table 8.4.3-10**

I-CSCF#2 forwards the INVITE request to the S-CSCF (S-CSCF#2) that will handle the session termination.

Table 8.4.3-10: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
   scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:icscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which the INVITE is routed to S-CSCF, and the mechanism by which S-CSCF retrieves the initial Request URI (which identifies the real destination of the session) is FFS.

11. **100 Trying (S-CSCF to I-CSCF) – see example in Table 8.4.3-11**

S-CSCF#2 responds to the INVITE request (10) with a 100 Trying provisional response.

Table 8.4.3-11: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
   scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

12. **Service Control**

S-CSCF#2 performs whatever service control logic is appropriate for this session attempt

S-CSCF#2 examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

13. **INVITE (S-S#1b to MT) – see example in Table 8.4.3-13**

S-CSCF#2 forwards the INVITE request, as determined by the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-13: INVITE (S-S#1b to MT)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

14. **100 Trying (MT to S-S#1b) – see example in Table 8.4.3-14**

S-CSCF#2 receives a 100 Trying provisional response to the INVITE request (13), as specified by the termination procedures.

Table 8.4.3-14: 100 Trying (MT to S-S#1b)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

15. **183 Session Progress (MT to S-S#1b) – see example in Table 8.4.3-15**

The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response to the INVITE request (13), as per the termination procedure.

Editor's Note: Diagram for S-S#1 should show I-CSCF#2 as a firewall and gateway, making it consistent with I-CSCF#1.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-15: 183 Session Progress (MT to S-S#1b)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

16. **183 Session Progress (S-CSCF to I-CSCF) – see example in Table 8.4.3-16**

   S-CSCF#2 forwards the 183 Session Progress provisional response to I-CSCF#2.

Table 8.4.3-16: 183 Session Progress (S-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

17. **183 Session Progress (I-CSCF to I-CSCF) – see example in Table 8.4.3-17**

I-CSCF#2 forwards the 183 Session Progress provisional response to I-CSCF#1.

Table 8.4.3-17: 183 Session Progress (I-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

18. **183 Session Progress (I-CSCF to S-CSCF) – see example in Table 8.4.3-18**

   I-CSCF#1 forwards the 183 Session Progress provisional response to S-CSCF#1.

Table 8.4.3-18: 183 Session Progress (I-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

   Record-Route: I-CSCF#1 determines the entry to the right of its own entry.

   Via: determined by I-CSCF#1.

19. **183 Session Progress (S-S#1b to MO) – see example in Table 8.4.3-19**

   S-CSCF#1 forwards the 183 Session Progress to the originator, as per the originating procedure.

   NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-19: 183 Session Progress (S-S#1b to MO)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

20. **PRACK (MO to S-S#1b) – see example in Table 8.4.3-20**

    The originator decides the final set of media streams, and includes this information in the PRACK request sent to S-CSCF#1 by the origination procedures.

    NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-20: PRACK (MO to S-S#1b)

```
PRACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home1.net, sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

21. **PRACK (S-CSCF to I-CSCF) – see example in Table 8.4.3-21**

S-CSCF#1 forwards the PRACK request to I-CSCF#1.

Table 8.4.3-21: PRACK (S-CSCF to I-CSCF)

```
PRACK sip:icscf.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

22. **PRACK (I-CSCF to I-CSCF) – see example in Table 8.4.3-22**

    I-CSCF#1 forwards the PRACK request to I-CSCF#2.

Table 8.4.3-22: PRACK (I-CSCF to I-CSCF)

```
PRACK sip:icscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:Token(sip:scscf.home2.net), sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

23. **PRACK (I-CSCF to S-CSCF) – see example in Table 8.4.3-23**

    I-CSCF#2 determines the routing information, and forwards the PRACK request to S-CSCF#2.

Table 8.4.3-23: PRACK (I-CSCF to S-CSCF)

```
PRACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

24. **PRACK (S-S#1b to MT) – see example in Table 8.4.3-24**

> S-CSCF#2 forwards the PRACK request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-24: PRACK (S-S#1b to MT)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

25. **200 OK (MT to S-S#1b) – see example in Table 8.4.3-25**

> The terminating endpoint responds to the PRACK request (24) with a 200 OK response.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-25: 200 OK (MT to S-S#1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

26. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.3-26**

S-CSCF#2 forwards the 200 OK response to I-CSCF#2.

Table 8.4.3-26: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

27. **200 OK (I-CSCF to I-CSCF) – see example in Table 8.4.3-27**

I-CSCF#2 forwards the 200 OK response to I-CSCF#1.

Table 8.4.3-27: 200 OK (I-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

28. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.3-28**

I-CSCF#1 forwards the 200 OK response to S-CSCF#1.

Table 8.4.3-28: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#1 determining the entry to the right of its own entry.

Via: determined by I-CSCF#1.

29. **200 OK (S-S#1b to MO) – see example in Table 8.4.3-29**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-29: 200 OK (S-S#1b to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

30. **COMET (MO to S-S#1b) – see example in Table 8.4.3-30**

When the originating endpoint has completed the resource reservation procedures, it sends the COMET request to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-30: COMET (MO to S-S#1b)

```
COMET sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home1.net, sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

31. **COMET (S-CSCF to I-CSCF) – see example in Table 8.4.3-31**

S-CSCF#1 forwards the COMET request to I-CSCF#1.

Table 8.4.3-31: COMET (S-CSCF to I-CSCF)

```
COMET sip:icscf.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

32. **COMET (I-CSCF to I-CSCF) – see example in Table 8.4.3-32**

I-CSCF#1 forwards the COMET request to I-CSCF#2.

Table 8.4.3-32: COMET (I-CSCF to I-CSCF)

```
COMET sip:icscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
   pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:Token(sip:scscf.home2.net), sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

33. **COMET (I-CSCF to S-CSCF) – see example in Table 8.4.3-33**

I-CSCF#2 forwards the COMET request to S-CSCF#2.

Table 8.4.3-33: COMET (I-CSCF to S-CSCF)

```
COMET sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
   scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

34. **COMET (S-S#1b to MT) – see example in Table 8.4.3-34**

S-CSCF#2 forwards the COMET request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-34: COMET (S-S#1b to MT)

```
COMET sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

35. **200 OK (MT to S-S#1b) – see example in Table 8.4.3-35**

    The terminating endpoint responds to the COMET request (34) with a 200 OK response.

    NOTE: There are a number of different termination flows, and the table represents a typical example of what one
             of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-35: 200 OK (MT to S-S#1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

36. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.3-36**

    S-CSCF#2 forwards the 200 OK response to I-CSCF#2.

Table 8.4.3-36: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

37. **200 OK (I-CSCF to I-CSCF) – see example in Table 8.4.3-37**

    I-CSCF#2 forwards the 200 OK response to I-CSCF#1.

Table 8.4.3-37: 200 OK (I-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

38. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.3-38**

    I-CSCF#1 forwards the 200 OK response to S-CSCF#1.

Table 8.4.3-38: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#1 determining the entry to the right of its own entry.

Via: determined by I-CSCF#1.

39. **200 OK (S-S#1b to MO) – see example in Table 8.4.3-39**

    S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-39: 200 OK (S-S#1b to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

40. **180 Ringing (MT to S-S#1b) – see example in Table 8.4.3-40**

    The terminating endpoint may optionally send a 180 Ringing provisional response indicating alerting is in progress. This response is sent by the termination procedure to S-CSCF#2.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-40: 180 Ringing (MT to S-S#1b)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
RSeq: 9022
Content-length: 0
```

41. **Service Control**

    The S-CSCF validates the service profile and performs any service control required for this subscriber.

42. **180 Ringing (S-CSCF to I-CSCF) – see example in Table 8.4.3-42**

    S-CSCF#2 forwards the 180 Ringing response to I-CSCF#2.

Table 8.4.3-42: 180 Ringing (S-CSCF to I-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

43. **180 Ringing (I-CSCF to I-CSCF) – see example in Table 8.4.3-43**

    I-CSCF#2 forwards the 180 Ringing response to I-CSCF#1.

Table 8.4.3-43: 180 Ringing (I-CSCF to I-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

44. **180 Ringing (I-CSCF to S-CSCF) – see example in Table 8.4.3-44**

    I-CSCF#1 forwards the 180 Ringing response to S-CSCF#1.

Table 8.4.3-44: 180 Ringing (I-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

Record-Route: formed by I-CSCF#1 determining the entry to the right of its own entry.

Via: determined by I-CSCF#1.

45. **Service Control**

The S-CSCF validates the service profile and performs any service control required for this subscriber.

46. **180 Ringing (S-S#1b to MO) – see example in Table 8.4.3-46**

S-CSCF#1 forwards the 180 Ringing response to the originator, per the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-46: 180 Ringing (S-S#1b to MO)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

47. **PRACK (MO to S-S#1b) – see example in Table 8.4.3-47**

The originator acknowledges the 180 Ringing provisional response (46) with a PRACK request.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-47: PRACK (MO to S-S#1b)

```
PRACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home1.net, sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Rack: 9022 127 INVITE
Content-length: 0
```

48. **PRACK (S-CSCF to I-CSCF) – see example in Table 8.4.3-48**

S-CSCF#1 forwards the PRACK request to I-CSCF#1.

Table 8.4.3-48: PRACK (S-CSCF to I-CSCF)

```
PRACK sip:icscf.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
   sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

49. **PRACK (I-CSCF to I-CSCF) – see example in Table 8.4.3-49**

    I-CSCF#1 forwards the PRACK request to I-CSCF#2.

Table 8.4.3-49: PRACK (I-CSCF to I-CSCF)

```
PRACK sip:icscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
   pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:Token(sip:scscf.home2.net), sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

50. **PRACK (I-CSCF to S-CSCF) – see example in Table 8.4.3-50**

    I-CSCF#2 determines the routing information, and forwards the PRACK request to S-CSCF#2.

Table 8.4.3-50: PRACK (I-CSCF to S-CSCF)

```
PRACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
   scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

51. **PRACK (S-S#1b to MT) – see example in Table 8.4.3-51**

    S-CSCF#2 forwards the PRACK request to the terminating endpoint.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-51: PRACK (S-S#1b to MT)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

52. **200 OK (MT to S-S#1b) – see example in Table 8.4.3-52**

    The terminating endpoint responds to the PRACK request (51) with a 200 OK response.

    NOTE: There are a number of different termination flows, and the table represents a typical example of what one
             of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-52: 200 OK (MT to S-S#1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

53. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.3-53**

    S-CSCF#2 forwards the 200 OK response to I-CSCF#2.

Table 8.4.3-53: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

54. **200 OK (I-CSCF to I-CSCF) – see example in Table 8.4.3-54**

    I-CSCF#2 forwards the 200 OK response to I-CSCF#1.

Table 8.4.3-54: 200 OK (I-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

55. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.3-55**

I-CSCF#1 forwards the 200 OK response to S-CSCF#1.

Table 8.4.3-55: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#1 determining the entry to the right of its own entry.

Via: determined by I-CSCF#1.

56. **200 OK (S-S#1b to MO) – see example in Table 8.4.3-56**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-56: 200 OK (S-S#1b to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

57. **200 OK (MT to S-S#1b) – see example in Table 8.4.3-57**

The final response to the INVITE (13), 200 OK, is sent by the terminating endpoint over the signalling path. This is typically generated when the subscriber has accepted the incoming session attempt. The response is sent to S-CSCF#2 per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-57: 200 OK (MT to S-S#1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

58. **Service Control**

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

59. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.3-59**

     The 200 OK response is forwarded to the I-CSCF#2.

Table 8.4.3-59: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

60. **200 OK (I-CSCF to I-CSCF) – see example in Table 8.4.3-60**

     The 200 OK response is forwarded to I-CSCF#1.

Table 8.4.3-60: 200 OK (I-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

61. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.3-61**

     The 200 OK response is forwarded to S-CSCF#1.

Table 8.4.3-61: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#1 determining the entry to the right of its own entry.

Via: determined by I-CSCF#1.

62. **Service Control**

S-CSCF#1 performs whatever service control logic is appropriate for this session completion

63. **200 OK (S-S#1b to MO) – see example in Table 8.4.3-63**

The 200 OK response is returned to the originating endpoint, by the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-63: 200 OK (S-S#1b to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

64. **ACK (MO to S-S#1b) – see example in Table 8.4.3-64**

The originating endpoint sends the final acknowledgement to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-64: ACK (MO to S-S#1b)

```
ACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home1.net, sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 ACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-length: 0
```

65. **ACK (S-CSCF to I-CSCF) – see example in Table 8.4.3-65**

S-CSCF#1 forwards the ACK request to I-CSCF#1.

Table 8.4.3-65: ACK (S-CSCF to I-CSCF)

```
ACK sip:icscf.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

66. **ACK (I-CSCF to I-CSCF) – see example in Table 8.4.3-66**

I-CSCF#1 forwards the ACK request to I-CSCF#2.

Table 8.4.3-66: ACK (I-CSCF to I-CSCF)

```
ACK sip:icscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:Token(sip:scscf.home2.net), sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

67. **ACK (I-CSCF to S-CSCF) – see example in Table 8.4.3-67**

I-CSCF#2 forwards the ACK request to S-CSCF#2.

Table 8.4.3-67: ACK (I-CSCF to S-CSCF)

```
ACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

68. **ACK (S-S#1b to MT) – see example in Table 8.4.3-68**

S-CSCF#2 forwards the ACK request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.3-68: ACK (S-S#1b to MT)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:icscf.home1.net,
    sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

8.4.4 (S-S#1c) Different network operators performing origination and termination, with configuration hiding by originating network operator

The Serving-CSCF handling session origination (S-CSCF#1) performs an analysis of the destination address, and determines that it belongs to a subscriber of a different operator. The originating network operator desires to keep their configuration hidden, so forwards the request through an I-CSCF (I-CSCF#1) to a well-known entry point in the destination operator's network, I-CSCF#2. I-CSCF#2 queries the HSS for current location information, and finds the S-CSCF assigned to the subscriber (S-CSCF#2), and forwards the request to S-CSCF#2. The terminating network operator does not desire to keep their configuration hidden, so I-CSCF#2 does not insert itself into the signaling path for future exchanges.

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

MO#1a Mobile origination, roaming, without firewall. The "Originating Network" of S-S#1c is therefore a visited network.

MO#1b Mobile origination, roaming, with firewall in home network. The "Originating Network" of S-S#1c is therefore a visited network.

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

PSTN-O PSTN origination. The "Originating Network" of S-S#1c is the home network. The element labeled 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#1a Mobile termination, roaming, without firewall. The "Terminating Network" of S-S#1c is a visited network.

MT#1b Mobile termination, roaming, with firewall in home network. The "Terminating Network" of S-S#1c is a visited network.

MT#2 Mobile termination, located in home service area. The "Terminating Network" of S-S#1c is the home network.

Figure 8.4.4 – S-S#1c

Procedure S-S#1c is as follows:

1. **INVITE (MO to S-S#1c) – see example in Table 8.4.4-1**

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

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-1: INVITE (MO to S-S#1c)

```
INVITE sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (S-S#1c to MO) – see example in Table 8.4.4-2**

   S-CSCF#1 responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.4.4-2: 100 Trying (S-S#1c to MO)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

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

   S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

   For this example, assume the subscriber is not allowed video.

4. **INVITE (S-CSCF to I-CSCF) – see example in Table 8.4.4-4**

   S-CSCF#1 performs an analysis of the destination address, and determines the network operator to whom the destination subscriber belongs. Since the originating operator desires to keep their internal configuration hidden, S-CSCF#1 forwards the INVITE request to I-CSCF#1.

Table 8.4.4-4: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:icscf.home1.net SIP/2.0
Via: SIP/2.0/UDP sip:scscf.home1.net SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home1.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Request-URI: set to the I-CSCF that will perform the translation needed to maintain configuration independence.

Route: updated to cause I-CSCF to forward the request to the proper terminating network operator. In the case of a TEL-URL, it has to be translated to a globally routable SIP-URL before sending the INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

5. **INVITE (I-CSCF to I-CSCF) – see example in Table 8.4.4-5**

I-CSCF#1 forwards the INVITE request to I-CSCF#2.

Table 8.4.4-5: INVITE (I-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

6. **100 Trying (I-CSCF to I-CSCF) – see example in Table 8.4.4-6**

   I-CSCF#2 respond to the INVITE request (5) with a 100 Trying provisional response.

Table 8.4.4-6: 100 Trying (I-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

7. **100 Trying (I-CSCF to S-CSCF) – see example in Table 8.4.4-7**

   I-CSCF#1 determines the Via header, and forwards the 100 Trying provisional response to S-CSCF#1.

Table 8.4.4-7: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

8. **Location Query**

I-CSCF#2 (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

9. **Location Response**

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

10. **INVITE (I-CSCF to S-CSCF) – see example in Table 8.4.4-10**

I-CSCF#2 forwards the INVITE request to the S-CSCF (S-CSCF#2) that will handle the session termination.

Table 8.4.4-10: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
   scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which the INVITE is routed to S-CSCF, and the mechanism by which S-CSCF retrieves the initial Request URI (which identifies the real destination of the session) is FFS.

11. **100 Trying (S-CSCF to I-CSCF) – see example in Table 8.4.4-11**

S-CSCF#2 responds to the INVITE request (10) with a 100 Trying provisional response.

Table 8.4.4-11: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
   scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

12. **Service Control**

S-CSCF#2 performs whatever service control logic is appropriate for this session attempt

S-CSCF#2 examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

13. **INVITE (S-S#1c to MT) – see example in Table 8.4.4-13**

S-CSCF#2 forwards the INVITE request, as determined by the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-13: INVITE (S-S#1c to MT)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

14. **100 Trying (MT to S-S#1c) – see example in Table 8.4.4-14**

S-CSCF#2 receives a 100 Trying provisional response to the INVITE request (13), as specified by the termination procedures.

Table 8.4.4-14: 100 Trying (MT to S-S#1c)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

15. **183 Session Progress (MT to S-S#1c) – see example in Table 8.4.4-15**

The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response to the INVITE request (13), as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-15: 183 Session Progress (MT to S-S#1c)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

16. **183 Session Progress (S-CSCF to I-CSCF) – see example in Table 8.4.4-16**

    S-CSCF#2 forwards the 183 Session Progress provisional response to I-CSCF#2.

Table 8.4.4-16: 183 Session Progress (S-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route:
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

17. **183 Session Progress (I-CSCF to I-CSCF) – see example in Table 8.4.4-17**

I-CSCF#2 forwards the 183 Session Progress provisional response to I-CSCF#1.

Table 8.4.4-17: 183 Session Progress (I-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net,SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route:
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

18. **183 Session Progress (I-CSCF to S-CSCF) – see example in Table 8.4.4-18**

I-CSCF#1 forwards the 183 Session Progress provisional response to S-CSCF#1.

Table 8.4.4-18: 183 Session Progress (I-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:sip:scscf.home2.net,sip:icscf.home1.net, sip:scscf.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Record-Route: I-CSCF#1 determines the entry to the right of its own entry.

Via: determined by I-CSCF#1.

19. **183 Session Progress (S-S#1c to MO) – see example in Table 8.4.4-19**

S-CSCF#1 forwards the 183 Session Progress to the originator, as per the originating procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-19: 183 Session Progress (S-S#1c to MO)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

20. **PRACK (MO to S-S#1c) – see example in Table 8.4.4-20**

The originator decides the final set of media streams, and includes this information in the PRACK request sent to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-20: PRACK (MO to S-S#1c)

```
PRACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home1.net, sip:scscf.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

21. **PRACK (S-CSCF to I-CSCF) – see example in Table 8.4.4-21**

S-CSCF#1 forwards the PRACK request to I-CSCF#1.

Table 8.4.4-21: PRACK (S-CSCF to I-CSCF)

```
PRACK sip:icscf.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf.home2.net,sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

22. **PRACK (I-CSCF to S-CSCF) – see example in Table 8.4.4-22**

I-CSCF#1 forwards the PRACK request to S-CSCF#2.

Table 8.4.4-22: PRACK (I-CSCF to S-CSCF)

```
PRACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

23. **PRACK (S-S#1c to MT) – see example in Table 8.4.4-23**

   S-CSCF#2 forwards the PRACK request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-23: PRACK (S-S#1c to MT)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
        scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

24. **200 OK (MT to S-S#1c) – see example in Table 8.4.4-24**

   The terminating endpoint responds to the PRACK request (24) with a 200 OK response.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-24: 200 OK (MT to S-S#1c)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

25. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.4-25**

S-CSCF#2 forwards the 200 OK response to I-CSCF#1.

Table 8.4.4-25: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

26. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.4-26**

I-CSCF#1 forwards the 200 OK response to S-CSCF#1.

Table 8.4.4-26: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#1 determining the entry to the right of its own entry.

Via: determined by I-CSCF#1.

27. **200 OK (S-S#1c to MO) – see example in Table 8.4.4-27**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-27: 200 OK (S-S#1c to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
    sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

28. **COMET (MO to S-S#1c) – see example in Table 8.4.4-28**

When the originating endpoint has completed the resource reservation procedures, it sends the COMET request to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-28: COMET (MO to S-S#1c)

```
COMET sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home1.net, sip:scscf.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

29. **COMET (S-CSCF to I-CSCF) – see example in Table 8.4.4-29**

S-CSCF#1 forwards the COMET request to I-CSCF#1.

Table 8.4.4-29: COMET (S-CSCF to I-CSCF)

```
COMET sip:icscf.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

30. **COMET (I-CSCF to S-CSCF) – see example in Table 8.4.4-30**

> I-CSCF#1 forwards the COMET request to S-CSCF#2.

Table 8.4.4-30: COMET (I-CSCF to S-CSCF)

```
COMET sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
   pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

31. **COMET (S-S#1c to MT) – see example in Table 8.4.4-31**

> S-CSCF#2 forwards the COMET request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-31: COMET (S-S#1c to MT)

```
COMET sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

32. **200 OK (MT to S-S#1c) – see example in Table 8.4.4-32**

The terminating endpoint responds to the COMET request (31) with a 200 OK response.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-32: 200 OK (MT to S-S#1c)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

33. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.4-33**

S-CSCF#2 forwards the 200 OK response to I-CSCF#1.

Table 8.4.4-33: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

34. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.4-34**

I-CSCF#1 forwards the 200 OK response to S-CSCF#1.

Table 8.4.4-34: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#1 determining the entry to the right of its own entry.

Via: determined by I-CSCF#1.

35. **200 OK (S-S#1c to MO) – see example in Table 8.4.4-35**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-35: 200 OK (S-S#1c to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

36. **180 Ringing (MT to S-S#1c) – see example in Table 8.4.4-36**

The terminating endpoint may optionally send a 180 Ringing provisional response indicating alerting is in progress. This response is sent by the termination procedure to S-CSCF#2.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-36: 180 Ringing (MT to S-S#1c)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
RSeq: 9022
Content-length: 0
```

37. **180 Ringing (S-CSCF to I-CSCF) – see example in Table 8.4.4-37**

S-CSCF#2 forwards the 180 Ringing response to I-CSCF#2.

Table 8.4.4-37: 180 Ringing (S-CSCF to I-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

38. **180 Ringing (I-CSCF to I-CSCF) – see example in Table 8.4.4-38**

    I-CSCF#2 forwards the 180 Ringing response to I-CSCF#1.

Table 8.4.4-38: 180 Ringing (I-CSCF to I-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

39. **180 Ringing (I-CSCF to S-CSCF) – see example in Table 8.4.4-39**

    I-CSCF#1 forwards the 180 Ringing response to S-CSCF#1.

Table 8.4.4-39: 180 Ringing (I-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:scscf.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

   Record-Route: formed by I-CSCF#1 determining the entry to the right of its own entry.

   Via: determined by I-CSCF#1.

40. **180 Ringing (S-S#1c to MO) – see example in Table 8.4.4-40**

    S-CSCF#1 forwards the 180 Ringing response to the originator, per the origination procedure.

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-40: 180 Ringing (S-S#1c to MO)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:scscf.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

41. **PRACK (MO to S-S#1c) – see example in Table 8.4.4-41**

The originator acknowledges the 180 Ringing provisional response (40) with a PRACK request.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-41: PRACK (MO to S-S#1c)

```
PRACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home1.net, sip:scscf.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Rack: 9022 127 INVITE
Content-length: 0
```

42. **PRACK (S-CSCF to I-CSCF) – see example in Table 8.4.4-42**

S-CSCF#1 forwards the PRACK request to I-CSCF#1.

Table 8.4.4-42: PRACK (S-CSCF to I-CSCF)

```
PRACK sip:icscf.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

43. **PRACK (I-CSCF to I-CSCF) – see example in Table 8.4.4-43**

I-CSCF#1 forwards the PRACK request to S-CSCF#2.

Table 8.4.4-43: PRACK (I-CSCF to S-CSCF)

```
PRACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

44. **PRACK (S-S#1c to MT) – see example in Table 8.4.4-44**

S-CSCF#2 forwards the PRACK request to the terminating endpoint.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-44: PRACK (S-S#1c to MT)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

45. **200 OK (MT to S-S#1c) – see example in Table 8.4.4-45**

The terminating endpoint responds to the PRACK request (44) with a 200 OK response.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-45: 200 OK (MT to S-S#1c)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

46. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.4-46**

S-CSCF#2 forwards the 200 OK response to I-CSCF#1.

Table 8.4.4-46: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

47. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.4-47**

    I-CSCF#1 forwards the 200 OK response to S-CSCF#1.

Table 8.4.4-47: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#1 determining the entry to the right of its own entry.

Via: determined by I-CSCF#1.

48. **200 OK (S-S#1c to MO) – see example in Table 8.4.4-48**

    S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-48: 200 OK (S-S#1c to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

49. **200 OK (MT to S-S#1c) – see example in Table 8.4.4-49**

    The final response to the INVITE (13), 200 OK, is sent by the terminating endpoint over the signalling path. This is typically generated when the subscriber has accepted the incoming session attempt. The response is sent to S-CSCF#2 per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-49: 200 OK (MT to S-S#1c)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

50. **Service Control**

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

51. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.4-51**

The 200 OK response is forwarded to the I-CSCF#2.

Table 8.4.4-51: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

52. **200 OK (I-CSCF to I-CSCF) – see example in Table 8.4.4-52**

The 200 OK response is forwarded to I-CSCF#1.

Table 8.4.4-52: 200 OK (I-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
    pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

53. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.4-53**

The 200 OK response is forwarded to S-CSCF#1.

Table 8.4.4-53: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#1 determining the entry to the right of its own entry.

Via: determined by I-CSCF#1.

54. **Service Control**

S-CSCF#1 performs whatever service control logic is appropriate for this session completion

55. **200 OK (S-S#1c to MO) – see example in Table 8.4.4-55**

The 200 OK response is returned to the originating endpoint, by the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-55: 200 OK (S-S#1c to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

56. **ACK (MO to S-S#1c) – see example in Table 8.4.4-56**

The originating endpoint sends the final acknowledgement to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-56: ACK (MO to S-S#1c)

```
ACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home1.net, sip:scscf.home2.net,
sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 ACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-length: 0
```

57. **ACK (S-CSCF to I-CSCF) – see example in Table 8.4.4-57**

S-CSCF#1 forwards the ACK request to I-CSCF#1.

Table 8.4.4-57: ACK (S-CSCF to I-CSCF)

```
ACK sip:icscf.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

58. **ACK (I-CSCF to S-CSCF) – see example in Table 8.4.4-58**

I-CSCF#1 forwards the ACK request to S-CSCF#2.

Table 8.4.4-58: ACK (I-CSCF to S-CSCF)

```
ACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP
pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

59. **ACK (S-S#1c to MT) – see example in Table 8.4.4-59**

S-CSCF#2 forwards the ACK request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.4-59: ACK (S-S#1c to MT)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home2.net, sip:icscf.home1.net, sip:Token(sip:scscf.home1.net)
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

8.4.5 (S-S#1d) Different network operators performing origination and termination, with configuration hiding by terminating network operator

The Serving-CSCF handling session origination (S-CSCF#1) performs an analysis of the destination address, and determines that it belongs to a subscriber of a different operator. S-CSCF#1 forwards the request to a well-known entry point in the destination operator's network, I-CSCF#2. I-CSCF#2 queries the HSS for current location information, and finds the S-CSCF assigned to the subscriber (S-CSCF#2), and forwards the request to S-CSCF#2. The terminating network operator desires to keep their configuration hidden, so I-CSCF#2 inserts itself into the signaling path for future exchanges.

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

MO#1a Mobile origination, roaming, without firewall. The "Originating Network" of S-S#1d is therefore a visited network.

MO#1b Mobile origination, roaming, with firewall in home network. The "Originating Network" of S-S#1d is therefore a visited network.

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

PSTN-O PSTN origination. The "Originating Network" of S-S#1d is the home network. The element labeled 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#1a Mobile termination, roaming, without firewall. The "Terminating Network" of S-S#1d is a visited network.

MT#1b Mobile termination, roaming, with firewall in home network. The "Terminating Network" of S-S#1d is a visited network.

MT#2 Mobile termination, located in home service area. The "Terminating Network" of S-S#1d is the home network.

Figure 8.4.5 – S-S#1d

Procedure S-S#1d is as follows:

1. **INVITE (MO to S-S#1d) – see example in Table 8.4.5-1**

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

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-1: INVITE (MO to S-S#1d)

```
INVITE sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (S-S#1d to MO) – see example in Table 8.4.5-2**

   S-CSCF#1 responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.4.5-2: 100 Trying (S-S#1d to MO)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

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

   S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

   For this example, assume the subscriber is not allowed video.

4. **INVITE (S-CSCF to I-CSCF) – see example in Table 8.4.5-4**

   S-CSCF#1 performs an analysis of the destination address, and determines the network operator to whom the destination subscriber belongs. S-CSCF#1 forwards the INVITE request to I-CSCF#2, the well-known entry point of the destination network.

Table 8.4.5-4: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP sip:scscf.home1.net SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home1.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Request-URI: set to the I-CSCF that will perform the translation needed to maintain configuration independence.

Via:, Record-Route: translated to maintain configuration independence of the home#1 operator.

5. **100 Trying (I-CSCF to I-CSCF) – see example in Table 8.4.5-5**

    I-CSCF#2 respond to the INVITE request (4) with a 100 Trying provisional response.

Table 8.4.5-5: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **Location Query**

    I-CSCF#2 (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

7. **Location Response**

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

8. **INVITE (I-CSCF to S-CSCF) – see example in Table 8.4.5-8**

    I-CSCF#2 forwards the INVITE request to the S-CSCF (S-CSCF#2) that will handle the session termination.

Table 8.4.5-8: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:icscf.home2.net, sip:scscf.home1.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which the INVITE is routed to S-CSCF, and the mechanism by which S-CSCF retrieves the initial Request URI (which identifies the real destination of the session) is FFS.

9. **100 Trying (S-CSCF to I-CSCF) – see example in Table 8.4.5-9**

    S-CSCF#2 responds to the INVITE request (8) with a 100 Trying provisional response.

Table 8.4.5-9: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

10. **Service Control**

    S-CSCF#2 performs whatever service control logic is appropriate for this session attempt

    S-CSCF#2 examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

    For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

11. **INVITE (S-S#1d to MT) – see example in Table 8.4.5-11**

    S-CSCF#2 forwards the INVITE request, as determined by the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-11: INVITE (S-S#1d to MT)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

12. **100 Trying (MT to S-S#1d) – see example in Table 8.4.5-12**

S-CSCF#2 receives a 100 Trying provisional response to the INVITE request (11), as specified by the termination procedures.

Table 8.4.5-12: 100 Trying (MT to S-S#1d)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

13. **183 Session Progress (MT to S-S#1d) – see example in Table 8.4.5-13**

The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response to the INVITE request (11), as per the termination procedure.

Editor's Note: Diagram for S-S#1 should show I-CSCF#2 as a firewall and gateway, making it consistent with I-CSCF#1.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-13: 183 Session Progress (MT to S-S#1d)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

14. **183 Session Progress (S-CSCF to I-CSCF) – see example in Table 8.4.5-14**

   S-CSCF#2 forwards the 183 Session Progress provisional response to I-CSCF#2.

Table 8.4.5-14: 183 Session Progress (S-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

15. **183 Session Progress (I-CSCF to S-CSCF) – see example in Table 8.4.5-15**

   I-CSCF#2 forwards the 183 Session Progress provisional response to S-CSCF#1.

Table 8.4.5-15: 183 Session Progress (I-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

16. **183 Session Progress (S-S#1d to MO) – see example in Table 8.4.5-16**

S-CSCF#1 forwards the 183 Session Progress to the originator, as per the originating procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-16: 183 Session Progress (S-S#1d to MO)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

17. **PRACK (MO to S-S#1d) – see example in Table 8.4.5-17**

The originator decides the final set of media streams, and includes this information in the PRACK request sent to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-17: PRACK (MO to S-S#1d)

```
PRACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

18. **PRACK (S-CSCF to I-CSCF) – see example in Table 8.4.5-18**

S-CSCF#1 forwards the PRACK request to I-CSCF#2.

Table 8.4.5-18: PRACK (S-CSCF to I-CSCF)

```
PRACK sip:icscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(sip:scscf.home2.net), sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

19. **PRACK (I-CSCF to S-CSCF) – see example in Table 8.4.5-19**

I-CSCF#2 determines the routing information, and forwards the PRACK request to S-CSCF#2.

Table 8.4.5-19: PRACK (I-CSCF to S-CSCF)

```
PRACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

20. **PRACK (S-S#1d to MT) – see example in Table 8.4.5-20**

S-CSCF#2 forwards the PRACK request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-20: PRACK (S-S#1d to MT)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

21. **200 OK (MT to S-S#1d) – see example in Table 8.4.5-21**

The terminating endpoint responds to the PRACK request (20) with a 200 OK response.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-21: 200 OK (MT to S-S#1d)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

22. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.5-22**

S-CSCF#2 forwards the 200 OK response to I-CSCF#2.

Table 8.4.5-22: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

23. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.5-23**

I-CSCF#2 forwards the 200 OK response to S-CSCF#1.

Table 8.4.5-23: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

24. **200 OK (S-S#1d to MO) – see example in Table 8.4.5-24**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-24: 200 OK (S-S#1d to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

25. **COMET (MO to S-S#1d) – see example in Table 8.4.5-25**

> When the originating endpoint has completed the resource reservation procedures, it sends the COMET request to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-25: COMET (MO to S-S#1d)

```
COMET sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

26. **COMET (S-CSCF to I-CSCF) – see example in Table 8.4.5-26**

> S-CSCF#1 forwards the COMET request to I-CSCF#2.

Table 8.4.5-26: COMET (S-CSCF to I-CSCF)

```
COMET sip:icscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(sip:scscf.home2.net), sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

27. **COMET (I-CSCF to S-CSCF) – see example in Table 8.4.5-27**

    I-CSCF#2 forwards the COMET request to S-CSCF#2.

Table 8.4.5-27: COMET (I-CSCF to S-CSCF)

```
COMET sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

28. **COMET (S-S#1d to MT) – see example in Table 8.4.5-28**

    S-CSCF#2 forwards the COMET request to the terminating endpoint, as per the termination procedure.

    NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-28: COMET (S-S#1d to MT)

```
COMET sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

29. **200 OK (MT to S-S#1d) – see example in Table 8.4.5-29**

The terminating endpoint responds to the COMET request (28) with a 200 OK response.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-29: 200 OK (MT to S-S#1d)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

30. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.5-30**

S-CSCF#2 forwards the 200 OK response to I-CSCF#2.

Table 8.4.5-30: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

31. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.5-31**

I-CSCF#2 forwards the 200 OK response to S-CSCF#1.

Table 8.4.5-31: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

32. **200 OK (S-S#1d to MO) – see example in Table 8.4.5-32**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-32: 200 OK (S-S#1d to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

33. **180 Ringing (MT to S-S#1d) – see example in Table 8.4.5-33**

The terminating endpoint may optionally send a 180 Ringing provisional response indicating alerting is in progress. This response is sent by the termination procedure to S-CSCF#2.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-33: 180 Ringing (MT to S-S#1d)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
   SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
RSeq: 9022
Content-length: 0
```

34. **180 Ringing (S-CSCF to I-CSCF) – see example in Table 8.4.5-34**

S-CSCF#2 forwards the 180 Ringing response to I-CSCF#2.

Table 8.4.5-34: 180 Ringing (S-CSCF to I-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

35. **180 Ringing (I-CSCF to S-CSCF) – see example in Table 8.4.5-35**

I-CSCF#2 forwards the 180 Ringing response to S-CSCF#1.

Table 8.4.5-35: 180 Ringing (I-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

36. **180 Ringing (S-S#1d to MO) – see example in Table 8.4.5-36**

S-CSCF#1 forwards the 180 Ringing response to the originator, per the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-36: 180 Ringing (S-S#1d to MO)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

37. **PRACK (MO to S-S#1d) – see example in Table 8.4.5-37**

The originator acknowledges the 180 Ringing provisional response (36) with a PRACK request.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-37: PRACK (MO to S-S#1d)

```
PRACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Rack: 9022 127 INVITE
Content-length: 0
```

38. **PRACK (S-CSCF to I-CSCF) – see example in Table 8.4.5-38**

    S-CSCF#1 forwards the PRACK request to I-CSCF#2.

Table 8.4.5-38: PRACK (S-CSCF to I-CSCF)

```
PRACK sip:icscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(sip:scscf.home2.net), sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

39. **PRACK (I-CSCF to S-CSCF) – see example in Table 8.4.5-39**

    I-CSCF#2 determines the routing information, and forwards the PRACK request to S-CSCF#2.

Table 8.4.5-39: PRACK (I-CSCF to S-CSCF)

```
PRACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

40. **PRACK (S-S#1d to MT) – see example in Table 8.4.5-40**

    S-CSCF#2 forwards the PRACK request to the terminating endpoint.

    NOTE: There are a number of different termination flows, and the table represents a typical example of what one
             of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-40: PRACK (S-S#1d to MT)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP cscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

41. **200 OK (MT to S-S#1d) – see example in Table 8.4.5-41**

The terminating endpoint responds to the PRACK request (40) with a 200 OK response.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-41: 200 OK (MT to S-S#1d)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

42. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.5-42**

S-CSCF#2 forwards the 200 OK response to I-CSCF#2.

Table 8.4.5-42: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

43. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.5-43**

I-CSCF#2 forwards the 200 OK response to S-CSCF#1.

Table 8.4.5-43: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

44. **200 OK (S-S#1d to MO) – see example in Table 8.4.5-44**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-44: 200 OK (S-S#1d to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

45. **200 OK (MT to S-S#1d) – see example in Table 8.4.5-45**

The final response to the INVITE (13), 200 OK, is sent by the terminating endpoint over the signalling path. This is typically generated when the subscriber has accepted the incoming session attempt. The response is sent to S-CSCF#2 per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-45: 200 OK (MT to S-S#1d)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Content-length: 0
```

46. **Service Control**

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

47. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.5-47**

The 200 OK response is forwarded to the I-CSCF#2.

Table 8.4.5-47: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

48. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.5-48**

The 200 OK response is forwarded to S-CSCF#1.

Table 8.4.5-48: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

Record-Route: formed by I-CSCF#2 reversing and translating all the entries to the left of its own entry.

49. **Service Control**

S-CSCF#1 performs whatever service control logic is appropriate for this session completion

50. **200 OK (S-S#1d to MO) – see example in Table 8.4.5-50**

The 200 OK response is returned to the originating endpoint, by the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-50: 200 OK (S-S#1d to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf.home2.net), sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

51. **ACK (MO to S-S#1d) – see example in Table 8.4.5-51**

The originating endpoint sends the final acknowledgement to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-51: ACK (MO to S-S#1d)

```
ACK sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf.home2.net, sip:Token(sip:scscf.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 ACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home1.net
Content-length: 0
```

52. **ACK (S-CSCF to I-CSCF) – see example in Table 8.4.5-52**

S-CSCF#1 forwards the ACK request to I-CSCF#2.

Table 8.4.5-52: ACK (S-CSCF to I-CSCF)

```
ACK sip:icscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(sip:scscf.home2.net), sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

53. **ACK (I-CSCF to S-CSCF) – see example in Table 8.4.5-53**

   I-CSCF#2 forwards the ACK request to S-CSCF#2.

Table 8.4.5-53: ACK (I-CSCF to S-CSCF)

```
ACK sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net
Record-Route: sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

54. **ACK (S-S#1d to MT) – see example in Table 8.4.5-54**

   S-CSCF#2 forwards the ACK request to the terminating endpoint, as per the termination procedure.

   NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.5-54: ACK (S-S#1d to MT)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home2.net, sip:icscf.home2.net, sip:scscf.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

8.4.6   (S-S#2) Single network operator performing origination and termination

The Serving-CSCF handling session 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 S-CSCF assigned to the subscriber (S-CSCF#2), and forwards the request to S-CSCF#2.

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

MO#1a Mobile origination, roaming, without firewall. The "Originating Network" of S-S#2 is therefore a visited network.

MO#1b Mobile origination, roaming, with firewall in home network. The "Originating Network" of S-S#2 is therefore a visited network.

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

PSTN-O PSTN origination. The "Originating Network" of S-S#2 is the home network. The element labeled 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#1a Mobile termination, roaming, without firewall. The "Terminating Network" of S-S#2 is a visited network.

MT#1b Mobile termination, roaming, with firewall in home network. The "Terminating Network" of S-S#2 is a visited network.

MT#2 Mobile termination, located in home service area. The "Terminating Network" of S-S#2 is the home network.

Figure 8.4.6 – S-S#2

Procedure S-S#2 is as follows:

1. **INVITE (MO to S-S#2) – see example in Table 8.4.6-1**

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

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-1: INVITE (MO to S-S#2)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (S-S#2 to MO) – see example in Table 8.4.6-2**

   S-CSCF#1 responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.4.6-2: 100 Trying (S-S#2 to MO)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

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

   S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

   For this example, assume the subscriber is not allowed video.

4. **INVITE (S-CSCF to I-CSCF) – see example in Table 8.4.6-4**

S-CSCF#1 performs an analysis of the destination address, and determines the network operator to whom the destination subscriber belongs. Since the originating operator does not desire to keep their internal configuration hidden, S-CSCF#1 forwards the INVITE request directly to to I-CSCF in the destination network.

Table 8.4.6-4: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP sip:scscf1.home.net SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

Request-URI: In the case where the Route header of the incoming INVITE request to S-CSCF contains a TEL-URL, it has to be translated to a globally routable SIP-URL before applying it as Request-URI of the outgoing INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: Need to verify that it is the Route header of the incoming INVITE request which may contain the TEL-URL, and not the Request-URI as indicated in the text of Tdoc N1-010353.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

5. **100 Trying (I-CSCF to S-CSCF) – see example in Table 8.4.6-5**

   I-CSCF responds to the INVITE request (4) by sending a 100 Trying provisional response to S-CSCF#1.

Table 8.4.6-5: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **Location Query**

I-CSCF (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

7. **Location Response**

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

8. **INVITE (I-CSCF to S-CSCF) – see example in Table 8.4.6-8**

I-CSCF forwards the INVITE request to the S-CSCF (S-CSCF#2) that will handle the session termination.

Table 8.4.6-8: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which the INVITE is routed to S-CSCF, and the mechanism by which S-CSCF retrieves the initial Request URI (which identifies the real destination of the session) is FFS.

NOTE: The I-CSCF does not add itself to the Record-Route header, as it has no need to remain in the signalling path once the session is established.

9. **100 Trying (S-CSCF to I-CSCF) – see example in Table 8.4.6-9**

S-CSCF#2 responds to the INVITE request (8) with a 100 Trying provisional response.

Table 8.4.6-9: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

10. **Service Control**

S-CSCF#2 performs whatever service control logic is appropriate for this session attempt

S-CSCF#2 examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

11. **INVITE (S-S#2 to MT) – see example in Table 8.4.6-11**

S-CSCF#2 forwards the INVITE request, as determined by the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-11: INVITE (S-S#2 to MT)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

12. **100 Trying (MT to S-S#2) – see example in Table 8.4.6-12**

S-CSCF#2 receives a 100 Trying provisional response to the INVITE request (11), as specified by the termination procedures.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-12: 100 Trying (MT to S-S#2)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

13. **183 Session Progress (MT to S-S#2) – see example in Table 8.4.6-13**

The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-13: 183 Session Progress (MT to S-S#2)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

14. **183 Session Progress (S-CSCF to I-CSCF) – see example in Table 8.4.6-14**

S-CSCF#2 forwards the 183 Session Progress provisional response to I-CSCF.

Table 8.4.6-14: 183 Session Progress (S-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

15. **183 Session Progress (I-CSCF to S-CSCF) – see example in Table 8.4.6-15**

     I-CSCF forwards the 183 Session Progress provisional response to S-CSCF#1.

Table 8.4.6-15: 183 Session Progress (I-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

16. **183 Session Progress (S-S#2 to MO) – see example in Table 8.4.6-16**

     S-CSCF#1 forwards the 183 Session Progress to the originator, as per the originating procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-16: 183 Session Progress (S-S#2 to MO)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Remote-Party-ID:
Anonymity:
Require
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

17. **PRACK (MO to S-S#2) – see example in Table 8.4.6-17**

The originator decides the final set of media streams, and includes this information in the PRACK request sent to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-17: PRACK (MO to S-S#2)

```
PRACK sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

18. **PRACK (S-CSCF to S-CSCF) – see example in Table 8.4.6-18**

S-CSCF#1 forwards the PRACK request to S-CSCF#2.

Table 8.4.6-18: PRACK (S-CSCF to S-CSCF)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

19. **PRACK (S-S#2 to MT) – see example in Table 8.4.6-19**

S-CSCF#2 forwards the PRACK request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-19: PRACK (S-S#2 to MT)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

20. **200 OK (MT to S-S#2) – see example in Table 8.4.6-20**

The terminating endpoint responds to the PRACK request (19) with a 200 OK response.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-20: 200 OK (MT to S-S#2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
Content-length: 0
```

21. **200 OK (S-CSCF to S-CSCF) – see example in Table 8.4.6-21**

    S-CSCF#2 forwards the 200 OK response to S-CSCF#1.

Table 8.4.6-21: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

22. **200 OK (S-S#2 to MO) – see example in Table 8.4.6-22**

    S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-22: 200 OK (S-S#2 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

23. **COMET (MO to S-S#2) – see example in Table 8.4.6-23**

    When the originating endpoint has completed the resource reservation procedures, it sends the COMET request to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-23: COMET (MO to S-S#2)

```
COMET sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

24. **COMET (S-CSCF to S-CSCF) – see example in Table 8.4.6-24**

S-CSCF#1 forwards the COMET request to S-CSCF#2.

Table 8.4.6-24: COMET (S-CSCF to S-CSCF)

```
COMET sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

25. **COMET (S-S#2 to MT) – see example in Table 8.4.6-25**

S-CSCF#2 forwards the COMET request to the terminating endpoint, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-25: COMET (S-S#2 to MT)

```
COMET sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

26. **200 OK (MT to S-S#2) – see example in Table 8.4.6-26**

   The terminating endpoint responds to the COMET request (25) with a 200 OK response.

   NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-26: 200 OK (MT to S-S#2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-length: 0
```

27. **200 OK (S-CSCF to S-CSCF) – see example in Table 8.4.6-27**

   S-CSCF#2 forwards the 200 OK response to S-CSCF#1.

Table 8.4.6-27: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

28. **200 OK (S-S#2 to MO) – see example in Table 8.4.6-28**

   S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-28: 200 OK (S-S#2 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

29. **180 Ringing (MT to S-S#2) – see example in Table 8.4.6-29**

The terminating endpoint may optionally send a 180 Ringing provisional response indicating alerting is in progress. This response is sent by the termination procedure to S-CSCF#2.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-29: 180 Ringing (MT to S-S#2)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
RSeq: 9022
Content-length: 0
```

30. **Service Control**

The S-CSCF validates the service profile and performs any service control required for this subscriber.

31. **180 Ringing (S-CSCF to I-CSCF) – see example in Table 8.4.6-31**

S-CSCF#2 forwards the 180 Ringing response to I-CSCF.

Table 8.4.6-31: 180 Ringing (S-CSCF to I-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

32. **180 Ringing (I-CSCF to S-CSCF) – see example in Table 8.4.6-32**

I-CSCF forwards the 180 Ringing response to S-CSCF#1.

Table 8.4.6-32: 180 Ringing (I-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

33. **Service Control**

   The S-CSCF validates the service profile and performs any service control required for this subscriber.

34. **180 Ringing (S-S#2 to MO) – see example in Table 8.4.6-34**

   S-CSCF#1 forwards the 180 Ringing response to the originator, per the origination procedure.

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-34: 180 Ringing (S-S#2 to MO)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

35. **PRACK (MO to S-S#2) – see example in Table 8.4.6-35**

   The originator acknowledges the 180 Ringing provisional response (34) with a PRACK request.

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-35: PRACK (MO to S-S#2)

```
PRACK sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack: 9022 127 INVITE
Content-length: 0
```

36. **PRACK (S-CSCF to S-CSCF) – see example in Table 8.4.6-36**

   S-CSCF#1 forwards the PRACK request to S-CSCF#2.

Table 8.4.6-36: PRACK (S-CSCF to S-CSCF)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

37. **PRACK (S-S#2 to MT) – see example in Table 8.4.6-37**

   S-CSCF#2 forwards the PRACK request to the terminating endpoint.

   NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-37: PRACK (S-S#2 to MT)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

38. **200 OK (MT to S-S#2) – see example in Table 8.4.6-38**

   The terminating endpoint responds to the PRACK request (37) with a 200 OK response.

   NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-38: 200 OK (MT to S-S#2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-length: 0
```

39. **200 OK (S-CSCF to S-CSCF) – see example in Table 8.4.6-39**

   S-CSCF#2 forwards the 200 OK response to S-CSCF#1.

Table 8.4.6-39: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

40. **200 OK (S-S#2 to MO) – see example in Table 8.4.6-40**

    S-CSCF#1 forwards the 200 OK response to the originating endpoint.

    NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-40: 200 OK (S-S#2 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

41. **200 OK (MT to S-S#2) – see example in Table 8.4.6-41**

    The final response, 200 OK, is sent by the terminating endpoint over the signalling path. This is typically generated when the subscriber has accepted the incoming session attempt. The response is sent to S-CSCF#2 per the termination procedure.

    NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-41: 200 OK (MT to S-S#2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-length: 0
```

42. **Service Control**

    S-CSCF#2 performs whatever service control logic is appropriate for this session completion

43. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.6-43**

    The 200 OK response is forwarded to the I-CSCF.

Table 8.4.6-43: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

44. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.6-44**

The 200 OK response is forwarded to S-CSCF#1.

Table 8.4.6-44: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

45. **Service Control**

S-CSCF#1 performs whatever service control logic is appropriate for this session completion

46. **200 OK (S-S#2 to MO) – see example in Table 8.4.6-46**

The 200 OK response is returned to the originating endpoint, by the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-46: 200 OK (S-S#2 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

47. **ACK (MO to S-S#2) – see example in Table 8.4.6-47**

The originating endpoint sends the final acknowledgement to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-47: ACK (MO to S-S#2)

```
ACK sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 ACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-length: 0
```

48. **ACK (S-CSCF to S-CSCF) – see example in Table 8.4.6-48**

   S-CSCF#1 forwards the ACK request to S-CSCF#2.

Table 8.4.6-48: ACK (S-CSCF to S-CSCF)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

49. **ACK (S-S#2 to MT) – see example in Table 8.4.6-49**

   S-CSCF#2 forwards the ACK request to the terminating endpoint, as per the termination procedure.

   NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 8.4.6-49: ACK (S-S#2 to MT)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

8.4.7   (S-S#3) PSTN Termination performed by home network of originator

The Serving-CSCF handling session origination performs an analysis of the destination address, and determines that it belongs will result in a PSTN termination. The request is therefore forwarded to a local BGCF. The BGCF performs further analysis of the destination address, combined with information of agreements between operators for optimum Gateway selection, and decides to do the PSTN termination locally. The BGCF therefore allocates a MGCF within the home network, and sends the request to it.

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

MO#1a Mobile origination, roaming, without firewall. The "Originating Network" of S-S#3 is therefore a visited network.

MO#1b Mobile origination, roaming, with firewall in home network. The "Originating Network" of S-S#3 is therefore a visited network.

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

PSTN-O PSTN origination. The "Originating Network" of S-S#3 is the home network. The element labeled S-CSCF#1 is the MGCF of the PSTN-O procedure.

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

PSTN-T PSTN termination.
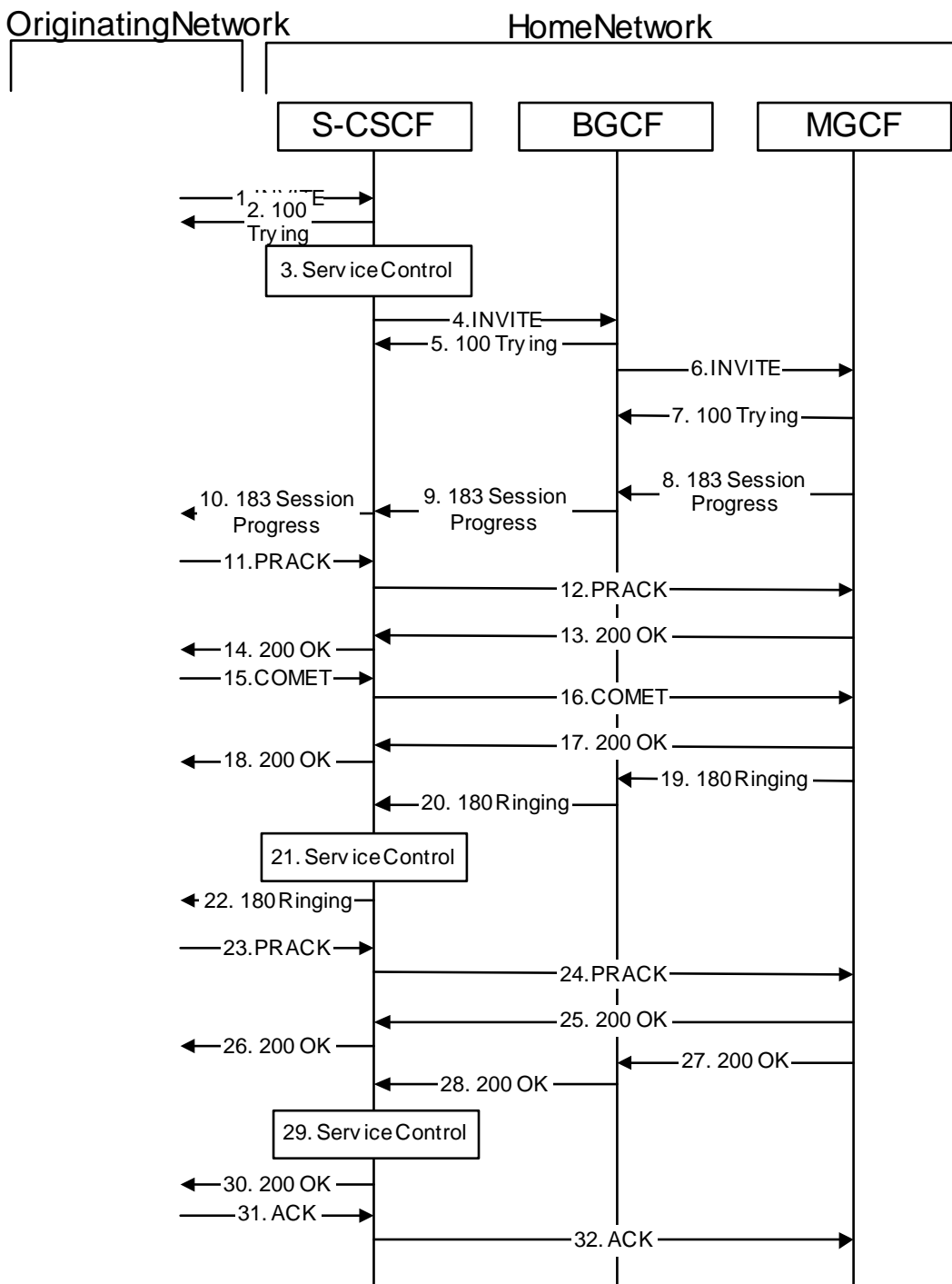
Figure 8.4.7 – S-S#3

Procedure S-S#3 is as follows:

1. **INVITE (MO to S-S#3) – see example in Table 8.4.7-1**

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

    NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-1: INVITE (MO to S-S#3)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (S-S#3 to MO) – see example in Table 8.4.7-2**

    S-CSCF#1 responds to the INVITE request (1) with a 100 Trying provisional response.

    NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-2: 100 Trying (S-S#3 to MO)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

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

    S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

For this example, assume the subscriber is not allowed video.

4. **INVITE (S-CSCF to BGCF) – see example in Table 8.4.7-4**

S-CSCF#1 performs an analysis of the destination address, and determines the destination is on the PSTN. S-CSCF forwards the INVITE request to the BGCF in the local network.

Table 8.4.7-4: INVITE (S-CSCF to BGCF)

```
INVITE sip:bgcf.home.net SIP/2.0
Via: SIP/2.0/UDP sip:scscf.home.net SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Route: In the case of a TEL-URL, it has to be translated to a globally routable SIP-URL before sending the INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

5. **100 Trying (BGCF to S-CSCF) – see example in Table 8.4.7-5**

BGCF sends a 100 Trying provisional response to S-CSCF.

Table 8.4.7-5: 100 Trying (BGCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **INVITE (BGCF to MGCF) – see example in Table 8.4.7-6**

BGCF analyzes the destination address, and allocates a MGCF to handle the termination. BGCF forwards the INVITE request to the MGCF.

Table 8.4.7-6: INVITE (BGCF to MGCF)

```
INVITE sip:mgcf39.home.net SIP/2.0
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

NOTE: The BGCF does not add itself to the Record-Route header, as it has no need to remain in the signalling path once the session is established.

7. **100 Trying (MGCF to BGCF) – see example in Table 8.4.7-7**

MGCF responds to the INVITE request (6) with a 100 Trying provisional response.

Table 8.4.7-7: 100 Trying (MGCF to BGCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

8. **183 Session Progress (MGCF to BGCF) – see example in Table 8.4.7-8**

The MGCF returns the media stream capabilities of the destination along the signalling path in a 183 Session Progress provisional response.

Table 8.4.7-8: 183 Session Progress (MGCF to BGCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:04/11@mgcf39.home.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

9. **183 Session Progress (BGCF to S-CSCF) – see example in Table 8.4.7-9**

    BGCF forwards the 183 Session Progress provisional response to S-CSCF.

Table 8.4.7-9: 183 Session Progress (BGCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

10. **183 Session Progress (S-S#3 to MO) – see example in Table 8.4.7-10**

    S-CSCF#1 forwards the 183 Session Progress to the originator, as per the originating procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-10: 183 Session Progress (S-S#3 to MO)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

11. **PRACK (MO to S-S#3) – see example in Table 8.4.7-11**

The originator decides the final set of media streams, and includes this information in the PRACK request sent to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-11: PRACK (MO to S-S#3)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:04/11@mgcf39.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

12. **PRACK (S-CSCF to MGCF) – see example in Table 8.4.7-12**

S-CSCF forwards the PRACK request to MGCF.

Table 8.4.7-12: PRACK (S-CSCF to MGCF)

```
PRACK sip:04/11@mgcf39.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

13. **200 OK (MGCF to S-CSCF) – see example in Table 8.4.7-13**

The MGCF responds to the PRACK request (12) with a 200 OK response.

Table 8.4.7-13: 200 OK (MGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:04/11@mgcf39.home.net
Content-length: 0
```

14. **200 OK (S-S#3 to MO) – see example in Table 8.4.7-14**

S-CSCF#1 forwards the 200 OK response to the originating endpoint.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-14: 200 OK (S-S#3 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

15. **COMET (MO to S-S#3) – see example in Table 8.4.7-15**

When the originating endpoint has completed the resource reservation procedures, it sends the COMET request to S-CSCF#1 by the origination procedures.

NOTE: There is a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-15: COMET (MO to S-S#3)

```
COMET sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:04/11@mgcf39.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

16. **COMET (S-CSCF to MGCF) – see example in Table 8.4.7-16**

    S-CSCF forwards the COMET request to MGCF.

Table 8.4.7-16: COMET (S-CSCF to MGCF)

```
COMET sip:mgcf39.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

17. **200 OK (MGCF to S-CSCF) – see example in Table 8.4.7-17**

    The MGCF responds to the COMET request (16) with a 200 OK response.

Table 8.3.4-17: 200 OK (MGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:04/11@mgcf39.home.net
Content-length: 0
```

18. **200 OK (S-S#3 to MO) – see example in Table 8.4.7-18**

    S-CSCF#1 forwards the 200 OK response to the originating endpoint.

    NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-18: 200 OK (S-S#3 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

19. **180 Ringing (MGCF to BGCF) – see example in Table 8.4.7-19**

    The MGCF may optionally send a 180 Ringing provisional response indicating alerting is in progress. This response is sent by the termination procedure to BGCF.

Table 8.4.7-19: 180 Ringing (MGCF to BGCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home.net
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:04/11@mgcf39.home.net
RSeq: 9022
Content-length: 0
```

20. **180 Ringing (BGCF to S-CSCF) – see example in Table 8.4.7-20**

    BGCF forwards the 180 Ringing response to S-CSCF.

Table 8.4.7-20: 180 Ringing (BGCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

21. **Service Control**

The S-CSCF validates the service profile and performs any service control required for this subscriber.

22. **180 Ringing (S-S#3 to MO) – see example in Table 8.4.7-22**

S-CSCF forwards the 180 Ringing response to the originator, per the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-22: 180 Ringing (S-S#3 to MO)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

23. **PRACK (MO to S-S#3) – see example in Table 8.4.7-23**

The originator acknowledges the 180 Ringing provisional response (22) with a PRACK request.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-23: PRACK (MO to S-S#3)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:04/11@mgcf39.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack: 9022 127 INVITE
Content-length: 0
```

24. **PRACK (S-CSCF to MGCF) – see example in Table 8.4.7-24**

S-CSCF forwards the PRACK request to MGCF.

Table 8.4.7-24: PRACK (S-CSCF to MGCF)

```
PRACK sip:04/11@mgcf39.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

25. **200 OK (MGCF to S-CSCF) – see example in Table 8.4.7-25**

The MGCF responds to the PRACK request (24) with a 200 OK response.

Table 8.4.7-25: 200 OK (MGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-length: 0
```

26. **200 OK (S-S#3 to MO) – see example in Table 8.4.7-26**

    S-CSCF forwards the 200 OK response to the originating endpoint.

    NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-26: 200 OK (S-S#3 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

27. **200 OK (MGCF to BGCF) – see example in Table 8.4.7-27**

    The final response, 200 OK, is sent by the MGCF over the signalling path when the subscriber has accepted the incoming session attempt.

Table 8.4.7-27: 200 OK (MGCF to BGCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:04/11@mgcf39.home.net
Content-length: 0
```

28. **200 OK (BGCF to S-CSCF) – see example in Table 8.4.7-28**

    The 200 OK response is forwarded to the S-CSCF.

Table 8.4.7-28: 200 OK (BGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

29. **Service Control**

S-CSCF performs whatever service control logic is appropriate for this session completion

30. **200 OK (S-S#3 to MO) – see example in Table 8.4.7-30**

The 200 OK is returned to the originating endpoint, by the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-30: 200 OK (S-S#3 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

31. **ACK (MO to S-S#3) – see example in Table 8.4.7-31**

The originating endpoint sends the final acknowledgement to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.7-31: ACK (MO to S-S#3)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:04/11@mgcf39.home.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 ACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-length: 0
```

32. **ACK (S-CSCF to MGCF) – see example in Table 8.4.7-32**

S-CSCF#1 forwards the ACK request to MGCF.

Table 8.4.7-32: ACK (S-CSCF to MGCF)

```
ACK sip:mgcf39.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

8.4.8 (S-S#4) PSTN Termination performed by different operator than origination

The Serving-CSCF handling session origination performs an analysis of the destination address, and determines that it belongs will result in a PSTN termination. The request is therefore forwarded to a local BGCF (BGCF#1). BGCF#1 performs further analysis of the destination address, combined with information of agreements between operators for optimum Gateway selection, and decides to do the PSTN termination in a different operator's network. BGCF#1 therefore forwards the request to a BGCF in the terminating operator's network, BGCF#2. BGCF#2 allocates a MGCF within the its network, and sends the request to it.

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

MO#1a Mobile origination, roaming, without firewall. The "Originating Network" of S-S#4 is therefore a visited network.

MO#1b Mobile origination, roaming, with firewall in home network. The "Originating Network" of S-S#4 is therefore a visited network.

MO#2 Mobile origination, located in home service area. The "Originating Network" of S-S#4 is therefore the home network.

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

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

PSTN-T PSTN termination.
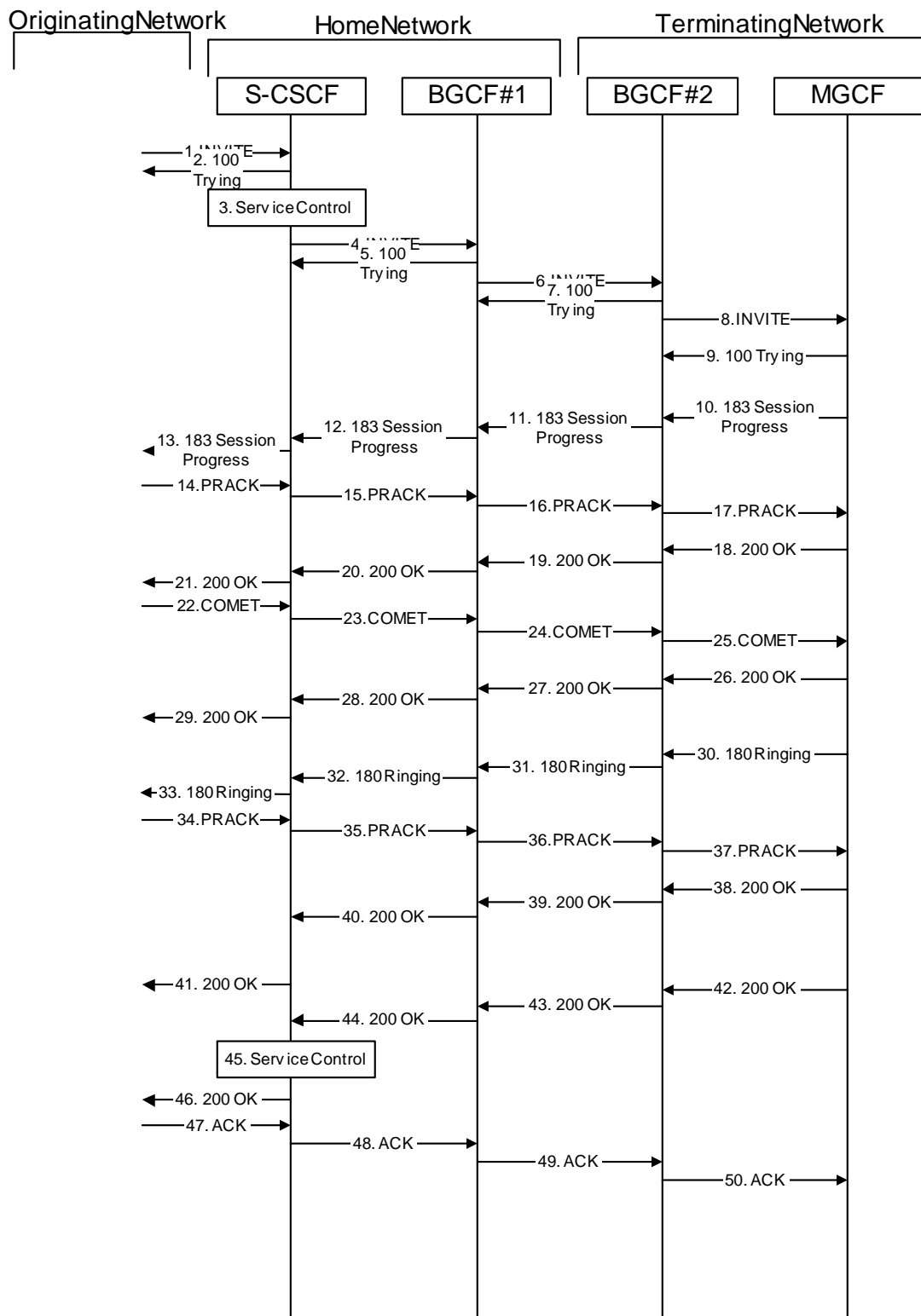
Figure 8.4.8 – S-S#4

Procedure S-S#4 is as follows:

1. **INVITE (MO to S-S#4) – see example in Table 8.4.8-1**

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

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-1: INVITE (MO to S-S#4)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
m=audio 3458 RTP/AVP 97 96 0 15
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (S-S#4 to MO) – see example in Table 8.4.8-2**

    S-CSCF#1 responds to the INVITE request (1) with a 100 Trying provisional response.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-2: 100 Trying (S-S#4 to MO)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

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

    S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

    For this example, assume the subscriber is not allowed video.

4. **INVITE (S-CSCF to BGCF) – see example in Table 8.4.8-4**

S-CSCF#1 performs an analysis of the destination address, and determines the destination is on the PSTN. S-CSCF#1 forwards the INVITE request to the BGCF in the local network.

Table 8.4.8-4: INVITE (S-CSCF to BGCF)

```
INVITE sip:bgcf1.home.net SIP/2.0
Via: SIP/2.0/UDP sip:scscf.home.net SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

5. **100 Trying (BGCF to S-CSCF) – see example in Table 8.4.8-5**

   BGCF#1 sends a 100 Trying provisional response to S-CSCF#1.

Table 8.4.8-5: 100 Trying (BGCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **INVITE (BGCF to BGCF) – see example in Table 8.4.8-6**

   BGCF#1 analyzes the destination address, and the inter-operator agreements for optimal PSTN termination, and selects the network operator that can best terminate this session. BGCF#1 forwards the INVITE request to the BGCF (BGCF#2) in the network that will handle the session termination.

Table 8.4.8-6: INVITE (BGCF to BGCF)

```
INVITE sip:bgcf2.term.net SIP/2.0
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:bgcf1.home.net, sip:scscf.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

7. **100 Trying (BGCF to BGCF) – see example in Table 8.4.8-7**

   BGCF#2 responds to the INVITE request (6) with a 100 Trying provisional response.

Table 8.4.8-7: 100 Trying (BGCF to BGCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

8. **INVITE (BGCF to MGCF) – see example in Table 8.4.8-8**

   BGCF#2 allocates a Media Gateway Controller, and forwards the INVITE request to that MGCF.

Table 8.4.8-8: INVITE (BGCF to MGCF)

```
INVITE sip:mgcf72.term.net SIP/2.0
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
a=
a=
a=
a=
```

9. **100 Trying (MGCF to BGCF) – see example in Table 8.4.8-9**

   MGCF sends a 100 Trying provisional response.

Table 8.4.8-9: 100 Trying (MGCF to BGCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

10. **183 Session Progress (MGCF to BGCF) – see example in Table 8.4.8-10**

    MGCF returns the media stream capabilities of the destination in a 183 Session Progress provisional response.

Table 8.4.8-10: 183 Session Progress (MGCF to BGCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:04/11@mgcf72.term.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=audio 0 RTP/AVP 97 96 0 15
```

11. **183 Session Progress (BGCF to BGCF) – see example in Table 8.4.8-11**

    BGCF#2 forwards the 183 Session Progress provisional response to BGCF#1.

Table 8.4.8-11: 183 Session Progress (BGCF to BGCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

12. **183 Session Progress (BGCF to S-CSCF) – see example in Table 8.4.8-12**

    BGCF#1 forwards the 183 Session Progress provisional response to S-CSCF.

Table 8.4.8-12: 183 Session Progress (BGCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

13. **183 Session Progress (S-S#4 to MO) – see example in Table 8.4.8-13**

     S-CSCF#1 forwards the 183 Session Progress response to the originator, as per the originating procedure.

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one
             of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-13: 183 Session Progress (S-S#4 to MO)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

14. **PRACK (MO to S-S#4) – see example in Table 8.4.8-14**

The originator decides the final set of media streams, and includes this information in the PRACK request sent to S-CSCF by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-14: PRACK (MO to S-S#4)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:bgcf1.home.net, sip:bgcf2.term.net, sip:04/11@mgcf72.term.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 128 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack: 9021 127 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=audio 0 RTP/AVP 97 96 0 15
```

15. **PRACK (S-CSCF to BGCF) – see example in Table 8.4.8-16**

S-CSCF forwards the PRACK request to BGCF#1.

Table 8.4.8-15: PRACK (S-CSCF to BGCF)

```
PRACK sip:bgcf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:bgcf2.term.net, sip:04/11@mgcf72.term.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

16. **PRACK (BGCF to BGCF) – see example in Table 8.4.8-16**

BGCF#1 forwards the PRACK request to BGCF#2.

Table 8.4.8-16: PRACK (BGCF to BGCF)

```
PRACK sip:bgcf2.term.net SIP/2.0
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:04/11@mgcf72.term.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

17. **PRACK (BGCF to MGCF) – see example in Table 8.4.8-17**

BGCF#2 forwards the PRACK request to the MGCF.

Table 8.4.8-17: PRACK (BGCF to MGCF)

```
PRACK sip:04/11@mgcf72.term.net SIP/2.0
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

18. **200 OK (MGCF to BGCF) – see example in Table 8.4.8-18**

The MGCF responds to the PRACK request (17) with a 200 OK response.

3GPP

Table 8.4.8-18: 200 OK (MGCF to BGCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:04/11@mgcf72.term.net
Content-length: 0
```

19. **200 OK (BGCF to BGCF) – see example in Table 8.4.8-19**

   BGCF#2 forwards the 200 OK response to BGCF#1.

Table 8.4.8-19: 200 OK (BGCF to BGCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

20. **200 OK (BGCF to S-CSCF) – see example in Table 8.4.8-20**

   BGCF#1 forwards the 200 OK response to S-CSCF.

Table 8.4.8-20: 200 OK (BGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

21. **200 OK (S-S#4 to MO) – see example in Table 8.4.8-21**

   S-CSCF forwards the 200 OK response to the originating endpoint.

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one
            of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-21: 200 OK (S-S#4 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

22. **COMET (MO to S-S#4) – see example in Table 8.4.8-22**

When the originating endpoint has completed the resource reservation procedures, it sends the COMET request to S-CSCF#1 by the origination procedures.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-22: COMET (MO to S-S#4)

```
COMET sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:bgcf1.home.net, sip:bgcf2.term.net, sip:04/11@mgcf72.term.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 129 COMET
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=audio 0 RTP/AVP 97 96 0 15
```

23. **COMET (S-CSCF to BGCF) – see example in Table 8.4.8-23**

S-CSCF forwards the COMET request to BGCF#1.

Table 8.4.8-23: COMET (S-CSCF to BGCF)

```
COMET sip:bgcf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:bgcf2.term.net, sip:04/11@mgcf72.term.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

24. **COMET (BGCF to BGCF) – see example in Table 8.4.8-24**

BGCF#1 forwards the COMET request to BGCF#2.

Table 8.4.8-24: COMET (BGCF to BGCF)

```
COMET sip:bgcf2.term.net SIP/2.0
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:04/11@mgcf72.term.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

25. **COMET (BGCF to MGCF) – see example in Table 8.4.8-25**

    BGCF#2 forwards the COMET request to the MGCF.

Table 8.4.8-25: COMET (BGCF to MGCF)

```
COMET sip:04/11@mgcf72.term.net SIP/2.0
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

26. **200 OK (MGCF to BGCF) – see example in Table 8.4.8-26**

    The MGCF responds to the COMET request (25) with a 200 OK response.

Table 8.4.8-26: 200 OK (MGCG to BGCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:04/11@mgcf72.term.net
Content-length: 0
```

27. **200 OK (BGCF to BGCF) – see example in Table 8.4.8-27**

   BGCF#2 forwards the 200 OK response to BGCF#1.

Table 8.4.8-27: 200 OK (BGCF to BGCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

28. **200 OK (BGCF to S-CSCF) – see example in Table 8.4.8-28**

   BGCF#1 forwards the 200 OK response to S-CSCF.

Table 8.4.8-28: 200 OK (BGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

29. **200 OK (S-S#4 to MO) – see example in Table 8.4.8-29**

   S-CSCF#1 forwards the 200 OK response to the originating endpoint.

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one
           of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-29: 200 OK (S-S#4 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

30. **180 Ringing (MGCF to BGCF) – see example in Table 8.4.8-30**

The MGCF may optionally send a 180 Ringing provisional response indicating alerting is in progress.

Table 8.4.8-30: 180 Ringing (MGCF to BGCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:04/11@mgcf72.term.net
RSeq: 9022
Content-length: 0
```

31. **180 Ringing (BGCF to BGCF) – see example in Table 8.4.8-31**

    BGCF#2 forwards the 180 Ringing response to BGCF#1.

Table 8.4.8-31: 180 Ringing (BGCF to BGCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

32. **180 Ringing (BGCF to S-CSCF) – see example in Table 8.4.8-32**

    BGCF#1 forwards the 180 Ringing response to S-CSCF.

Table 8.4.8-32: 180 Ringing (BGCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

33. **180 Ringing (S-S#4 to MO) – see example in Table 8.4.8-33**

    S-CSCF#1 forwards the 180 Ringing response to the originator, per the origination procedure.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-33: 180 Ringing (S-S#4 to MO)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

34. **PRACK (MO to S-S#4) – see example in Table 8.4.8-34**

The originator acknowledges the 180 Ringing provisional response (33) with a PRACK request.

NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-34: PRACK (MO to S-S#4)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:bgcf1.home.net, sip:bgcf2.term.net, sip:04/11@mgcf72.term.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack: 9022 127 INVITE
Content-length: 0
```

35. **PRACK (S-CSCF to BGCF) – see example in Table 8.4.8-35**

S-CSCF#1 forwards the PRACK request to BGCF#1.

Table 8.4.8-35: PRACK (S-CSCF to BGCF)

```
PRACK sip:bgcf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:bgcf2.term.net, sip:04/11@mgcf72.term.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

36. **PRACK (BGCF to BGCF) – see example in Table 8.4.8-36**

BGCF#1 forwards the PRACK request to BGCF#2.

Table 8.4.8-36: PRACK (BGCF to BGCF)

```
PRACK sip:bgcf2.term.net SIP/2.0
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:04/11@mgcf72.term.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

37. **PRACK (BGCF to MGCF) – see example in Table 8.4.8-37**

   BGCF#2 forwards the PRACK request to the MGCF.

Table 8.4.8-37: PRACK (BGCF to MGCF)

```
PRACK sip:04/11@mgcf72.term.net SIP/2.0
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

38. **200 OK (MGCF to BGCF) – see example in Table 8.4.8-38**

   The MGCF responds to the PRACK request (37) with a 200 OK response.

Table 8.4.8-38: 200 OK (MGCF to BGCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:04/11@mgcf72.term.net
Content-length: 0
```

39. **200 OK (BGCF to BGCF) – see example in Table 8.4.8-39**

   BGCF#2 forwards the 200 OK response to BGCF#1.

Table 8.4.8-39: 200 OK (BGCG to BGCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

40. **200 OK (BGCF to S-CSCF) – see example in Table 8.4.8-40**

BGCF#1 forwards the 200 OK response to S-CSCF.

Table 8.4.8-40: 200 OK (BGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

41. **200 OK (S-S#4 to MO) – see example in Table 8.4.8-41**

     S-CSCF forwards the 200 OK to the originating endpoint.

     NOTE: There are a number of different origination flows, and the table represents a typical example of what one
              of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-41: 200 OK (S-S#4 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

42. **200 OK (MGCF to BGCF) – see example in Table 8.4.8-42**

     The final response, 200 OK, is sent by the MGCF when the subscriber has accepted the incoming session
     attempt.

Table 8.4.8-42: 200 OK (MGCF to BGCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
From:
To:
Call-ID:
CSeq: 127 INVITE
Contact: sip:04/11@mgcf72.term.net
Content-length: 0
```

43. **200 OK (BGCF to BGCF) – see example in Table 8.4.8-43**

     BGCF#2 forwards the 200 OK final response to BGCF#1.

Table 8.4.8-43: 200 OK (BGCF to BGCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

44. **200 OK (BGCF to S-CSCF) – see example in Table 8.4.8-44**

   BGCF#1 forwards the 200 OK final response to S-CSCF.

Table 8.4.8-44: 200 OK (BGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

45. **Service Control**

   S-CSCF performs whatever service control logic is appropriate for this session completion

46. **200 OK (S-S#4 to MO) – see example in Table 8.4.8-46**

   The 200 OK response is returned to the originating endpoint, by the origination procedure.

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-46: 200 OK (S-S#4 to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

47. **ACK (MO to S-S#4) – see example in Table 8.4.8-47**

   The originating endpoint sends the final acknowledgement to S-CSCF by the origination procedures.

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 is assumed.

Table 8.4.8-47: ACK (MO to S-S#4)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:bgcf1.home.net, sip:bgcf2.term.net, sip:04/11@mgcf72.term.net
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 ACK
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-length: 0
```

48. **ACK (S-CSCF to BGCF) – see example in Table 8.4.8-48**

   S-CSCF forwards the ACK request to BGCF#1.

Table 8.4.8-48: ACK (S-CSCF to BGCF)

```
ACK sip:bgcf2.term.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:bgcf2.term.net, sip:04/11@mgcf72.term.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

49. **ACK (BGCF to BGCF) – see example in Table 8.4.8-49**

BGCF#1 forwards the ACK request to BGCF#2.

Table 8.4.8-49: ACK (BGCF to BGCF)

```
ACK sip:bgcf2.term.net SIP/2.0
Via: SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:04/11@mgcf72.term.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

50. **ACK (BGCF to MGCF) – see example in Table 8.4.8-50**

BGCF#2 forwards the ACK request to the MGCF.

Table 8.4.8-50: ACK (BGCF to MGCF)

```
ACK sip:04/11@mgcf72.term.net SIP/2.0
Via: SIP/2.0/UDP bgcf2.term.net, SIP/2.0/UDP bgcf1.home.net, SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP
    pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:bgcf2.term.net, sip:bgcf1.home.net, sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

8.5 Sample MM flows: addition of additional media streams

8.5.1 Sample MM session flow - addition of another media originator and terminator are both roaming and operated by different networks (without I-CSCF providing configuration independence).

Both networks are without I-CSCF providing configuration independence. The UE has already established an IM CN voice session and is generating an invite to add another video media to the already established session.
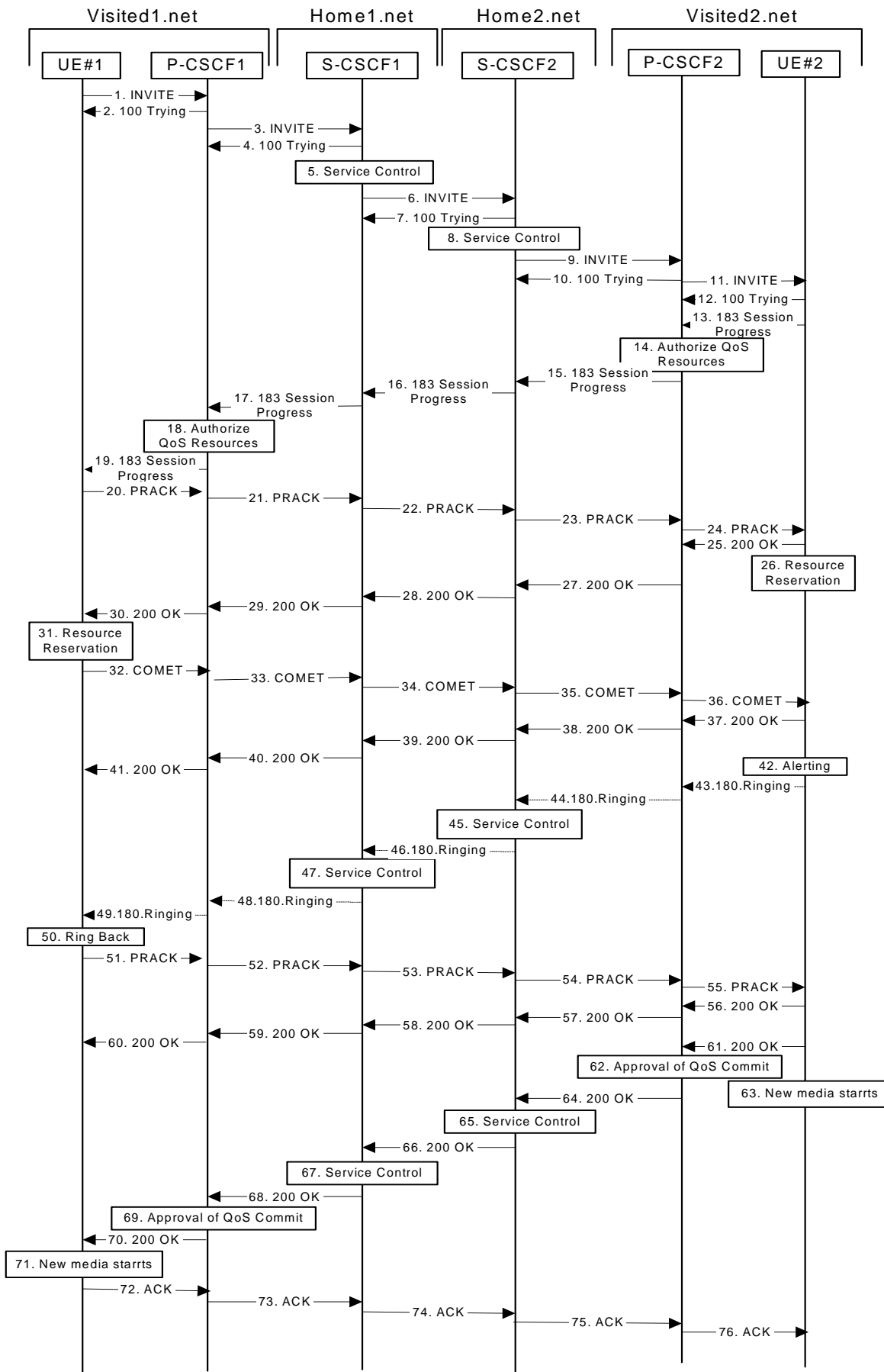
Figure 8.5.1 Sample MM session flow—Addition of another media

1. **INVITE (UE1 to P-CSCF1) – see example in Table 8.5.1-1**

   UE#1 sends a SIP INVITE request, containing new SDP for the new video media and including the original SDP, to P-CSCF1, which is pcscf1.visited1.net in its visited network.

Table 8.5.1-1 INVITE (UE1 to P-CSCF1)

```
INVITE sip:token1@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111;time=36123E5B;seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 132 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907166275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=video 9544 RTP/AVP 31
a=rtpmap:31 H261/90000
a=qos:mandatory sendrecv
```

Request-URI: contains the keyed number from the user.

Via: contains the IP address or FQDN of the originating UE.

Remote-Party-ID: contains the public identity of the UE. The Display name is optional.

From:, To:, Call-ID: follow the recommendations of draft-ietf-sip-privacy-01, even though anonymity is not being requested for this session.

Cseq: is a random starting number.

Contact: is the IP address or FQDN of the originating UE.

2. **100 Trying (P-CSCF1 to UE1) - see example in Table 8.5.1-2**

   P-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.5.1-2: 100 Trying (P-CSCF1 to UE1)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **INVITE (P-CSCF1 to S-CSCF1) - see example in Table 8.5.1-3**

   The INVITE request is sent by the P-CSCF to the next hop scscf1.home1.net, which is in UE's home network. Because this a re-invite, so the I-CSCF1 is not involved in sip transaction.

Table 8.5.1-3: INVITE (P-CSCF1 to S-CSCF1)

```
INVITE sip:scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home2.net, sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

Route: P-CSCF knows the request routing from the previous sip transactions.

Contact: a locally defined value that identifies the UE.

Request-URI: the first component in the remembered Path header from Registration.

4. **100 Trying (S-CSCF1 to P-CSCF1) - see example in Table 8.5.1-4**

   S-CSCF sends the 100 Trying provisional response to P-CSCF.

Table 8.5.1-4: 100 Trying (S-CSCF1 to P-CSCF1)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

5. **Service Control**

   S-CSCF validates the service profile, and performs any origination service control required for this subscriber.

   S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

6. **INVITE (S-CSCF1 to S-CSCF2) - see example in Table 8.5.1-6**

   S-CSCF1 sends the INVITE request to UE's serving CSCF-cscf2.home2.net, which is in the callee (UE2)'s home network. Because this is a re-invite, so the I-CSCF2 is not involved in the sip transaction.

Table 8.5.1-6: INVITE (S-CSCF1 to S-CSCF2)

```
INVITE sip:scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf1.home1.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

7. **100 Trying (S-CSCF2 to S-CSCF1) - see example in Table 8.5.1-7**

    S-CSCF1 receives a 100 Trying provisional response, as specified by the S-CSCF to S-CSCF procedures.

    NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.5.1-7: 100 Trying (S-CSCF2 to S-CSCF1)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

8. **Service Control**

    S-CSCF2 validates the service profile, and performs any origination service control required for this subscriber.

    S-CSCF#2 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

9. **INVITE (S-CSCF2 to P-CSCF2) - see example in Table 8.5.1-9**

    S-CSCF2 forwards the INVITE request tocallee's P-CSCF pcscf2.visited2.net which is in the UE2's visited network, called visited2.net

Editor's Note: Need to represent the ability of the S-CSCF to take part in the negotiation of the media streams (by removing entries) in some manner.

Editor's Note: Need for additional headers to transport e.g. Billing-Correlation-Identifier is FFS.

Table 8.5.1-9: INVITE (S-CSCF2 to P-CSCF2)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

10. **100 Trying (P-CSCF2 to S-CSCF2) - see example in Table 8.5.1-10**

    P-CSCF sends a 100 Trying provisional response back to S-CSCF2.

    NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of
             what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.5.1-10: 100 Trying (P-CSCF2 to S-CSCF2)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

11. **INVITE (P-CSCF2 to UE2) - see example in Table 8.5.1-11**

    P-CSCF determines the UE address from the value of the Request-URI, and forwards the INVITE request to the
    UE.

Table 8.5.1-11: INVITE (P-CSCF2 to UE2)

```
INVITE sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token2
Media-Authorization: 00200001001001017063663322e78797a2e6e6574000c020133315331343363233
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token2@pcscf2.visited2.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14623".

12. **100 Trying (UE2 to P-CSCF2) - see example in Table 8.5.1-12**

P-CSCF receives a 100 Trying provisional response back to S-CSCF2.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.5.1-12: 100 Trying (UE2 to P-CSCF2)

```
SIP/2.0 100 Trying
Via: pcscf2.visited2.net;branch=token2 SIP/2.0/UDP,
From:
To:
Call-ID:
CSeq:
```

13. **183 Session Progress (UE2 to P-CSCF2) - see example in Table 8.5.1-13**

The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response

Table 8.5.1-13: 183 Session Progress response (UE2 to P-CSCF2)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token2
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9022
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907166275 0
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=video 7544 RTP/AVP 31
a=rtpmap:31 H261/90000
a=qos:mandatory sendrecv confirm
```

14. **Authorize QoS Resources**

   P-CSCF2 authorizes the resources necessary for this new media

15. **183 Session Progress (P-CSCF2 to S-CSCF2) - see example in Table 8.5.1-15**

   P-CSCF2 forwards the 183 Session Progress response to S-CSCF2

Table 8.5.1-15: 183 Session Progress (P-CSCF2 to S-CSCF2)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

   Record-Route: header entries to the left of I-CSCF's entry are reversed and translated.

16. **183 Session Progress (S-CSCF2 to S-CSCF1) - see example in Table 8.5.1-16**

S-CSCF2 forwards the 183 Session Progress response to caller's S-CSCF

Table 8.5.1-16: 183 Session Progress (S-CSCF2 to S-CSCF1)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

17. **183 Session Progress (S-CSCF1 to P-CSCF1) - see example in Table 8.5.1-17**

S-CSCF1 forwards the 183 Session Progress response to the caller's P-CSCF

Table 8.5.1-17: 183 Session Progress (S-CSCF1 to P-CSCF1)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

18. **Authorize QoS Resources**

P-CSCF1 authorizes the resources necessary for this new media

19. **183 Session Progress (P-CSCF1 to UE1) - see example in Table 8.5.1-19**

P-CSCF forwards the 183 Session Progress response to the originating endpoint

Table 8.5.1-19: 183 Session Progress (P-CSCF1 to UE1)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Media-Authorization: 0020000100100101706366312e78797a2e6e6574000c02013942563330373400
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited1.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf1.xyz.net" with credentials "9BV3074".

20. **PRACK (UE1 to P-CSCF1) - see example in Table 8.5.1-20**

The originating endpoint sends a PRACK request containing the final SDP to be used in this session to it's P-CSCF

Table 8.5.1-20: PRACK (UE1 to P-CSCF1)

```
PRACK sip:token3@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 133 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 9022 132 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=video 9544 RTP/AVP 31
a=rtpmap:31 H261/90000
a=qos:mandatory sendrecv
```

21. **PRACK (P-CSCF1 to S-CSCF1) - see example in Table 8.5.1-21**

   P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

   P-CSCF1 forwards the PRACK request to S-CSCF1.

Table 8.5.1-21: PRACK (P-CSCF1 to S-CSCF1)

```
PRACK sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
Rack:
Content-Type:
Content-length: (…)

v=0
o=-
s=-
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

22. **PRACK (S-CSCF1 to S-CSCF2) - see example in Table 8.5.1-22**

   S-CSCF1 forwards the PRACK request to S-CSCF2.

Table 8.5.1-22: PRACK (S-CSCF1 to S-CSCF2)

```
PRACK sip: scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: SIP/2.0/UDP scscf1.home1.net
From:
To:
Call-ID:
Cseq:
Contact: sip%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
Rack:
Content-Type:
Content-length: (…)

v=0
o=-
s=-
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

23. **PRACK (S-CSCF2 to P-CSCF2) - see example in Table 8.5.1-23**

    S-CSCF2 forwards the PRACK request to P-CSCF2.

Table 8.5.1-23: PRACK (S-CSCF2 to P-CSCF2)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.visited2.net
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

24. **PRACK (P-CSCF2 to UE2) - see example in Table 8.5.1-24**

    P-CSCF2 forwards the PRACK request to callee UE2.

Table 8.5.1-24: PRACK (P-CSCF2 to UE2)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token4
From:
To:
Call-ID:
Cseq:
Contact: token4@pcscf2.visited2.net
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

25. **200 OK (UE2 to P-CSCF2) - see example in Table 8.5.1-25**

   UE acknowledges the PRACK request with a 200 OK response.

Table 8.5.1-25: 200 OK (UE2 to P-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token4
From:
To:
Call-ID:
CSeq: 133 Rrack
Content-length: 0
```

26. **Resource Reservation**

   UE2 initiates the reservation procedures for the new media.

27. **200 OK (P-CSCF2 to S-CSCF2) - see example in Table 8.5.1-27**

   P-CSCF forwards the 200 OK response to S-CSCF.

Table 8.5.1-27: 200 OK (P-CSCF2 to S-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
From:
To:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home2.net
Call-ID:
CSeq:
Content-length:
```

28. **200 OK (S-CSCF2 to S-CSCF1) - see example in Table 8.5.1-28**

   S-CSCF2 forwards the 200 OK response to the originator's S-CSCF, scscf1.home1.net.

Table 8.5.1-28: 200 OK (S-CSCF2 to S-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

29. **200 OK (S-CSCF1 to P-CSCF1) - see example in Table 8.5.1-29**

S-CSCF1 forwards the 200 OK response to the originator's P-CSCF1.

Table 8.5.1-29: 200 OK (S-CSCF1 to P-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

30. **200 OK (P-CSCF1 to UE1) - see example in Table 8.5.1-30**

S-CSCF forwards the 200 OK response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.5.1-30: 200 OK (P-CSCF1 to UE1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token5@pcscf1.visited1.net
Content-Length:
```

31. **Resource Reservation**

UE1 initiates the reservation procedures for the new media.

32. **COMET (UE1 to P-CSCF1) - see example in Table 8.5.1-32**

When the resource reservation is completed, UE sends the COMET request to the terminating endpoint, via the signalling path established by the INVITE request. The request is sent first to P-CSCF.

Table 8.5.1-32: COMET (UE1 to P-CSCF1)

```
COMET sip:token5@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 134 COMET
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=video 9544 RTP/AVP 31
a=rtpmap:31 H261/90000
a=qos:success sendonly
```

33. **COMET (P-CSCF1 to S-CSCF1) - see example in Table 8.5.1-33**

    P-CSCF1 forwards the COMET request to S-CSCF1.

Table 8.5.1-33: COMET (P-CSCF1 to S-CSCF1)

```
COMET sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
Content-Type:
Content-length: (…)

v=0
o=-
s=-
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

34. **COMET (S-CSCF1 to S-CSCF2) - see example in Table 8.5.1-34**

    S-CSCF1 forwards the COMET request to S-CSCF2.

Table 8.5.1-34: COMET (S-CSCF1 to S-CSCF2)

```
COMET sip: scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: SIP/2.0/UDP scscf1.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length: (…)

v=0
o=-
s=-
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

35. **COMET (S-CSCF2 to P-CSCF2) - see example in Table 8.5.1-35**

   S-CSCF2 forwards the COMET request to P-CSCF2.

Table 8.5.1-35: COMET (S-CSCF2 to P-CSCF2)

```
COMET sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net
From:
To:
Call-ID:
Cseq:
Contact: Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

36. **COMET (P-CSCF2 to UE2) - see example in Table 8.5.1-36**

   P-CSCF forwards the COMET request to UE2.

Table 8.5.1-36: COMET (P-CSCF2 to UE2)

```
COMET sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token5
From:
To:
Call-ID:
Cseq:
Contact: token5@pcscf2.visited2.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

37. **200 OK (UE2 to P-CSCF2) - see example in Table 8.5.1-37**

UE acknowledges the COMET request with a 200 OK response.

Table 8.5.1-37: 200 OK (UE2 to P-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token5
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

38. **200 OK (P-CSCF2 to S-CSCF2) - see example in Table 8.5.1-38**

P-CSCF2 forwards the 200 OK response to S-CSCF2.

Table 8.5.1-38: 200 OK (P-CSCF2 to S-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
From:
To:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home2.net
Call-ID:
CSeq:
Content-length:
```

39. **200 OK (S-CSCF2 to S-CSCF1) - see example in Table 8.5.1-39**

S-CSCF2 forwards the 200 OK response to the originator's serving CSCF.

Table 8.5.1-39: 200 OK (S-CSCF2 to S-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

40. **200 OK (S-CSCF1 to P-CSCF1) - see example in Table 8.5.1-40**

   S-CSCF1 forwards the 200 OK response to the P-CSCF1.

Table 8.5.1-40: 200 OK (S-CSCF1 to P-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

41. **200 OK (P-CSCF1 to UE1) - see example in Table 8.5.1-41**

   P-CSCF1 forwards the 200 OK response to UE1.

Table 8.5.1-41: 200 OK (P-CSCF1 to UE1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token6@pcscf1.visited1.net
Content-Length:
```

42. **Alerting**

   UE#2 may optionally delay the session establishment in order to alert the subscriber to the incoming additional media.

43. **180 Ringing (UE2 to P-CSCF2) – see example in Table 8.5.1-43**

   Before proceeding with session establishment, the UE waits for two events. First, the resource reservation initiated in step #26 must complete successfully. Second, the resource reservation initiated by the originating endpoint must complete successfully (which is indicated by message #31 received by UE). The UE may now immediately accept the session or alert the destination subscriber of an incoming session attempt; if the latter it indicates this to the calling party by a 180 Ringing provisional response sent to P-CSCF.

Table 8.5.1-43: 180 Ringing (UE2 to P-CSCF2)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token2
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9023
Content-length: 0
```

44. **180 Ringing (P-CSCF2 to S-CSCF2) - see example in Table 8.5.1-44**

P-CSCF2 forwards the 180 Ringing response to S-CSCF2.

Table 8.5.1-44: 180 Ringing (P-CSCF2 to S-CSCF2)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Content-length:
```

45. **Service Control**

S-CSCF2 performs whatever service control is appropriate for this ringing.

46. **180 Ringing (S-CSCF2 to S-CSCF1) - see example in Table 8.5.1-46**

S-CSCF forwards the 180 Ringing response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.5.1-46: 180 Ringing (S-CSCF2 to S-CSCF1)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Content-length:
```

47. **Service Control**

S-CSCF1 performs whatever service control is appropriate for this ringing.

48. **180 Ringing (S-CSCF1 to P-CSCF1) - see example in Table 8.5.1-48**

S-CSCF1 forwards the 180 Ringing response to the P-CSCF1.

Table 8.5.1-48: 180 Ringing (S-CSCF1 to P-CSCF1)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Content-length:
```

49. **180 Ringing (P-CSCF1 to UE1) - see example in Table 8.5.1-49**

P-CSCF forwards the 180 Ringing response to the UE1

Table 8.5.1-49: 180 Ringing (P-CSCF1 to UE1)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token6@pcscf1.visited.net
Content-Length:
```

50. **Ringback**

    UE1 indicates to the originator that the media addition is being delayed due to alerting. Typically this involves playing a ringback sequence.

51. **PRACK (UE1 to P-CSCF1) - see example in Table 8.5.1-51**

    The originating endpoint sends a PRACK request for the Ringing response to the terminator.

Table 8.5.1-51: PRACK (UE1 to P-CSCF1)

```
PRACK sip:token6@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 135 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 9023 132 INVITE
Content-length:0
```

52. **PRACK (P-CSCF1 to S-CSCF1) - see example in Table 8.5.1-52**

    P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

    P-CSCF1 forwards the PRACK request to S-CSCF1.

Table 8.5.1-52: PRACK (P-CSCF1 to S-CSCF1)

```
PRACK sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

53. **PRACK (S-CSCF1 to S-CSCF2) - see example in Table 8.5.1-53**

    S-CSCF1 forwards the PRACK request to S-CSCF2.

Table 8.5.1-53: PRACK (S-CSCF1 to S-CSCF2)

```
PRACK sip: scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: SIP/2.0/UDP scscf1.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length: 0
```

54. **PRACK (S-CSCF2 to P-CSCF2) - see example in Table 8.5.1-54**

S-CSCF2 forwards the PRACK request to P-CSCF2.

Table 8.5.1-54: PRACK (S-CSCF2 to P-CSCF2)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

55. **PRACK (P-CSCF2 to UE2) - see example in Table 8.5.1-55**

P-CSCF2 forwards the PRACK request to callee UE2.

Table 8.5.1-55: PRACK (P-CSCF2 to UE2)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token7
From:
To:
Call-ID:
Cseq:
Contact: token7@pcscf2.visited2.net
Rack:
Content-length:
```

56. **200 OK (UE2 to P-CSCF2) - see example in Table 8.5.1-56**

UE2 acknowledges the PRACK request with a 200 OK response.

Table 8.5.1-56: 200 OK (UE2 to P-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token7
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

57. **200 OK (P-CSCF2 to S-CSCF2) - see example in Table 8.5.1-57**

P-CSCF2 forwards the 200 OK response to S-CSCF2.

Table 8.5.1-57: 200 OK (P-CSCF2 to S-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net,
SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
From:
To:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home2.net
Call-ID:
CSeq:
Content-length:
```

58. **200 OK (S-CSCF2 to S-CSCF1) - see example in Table 8.5.1-58**

S-CSCF2 forwards the 200 OK response to the originator's serving CSCF.

Table 8.5.1-58: 200 OK (S-CSCF2 to S-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

59. **200 OK (S-CSCF1 to P-CSCF1) - see example in Table 8.5.1-59**

S-CSCF1 forwards the 200 OK response to the P-CSCF1.

Table 8.5.1-59: 200 OK (S-CSCF1 to P-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

60. **200 OK (P-CSCF1 to UE1) - see example in Table 8.5.1-60**

P-CSCF1 forwards the 200 OK response to UE1.

Table 8.5.1-60: 200 OK (P-CSCF1 to UE1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token8@pcscf1.visited1.net
Content-Length:
```

61. **200 OK (UE2 to P-CSCF2) - see example in Table 8.5.1-61**

UE acknowledges the Invite request with a 200 OK response.

Table 8.5.1-61: 200 OK (UE2 to P-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token2
From:
To:
Call-ID:
CSeq: 132 Invite
Content-length: 0
```

62. **Approval of QoS Commit**

   P-CSCF2 approves the commitment of the QoS resources for this additional media

63. **New media can start here.**

64. **200 OK (P-CSCF2 to S-CSCF2) - see example in Table 8.5.1-64**

   P-CSCF2 forwards the 200 OK response to S-CSCF2.

Table 8.5.1-64: 200 OK (P-CSCF2 to S-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

65. **Service Control**

   S-CSCF2 performs whatever service control is required for the session completion

66. **200 OK (S-CSCF2 to S-CSCF1) - see example in Table 8.5.1-66**

   S-CSCF forwards the 200 OK response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.5.1-66: 200 OK (S-CSCF2 to S-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

67. **Service Control**

   S-CSCF1 performs whatever service control is required for the session completion

68. **200 OK (S-CSCF1 to P-CSCF1) - see example in Table 8.5.1-68**

   S-CSCF1 forwards the 200 OK response to the P-CSCF1.

Table 8.5.1-68: 200 OK (S-CSCF1 to P-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP SIP/2.0/UDP pcscf1.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

69. **Approval of QoS Commit**

    P-CSCF1 approves the commitment of the QoS resources for this additional media

70. **200 OK (P-CSCF1 to UE1) - see example in Table 8.5.1-70**

    P-CSCF forwards the 200 OK response to the UE1

Table 8.5.1-70: 200 OK (P-CSCF1 to UE1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token7@pcscf1.visited.net
Content-Length:
```

71. **New media can start here**

72. **ACK (UE1 to P-CSCF1) - see example in Table 8.5.1-72**

    UE1 forwards the ACK request to P-CSCF1.

Table 8.5.1-72: ACK (UE1 to P-CSCF1)

```
ACK sip:token9@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 132 ACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: (…)
```

73. **ACK (P-CSCF1 to S-CSCF1) - see example in Table 8.5.1-73**

    P-CSCF1 adds a Route header, with the saved value from the previous response. P-CSCF1 identifies the proper saved value by the Request-URI.

    P-CSCF1 forwards the ACK request to S-CSCF1.

Table 8.5.1-73: ACK (P-CSCF1 to S-CSCF1)

```
ACK sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length: (…)
```

74. **ACK (S-CSCF1 to S-CSCF2) - see example in Table 8.5.1-74**

S-CSCF1 forwards the ACK request to S-CSCF2.

Table 8.5.1-74: ACK (S-CSCF1 to S-CSCF2)

```
ACK sip: scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: SIP/2.0/UDP scscf1.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length: (…)
```

75. **ACK (S-CSCF2 to P-CSCF2) - see example in Table 8.5.1-75**

   S-CSCF2 forwards the ACK request to P-CSCF2.

Table 8.5.1-75: ACK (S-CSCF2 to P-CSCF2)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

76. **ACK (P-CSCF2 to UE2) - see example in Table 8.5.1-76**

   P-CSCF forwards the ACK request to UE2.

Table 8.5.1-76: ACK (P-CSCF2 to UE2)

```
ACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token5
From:
To:
Call-ID:
Cseq:
Contact: token5@pcscf2.visited2.net
Content-length:
```

8.5.2 Sample MM session flow - addition of another media originator and terminator are both roaming and operated by different networks (with I-CSCF providing configuration independence).

Both networks are with I-CSCF providing configuration independence. The UE has already established an IM CN voice session and is generating an invite to add another video media to the already established session.
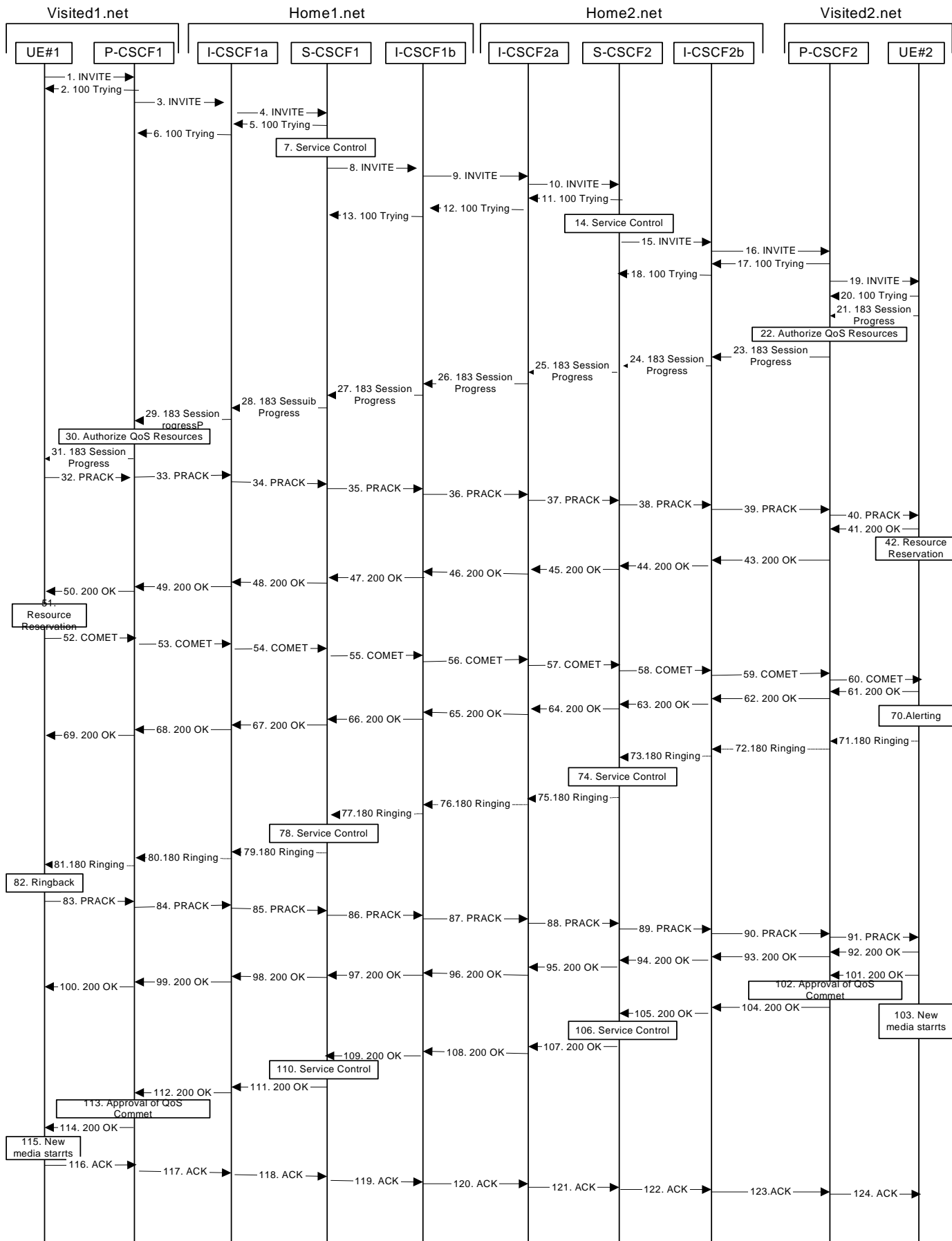
Figure 8.5.2 Sample MM session flow-additional of another media with I-CSCF firewall

1. **INVITE (UE1 to P-CSCF1) - see example in Table 8.5.2-1**

   UE sends the Re-INVITE request, containing another media description in SDP, to the P-CSCF determined via the CSCF discovery mechanism. An example is contained in table 8.5.2-1.

   Editor's Note: Certain fields in the SDP carry no information. In particular the "o=", "s=" fields and "t=". These are, however, mandatory fields within SDP. Does 3GPP wish to define a non-standard version of SDP that removes these, and if so, how does this interwork with outside SIP networks that use standard SDP.

Table 8.5.2-1: INVITE (UE1 to P-CSCF1)

```
INVITE sip:token1@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111;time=36123E5B;seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 132 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907166275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=video 9544 RTP/AVP 31
a=rtpmap:31 H261/90000
a=qos:mandatory sendrecv
```

Editor's Note: All tables need a sentence of text referencing the table. "An example is contained in table 8.w.x.y-z."

Request-URI: contains the keyed number from the user.

Via: contains the IP address or FQDN of the originating UE.

Remote-Party-ID: contains the public identity of the UE. The Display name is optional.

From:, To:, Call-ID: follow the recommendations of draft-ietf-sip-privacy-01, even though anonymity is not being requested for this session.

Cseq: is a random starting number.

Contact: is the IP address or FQDN of the originating UE.

2. **100 Trying (P-CSCF1 to UE1) - see example in Table 8.5.2-2**

   P-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.5.2-2: 100 Trying (P-CSCF1 to UE1)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **INVITE (P-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-3**

P-CSCF1 forwards the INVITE to the next hop name/address, as determined from previous response messages

Table 8.5.2-3: INVITE (P-CSCF1 to I-CSCF1a)

```
INVITE sip:icscf1a.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(scscf1.home2.net), sip:icscf1b.home1.net, sip:icscf2a.home2.net,
    sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

Route: P-CSCF remembers (from the previous response messages) the request routing for this UE. This becomes a Route header in the request. The initial Request-URI is appended to the Route header.

Contact: a locally defined value that identifies the UE.

Request-URI: the first component in the remembered Path header from Registration.

Editor's Note: Need to represent the ability of the P-CSCF to take part in the negotiation of the media streams (by removing entries) in some manner.

4. **INVITE (I-CSCF1a to S-CSCF1) - see example in Table 8.5.2-4**

I-CSCF1a performs the firewall function and forwards the invite to S-CSCF1.

Table 8.5.2-4: INVITE (I-CSCF1a to S-CSCF1)

```
INVITE sip:scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf1b.home1.net, sip:icscf2a.home2.net, sip:token(scscf2.home2.net),
    sip:icscf2b.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf1a.home1.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

5. **100 Trying (S-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-5**

   S-CSCF1 sends the 100 Trying provisional response to P-CSCF1 through I-CSCF1a.

Table 8.5.2-5: 100 Trying (S-CSCF1 to I-CSCF1a)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
[5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

6. **100 Trying (I-CSCF1a to P-CSCF1) - see example in Table 8.5.2-6**

   I-CSCF1a forwards the 100 Trying provisional response to P-CSCF1.

Table 8.5.2-6: 100 Trying (I-CSCF1a to P-CSCF1)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

7. **Service Control**

   S-CSCF1 validates the service profile, and performs any origination service control required for this subscriber.

8. **INVITE (S-CSCF1 to I-CSCF1b) - see example in Table 8.5.2-8**

   S-CSCF1 recognizes that this invite applies to an existing session. It therefore forwards the INVITE along the existing path to I-CSCF1b.

Table 8.5.2-8: INVITE (S-CSCF1 to I-CSCF1b)

```
INVITE sip:icscf1b.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2a.home2.net, sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf1.home1.net, sip: icscf1a.home1.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

9. **INVITE (I-CSCF1b to I-CSCF2a) - see example in Table 8.5.2-9**

   I-CSCF1b forwards the INVITE request to the next hop I-CSCF2a and performs the firewall function

Table 8.5.2-9: INVITE (I-CSCF1b to I-CSCF2a)

```
INVITE sip:icscf2a.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf1b.home1.net sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

10. **INVITE (I-CSCF2a to S-CSCF2) - see example in Table 8.5.2-10**

    I-CSCF2a forwards the INVITE request to S-CSCF2

Table 8.5.2-10: INVITE (I-CSCF2a to S-CSCF2)

```
INVITE sip:scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2b.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf2a.home2.net, sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net),
sip:icscf1a.home1.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

11. **100 Trying (S-CSCF2 to I-CSCF2a) - see example in Table 8.5.2-11**

    S-CSCF2 sends a 100 Trying provisional response back to S-CSCF1 through I-CSCF2a

Table 8.5.2-11: 100 Trying (S-CSCF2 to I-CSCF2a)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

12. **100 Trying (I-CSCF2a to I-CSCF1b) - see example in Table 8.5.2-12**

    I-CSCF2a forwards a 100 Trying provisional response to the upstream next hop I-CSCF1b.

Table 8.5.2-12: 100 Trying (I-CSCF2a to I-CSCF1b)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

13. **100 Trying (I-CSCF1b to S-CSCF1) - see example in Table 8.5.2-13**

    I-CSCF forwards a 100 Trying provisional response to the S-CSCF1.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 8.5.2-13: 100 Trying (I-CSCF1b to S-CSCF1)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

14. **Service Control**

    S-CSCF2 validates the service profile, and performs any origination service control required for this subscriber.

15. **INVITE (S-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-15**

    S-CSCF2 recognizes that this invite applies to an existing session. It therefore forwards the INVITE along the existing path to I-CSCF2b.

Table 8.5.2-15: INVITE (S-CSCF2 to I-CSCF2b)

```
INVITE sip: icscf2b.home2.net
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf2.home2.net, sip:icscf2a.home2.net, sip:icscf1b.home1.net,
    sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

16. **INVITE (I-CSCF2b to P-CSCF2) - see example in Table 8.5.2-16**

    I-CSCF2b performs the firewall function and forwards the INVITE request to P-CSCF2

Table 8.5.2-16: INVITE (I-CSCF2 to P-CSCF2)

```
INVITE sip: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

17. **100 Trying (P-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-17**

    P-CSCF2 sends a 100 Trying provisional response back to S-CSCF2 through I-CSCF2b.

Table 8.5.2-17: 100 Trying (P-CSCF2 to I-CSCF2b)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

18. **100 Trying (I-CSCF2b to S-CSCF2) - see example in Table 8.5.2-18**

    I-CSCF2b forwards a 100 Trying provisional response back to S-CSCF2.

Table 8.5.2-18: 100 Trying (I-CSCF2b to S-CSCF2)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP
    icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net,
    SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

19. **INVITE (P-CSCF2 to UE2) - see example in Table 8.5.2-19**

P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.5.2-19: INVITE (P-CSCF2 to UE2)

```
INVITE sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token2
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363233
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token2@pcscf2.visited2.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14623".

20. **100 Trying (UE2 to P-CSCF2) - see example in Table 8.5.2-20**

UE2 sends a 100 Trying provisional response back to P-CSCF2.

Table 8.5.2-20: 100 Trying (UE2 to P-CSCF2)

```
SIP/2.0 100 Trying
Via: pcscf2.visited2.net;branch=token2 SIP/2.0/UDP
From:
To:
Call-ID:
CSeq:
```

21. **183 Session Progress (UE2 to P-CSCF2) - see example in Table 8.5.2-21**

The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response.

Table 8.5.2-21: 183 Session Progress response (UE2 to P-CSCF2)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token2
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9022
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907166275 0
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
m=video 7544 RTP/AVP 31
a=rtpmap:31 H261/90000
a=qos:mandatory sendrecv confirm
```

22. **Authorize QoS Resources**

    P-CSCF2 authorizes the resources necessary for this new media

23. **183 Session Progress (P-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-23**

    P-CSCF forwards the 183 Session Progress response to P-CSCF

Table 8.5.2-23: 183 Session Progress (P-CSCF2 to I-CSCF2b)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.visited2.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

24. **183 Session Progress (I-CSCF2b to S-CSCF2) - see example in Table 8.5.2-24**

I-CSCF2b forwards the 183 Session Progress response to S-CSCF2

Table 8.5.2-24: 183 Session Progress (I-CSCF2b to S-CSCF2)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:scscf2.home2.net, sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

25. **183 Session Progress (S-CSCF2 to I-CSCF2a) - see example in Table 8.5.2-25**

S-CSCF2 forwards the 183 Session Progress response to I-CSCF2a

Table 8.5.2-25: 183 Session Progress (S-CSCF2 to I-CSCF2a)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:scscf2.home2.net, sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

26. **183 Session Progress (I-CSCF2a to I-CSCF1b) - see example in Table 8.5.2-26**

I-CSCF2a forwards the 183 Session Progress response to I-CSCF1b

Table 8.5.2-26: 183 Session Progress (I-CSCF2a to I-CSCF1b)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

27. **183 Session Progress (I-CSCF1b to S-CSCF1) - see example in Table 8.5.2-27**

    I-CSCF1b forwards the 183 Session Progress response to the S-CSCF1

Table 8.5.2-27: 183 Session Progress (I-CSCF1b to S-CSCF1)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:sip:scscf1.home1.net, sip:icscf1a.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

28. **183 Session Progress (S-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-28**

S-CSCF1 forwards the 183 Session Progress response to I-CSCF1a.

Table 8.5.2-28: 183 Session Progress (S-CSCF1 to I-CSCF1a)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

29. **183 Session Progress (I-CSCF1a to P-CSCF1) - see example in Table 8.5.2-29**

I-CSCF1a forwards the 183 Session Progress response to P-CSCF1

Table 8.5.2-29: 183 Session Progress (I-CSCF1a to P-CSCF1)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

30. **Authorize QoS Resources**

   P-CSCF1 authorizes the resources necessary for this new media

31. **183 Session Progress (P-CSCF1 to UE1) - see example in Table 8.5.2-31**

   P-CSCF1 forwards the 183 Session Progress response to the originating endpoint

Table 8.5.2-31: 183 Session Progress (P-CSCF1 to UE1)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Media-Authorization: 0020000100100101706366312e78797a2e6e6574000c02013942563330373400
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited1.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

   Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf1.xyz.net" with credentials "9BV3074".

32. **PRACK (UE1 to P-CSCF1) - see example in Table 8.5.2-32**

   The originator decides the final set of media streams for this media addition, and sends the Final SDP to P-CSCF1.

Table 8.5.2-32: PRACK (UE1 to P-CSCF1)

```
PRACK sip:token3@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111;time=36123E5B;seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 133 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 9022 132 Invite
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907166275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv
m=video 9544 RTP/AVP 31
a=rtpmap:31 H261/90000
a=qos:mandatory sendrecv
```

33. **PRACK (P-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-33**

    The PRACK request is forwarded through this I-CSCF to the S-CSCF.

Table 8.5.2-33: PRACK (P-CSCF1 to I-CSCF1a)

```
PRACK sip:icscf1a.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(scscf1.home2.net), sip:icscf1b.home1.net, sip:icscf2a.home2.net,
    sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

34. **PRACK (I-CSCF1a to S-CSCF1) - see example in Table 8.5.2-34**

    The PRACK request is forwarded through this I-CSCF1a to the S-CSCF1.

Table 8.5.2-34: PRACK (I-CSCF1a to S-CSCF1)

```
PRACK sip:scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf1b.home1.net, sip:icscf2a.home2.net, sip:token(scscf2.home2.net),
    sip:icscf2b.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

35. **PRACK (S-CSCF1 to I-CSCF1b) - see example in Table 8.5.2-35**

S-CSCF1 forwards the PRACK request to I-CSCF1b

Table 8.5.2-35: PRACK (S-CSCF1 to I-CSCF1b)

```
PRACK sip:icscf1b.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2a.home2.net, sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

36. **PRACK (I-CSCF1b to I-CSCF2a) - see example in Table 8.5.2-36**

I-CSCF1b forwards the PRACK request to I-CSCF2a

Table 8.5.2-36: PRACK (I-CSCF1b to I-CSCF2a)

```
PRACK sip:icscf2a.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf1b.home1.net sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

37. **PRACK (I-CSCF2a to S-CSCF2) - see example in Table 8.5.2-37**

    I-CSCF2a forwards the PRACK request to S-CSCF2

Table 8.5.2-37: PRACK (I-CSCF2a to S-CSCF2)

```
PRACK sip:scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2b.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf2a.home2.net, sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net),
    sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

38. **PRACK (S-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-38**

    S-CSCF2 forwards the PRACK request to I-CSCF2b

Table 8.5.2-38: PRACK (S-CSCF2 to I-CSCF2b)

```
PRACK sip: icscf2b.home2.net
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf2.home2.net, sip:icscf2a.home2.net, sip:icscf1b.home1.net,
    sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

39. **PRACK (I-CSCF2b to P-CSCF2) - see example in Table 8.5.2-39**

   I-CSCF1b forwards the PRACK request to P-CSCF2

Table 8.5.2-39: PRACK (I-CSCF2b to P-CSCF2)

```
PRACK sip: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

40. **PRACK (P-CSCF2 to UE2) - see example in Table 8.5.2-40**

   P-CSCF2 and forwards the PRACK request to the UE2.

Table 8.5.2-40: PRACK (P-CSCF2 to UE2)

```
PRACK sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.visited2.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

41. **200 OK (UE2 to P-CSCF2) - see example in Table 8.5.2-41**

    UE2 acknowledges the PRACK request with a 200 OK response.

Table 8.5.2-41: 200 OK (UE2 to P-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token3
From:
To:
Call-ID:
CSeq: 133 Prack
Content-length: 0
```

42. **Resource Reservation**

    After determining the final set of media streams for this additional media, UE2 initiates the reservation procedures for the additional resources needed for this new media.

43. **200 OK (P-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-43**

    P-CSCF2 forwards the 200 OK response to I-CSCF2b.

Table 8.5.2-43: 200 OK (P-CSCF2 to I-CSCF2b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
   icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
   SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
   sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

44. **200 OK (I-CSCF2b to S-CSCF2) - see example in Table 8.5.2-44**

    I-CSCF2b forwards the 200 OK response to S-CSCF2.

Table 8.5.2-44: 200 OK (I-CSCF2b to S-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, scscf2.home2.net, sip:icscf2a.home2.net, sip:icscf1b.home1.net,
    sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

45. **200 OK (S-CSCF2 to I-CSCF2a) - see example in Table 8.5.2-45**

   S-CSCF2 forwards the 200 OK response to I-CSCF2a

Table 8.5.2-45: 200 OK (S-CSCF2 to I-CSCF2a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

46. **200 OK (I-CSCF2a to I-CSCF1b) - see example in Table 8.5.2-46**

   I-CSCF2a forwards the 200 OK response to I-CSCF1b.

Table 8.5.2-46: 200 OK (I-CSCF2a to I-CSCF1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited1.net
Content-Length:
```

47. **200 OK (I-CSCF1b to S-CSCF1) - see example in Table 8.5.2-47**

   S-CSCF forwards the 200 OK response to S-CSCF1.

Table 8.5.2-47: 200 OK (I-CSCF1b to S-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited1.net
Content-Length:
```

48. **200 OK (S-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-48**

     S-CSCF1 forwards the 200 OK response to I-CSCF1a.

Table 8.5.2-48: 200 OK (S-CSCF1 to I-CSCF1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
   sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

49. **200 OK (I-CSCF1a to P-CSCF1) - see example in Table 8.5.2-49**

     I-CSCF1 forwards the 200 OK response to P-CSCF1.

Table 8.5.2-49: 200 OK (I-CSCF1a to P-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
   sip:icscf1b.home1.net, token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

50. **200 OK (P-CSCF1 to UE1) - see example in Table 8.5.2-50**

     P-CSCF1 forwards the 200 OK response to the originator

Table 8.5.2-50: 200 OK (P-CSCF1 to UE1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token4@pcscf1.visited1.net
Content-Length:
```

51. **Resource Reservation**

     After determining the final set of media streams for this additional media, UE1 initiates the reservation procedures for the additional resources needed for this new media.

52. **COMET (UE1 to P-CSCF1) - see example in Table 8.5.2-52**

     After determining the final set of media streams for this additional media, UE1 initiates the reservation p procedures for the additional resources needed for this new media.

Table 8.5.2-52: COMET (UE1 to P-CSCF1)

```
COMET sip:token4@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111;time=36123E5B;seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 134 COMET
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907166275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:success sendonly
m=video 9544 RTP/AVP 31
a=rtpmap:31 H261/90000
a=qos:success sendonly
```

53. **COMET (P-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-53**

    The COMET request is forwarded through this P-CSCF1 to the I-CSCF1a.

Table 8.5.2-53: COMET (P-CSCF1 to I-CSCF1a)

```
COMET sip:icscf1a.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(scscf1.home2.net), sip:icscf1b.home1.net, sip:icscf2a.home2.net,
    sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

54. **COMET (I-CSCF1a to S-CSCF1) - see example in Table 8.5.2-54**

    The COMET request is forwarded through this I-CSCF1a to the S-CSCF1.

Table 8.5.2-54: COMET (I-CSCF1a to S-CSCF1)

```
COMET sip:scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf1b.home1.net, sip:icscf2a.home2.net, sip:token(scscf2.home2.net),
    sip:icscf2b.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
m=
a=
a=
```

55. **COMET (S-CSCF1 to I-CSCF1b) - see example in Table 8.5.2-55**

S-CSCF1 forwards the COMET request to I-CSCF1b.

Table 8.5.2-55: COMET (S-CSCF1 to I-CSCF1b)

```
COMET sip:icscf1b.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2a.home2.net, sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf1.home1.net, sip: icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

56. **COMET (I-CSCF1b to I-CSCF2a) - see example in Table 8.5.2-56**

I-CSCF1b forwards the COMET request, to I-CSCF2a

Table 8.5.2-56: COMET (I-CSCF1b to I-CSCF2a)

```
COMET sip:icscf2a.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf1b.home1.net sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

57. **COMET (I-CSCF2a to S-CSCF2) - see example in Table 8.5.2-57**

I-CSCF2a forwards the COMET request to S-CSCF2.

Table 8.5.2-57: COMET (I-CSCF2a to S-CSCF2)

```
COMET sip:scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2b.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf2a.home2.net, sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net),
    sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

58. **COMET (S-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-58**

S-CSCF2 forwards the COMET request to I-CSCF2b.

Table 8.5.2-58: COMET (S-CSCF2 to I-CSCF2b)

```
COMET sip: icscf2b.home2.net
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaaa%3abbbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf2.home2.net, sip:icscf2a.home2.net, sip:icscf1b.home1.net,
    sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

59. **COMET (I-CSCF2b to P-CSCF2) - see example in Table 8.5.2-59**

   I-CSCF2b forwards the COMET request to P-CSCF2

Table 8.5.2-59: COMET (I-CSCF2b to P-CSCF2)

```
COMET sip: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbbb%5d@pcscf2.visited2.net
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

60. **COMET (P-CSCF2 to UE2) - see example in Table 8.5.2-60**

   P-CSCF2 determines the UE address from the value of the Request-URI and forwards the COMET request to the UE2.

Table 8.5.2-60: COMET (P-CSCF2 to UE2)

```
COMET sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token4
From:
To:
Call-ID:
Cseq:
Contact: token4@pcscf2.visited2.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
m=
a=
a=
```

61. **200 OK (UE2 to P-CSCF2) - see example in Table 8.5.2-61**

   UE2 acknowledges the COMET request with a 200 OK response.

Table 8.5.2-61: 200 OK (UE2 to P-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token4
From:
To:
Call-ID:
CSeq: 134 Comet
Content-length: 0
```

62. **200 OK (P-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-62**

Table 8.5.2-62: 200 OK (P-CSCF2 to I-CSCF2b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

63. **200 OK (I-CSCF2b to S-CSCF2) - see example in Table 8.5.2-63**

Table 8.5.2-63: 200 OK (I-CSCF2b to S-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, scscf2.home2.net, sip:icscf2a.home2.net, sip:icscf1b.home1.net,
    sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

64. **200 OK (S-CSCF2 to I-CSCF2a) - see example in Table 8.5.2-64**

Table 8.5.2-64: 200 OK (S-CSCF2 to I-CSCF2a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

65. **200 OK (I-CSCF2a to I-CSCF1b) - see example in Table 8.5.2-65**

Table 8.5.2-65: 200 OK (I-CSCF2a to I-CSCF1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited1.net
Content-Length:
```

66. **200 OK (I-CSCF1b to S-CSCF1) - see example in Table 8.5.2-66**

Table 8.5.2-66: 200 OK (I-CSCF1b to S-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited1.net
Content-Length:
```

67. **200 OK (S-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-67**

Table 8.5.2-67: 200 OK (S-CSCF1 to I-CSCF1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

68. **200 OK (I-CSCF1a to P-CSCF1) - see example in Table 8.5.2-68**

Table 8.5.2-68: 200 OK (I-CSCF1a to P-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

69. **200 OK (P-CSCF1 to UE1) - see example in Table 8.5.2-69**

Table 8.5.2-69: 200 OK (P-CSCF1 to UE1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token5@pcscf1.visited1.net
Content-Length:
```

70. **Alerting**

UE2 may optionally delay the session establishment in order to alert the subscriber to the incoming additional media.

71. **180 Ringing (UE2 to P-CSCF2) - see example in Table 8.5.2-71**

If UE2 performs alerting, it sends a ringing indication to the originator via the signaling path. The message is sent first to P-CSCF2.

Table 8.5.2-71: 180 Ringing (UE2 to P-CSCF2)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token2
Require: 100rel
From:
To:
Call-ID:
CSeq: 132 Invite
Rseq: 9023
Content-length: 0
```

72. **180 Ringing (P-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-72**

Table 8.5.2-72: 180 Ringing (P-CSCF2 to I-CSCF2b)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Rseq:
Content-length:
```

73. **180 Ringing (I-CSCF2b to S-CSCF2) - see example in Table 8.5.2-73**

Table 8.5.2-73: 180 Ringing (I-CSCF2b to S-CSCF2)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, scscf2.home2.net, sip:icscf2a.home2.net, sip:icscf1b.home1.net,
    sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Rseq:
Content-length:
```

74. **Service Control**

S-CSCF2 performs whatever service control is appropriate for this ringing

75. **180 Ringing (S-CSCF2 to I-CSCF2a) - see example in Table 8.5.2-75**

Table 8.5.2-75: 180 Ringing (S-CSCF2 to I-CSCF2a)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Rseq:
Content-length:
```

76. **180 Ringing (I-CSCF2a to I-CSCF1b) - see example in Table 8.5.2-76**

Table 8.5.2-76: 180 Ringing (I-CSCF2a to I-CSCF1b)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Rseq:
Contact:
Content-Length:
```

77. **180 Ringing (I-CSCF1b to S-CSCF1) - see example in Table 8.5.2-77**

Table 8.5.2-77: 180 Ringing (I-CSCF1b to S-CSCF1)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.netRequire:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
Rseq:
Content-Length:
```

78. **Service Control**

S-CSCF1 performs whatever service control is appropriate for this ringing

79. **180 Ringing (S-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-79**

Table 8.5.2-79: 180 Ringing (S-CSCF1 to I-CSCF1a)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Rseq:
Content-length: 0
```

80. **180 Ringing (I-CSCF1a to P-CSCF1) - see example in Table 8.5.2-80**

Table 8.5.2-80: 180 Ringing (I-CSCF1a to P-CSCF1)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, token(sip:scscf1.home1.net), sip:icscf1a.home1.net
Require:
From:
To:
Call-ID:
CSeq:
Rseq:
Content-length:
```

81. **180 Ringing (P-CSCF1 to UE1) - see example in Table 8.5.2-81**

Table 8.5.2-81: 180 Ringing (P-CSCF1 to UE1)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token5@pcscf1.visited1.net
Content-Length:
```

82. **Ringback**

UE#1 indicates to the originator that the media addition is being delayed due to alerting. Typically this involves playing a ringback sequence.

83. **PRACK (UE1 to P-CSCF1) - see example in Table 8.5.2-83**

The originator sends ACK to the terminator for the Ringing response.

Table 8.5.2-83: PRACK (UE1 to P-CSCF1)

```
PRACK sip:token5@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111;time=36123E5B;seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 135 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 9023 132 Invite
Content-Type:
Content-length:
```

84. **PRACK (P-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-84**

The PRACK request is forwarded through this I-CSCF 1to the S-CSCF.

Table 8.5.2-84: PRACK (P-CSCF1 to I-CSCF1a)

```
PRACK sip:icscf1a.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(scscf1.home2.net), sip:icscf1b.home1.net, sip:icscf2a.home2.net,
    sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
Rack:
Content-Type:
Content-length:
```

85. **PRACK (I-CSCF1a to S-CSCF1) - see example in Table 8.5.2-85**

   The PRACK request is forwarded through this I-CSCF1a to the S-CSCF1.

Table 8.5.2-85: PRACK (I-CSCF1a to S-CSCF1)

```
PRACK sip:scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf1b.home1.net, sip:icscf2a.home2.net, sip:token(scscf2.home2.net),
    sip:icscf2b.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:
```

86. **PRACK (S-CSCF1 to I-CSCF1b) - see example in Table 8.5.2-86**

   S-CSCF1 forwards the PRACK request to I-CSCF1b

Table 8.5.2-86: PRACK (S-CSCF1 to I-CSCF1b)

```
PRACK sip:icscf1b.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2a.home2.net, sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:
```

87. **PRACK (I-CSCF1b to I-CSCF2a) - see example in Table 8.5.2-87**

   I-CSCF1b forwards the PRACK request to I-CSCF2a

Table 8.5.2-87: PRACK (I-CSCF1b to I-CSCF2a)

```
PRACK sip:icscf2a.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf1b.home1.net sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:
```

88. **PRACK (I-CSCF2a to S-CSCF2) - see example in Table 8.5.2-88**

    I-CSCF2a forwards the PRACK request to S-CSCF2

Table 8.5.2-88: PRACK (I-CSCF2a to S-CSCF2)

```
PRACK sip:scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2b.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf2a.home2.net, sip:icscf1b.home1.net, sip:token(scscf1.home1.net),
    sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:
```

89. **PRACK (S-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-89**

    S-CSCF2 forwards the PRACK request to I-CSCF2b

Table 8.5.2-89: PRACK (S-CSCF2 to I-CSCF2b)

```
PRACK sip: icscf2b.home2.net
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf2.home2.net, sip:icscf2a.home2.net, sip:icscf1b.home1.net,
    sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:
```

90. **PRACK (I-CSCF2b to P-CSCF2) - see example in Table 8.5.2-90**

    I-CSCF1b forwards the PRACK request to P-CSCF2

Table 8.5.2-90: PRACK (I-CSCF2b to P-CSCF2)

```
PRACK sip: sip%5b5555%3a%3aeee%3afff%3aaaaa%3abbbb%5d@pcscf2.visited2.net
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-Type:
Content-length:
```

91. **PRACK (P-CSCF2 to UE2) - see example in Table 8.5.2-91**

   P-CSCF2 and forwards the PRACK request to the UE2.

Table 8.5.2-91: PRACK (P-CSCF2 to UE2)

```
PRACK sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token6
From:
To:
Call-ID:
Cseq:
Contact: token6@pcscf2.visited2.net
Content-Type:
Content-length:
```

92. **200 OK (UE2 to P-CSCF2) - see example in Table 8.5.2-92**

   UE2 acknowledges the PRACK request with a 200 OK response.

Table 8.5.2-92: 200 OK (UE2 to P-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token6
From:
To:
Call-ID:
CSeq: 135 PRACK
Content-length: 0
```

93. **200 OK (P-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-93**

   P-CSCF2 forwards the 200 OK response to I-CSCF2b.

Table 8.5.2-93: 200 OK (P-CSCF2 to I-CSCF2b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

94. **200 OK (I-CSCF2b to S-CSCF2) - see example in Table 8.5.2-94**

I-CSCF2b forwards the 200 OK response to S-CSCF2.

Table 8.5.2-94: 200 OK (I-CSCF2b to S-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, scscf2.home2.net, sip:icscf2a.home2.net, sip:icscf1b.home1.net,
    sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

95. **200 OK (S-CSCF2 to I-CSCF2a) - see example in Table 8.5.2-95**

S-CSCF2 forwards the 200 OK response to I-CSCF2a

Table 8.5.2-95: 200 OK (S-CSCF2 to I-CSCF2a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

96. **200 OK (I-CSCF2a to I-CSCF1b) - see example in Table 8.5.2-96**

I-CSCF2a forwards the 200 OK response to I-CSCF1b.

Table 8.5.2-96: 200 OK (I-CSCF2a to I-CSCF1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited1.net
Content-Length:
```

97. **200 OK (I-CSCF1b to S-CSCF1) - see example in Table 8.5.2-97**

S-CSCF forwards the 200 OK response to S-CSCF1.

Table 8.5.2-97: 200 OK (I-CSCF1b to S-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited1.net
Content-Length:
```

98. **200 OK (S-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-98**

    S-CSCF1 forwards the 200 OK response to I-CSCF1a.

Table 8.5.2-98: 200 OK (S-CSCF1 to I-CSCF1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

99. **200 OK (I-CSCF1a to P-CSCF1) - see example in Table 8.5.2-99**

    I-CSCF1 forwards the 200 OK response to P-CSCF1.

Table 8.5.2-99: 200 OK (I-CSCF1a to P-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

100. **200 OK (P-CSCF1 to UE1) - see example in Table 8.5.2-100**

    P-CSCF1 forwards the 200 OK response to the originator

Table 8.5.2-100: 200 OK (P-CSCF1 to UE1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token7@pcscf1.visited1.net
Content-Length:
```

101. **200 OK (UE2 to P-CSCF2) - see example in Table 8.5.2-101**

    UE acknowledges the Invite request with a 200 OK response.

Table 8.5.2-101: 200 OK (UE2 to P-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token2
From:
To:
Call-ID:
CSeq: 132 Invite
Content-length: 0
```

102. **Approval of QoS Commet**

   P-CSCF2 approves the commitment of the QoS resources for this additional media.

103. **UE2 can start the new media**

104. **200 OK (P-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-104**

   P-CSCF2 forwards the 200 OK response to I-CSCF2b.

Table 8.5.2-104: 200 OK (P-CSCF2 to I-CSCF2b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

105. **200 OK (I-CSCF2b to S-CSCF2) - see example in Table 8.5.2-105**

   I-CSCF2b forwards the 200 OK response to S-CSCF2.

Table 8.5.2-105: 200 OK (I-CSCF2b to S-CSCF2)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, scscf2.home2.net, sip:icscf2a.home2.net, sip:icscf1b.home1.net,
sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

106. **Service Control**

   S-CSCF2 performs whatever service control is required for the session completion

107. **200 OK (S-CSCF2 to I-CSCF2a) - see example in Table 8.5.2-107**

   S-CSCF2 forwards the 200 OK response to I-CSCF2a.

Table 8.5.2-107: 200 OK (S-CSCF2 to I-CSCF2a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

108. **200 OK (I-CSCF2a to I-CSCF1b) - see example in Table 8.5.2-108**

    S-CSCF forwards the 200 OK response to I-CSCF1b.

Table 8.5.2-108: 200 OK (I-CSCF2a to I-CSCF1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

109. **200 OK (I-CSCF1b to S-CSCF1) - see example in Table 8.5.2-109**

    I-CSCF1b forwards the 200 OK response to S-CSCF1.

Table 8.5.2-109: 200 OK (I-CSCF2a to I-CSCF1b)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.visited1.net
Content-Length:
```

110. **Service Control**

    S-CSCF1 performs whatever service control is required for the session completion

111. **200 OK (S-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-111**

    S-CSCF1 forwards the 200 OK response to I-CSCF1a.

Table 8.5.2-111: 200 OK (S-CSCF1 to I-CSCF1a)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

112. **200 OK (I-CSCF1a to P-CSCF1) - see example in Table 8.5.2-112**

    I-CSCF1a forwards the 200 OK response to P-CSCF1.

Table 8.5.2-112: 200 OK (I-CSCF1a to P-CSCF1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

113. **Approval of QoS Commit**

    P-CSCF1 approves the commitment of the QoS resources for this additional media.

114. **200 OK (P-CSCF1 to UE1) - see example in Table 8.5.2-114**

    S-CSCF forwards the 200 OK response to UE1.

Table 8.5.2-114: 200 OK (P-CSCF1 to UE1)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token8@pcscf1.visited1.net
Content-Length:
```

115. **UE1 can start new media**

116. **ACK (UE1 to P-CSCF1) - see example in Table 8.5.2-116**

    UE1 responds to the final response with a SIP ACK message, which is passed to the destination via the signalling path. The message is sent first to P-CSCF1.

Table 8.5.2-116: ACK (UE1 to P-CSCF1)

```
ACK sip:token8@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111;time=36123E5B;seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 132 ACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: (…)
```

117. **ACK (P-CSCF1 to I-CSCF1a) - see example in Table 8.5.2-117**

Table 8.5.2-117: ACK (P-CSCF1 to I-CSCF1a)

```
ACK sip:icscf1a.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(scscf1.home2.net), sip:icscf1b.home1.net, sip:icscf2a.home2.net,
    sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
```

118. **ACK (I-CSCF1a to S-CSCF1) - see example in Table 8.5.2-118**

Table 8.5.2-118: ACK (I-CSCF1a to S-CSCF1)

```
ACK sip:scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf1b.home1.net, sip:icscf2a.home2.net, sip:token(scscf2.home2.net),
    sip:icscf2b.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

119. **ACK (S-CSCF1 to I-CSCF1b) - see example in Table 8.5.2-119**

Table 8.5.2-119: ACK (S-CSCF1 to I-CSCF1b)

```
ACK sip:icscf1b.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2a.home2.net, sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf1.home1.net, sip: icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

120. **ACK (I-CSCF1b to I-CSCF2a) - see example in Table 8.5.2-120**

Table 8.5.2-120: ACK (I-CSCF1b to I-CSCF2a)

```
ACK sip:icscf2a.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(scscf2.home2.net), sip:icscf2b.home2.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf1b.home1.net sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

121. **ACK (I-CSCF2a to S-CSCF2) - see example in Table 8.5.2-121**

Table 8.5.2-121: ACK (I-CSCF2a to S-CSCF2)

```
ACK sip:scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP
    token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2b.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited2.net
Record-Route: sip:icscf2a.home2.net, sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net),
    sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

122. **ACK (S-CSCF2 to I-CSCF2b) - see example in Table 8.5.2-122**

Table 8.5.2-122: ACK (S-CSCF2 to I-CSCF2b)

```
ACK sip: icscf2b.home2.net
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP token(scscf1.home1.net), SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf2.home2.net, sip:icscf2a.home2.net, sip:icscf1b.home1.net,
    sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

123. **ACK (I-CSCF2b to P-CSCF2) - see example in Table 8.5.2-123**

Table 8.5.2-123: ACK (S-CSCF2 to I-CSCF2b)

```
ACK sip: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP token(scscf2.home2.net), SIP/2.0/UDP
    icscf2a.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP token(scscf1.home1.net),
    SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf2b.home2.net, sip:token(scscf2.home2.net), sip:icscf2a.home2.net,
    sip:icscf1b.home1.net, sip:token(sip:scscf1.home1.net), sip:icscf1a.home1.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

124. **ACK (P-CSCF2 to UE2) - see example in Table 8.5.2-124**

Table 8.5.2-124: ACK (P-CSCF2 to UE2)

```
ACK sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token9@pcscf2.visited2.net
Content-Type:
Content-length:
```

9 Signalling flows for session termination

9.1 Mobile Terminal initiated session release

9.2 PSTN initiated session release

10 Procedures to enable enhanced multimedia services

10.1 Session hold and resume procedures

10.1.1 Introduction

This clause gives information flows for the procedures for placing sessions on hold that were previously established by the mechanisms of clause 8, and resuming the session afterwards. Two cases are presented: mobile-to-mobile (UE-UE), and a UE-initiated hold of a UE-PSTN session.

For a multi-media session, it is possible to place a subset of the media streams on hold while maintaining the others.

10.1.2 Mobile-to-mobile session hold and resume procedures

An IM session was previously established between an initiating UE and a terminating UE. Each of these UEs has an associated P-CSCF in the same network where they are currently located (either home or roaming), and a S-CSCF assigned in their home network . These functional elements co-operate to clear the session, and the procedures are independent of whether they are located in the home or visited networks.

The hold and resume procedures are identical whether the UE that initiated the session also initiates the session-hold, or whether the UE that terminated the session initiates the session-hold.

When a media stream has been placed on hold, it shall not be resumed by any endpoint other than the one that placed it on hold.

These procedures show only one combination of Mobile-Originated, Serving-to-Serving, and Mobile-Terminated procedures, MO#2, S-S#2, and MT#2. These procedures do not show the use of optional I-CSCFs. If an I-CSCF was included in the signalling path during the session establishment procedure, it would continue to be used in any subsequent flows such as the ones described in this clause. Procedures at the I-CSCFs are identical to those described for the BYE, PRACK, and COMET requests and responses described in other clauses.

The procedures for placing a media stream on hold, and later resuming the media stream, are as shown in Figure 10.1.2-1:
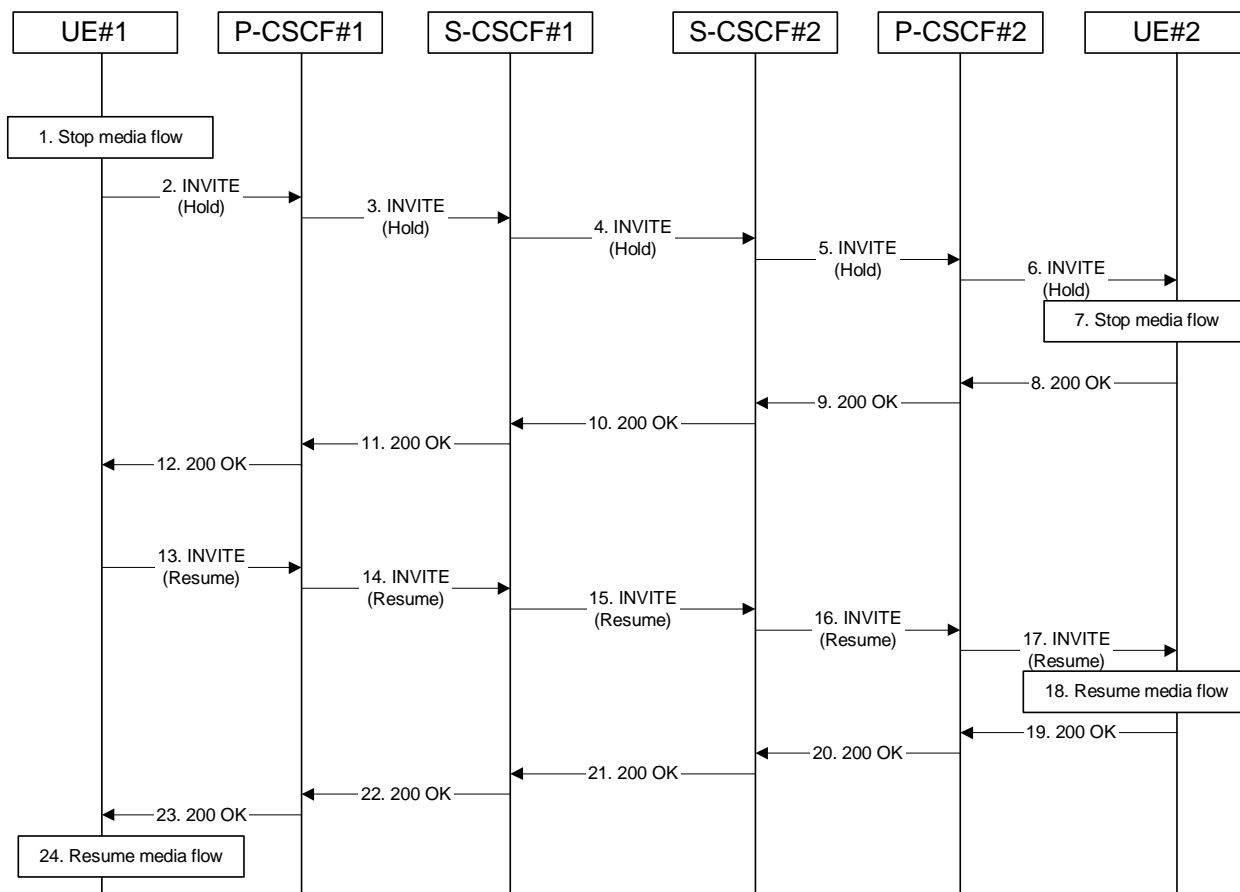
Figure 10.1.2-1: Mobile to mobile session hold and resume

Information flow procedures are as follows:

1. **Stop Media Flow**

   UE#1 detects a request from the subscriber to place a media stream on hold. UE#1 stops sending the media stream to the remote endpoint, but keeps the resources for the session reserved.

2. **INVITE(Hold) (UE to P-CSCF) – see example in Table 10.1.2-2**

   UE#1 sends a Hold request to its proxy, P-CSCF#1.

Table 10.1.2-2: INVITE(Hold) (UE to P-CSCF)

```
INVITE sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 0::0:0:0:0
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

SDP: Same SDP as negotiated during the session setup, but with the "c=" line having an IP address of zeroes. The IP address of zeroes indicates a session hold request.

3. **INVITE (Hold) (P-CSCF to S-CSCF) – see example in Table 10.1.2-3**

   P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

   P-CSCF#1 forwards the Hold request to S-CSCF#1.

Table 10.1.2-3: INVITE(Hold) (P-CSCF to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

4. **INVITE(Hold) (S-CSCF to S-CSCF) – see example in Table 10.1.2-4**

S-CSCF#1 forwards the Hold request to S-CSCF#2.

Table 10.1.2-4: INVITE(Hold) (S-CSCF to S-CSCF)

```
INVITE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

5. **INVITE(Hold) (S-CSCF to P-CSCF) – see example in Table 10.1.2-5**

S-CSCF#2 forwards the Hold request to P-CSCF#2.

Table 10.1.2-5: INVITE(Hold) (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

6. **INVITE(Hold) (P-CSCF to UE) – see example in Table 10.1.2-6**

P-CSCF#2 forwards the Hold request to UE#2.

Table 10.1.2-6: INVITE(Hold) (P-CSCF to UE)

```
INVITE sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

7. **Stop Media flow**

   UE#2 stops sending the media stream to the remote endpoint, but keeps the resources for the session reserved.

8. **200-OK (UE to P-CSCF) – see example in Table 10.1.2-8**

   UE#2 acknowledges receipt of the Hold request (6) with a 200-OK final response, sent to P-CSCF#2.

Table 10.1.2-8: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::eee:fff:aaa:bbb
s=-
c= IN IP6 0::0:0:0:0
b=AS:64
t=907165275 0
m=audio 6402 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
```

9. **200-OK (P-CSCF to S-CSCF) – see example in Table 10.1.2-9**

   P-CSCF#2 forwards the 200 OK final response to S-CSCF#2.

Table 10.1.2-9: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

10. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.1.2-10**

S-CSCF#2 forwards the 200 OK final response to S-CSCF#1.

Table 10.1.2-10: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

11. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.1.2-11**

S-CSCF#1 forwards the 200 OK final response to P-CSCF#1.

Table 10.1.2-11: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

12. **200-OK (P-CSCF to UE) – see example in Table 10.1.2-12**

P-CSCF#1 forwards the 200 OK final response to UE#1.

Table 10.1.2-12: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

13. **INVITE(Resume) (UE to P-CSCF) – see example in Table 10.1.2-13**

UE#1 detects a request from the subscriber to resume the media stream previously placed on hold. UE#1 sends a Resume request to its proxy, P-CSCF#1.

Table 10.1.2-13: INVITE(Resume) (UE to P-CSCF)

```
INVITE sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

SDP: Same SDP as negotiated during the session setup, but with the "c=" line having the actual IP address.

14. **INVITE(Resume) (P-CSCF to S-CSCF) – see example in Table 10.1.2-14**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#1 forwards the Resume request to S-CSCF#1.

Table 10.1.2-14: INVITE(Resume) (P-CSCF to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

15. **INVITE(Resume) (S-CSCF to S-CSCF) – see example in Table 10.1.2-15**

S-CSCF#1 forwards the Resume request to S-CSCF#2.

Table 10.1.2-15: INVITE(Resume) (S-CSCF to S-CSCF)

```
INVITE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

16. **INVITE(Resume) (S-CSCF to P-CSCF) – see example in Table 10.1.2-16**

S-CSCF#2 forwards the Resume request to P-CSCF#2.

Table 10.1.2-16: INVITE(Resume) (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

17. **INVITE(Resume) (P-CSCF to UE) – see example in Table 10.1.2-17**

P-CSCF#2 forwards the Resume request to UE#2.

Table 10.1.2-17: INVITE(Resume) (P-CSCF to UE)

```
INVITE sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

18. **Resume media flow**

UE#2 resumes sending the media stream to the remote endpoint.

19. **200-OK (UE to P-CSCF) – see example in Table 10.1.2-19**

UE#2 acknowledges receipt of the Resume request (17) with a 200-OK final response, sent to P-CSCF#2.

Table 10.1.2-19: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::eee:fff:aaa:bbb
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=audio 6402 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
```

20. **200-OK (P-CSCF to S-CSCF) – see example in Table 10.1.2-20**

P-CSCF#2 forwards the 200 OK final response to S-CSCF#2.

Table 10.1.2-20: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

21. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.1.2-21**

S-CSCF#2 forwards the 200 OK final response to S-CSCF#1.

Table 10.1.2-21: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

22. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.1.2-22**

S-CSCF#1 forwards the 200 OK final response to P-CSCF#1.

Table 10.1.2-22: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

23. **200-OK (P-CSCF to UE) – see example in Table 10.1.2-23**

P-CSCF#1 forwards the 200 OK final response to UE#1.

Table 10.1.2-23: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

24. **Resume Media Flow**

UE#1 resumes sending the media stream to the remote endpoint.

10.1.3 Mobile-initiated hold and resume of a mobile-PSTN session

An IM session was previously established between an initiating UE and a MGCF acting as a gateway for a session terminating on the PSTN, or between an initiating MGCF acting as a gateway for a session originating on the PSTN to a terminating UE. The UE has an associated P-CSCF in the same network where it is currently located (either home or roaming), an S-CSCF assigned in its home network, and a BGCF that chooses the MGCF. These functional elements

co-operate to clear the session, and the procedures are independent of whether they are located in the subscriber's home or visited networks. Therefore there is no distinction in this clause of home network vs. visited network.

The session hold and resume procedure is similar whether the UE initiated the session to the PSTN, or if the PSTN initiated the session to the UE. The only difference is the optional presence of the BGCF in the case of a session initiated by the UE. Note that the BGCF might or might not be present in the signalling path after the first INVITE is routed.

These procedures show only one combination of Mobile-Originated, Serving-to-Serving, and Mobile-Terminated procedures, MO#2, S-S#3, and PSTN-T. These procedures do not show the use of optional I-CSCFs, or the use of the BGCF in achieving network configuration independence. If an I-CSCF/BGCF was included in the signalling path during the session establishment procedure, it would continue to be used in any subsequent flows such as the ones described in this clause. Procedures at the I-CSCFs are identical to those described for the BYE, PRACK, and COMET requests and responses described in other clauses.

The procedures for placing a media stream on hold, and later resuming the media stream, are as shown in Figure 10.1.3-1:
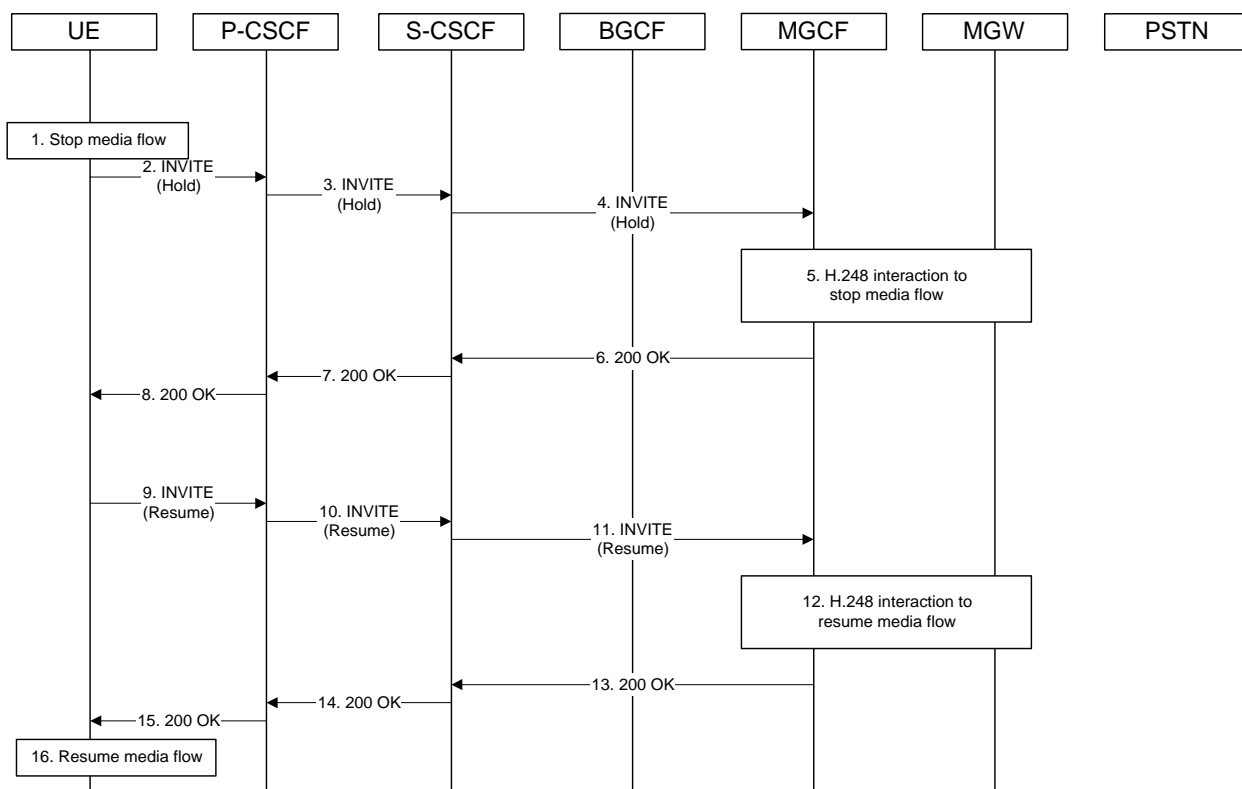
Figure 10.1.3-1: Mobile to PSTN session hold and resume

Information flow procedures are as follows:

1. **Stop Media Flow**

   UE#1 detects a request from the subscriber to place a media stream on hold. UE#1 stops sending the media stream to the remote endpoint, but keeps the resources for the session reserved.

2. **INVITE (Hold) (UE to P-CSCF) – see example in 10.1.3-2**

   UE sends a Hold request to its proxy, P-CSCF.

Table 10.1.3-2: INVITE (Hold) (UE to P-CSCF)

```
INVITE sip:token6@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 0::0:0:0:0
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

SDP: Same SDP as negotiated during the session setup, but with the "c=" line having an IP address of zeroes. The IP address of zeroes indicates a session hold request.

3. **INVITE (Hold) (P-CSCF to S-CSCF) – see example in Table 10.1.3-3**

   P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

   P-CSCF forwards the Hold request to S-CSCF.

Table 10.1.3-3: INVITE (Hold) (P-CSCF to S-CSCF)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:mgcf.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

4. **INVITE (Hold) (S-CSCF to MGCF) – see example in Table 10.1.3-4**

   S-CSCF forwards the Hold request to MGCF.

Table 10.1.3-4: INVITE (Hold) (S-CSCF to MGCF)

```
INVITE sip:mgcf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

5. **H.248 Interaction to Stop Media flow**

   MGCF initiates a H.248 interaction with MGW instructing it to stop sending the media stream, but to keep the resources for the session reserved.

6. **200-OK (MGCF to S-CSCF) – see example in Table 10.1.3-6**

   MGCF acknowledges receipt of the Hold request (4) with a 200-OK final response, sent to S-CSCF.

Table 10.1.3-6: 200 OK (MGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::eee:fff:aaa:bbb
s=-
c= IN IP6 0::0:0:0:0
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
```

7. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.1.3-7**

S-CSCF forwards the 200 OK final response to P-CSCF.

Table 10.1.3-7: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

8. **200-OK (P-CSCF to UE) – see example in Table 10.1.3-8**

   P-CSCF forwards the 200 OK final response to UE.

Table 10.1.3-8: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

9. **INVITE(Resume) (UE to P-CSCF) – see example in Table 10.1.3-9**

   UE detects a request from the subscriber to resume the media stream previously placed on hold. UE sends a Resume request to its proxy, P-CSCF.

Table 10.1.3-9: INVITE(Resume) (UE to P-CSCF)

```
INVITE sip:token6@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

SDP: Same SDP as negotiated during the session setup, but with the "c=" line having the actual IP address.

10. **INVITE(Resume) (P-CSCF to S-CSCF) – see example in Table 10.1.3-10**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF forwards the Resume request to S-CSCF.

Table 10.1.3-10: INVITE(Resume) (P-CSCF to S-CSCF)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:mgcf.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

11. **INVITE(Resume) (S-CSCF to MGCF) – see example in Table 10.1.3-11**

   S-CSCF forwards the Resume request to MGCF.

Table 10.1.3-11: INVITE(Resume) (S-CSCF to MGCF)

```
INVITE sip:mggcf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

12. **H.248 Interaction to Resume media flow**

   MGCF initiates a H.248 interaction with MGW instructing it to resume sending the media stream.

13. **200-OK (MGCF to S-CSCF) – see example in Table 10.1.3-13**

   MGCF acknowledges receipt of the Resume request (11) with a 200-OK final response, sent to S-CSCF.

Table 10.1.3-13: 200 OK (MGCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::eee:fff:aaa:bbb
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=audio 6402 RTP/AVP 97
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
```

14. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.1.3-14**

   S-CSCF forwards the 200 OK final response to P-CSCF.

Table 10.1.3-14: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

15. **200-OK (P-CSCF to UE) – see example in Table 10.1.3-15**

    P-CSCF forwards the 200 OK final response to UE.

Table 10.1.3-15: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

16. **Resume Media Flow**

    UE resumes sending the media stream to the remote endpoint

10.2 Initiating and destination party identification

10.2.1 Introduction

When the UE (or MGCF) initiates a session in the IM CN subsystem, it shall determine, based on preferences of the initiating user, whether its identity is to be made available to the destination, or to remain anonymous. Three cases shall be distinguished:

1. The initiating user desires his/her identity to be anonymous

2. The initiating user desires to be identified as the initiator of the session

3. The initiating user did not state a preference for this session.

The values of the headers "From", "To", "Call-ID", "Remote-Party-ID", and "Contact" shall be based on the decision above.

When the UE (or MGCF) receives an incoming session request in the IM CN subsystem, it shall determine, based on preferences of the destination user, whether its identity is to be made available to the initiator or to remain anonymous. Three cases shall be distinguished:

4. The destination user desires his/her identity to be anonymous

5. The destination user desires to be identified as the destination of the session

6. The destination user did not state a preference for this session.

The values of the header "Remote-Party-ID" shall be based on the decision above.

The rules for processing the header values at a proxy are given in draft-ietf-sip-privacy-01.

10.2.2   IM sessions with session initiator desiring anonymity

If the initiating user desires the session to be anonymous, the following rules shall be followed in generating header values:

From:	UE shall provide a cryptographically random identifier for the userinfo, and a non-identifying hostname, e.g. "localhost" in the host name. Username shall be empty.
To:	If a telephone number is used in the addr-spec, the UE shall provide a full E.164 number including the country code. Otherwise, the UE shall provide a cryptographically random identifier for the userinfo, different from the value of the From header, and a non-identifying hostname, e.g. "localhost" in the host name.
Call-ID:	UE shall provide a cryptographically random identifier for the userinfo, and a non-identifying hostname, e.g. "localhost" in the host name. This may be identical to the "From" header value.
Contact:	The userinfo shall either be empty, or be the same cryptographically random identifier that appears in the From header. The hostname shall be an IP address rather than an FQDN.
Remote-Party-ID:	UE shall include the subscriber identity and URL in the Remote-Party-ID header, with a tag "privacy=full"

An example of an initial INVITE request following the rules for an anonymous session is given in Table 10.2.2-1. This revised information would appear as step #1 of MO#1a (Subclause 8.2.2), MO#1b (Subclause 8.2.3), MO#2 (Subclause 8.2.4), and step #4 of PSTN-O (Subclause 8.2.5).

Table 10.2.2-1: INVITE (Anonymous session)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=full
Proxy-Require: privacy
Anonymity: Off
From: sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(+1-212-555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

From: Contains a cryptographically random identifier for the userinfo, and a non-identifying hostname ("localhost") for the hostname. Username is empty.

To: Contains a cryptographically random identifier for the userinfo, distinct from the value of the From header, and a non-identifying hostname ("localhost") for the hostname. Username is empty.

Call-ID: Contains a cryptographically random identifier for the useringo, and a non-identifying hostname ("localhost") for the hostname. This value is, in this example, identical to the From header value.

The values of From, To, Call-ID, and Remote-Party-ID, as given above, are carried through the INVITE sequence, through the S-CSCF serving the destination subscriber. When S-CSCF#2 forwards the INVITE request to the termination procedure (step #11 of S-S#1a, step #13 of S-S#1b, step#11 of S-S#2, step#4 of MT#1a, step#4 of MT#1b, step#4 of MT#2), the Remote-Party-ID header is updated with a private URL. An example of this INVITE request is given in table 10.2.2-2.

Table 10.2.2-2: INVITE (S-S to MT)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
Supported:
Remote-Party-ID: <sip:token(tel:+1-212-555-1111)@scscf.home2.net;user=private>
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Remote-Party-ID: Contains a cryptographically random identifier for the userinfo, generated from the originating subscriber information, and the hostname identifying the S-CSCF that generated the userinfo string. Username is empty.

When an I-CSCF is used to maintain configuration independence, it may (based on operator preferences) update the Remote-Party-ID header in order to hide the S-CSCF address. This occurs in MT#1b step #5. If so, it generates a new private URL with its own hostname. An example of this INVITE request is given in table 10.2.2-3.

Table 10.2.2-3: INVITE (I-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf.home2.net, sip:scscf.home1.net
Supported:
Remote-Party-ID: <sip:token(sip:token(tel:+1-212-555-1111)@scscf.home2.net;
user=private)@icscf.home2.net;user=private>
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Remote-Party-ID: Contains a cryptographically random identifier for the userinfo, generated from the originating subscriber information, and the hostname identifying the I-CSCF that generated the userinfo string. Username is empty.

10.2.3 IM sessions with session initiator being identified

If the initiating user desires to be identified as the session originator, the following rules shall be followed in generating header values:

From:	UE may provide a cryptographically random identifier for the userinfo, and a non-identifying hostname, e.g. "localhost" in the host name. The username may be provided as an identification string.
To:	If a telephone number is used in the addr-spec, the UE may provide a full E.164 number including the country code. Otherwise, the UE may provide a cryptographically random identifier for the userinfo, which shall be different from the value of the From header, and a non-identifying hostname, e.g. "localhost" in the host name. The username may be provided as an identification string.
Call-ID:	UE may provide a cryptographically random identifier for the userinfo, and a non-identifying hostname, e.g. "localhost" in the host name. This may be identical to the "From" header value.
Contact:	The userinfo may either be empty, or be the same cryptographically random identifier that appears in the From header. The hostname may be an IP address or an FQDN.
Remote-Party-ID:	UE shall include the subscriber identity and URL in the Remote-Party-ID header, with a tag "privacy=off"

An example of an initial INVITE request following the rules for an identified session is given in Table 10.2.3-1.

Table 10.2.3-1: INVITE (Identified session)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>; privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <tel:+1-212-555-1111>;tag=171828
To: sip:555-2222@home.net;user=phone
Call-ID: B36(SHA-1(+1-212-555-1111;time=36123E5B;seq=72))@[5555::aaa:bbb:ccc:ddd]
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

An additional example of information flows for this situation are those contained in Clause 8.

10.2.4 IM sessions without initiator preference for anonymity

If the initiating user did not state a preference for whether the session be anonymous, local policies and regulations may force the network operator to make it anonymous. Therefore, the following rules shall be followed in generating header values:

From:	UE shall provide a cryptographically random identifier for the userinfo, and a non-identifying hostname, e.g. "localhost" in the host name. Username may be empty, or may provide an identifying string. Any identifying string should not identify the subscriber.
To:	If a telephone number is used in the addr-spec, the UE shall provide a full E.164 number including the country code. Otherwise, the UE shall provide a cryptographically random identifier for the userinfo, different from the value of the From header, and a non-identifying hostname, e.g. "localhost" in the host name.
Call-ID:	UE shall provide a cryptographically random identifier for the userinfo, and a non-identifying hostname, e.g. "localhost" in the host name. This may be identical to the "From" header value.
Contact:	The userinfo shall either be empty, or be the same cryptographically random identifier that appears in the From header. The hostname shall be an IP address rather than an FQDN.
Remote-Party-ID:	UE shall include the subscriber identity and URL in the Remote-Party-ID header, without a "privacy" tag.

An example of an initial INVITE request following the rules for an unspecified session is given in Table 10.2.4-1.

Table 10.2.4-1: INVITE (Unspecified session)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(+1-212-555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

From: Contains a cryptographically random identifier for the userinfo, and a non-identifying hostname ("localhost") for the hostname. Username contains a string that does not identify the subscriber.

To: Contains a cryptographically random identifier for the userinfo, distinct from the value of the From header, and a non-identifying hostname ("localhost") for the hostname. Username is empty.

Call-ID: Contains a cryptographically random identifier for the useringo, and a non-identifying hostname ("localhost") for the hostname. This value is, in this example, identical to the From header value.

The values of From, To, Call-ID, and Remote-Party-ID, as given above, are carried through the INVITE sequence, through the S-CSCF serving the destination subscriber.

Based on local policy or regulatory requirements, the S-CSCF serving the destination subscriber may either allow the identification information to be given to the destination (by following the example in subclause 10.2.2), or may restrict it (by following the example in subclause 10.2.1).

10.2.5   IM sessions with destination requesting anonymity

If the destination user desires the session to be anonymous, the UE shall indicate this in the value of the Remote-Party-ID header in the first non-100 response to the initial INVITE. An example of this response from UE to P-CSCF (step#8 of MT#1a, step#10 of MT#1b, step#8 of MT#2), is given in Table 10.2.5-1.

Table 10.2.5-1: 183 Session Progress (UE to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=full
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
```

Remote-Party-ID: identifies the answering subscriber. It contains the public identifier URL, and the name of the answering party. The tag "privacy=full" is appended to indicate Anonymity is requested.

The value of the Remote-Party-ID header is carried through the 183-Session-Progress sequence, to the S-CSCF serving the initiating subscriber. When S-CSCF#1 forwards the 183-Session-Progress response to the originating procedure (step#16 of S-S#1a, step#19 of S-S#1b, step#16 of S-S#2, also step#9 of MO#1a, step#11 of MO#1b, step#9 of MO#2), the Remote-Party-ID header is updated with a private URL. An example of this 183-Session-Progress response is given in table 10.2.5-2.

Table 10.2.5-2: 183 Session Progress (S-SCSF to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID: <sip:token(tel:+1-212-555-2222)@scscf.home1.net;user=private>
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
```

> **Remote-Party-ID:** Contains a cryptographically random identifier for the userinfo, generated from the originating subscriber information, and the hostname identifying the S-CSCF that generated the userinfo string. Username is empty.

When an I-CSCF is used to maintain configuration independence, it may (based on operator preferences) update the Remote-Party-ID header in order to hide the S-CSCF address. This occurs in MO#1b step #12. If so, it generates a new private URL with its own hostname. An example of this INVITE request is given in table 10.2.5-3.

Table 10.2.5-3: 183 Session Progress (S-SCSF to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Remote-Party-ID: <sip:token(sip:token(tel:+1-212-555-2222)@scscf.home1.net;
    user=private)@icscf.home1.net;user=private>
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
```

> **Remote-Party-ID:** Contains a cryptographically random identifier for the userinfo, generated from the originating subscriber information, and the hostname identifying the I-CSCF that generated the userinfo string. Username is empty.

10.2.6   IM sessions with destination party being identified

If the destination user desires to be identified, the privacy tag of the Remote-Party-ID header indicate "privacy=off".
An example of this response from UE to P-CSCF (step#8 of MT#1a, step#10 of MT#1b, step#8 of MT#2), is given in
Table 10.2.6-1.

Table 10.2.6-1: 183 Session Progress (UE to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
```

Remote-Party-ID: identifies the answering subscriber. It contains the public identifier URL, and the name of the
answering party. The tag "privacy=off" is appended to indicate Anonymity is not requested.

An additional example of information flows for this situation are those contained in Clause 8

10.2.7   IM sessions without destination preference for anonymity

If the destination user did not state a preference for whether the session be anonymous, local policies and regulations
may force the network operator to make it anonymous. The destination UE indicates its lack of preference by not
providing a "privacy" tag on the Remote-Party-ID header. An example of this response from UE to P-CSCF (step#8 of
MT#1a, step#10 of MT#1b, step#8 of MT#2), is given in Table 10.2.7-1.

Table 10.2.7-1: 183 Session Progress (UE to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
```

Remote-Party-ID: identifies the answering subscriber. It contains the public identifier URL, and the name of the answering party. The lack of a tag "privacy=" indicates lack of a preference for an anonymous or identified session..

Based on local policy or regulatory requirements, the S-CSCF serving the originating subscriber may either allow the identification information to be given to the initiator (by following the example in subclause 10.2.6), or may restrict it (by following the example in subclause 10.2.5).

10.3 Procedures for codec and media flow negotiations

10.3.1 Introduction

This clause gives information flows for the procedures for determining the set of mutually-supported codecs between the endpoints of a multi-media session, determining the initial codecs to be used for the multi-media session, and the procedures for changing between codecs when multiple ones are supported.

Editor's note: If transcoding is to be supported, these procedures need to be adjusted.

10.3.2 Codec or media flow change within the existing reservation

After the multi-media session is established, it is possible for either endpoint to change the set of media flows or codec for a media flow. If the change is within the resources already reserved, then it is only necessary to synchronise the change with the other endpoint. Note that an admission control decision will not fail if the new resource request is within the existing reservation.

The information flow for changing a codec within an existing reservation is given in Figure 10.3.2-1.
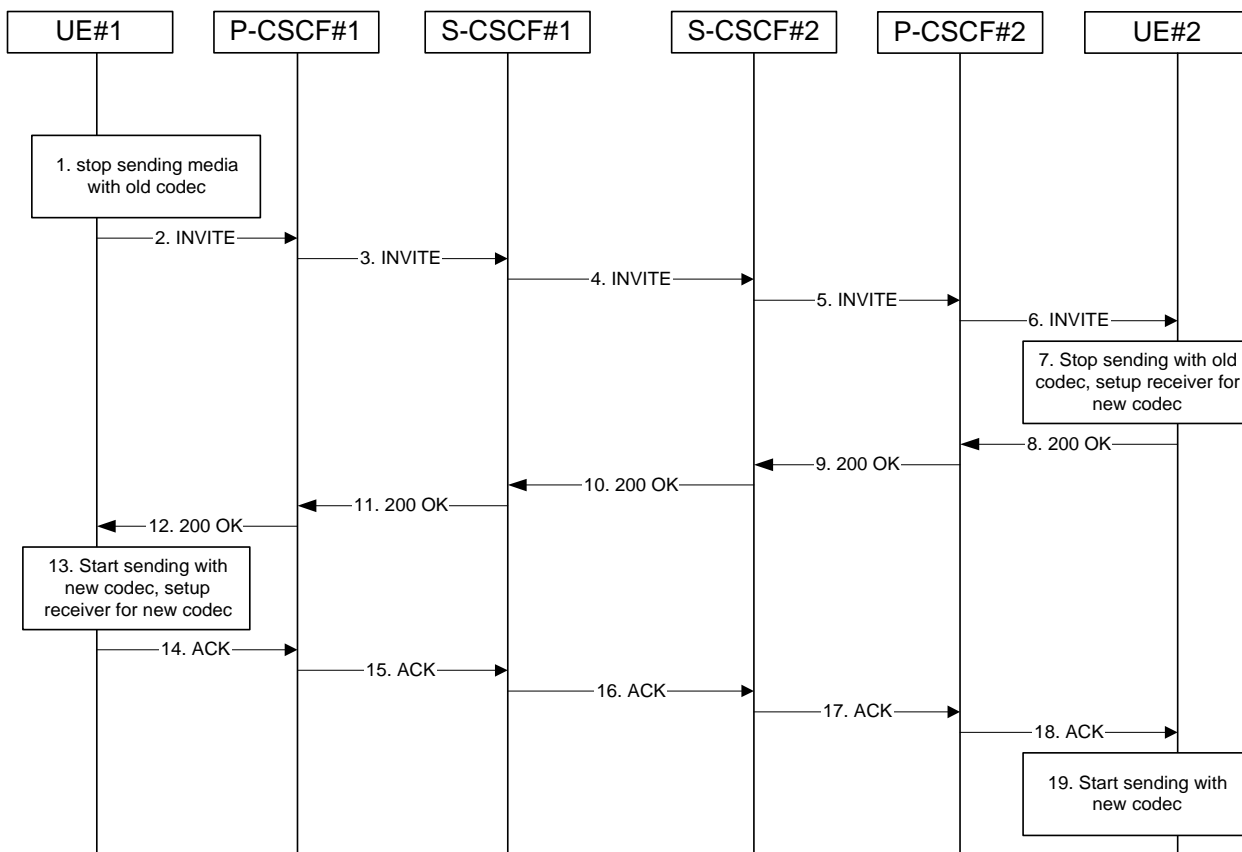
Figure 10.3.2-1: Codec or media flow change - same reservation

For this example, we assume the session was established with authorization for two codecs, AMR and G726-32, but that AMR was initially chosen for the media. UE#1 now desires to change the media to use G726-32.

The detailed procedure is as follows:

1. **UE#1 stops sending media with old codec.**

   UE#1 determines that a new media stream is desired, or that a change is needed in the codec in use for an existing media stream. UE#1 evaluates the impact of this change, and determines the existing resources reserved for the session are adequate. UE#1 builds a revised SDP that includes all the common media flows determined by the initial negotiation, but assigns a codec and port number only to those to be used onward. UE#1 stops transmitting media streams on those to be dropped from the session.

2. **INVITE (UE to P-CSCF) – see example in Table 10.3.2-2**

   UE#1 sends the INVITE request to P-CSCF#1 containing this SDP

Table 10.3.2-2: INVITE (UE to P-CSCF)

```
INVITE sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555:: aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP
m=video 0 RTP/AVP
m=audio 3456 RTP/AVP 96
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

SDP The SDP contains the revised set of codecs desired by UE#1

3. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.3.2-3**

   P-CSCF#1 forwards the INVITE request to S-CSCF#1

Table 10.3.2-3: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

4. **INVITE (S-CSCF to S-CSCF) – see example in Table 10.3.2-4**

S-CSCF#1 forwards the INVITE request, through the S-CSCF to S-CSCF Session Flow Procedures, to S-CSCF#2.

Table 10.3.2-4: INVITE (S-CSCF to S-CSCF)

```
INVITE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

5. **INVITE (S-CSCF to P-CSCF) – see example in Table 10.3.2-5**

   S-CSCF#3 forwards the INVITE request to P-CSCF#2.

Table 10.3.2-5: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

6. **INVITE (P-CSCF to UE) – see example in Table 10.3.2-6**

   P-CSCF#2 forwards the INVITE request to UE#2

Table 10.3.2-6: INVITE (P-CSCF to UE)

```
INVITE sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
Media-Authorization: 00200001001001706366322e78797a2e6e6574000c020133315331343363231
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

7. **UE#2 stops sending with old codec, and initializes receiver for new codec**

UE#2 receives the INVITE message, and agrees that it is a change within the previous resource reservation. UE#2 stops sending the media streams to be deleted, and initialises its media receivers for the new codec.

UE#2 may optionally perform an alerting function at this point, and respond to UE#1 with a 180 Ringing provisional response (not shown in figure). When it is ready for the new media stream, UE#2 responds with a 200 OK.

8. **200 OK (UE to P-CSCF) – see example in Table 10.3.2-8**

UE#2 responds to the INVITE request (6) with a 200 OK response, sent to P-CSCF#2

Table 10.3.2-8: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq: 131 INVITE
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::eee:fff:aaa:bbb
s=-
c= IN IP6 5555:: eee:fff:aaa:bbb
t=907165275 0
m=video 0 RTP/AVP
m=video 0 RTP/AVP
m=audio 6544 RTP/AVP 96
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP
```

9. **200 OK (P-CSCF to S-CSCF) – see example in Table 10.3.2-9**

P-CSCF#2 sends the 200 OK response to S-CSCF#2

Table 10.3.2-9: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

10. **200 OK (S-CSCF to S-CSCF) – see example in Table 10.3.2-10**

S-CSCF#2 sends the 200 OK response to S-CSCF#1

Table 10.3.2-10: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

11. **200 OK (S-CSCF to P-CSCF) – see example in Table 10.3.2-11**

S-CSCF#1 sends the 200 OK response to P-CSCF#1

Table 10.3.2-11: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

12. **200 OK (P-CSCF to UE) – see example in Table 10.3.2-12**

   P-CSCF#1 sends the 200 OK response to UE#1

Table 10.3.2-12: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token6@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

   P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

   Contact: a locally unique token to identify the saved routing information

13. **UE#1 starts sending with new codec, and initializes receiver for new codec**

   UE#1 starts sending media using the new codecs. UE#1 also releases any excess resources no longer needed.

14. **ACK (UE to P-CSCF) – see example in Table 10.3.2-14**

   UE#1 sends the ACK request to UE#2, along the signalling path established by the INVITE request

Table 10.3.2-14: ACK (UE to P-CSCF)

```
ACK sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 ACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

15. **ACK (P-CSCF to S-CSCF) – see example in Table 10.3.2-15**

   P-CSCF#1 sends the ACK request to S-CSCF#1, along the signalling path established by the INVITE request

Table 10.3.2-15: ACK (P-CSCF to S-CSCF)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200 OK response (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

16. **ACK (S-CSCF to S-CSCF) – see example in Table 10.3.2-16**

   S-CSCF#1 sends the ACK request to S-CSCF#2, along the signalling path established by the INVITE request

Table 10.3.2-16: ACK (S-CSCF to S-CSCF)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Length:
```

17. **ACK (S-CSCF to P-CSCF) – see example in Table 10.3.2-17**

   S-CSCF#2 sends the ACK request to P-CSCf#2, along the signalling path established by the INVITE request.

Table 10.3.2-17: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

18. **ACK (P-CSCF to UE) – see example in Table 10.3.2-18**

   P-CSCF#2 sends the ACK request to UE#2, along the signalling path established by the INVITE request

Table 10.3.2-18: ACK (P-CSCF to UE)

```
ACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token5
From:
To:
Call-ID:
Cseq:
Contact: token5@pcscf2.home.net
Content-length:
```

   P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

   Contact: a locally unique token to identify the saved routing information.

   Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

19. **UE#2 starts sending with new codec**

   UE#2 starts sending media using the new codecs. UE#2 also releases any excess resources no longer needed.

10.3.3   Codec or media flow change requiring new resources and/or authorisation

After the multi-media session is established, it is possible for either endpoint to change the set of media flows or codec for a media flow. If the change requires additional resources beyond those previously reserved, then it is necessary to perform the resource reservation and bearer establishment procedures. If the reservation request fails for whatever reason, the original multi-media session remains in progress.

An example information flow for a codec or media flow change requiring new resources and/or authorization is given in Figure 10.3.2-1. This example shows mobile originated while in home network, establishing a session with another mobile served by the same network operator, also in its home network (MO#2, S-S#2, MT#2). Other configurations may include I-CSCFs in the signaling path; procedures at the I-CSCFs are identical to those described for the BYE, PRACK, and COMET requests and responses described in other clauses.
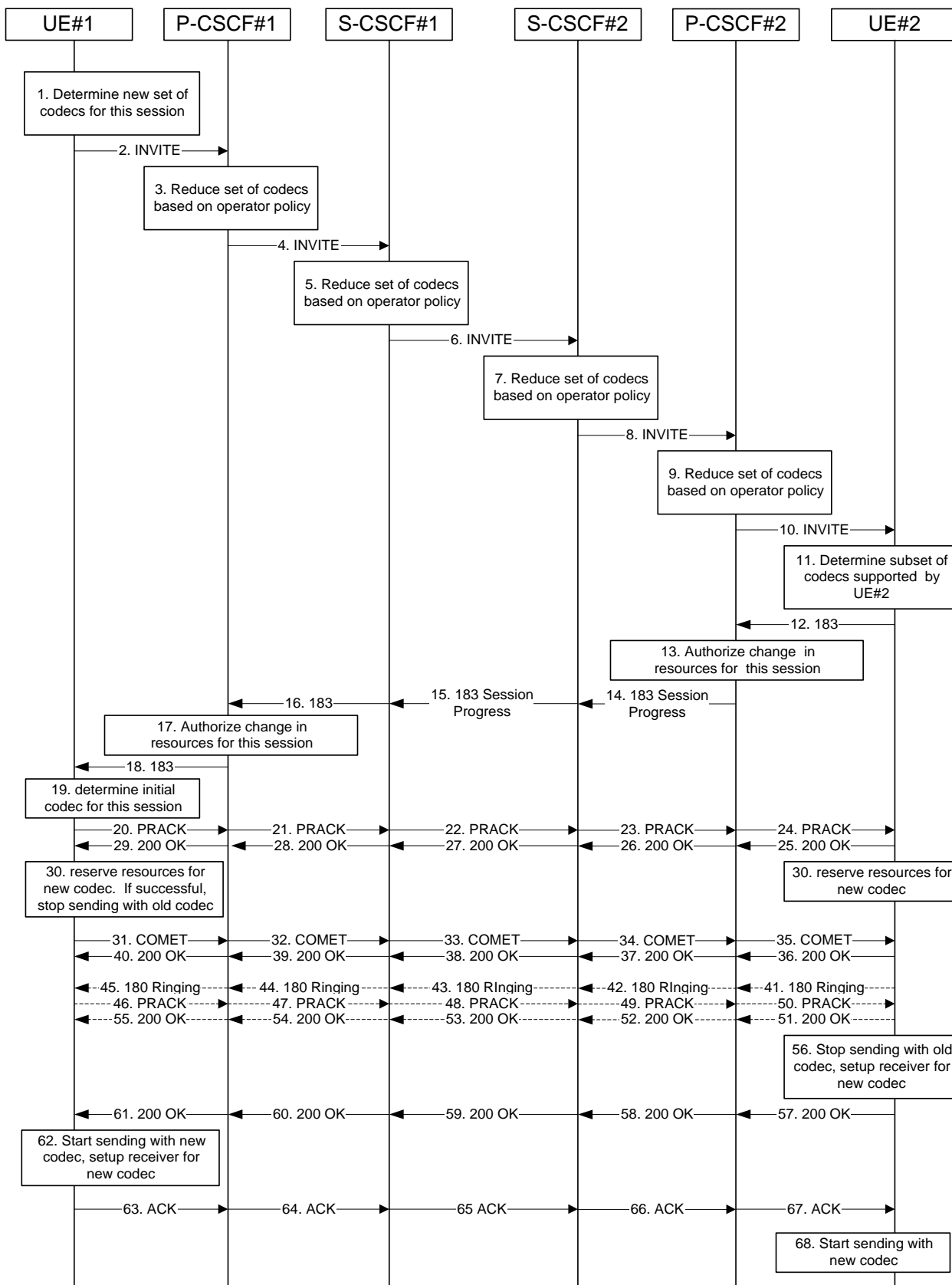
Figure 10.3.3-1: Codec or media flow change - new reservation

The detailed procedure is as follows:

1. **Determine new set of codecs for this session**

UE#1 determines the revised set of codecs that it is wishes to support for this session. It builds a SDP containing bandwidth requirements and characteristics of each, and assigns local port numbers for each possible media flow. Multiple media flows may be offered, and for each media flow (m= line in SDP), there may be multiple codec choices offered.

For this example, assume UE#1 originally established the session using audio (AMR) only, and now wishes to change to stereo (using the L16 2-channel codec, RTP/AVP code 10) and add an additional video media stream (MPV).

2. **INVITE (UE to P-CSCF) – see example in Table 10.3.3-2**

UE#1 sends the INVITE request to P-CSCF#1 containing this SDP

Table 10.3.3-2: INVITE (UE to P-CSCF)

```
INVITE sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 10
a=qos:mandatory sendrecv
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

SDP The SDP contains the revised set of codecs desired by UE#1

3. **P-CSCF reduces set of supported codecs based on operator policy**

P-CSCF#1 examines the media parameters, and removes any choices that the network operator decides based on local policy, not to allow on the network.

4. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.3.3-4**

P-CSCF#1 forwards the INVITE request to S-CSCF#1

Table 10.3.3-4: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

5. **S-CSCF reduces set of supported codecs based on operator policy**

    S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

6. **INVITE (S-CSCF to S-CSCF) – see example in Table 10.3.3-6**

    S-CSCF#1 forwards the INVITE request, through the S-CSCF to S-CSCF Session Flow Procedures, to S-CSCF#2.

Table 10.3.3-6: INVITE (S-CSCF to S-CSCF)

```
INVITE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

7. **S-CSCF reduces set of supported codecs based on operator policy**

S-CSCF#2 examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

8. **INVITE (S-CSCF to P-CSCF) – see example in Table 10.3.3-8**

   S-CSCF#3 forwards the INVITE request to P-CSCF#2.

Table 10.3.3-8: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

9. **P-CSCF reduces set of supported codecs based on operator policy**

   P-CSCF#2 examines the media parameters, and removes any that the network operator decides, based on local policy, not to allow on the network.

10. **INVITE (P-CSCF to UE) – see example in Table 10.3.3-10**

    P-CSCF#2 forwards the INVITE request to UE#2

Table 10.3.3-10: INVITE (P-CSCF to UE)

```
INVITE sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

11. **Determine subset of codecs supported by UE#2**

UE#2 determines the subset of codecs that it is capable of supporting for this session. It determines the intersection of those it supports with those appearing in the SDP in the INVITE request. For each media flow that is not supported, UE#2 inserts a SDP entry for media (m= line) with port=0. For each media flow that is supported, UE#2 inserts a SDP entry with an assigned port and with the codecs in common with those in the SDP from UE#1.

For this example, assume UE#2 supports all those requested by UE#1

12. **183 Session Progress (UE to P-CSCF) – see example in Table 10.3.3-12**

UE#2 returns a 183 Session Progress response, containing SDP listing common media flows and codecs, to P-CSCF#2.

Table 10.3.3-12: 183 Session Progress (UE to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf2.visited.net;branch=token3
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 18
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::eee:fff:aaa:bbb
s=-
c= IN IP6 5555:: eee:fff:aaa:bbb
t=907165275 0
m=video 6540 RTP/AVP 99
a=qos:mandatory sendrecv confirm
a=rtpmap:99:MPV
m=audio 6544 RTP/AVP 10
a=qos:mandatory sendrecv confirm
```

SDP The SDP contains the subset of codecs supported by UE#2.

13. **Authorize resources for common codecs for this session**

P-CSCF#2 authorises the QoS resources for the common media flows and codec choices.

14. **183 Session Progress (P-CSCF to S-CSCF) - see example in Table 10.3.3-14**

P-CSCF#2 forwards the 183 Session Progress response to S-CSCF#2.

Table 10.3.3-14: 183 Session Progress (P-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaaa%3abbbb%5d@pcscf2.home.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

15. **183 Session Progress (S-CSCF to S-CSCF) – see example in Table 10.3.3-15**

   S-CSCF#2 forwards the 183 Session Progress response to S-CSCF#1

Table 10.3.3-15: 183 Session Progress (S-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

16. **183 Session Progress (S-CSCF to P-CSCF) – see example in Table 10.3.3-16**

   S-CSCF#1 forwards the 183 Session Progress response to P-CSCF#1

Table 10.3.3-16: 183 Session Progress (S-CSCF to P-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

17. **Authorize resources for common codecs for this session**

    P-CSCF#1 authorises the QoS resources for the remaining media flows and codec choices.

18. **183 Session Progress (P-CSCF to UE) – see example in Table 10.3.3-18**

    P-CSCF#1 forwards the 183 Session Progress response to UE#1

Table 10.3.3-18: 183 Session Progress (P-CSCF to UE)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Media-Authorization: 00200001001001017063666312e78797a2e6e6574000c02013942563330373200
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token1@pcscf1.home.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf1.home.net, sip:scscf2.home.net,
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
```

Contact: a locally unique token to identify the saved routing information.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf1.xyz.net" with credentials "9BV3072".

19. **Determine revised codec(s) for this session**

UE#1 determines which media flows should be used for this session, and which codecs should be used for each of those media flows. If there was any change in media flows, or if there was more than one choice of codec for a media flow, then UE#1 must include an SDP in the PRACK message sent to UE#2.

For this example, assume UE#1 chooses L10 for stereo audio and MPV for video, so no changes are made to the SDP.

20. **PRACK (UE to P-CSCF) – see example in Table 10.3.3-20**

UE#1 sends the PRACK request to UE#2, along the signalling path established by the INVITE request

Table 10.3.3-20: PRACK (UE to P-CSCF)

```
PRACK sip:token1@pcscf1.visited.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 132 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 18 131 INVITE
Content-length: 0
```

Request-URI: takes the value of the Contact header of the received 183 Session Progress response.

Via:, Contact: take the value of either the IP address of RQDN of the originating UE.

From:, To:, Call-ID: copied from the 183 Session Progress response so that they include any tag parameter.

Cseq: takes a higher value than that in the previous request.

21. **PRACK (P-CSCF to S-CSCF) – see example in Table 10.3.3-21**

P-CSCF#1 sends the PRACK request to S-CSCF#1, along the signalling path established by the INVITE request

Table 10.3.3-21: PRACK (P-CSCF to S-CSCF)

```
PRACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack:
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the previous response (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

22. **PRACK (S-CSCF to S-CSCF) – see example in Table 10.3.3-22**

S-CSCF#1 sends the PRACK request to S-CSCF#2, along the signalling path established by the INVITE request

Table 10.3.3-22: PRACK (S-CSCF to S-CSCF)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

Request-URI: the first component of the Route header. This will vary according to which S-CSCF to S-CSCF flow is used.

23. **PRACK (S-CSCF to P-CSCF) – see example in Table 10.3.3-23**

S-CSCF#2 sends the PRACK request to P-CSCf#2, along the signalling path established by the INVITE request.

Table 10.3.3-23: PRACK (S-CSCF to P-CSCF)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

24. **PRACK (P-CSCF to UE) – see example in Table 10.3.3-24**

P-CSCF#2 sends the PRACK request to UE#2, along the signalling path established by the INVITE request

Table 10.3.3-24: PRACK (P-CSCF to UE)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token2
From:
To:
Call-ID:
Cseq:
Contact: token2@pcscf2.home.net
Rack:
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

25. **200 OK (UE to P-CSCF) – see example in Table 10.3.3-25**

UE#2 responds to the PRACK request (24) with a 200 OK response to P-CSCF#2

Table 10.3.3-25: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token2
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

26. **200 OK (P-CSCF to S-CSCF) – see example in Table 10.3.3-26**

   P-CSCF#2 sends the 200 OK response to S-CSCF#2

Table 10.3.3-26: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

27. **200 OK (S-CSCF to S-CSCF) – see example in Table 10.3.3-27**

   S-CSCF#2 sends the 200 OK response to S-CSCF#1

Table 10.3.3-27: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

28. **200 OK (S-CSCF to P-CSCF) – see example in Table 10.3.3-28**

   S-CSCF#1 sends the 200 OK response to P-CSCF#1

Table 10.3.3-28: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

29. **200 OK (P-CSCF to UE) – see example in Table 10.3.3-29**

   P-CSCF#1 sends the 200 OK response to UE#1

Table 10.3.3-29: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf1.home.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

30. **Reserve resources for new media streams**

UE#1 and UE#2 reserve the resources needed for the added or changed media flows. If the reservation is successfully completed by UE#1, it stops transmitting any deleted media streams

31. **COMET (UE to P-CSCF) – see example in Table 10.3.3-31**

UE#1 sends the COMET request to P-CSCF#1

Table 10.3.3-31: COMET (UE to P-CSCF)

```
COMET sip:token1@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 133 COMET
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:success send
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 10
a=qos:success send
```

Request-URI: takes the value of the Contact header of the received 183 Session Progress response.

Via:, Contact: take the value of either the IP address or FQDN of the originating UE.

From:, To:, Call-ID: copied from the 183 Session Progress response so that they include any tag parameters.

CSeq: takes a higher value than that in the previous request.

The SDP indicates that the resource reservation was successful.

32. **COMET (P-CSCF to S-CSCF) – see example in Table 10.3.3-32**

P-CSCF#1 sends the COMET request to S-CSCF#1

Table 10.3.3-32: COMET (P-CSCF to S-CSCF)

```
COMET sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

Request-URI: the first component of the saved Route header.

Route: saved from the 183 Session Progress response (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

33. **COMET (S-CSCF to S-CSCF) – see example in Table 10.3.3-33**

S-CSCF#1 sends the COMET request to S-CSCF#2

Table 30.3.3-33: COMET (MO#2 to S-S)

```
COMET sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

Request-URI: the first component of the Route header. This will vary according to which S-CSCF to S-CSCF flow is used.

34. **COMET (S-CSCF to P-CSCF) – see example in Table 10.3.3-34**

S-CSCF#2 sends the COMET request to P-CSCF#2

Table 10.3.3-34: COMET (S-CSCF to P-CSCF)

```
COMET sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

35. **COMET (P-CSCF to UE) – see example in Table 10.3.3-35**

   P-CSCF#2 sends the COMET request to UE#2

Table 10.3.3-35: COMET (P-CSCF to UE)

```
COMET sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token6
From:
To:
Call-ID:
Cseq:
Contact: token6@pcscf2.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

36. **200 OK (UE to P-CSCF) – see example in Table 10.3.3-36**

   UE#2 responds to the COMET request (35) with a 200 OK response, sent to P-CSCF#2

Table 10.3.3-36: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token6
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

37. **200 OK (P-CSCF to S-CSCF) – see example in Table 10.3.3-37**

    P-CSCF#2 sends the 200 OK response to S-CSCF#2

Table 10.3.3-37: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

38. **200 OK (S-CSCF to S-CSCF) – see example in Table 10.3.3-38**

    S-CSCF#2 sends the 200 OK response to S-CSCF#1

Table 10.3.3-38: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

39. **200 OK (S-CSCF to P-CSCF) – see example in Table 10.3.3-39**

    S-CSCF#1 sends the 200 OK response to P-CSCF#1

Table 10.3.3-39: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

40. **200 OK (P-CSCF to UE) – see example in Table 10.3.3-40**

    P-CSCF#1 sends the 200 OK response to UE#1

Table 10.3.3-40: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token3@pcscf1.home.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

41. **180 Ringing (UE to P-CSCF) – see example in Table 10.3.3-41**

Depending on the type of codec change being performed, alerting may be required at the destination UE. If so, UE#2 sends a 180 Ringing provisional response to the originator, through P-CSCF#2.

Table 10.3.3-41: 180 Ringing (UE to P-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
Require: 100rel
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 19
Content-length: 0
```

42. **180 Ringing (P-CSCF to S-CSCF) – see example in Table 10.3.3-42**

P-CSCF#2 sends the 180 Ringing response to S-CSCF#2

Table 10.3.3-42: 180 Ringing (P-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

43. **180 Ringing (S-CSCF to S-CSCF) – see example in Table 10.3.3-43**

S-CSCF#2 sends the 180 Ringing response to S-CSCF#1

Table 10.3.3-43: 180 Ringing (S-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

44. **180 Ringing (S-CSCF to P-CSCF) – see example in Table 10.3.3-44**

S-CSCF#1 sends the 180 Ringing response to P-CSCF#1

Table 10.3.3-44: 180 Ringing (S-CSCF to P-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

45. **180 Ringing (P-CSCF to UE) – see example in Table 10.3.3-45**

P-CSCF#1 sends the 180 Ringing response to UE#1

Table 10.3.3-45: 180 Ringing (P-CSCF to UE)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token4@pcscf1.home.net
RSeq:
Content-length:
```

Editor's Note: Additional QoS interactions to handle one-way media at this point (e.g. for PSTN ringback and announcements) is for further study.

46. **PRACK (UE to P-CSCF) – see example in Table 10.3.3-46**

UE#1 sends the PRACK request to UE#2, along the signalling path established by the INVITE request

Table 10.3.3-46: PRACK (UE to P-CSCF)

```
PRACK sip:token4@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 19 131 INVITE
Content-length: 0
```

Request-URI: takes the value of the Contact header of the 180 Ringing response.

Via:, Contact: take the value of either the IP address or FQDN of the UE.

From:, To:, Call-ID: copied from the 180 Ringing response so that they include any revised tag parameters.

Cseq: takes a higher value than in the previous request.

47. **PRACK (P-CSCF to S-CSCF) – see example in Table 10.3.3-47**

P-CSCF#1 sends the PRACK request to S-CSCF#1, along the signalling path established by the INVITE request

Table 10.3.3-47: PRACK (P-CSCF to S-CSCF)

```
PRACK sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack:
Content-length:
```

Route: P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

48. **PRACK (S-CSCF to S-CSCF) – see example in Table 10.3.3-48**

S-CSCF#1 sends the PRACK request to S-CSCF#2, along the signalling path established by the INVITE request

Table 10.3.3-48: PRACK (S-CSCF to S-CSCF)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

49. **PRACK (S-CSCF to P-CSCF) – see example in Table 10.3.3-49**

S-CSCF#2 sends the PRACK request to P-CSCf#2, along the signalling path established by the INVITE request.

Table 10.3.3-49: PRACK (S-CSCF to P-CSCF)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

50. **PRACK (P-CSCF to UE) – see example in Table 10.3.3-50**

P-CSCF#2 sends the PRACK request to UE#2, along the signalling path established by the INVITE request

Table 10.3.3-50: PRACK (P-CSCF to UE)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token4
From:
To:
Call-ID:
Cseq:
Contact: token4@pcscf2.home.net
Rack:
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

51. **200 OK (UE to P-CSCF) – see example in Table 10.3.3-51**

UE#2 responds to the PRACK request (50) with a 200 OK response to P-CSCF#2

Table 10.3.3-51: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token4
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

52. **200 OK (P-CSCF to S-CSCF) – see example in Table 10.3.3-52**

P-CSCF#2 sends the 200 OK response to S-CSCF#2

Table 10.3.3-52: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

53. **200 OK (S-CSCF to S-CSCF) – see example in Table 10.3.3-53**

    S-CSCF#2 sends the 200 OK response to S-CSCF#1

Table 10.3.3-53: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

54. **200 OK (S-CSCF to P-CSCF) – see example in Table 10.3.3-54**

    S-CSCF#1 sends the 200 OK response to P-CSCF#1

Table 10.3.3-54: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

55. **200 OK (P-CSCF to UE) – see example in Table 10.3.3-55**

    P-CSCF#1 sends the 200 OK response to UE#1

Table 10.3.3-55: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token5@pcscf1.home.net
Content-Length:
```

   P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

   Contact: a locally unique token to identify the saved routing information

56. **Perform Codec change**

    UE#2 stops sending the media streams to be deleted, and initialises its media receivers for the new codec

57. **200 OK (UE to P-CSCF) – see example in Table 10.3.3-57**

    UE#2 responds to the INVITE request (10) with a 200 OK response, sent to P-CSCF#2

Table 10.3.3-57: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq: 131 INVITE
Content-length: 0
```

58. **200 OK (P-CSCF to S-CSCF) – see example in Table 10.3.3-58**

P-CSCF#2 sends the 200 OK response to S-CSCF#2

Table 10.3.3-58: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

59. **200 OK (S-CSCF to S-CSCF) – see example in Table 10.3.3-59**

S-CSCF#2 sends the 200 OK response to S-CSCF#1

Table 10.3.3-59: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

60. **200 OK (S-CSCF to P-CSCF) – see example in Table 10.3.3-60**

S-CSCF#1 sends the 200 OK response to P-CSCF#1

Table 10.3.3-60: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

61. **200 OK (P-CSCF to UE) – see example in Table 10.3.3-61**

P-CSCF#1 sends the 200 OK response to UE#1

Table 10.3.3-61: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token6@pcscf1.home.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

62. **Start using new codec**

UE#1 starts sending media using the new codecs. UE#1 also releases any excess resources no longer needed

63. **ACK (UE to P-CSCF) – see example in Table 10.3.3-63**

UE#1 sends the ACK request to UE#2, along the signalling path established by the INVITE request

Table 10.3.3-63: ACK (UE to P-CSCF)

```
ACK sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 ACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

64. **ACK (P-CSCF to S-CSCF) – see example in Table 10.3.3-64**

P-CSCF#1 sends the ACK request to S-CSCF#1, along the signalling path established by the INVITE request

Table 10.3.3-64: ACK (P-CSCF to S-CSCF)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200 OK response (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

65. **ACK (S-CSCF to S-CSCF) – see example in Table 10.3.3-65**

S-CSCF#1 sends the ACK request to S-CSCF#2, along the signalling path established by the INVITE request

Table 10.3.3-65: ACK (S-CSCF to S-CSCF)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Length:
```

66. **ACK (S-CSCF to P-CSCF) – see example in Table 10.3.3-66**

S-CSCF#2 sends the ACK request to P-CSCf#2, along the signalling path established by the INVITE request.

Table 10.3.3-66: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

67. **ACK (P-CSCF to UE) – see example in Table 10.3.3-67**

P-CSCF#2 sends the ACK request to UE#2, along the signalling path established by the INVITE request

Table 10.3.3-67: ACK (P-CSCF to UE)

```
ACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token5
From:
To:
Call-ID:
Cseq:
Contact: token5@pcscf2.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

68. **Start using new codec**

UE#2 starts sending media using the new codecs. UE#2 also releases any excess resources no longer needed

10.3.4 Error in changing codec or media flow within an existing reservation

After the multi-media session is established, it is possible for either endpoint to change the set of media flows or codec for a media flow. If the change is within the resources already reserved, then it is only necessary to synchronise the change with the other endpoint. Note that an admission control decision will not fail if the new resource request is within the existing reservation.

However, it is possible the destination UE can no longer support the requested codec, due to, for example, other simultaneous sessions involving the destination UE. The destination UE therefore has the ability to refuse the codec change.

The information flow for refusing a codec change within an existing reservation is given in Figure 10.3.4-1.
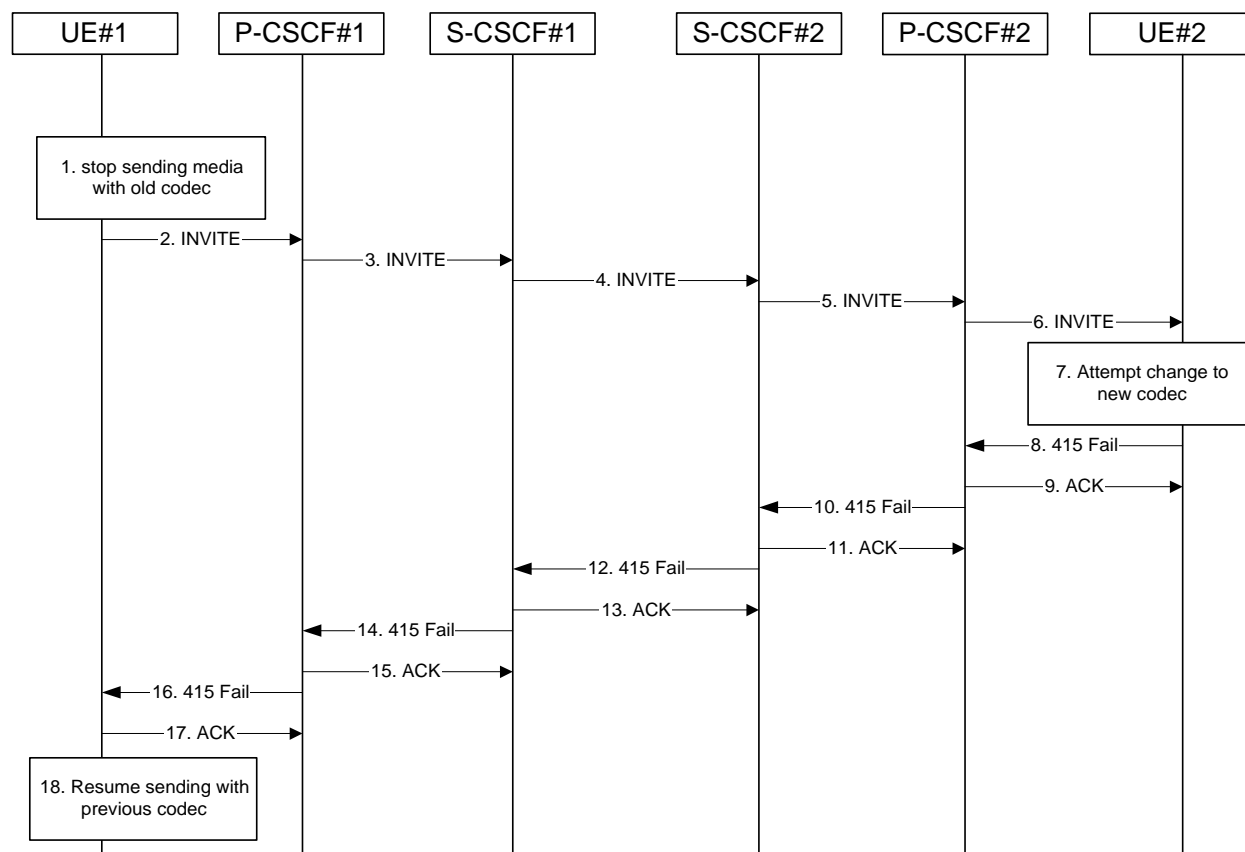
Figure 10.3.4-1: Error changing codec or media flow – within previous reservation

For this example, we assume the session was established with authorization for two codecs, AMR and G726-32, but that AMR was initially chosen for the media. UE#1 now desires to change the media to use G726-32.

The detailed procedure is as follows:

1. **UE#1 stops sending media with old codec.**

   UE#1 determines that a new media stream is desired, or that a change is needed in the codec in use for an existing media stream. UE#1 evaluates the impact of this change, and determines the existing resources reserved for the session are adequate. UE#1 builds a revised SDP that includes all the common media flows determined by the initial negotiation, but assigns a codec and port number only to those to be used onward. UE#1 stops transmitting media streams on those to be dropped from the session.

2. **INVITE (UE to P-CSCF) – see example in Table 10.3.4-2**

   UE#1 sends the INVITE request to P-CSCF#1 containing this SDP

Table 10.3.4-2: INVITE (UE to P-CSCF)

```
INVITE sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555:: aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP
m=video 0 RTP/AVP
m=audio 3456 RTP/AVP 96
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

SDP The SDP contains the revised set of codecs desired by UE#1

3. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.3.4-3**

   P-CSCF#1 forwards the INVITE request to S-CSCF#1

Table 10.3.4-3: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

4. **INVITE (S-CSCF to S-CSCF) – see example in Table 10.3.4-4**

S-CSCF#1 forwards the INVITE request, through the S-CSCF to S-CSCF Session Flow Procedures, to S-CSCF#2.

Table 10.3.4-4: INVITE (S-CSCF to S-CSCF)

```
INVITE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

5. **INVITE (S-CSCF to P-CSCF) – see example in Table 10.3.4-5**

S-CSCF#3 forwards the INVITE request to P-CSCF#2.

Table 10.3.4-5: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

6. **INVITE (P-CSCF to UE) – see example in Table 10.3.4-6**

P-CSCF#2 forwards the INVITE request to UE#2

Table 10.3.4-6: INVITE (P-CSCF to UE)

```
INVITE sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
m=
m=
a=
m=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

7. **UE#2 attempts to change to new codec**

UE#2 receives the INVITE message, and agrees that it is a change within the previous resource reservation. UE#2 encounters a failure attempting to change to the new codec, due to, e.g., internal resources that were available when the session was initiated but which are no longer available.

8. **415 Unsupported Media Type (UE to P-CSCF) – see example in Table 10.3.4-8**

UE#2 responds to the INVITE request (6) with a 415 Unsupported Media Type response, sent to P-CSCF#2

Table 10.3.4-8: 415 Unsupported Media Type (UE to P-CSCF)

```
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq: 131 INVITE
Content-length: 0
```

9. **ACK (P-CSCF to UE) – see example in Table 10.3.4-9**

P-CSCF#2 acknowledges the error response by sending an ACK request to UE#2

Table 10.3.4-9: ACK (P-CSCF to UE)

```
ACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

10. **415 Unsupported Media Type (P-CSCF to S-CSCF) – see example in Table 10.3.4-10**

   P-CSCF#2 sends the 415 Unsupported Media Type response to S-CSCF#2

Table 10.3.4-10: 415 Unsupported Media Type (P-CSCF to S-CSCF)

```
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
   SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

11. **ACK (S-CSCF to P-CSCF) – see example in Table 10.3.4-11**

   S-CSCF#2 acknowledges the error response by sending an ACK request to P-CSCf#2, along the signalling path
   established by the INVITE request.

Table 10.3.4-11: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

12. **415 Unsupported Media Type (S-CSCF to S-CSCF) – see example in Table 10.3.4-10**

   S-CSCF#2 sends the 415 Unsupported Media Type response to S-CSCF#1

Table 10.3.4-10: 415 Unsupported Media Type (S-CSCF to S-CSCF)

```
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

13. **ACK (S-CSCF to S-CSCF) – see example in Table 10.3.4-13**

   S-CSCF#1 acknowledges the error response by sending an ACK request to S-CSCF#2

Table 10.3.4-13: ACK (S-CSCF to S-CSCF)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net
From:
To:
Call-ID:
Cseq:
Content-Length:
```

14. **415 Unsupported Media Type (S-CSCF to P-CSCF) – see example in Table 10.3.4-11**

   S-CSCF#1 sends the 415 Unsupported Media Type response to P-CSCF#1

Table 10.3.4-11: 415 Unsupported Media Type (S-CSCF to P-CSCF)

```
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

15. **ACK (P-CSCF to S-CSCF) – see example in Table 10.3.4-15**

   P-CSCF#1 acknowledges the error response by sending an ACK request to S-CSCF#1

Table 10.3.4-15: ACK (P-CSCF to S-CSCF)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

16. **415 Unsupported Media Type (P-CSCF to UE) – see example in Table 10.3.4-16**

   P-CSCF#1 sends the 415 Unsupported Media Type response to UE#1

Table 10.3.4-16: 415 Unsupported Media Type (P-CSCF to UE)

```
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

17. **ACK (UE to P-CSCF) – see example in Table 10.3.4-17**

   UE#1 acknowledges the error response by sending an ACK request to P-CSCF#1

Table 10.3.4-17: ACK (UE to P-CSCF)

```
ACK sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
Cseq:
Content-length:
```

18. **UE#1 resumes sending with previous codec**

   UE#1 resumes sending media using the previous codecs.

10.3.5 Error changing codec or media flows requiring new resources and/or authorisation

After the multi-media session is established, it is possible for either endpoint to change the set of media flows or codec for a media flow. If the change requires additional resources beyond those previously reserved, then it is necessary to perform the resource reservation and bearer establishment procedures. If the reservation request fails for whatever reason, the original multi-media session remains in progress.

If the destination UE is unable, or unwilling, to change to the new set of codecs, it may return a 415 Unsupported Media Type error response.

If the P-CSCF and/or S-CSCF disallow a particular media flow or codec appearing in the SDP from the initiating UE, and it is the last codec in the last media flow, the CSCF shall return a 415 Unsupported Media Type error response.

An example information flow for an error changing codec or media flow requiring new resources and/or authorization is given in Figure 10.3.5-1. This is the case where the UE rejects the codec change; rejection by a CSCF is a subset of these information flows.

This example shows mobile originated while in home network, establishing a session with another mobile served by the same network operator, also in its home network (MO#2, S-S#2, MT#2). Other configurations may include I-CSCFs in the signaling path; procedures at the I-CSCFs are identical to those described for the BYE, PRACK, and COMET requests and responses described in other clauses.
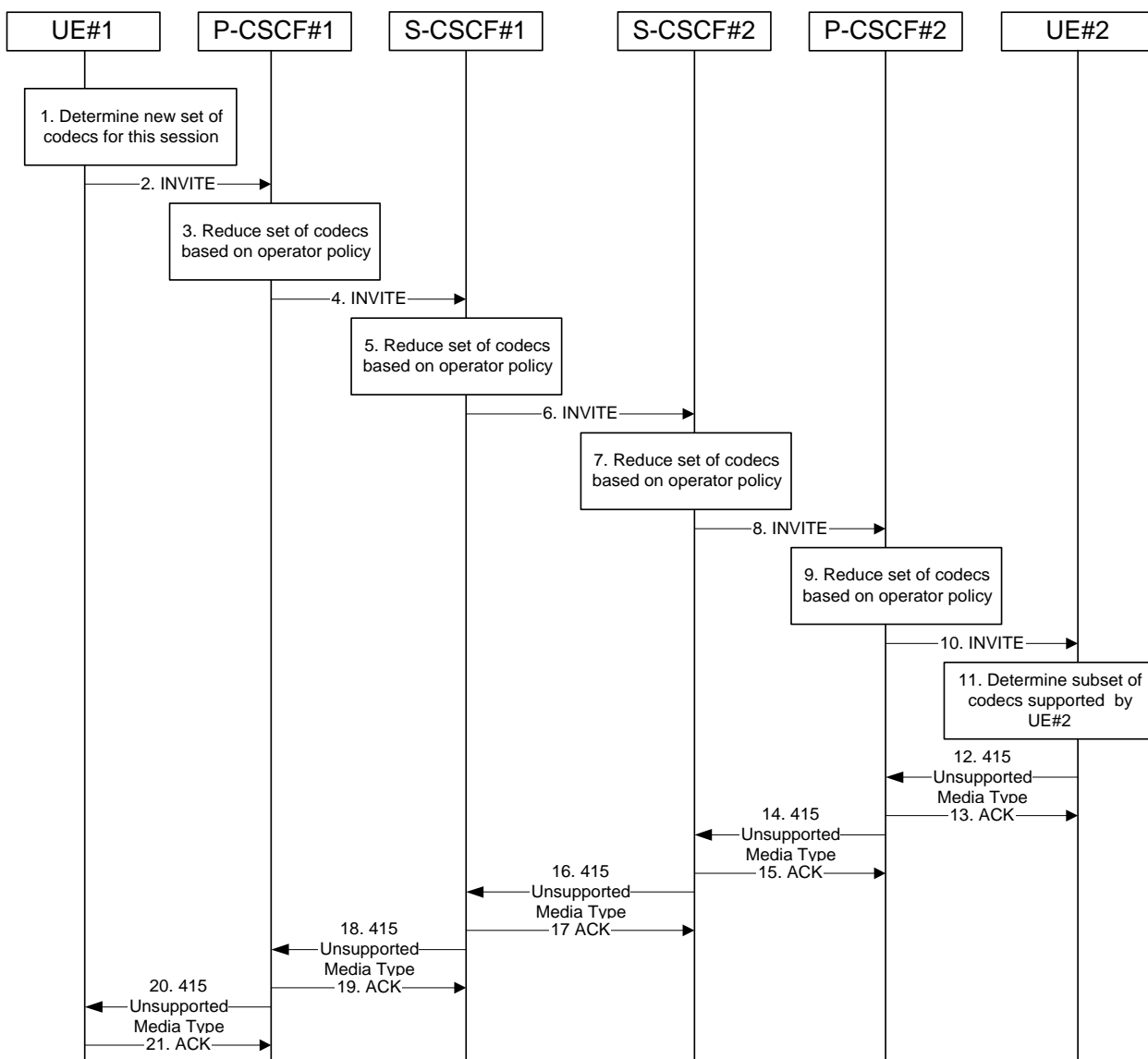
Figure 10.3.5-1: Error changing Codec or media flows needing a new reservation

The detailed procedure is as follows:

1. **Determine new set of codecs for this session**

UE#1 determines the revised set of codecs that it is wishes to support for this session. It builds a SDP containing bandwidth requirements and characteristics of each, and assigns local port numbers for each possible media flow. Multiple media flows may be offered, and for each media flow (m= line in SDP), there may be multiple codec choices offered.

For this example, assume UE#1 originally established the session using audio (AMR) only, and now wishes to change to stereo (using the L16 2-channel codec, RTP/AVP code 10) and add an additional video media stream (MPV).

2. **INVITE (UE to P-CSCF) – see example in Table 10.3.5-2**

UE#1 sends the INVITE request to P-CSCF#1 containing this SDP

Table 10.3.5-2: INVITE (UE to P-CSCF)

```
INVITE sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 10
a=qos:mandatory sendrecv
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

SDP The SDP contains the revised set of codecs desired by UE#1

3. **P-CSCF reduces set of supported codecs based on operator policy**

P-CSCF#1 examines the media parameters, and removes any choices that the network operator decides based on local policy, not to allow on the network.

4. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.3.5-4**

P-CSCF#1 forwards the INVITE request to S-CSCF#1

Table 10.3.5-4: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

5. **S-CSCF reduces set of supported codecs based on operator policy**

   S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

6. **INVITE (S-CSCF to S-CSCF) – see example in Table 10.3.5-6**

   S-CSCF#1 forwards the INVITE request, through the S-CSCF to S-CSCF Session Flow Procedures, to S-CSCF#2.

Table 10.3.5-6: INVITE (S-CSCF to S-CSCF)

```
INVITE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

7. **S-CSCF reduces set of supported codecs based on operator policy**

S-CSCF#2 examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

8. **INVITE (S-CSCF to P-CSCF) – see example in Table 10.3.5-8**

   S-CSCF#3 forwards the INVITE request to P-CSCF#2.

Table 10.3.5-8: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

9. **P-CSCF reduces set of supported codecs based on operator policy**

   P-CSCF#2 examines the media parameters, and removes any that the network operator decides, based on local policy, not to allow on the network.

10. **INVITE (P-CSCF to UE) – see example in Table 10.3.5-10**

   P-CSCF#2 forwards the INVITE request to UE#2

Table 10.3.5-10: INVITE (P-CSCF to UE)

```
INVITE sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
t=
m=
a=
a=
m=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

11. **Determine subset of codecs supported by UE#2**

UE#2 determines the subset of codecs that it is capable of supporting for this session. It determines the intersection of those it supports with those appearing in the SDP in the INVITE request. For each media flow that is not supported, UE#2 inserts a SDP entry for media (m= line) with port=0. For each media flow that is supported, UE#2 inserts a SDP entry with an assigned port and with the codecs in common with those in the SDP from UE#1.

For this example, assume UE#2 does not supports any of those requested by UE#1

12. **415 Unsupported Media Type (UE to P-CSCF) – see example in Table 10.3.5-12**

UE#2 responds to the INVITE request (10) with a 415 Unsupported Media Type response, sent to P-CSCF#2

Table 10.3.5-12: 415 Unsupported Media Type (UE to P-CSCF)

```
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

13. **ACK (P-CSCF to UE) – see example in Table 10.3.5-13**

P-CSCF#2 responds to the 415 Unsupported Media Type error (12) by sending an ACK request to UE#2

Table 10.3.5-13: ACK (P-CSCF to UE)

```
ACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

14. **415 Unsupported Media Type (P-CSCF to S-CSCF) – see example in Table 10.3.5-14**

P-CSCF#2 sends the 415 Unsupported Media Type response to S-CSCF#2

Table 10.3.5-14: 415 Unsupported Media Type (P-CSCF to S-CSCF)

```
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

15. **ACK (S-CSCF to P-CSCF) – see example in Table 10.3.5-15**

S-CSCF#2 responds to the 415 Unsupported Media Type error by sending an ACK request to P-CSCf#2, along the signalling path established by the INVITE request.

Table 10.3.5-15: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

16. **415 Unsupported Media Type (S-CSCF to S-CSCF) – see example in Table 10.3.5-16**

    S-CSCF#2 sends the 415 Unsupported Media Type response to S-CSCF#1

Table 10.3.5-16: 415 Unsupported Media Type (S-CSCF to S-CSCF)

```
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

17. **ACK (S-CSCF to S-CSCF) – see example in Table 10.3.5-17**

    S-CSCF#1 acknowledges the error indication (16) by sending an ACK request to S-CSCF#2, along the signalling path established by the INVITE request

Table 10.3.5-17: ACK (S-CSCF to S-CSCF)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net
From:
To:
Call-ID:
Cseq:
Content-Length:
```

18. **415 Unsupported Media Type (S-CSCF to P-CSCF) – see example in Table 10.3.5-18**

    S-CSCF#1 sends the 415 Unsupported Media Type response to P-CSCF#1

Table 10.3.5-18: 415 Unsupported Media Type (S-CSCF to P-CSCF)

```
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

19. **ACK (P-CSCF to S-CSCF) – see example in Table 10.3.5-19**

    P-CSCF#1 acknowledges the error response (18) by sending an ACK request to S-CSCF#1

Table 10.3.5-19: ACK (P-CSCF to S-CSCF)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

20. **415 Unsupported Media Type (P-CSCF to UE) – see example in Table 10.3.5-16**

P-CSCF#1 sends the 415 Unsupported Media Type response to UE#1

Table 10.3.5-16: 415 Unsupported Media Type (P-CSCF to UE)

```
SIP/2.0 415 Unsupported Media Type
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:
```

21. **ACK (UE to P-CSCF) – see example in Table 10.3.5-21**

UE#1 acknowledges the error response by sending an ACK request to P-CSCF#1

Table 10.3.5-21: ACK (UE to P-CSCF)

```
ACK sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
Cseq:
Content-length:
```

10.4   Session redirection procedures

10.4.1   Introduction

This clause gives information flows for the procedures for performing session redirection. The decision to redirect a session to a different destination may be made for different reasons by a number of different functional elements, and at different points in the establishment of the session.

Three cases of session redirection prior to bearer establishment are presented, and one case of session redirection after bearer establishment.

These cases enable the typical services of "Session Forward Unconditional", "Session Forward Busy", "Session Forward Variable", "Selective Session Forwarding", and "Session Forward No Answer", though it is important to recognise that the implementation is significantly different from the counterparts in the CS domain.

10.4.2   Session redirection initiated by S-CSCF to IM CN subsystem

One of the functional elements in a basic session flow that may initiate a redirection is the S-CSCF of the destination subscriber. The subscriber profile information obtained from the HSS by the 'Cx-pull' during registration may contain complex logic and triggers causing session redirection. S-CSCF#2 sends the SIP INVITE request to the I-CSCF for the new destination (I-CSCF#F in the diagram), who forwards it to S-CSCF#F, who forwards it to the new destination.

In cases when the destination subscriber is not currently registered in the IM CN subsystem, the I-CSCF may assign a temporary S-CSCF to perform the service control on behalf of the intended destination. This temporary S-CSCF takes the role of S-CSCF#2 in the following information flow.

The service implemented by this information flow is typically "Session Forward Unconditional", "Session Forward Variable" or "Selective Session Forwarding". S-CSCF#2 may also make use of knowledge of current sessions in progress at the UE, and implement "Session Forwarding Busy" in this way.

There are 9 distinct information flows for this session redirection, as follows:

- Single network operator performing origination, forwarding, and termination

- One network operator performing origination and forwarding, separate network operator performing termination, with a firewall between to maintain configuration independence

- One network operator performing origination and forwarding, separate network operator performing termination, without a firewall between

- One network operator performing origination, second network operator performing forwarding and termination, with a firewall between to maintain configuration independence

- One network operator performing origination, second network operator performing forwarding and termination, without a firewall between

- One network operator performing origination, second network operator performing forwarding, and third network operator performing termination, without any firewalls between them

- One network operator performing origination, second network operator performing forwarding, and third network operator performing termination, with a firewall between first two to maintain configuration independence

- One network operator performing origination, second network operator performing forwarding, and third network operator performing termination, with a firewall between second and third to maintain configuration independence

- One network operator performing origination, second network operator performing forwarding, and third network operator performing termination, with a firewall between all three to maintain configuration independence

Further, it is possible that a session will be redirected multiple times, so the above list generalizes to include multiple forwarding elements.

All of these Session-Redirection procedures can be combined with MO#1a, MO#1b, or MO#2 for session origination, and with MT#1a, MT#1b, or MT#2 for session termination.

Only the first case is shown here, with a single network operator performing origination, forwarding, and termination. The additional cases can be derived from the procedures shown here and in S-S#1a, and S-S#1b.

This case is shown in the information flow in Figure 10.4.2-1
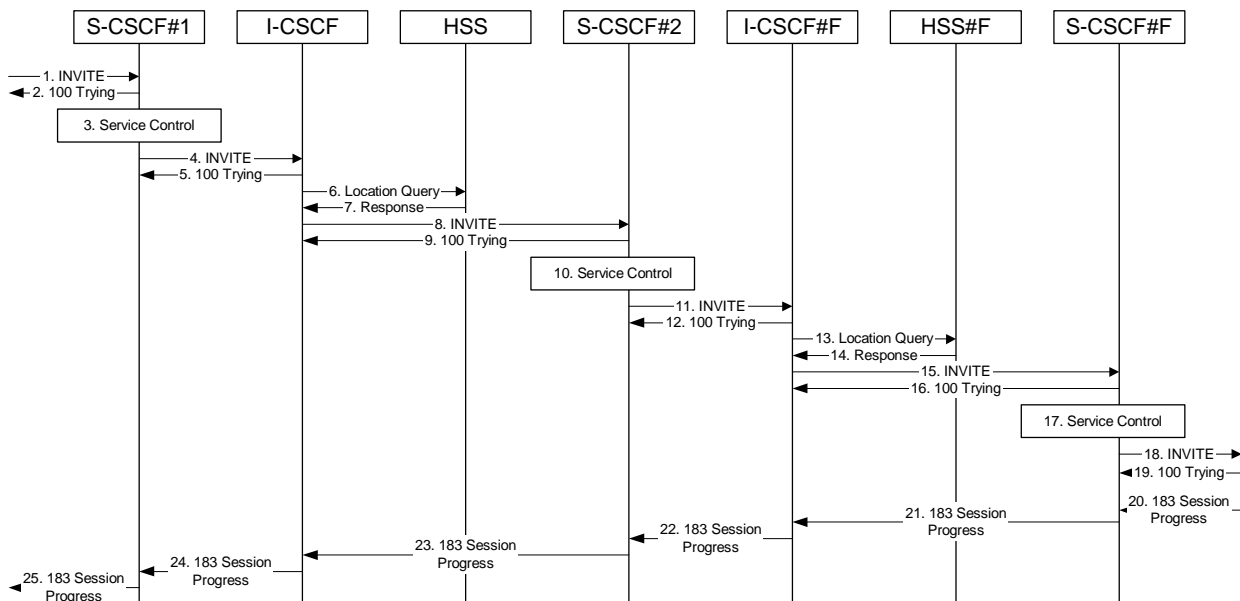
Figure 10.4.2-1: Session redirection initiated by S-CSCF to IM CN subsystem

The IM CN subsystem - Session Redirection Procedure is as follows:

1. **INVITE (MO to S-CSCF) – see example in Table 10.4.2-1**

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

   NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 10.4.2-1: INVITE (MO to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (S-CSCF to MO) – see example in Table 10.4.2-2**

   S-CSCF#1 responds to the INVITE request (1) with a 100 Trying provisional response.

Table 10.4.2-2: 100 Trying (S-CSCF to MO)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

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

4. **INVITE (S-CSCF to I-CSCF) – see example in Table 10.4.2-4**

   S-CSCF#1 performs an analysis of the destination address, and determines the network operator to whom the destination subscriber belongs. Since it is a destination served by the same network operator, S-CSCF#1 forwards the INVITE request directly to to I-CSCF in the same network.

Table 10.4.2-4: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-3333@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP sip:scscf1.home.net SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Request-URI: In the case where the Route header of the incoming INVITE request to S-CSCF contains a TEL-URL, it has to be translated to a globally routable SIP-URL before applying it as Request-URI of the outgoing INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: Need to verify that it is the Route header of the incoming INVITE request which may contain the TEL-URL, and not the Request-URI as indicated in the text of Tdoc N1-010353.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

5. **100 Trying (I-CSCF to S-CSCF) – see example in Table 10.4.2-5**

   I-CSCF responds to the INVITE request (4) by sending a 100 Trying provisional response to S-CSCF#1.

Table 10.4.2-5: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **Location Query**

   I-CSCF (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

7. **Location Response**

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

8. **INVITE (I-CSCF to S-CSCF) – see example in Table 10.4.2-8**

   I-CSCF forwards the INVITE request to the S-CSCF (S-CSCF#2) that will handle the session termination.

Table 10.4.2-8: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which the INVITE is routed to S-CSCF, and the mechanism by which S-CSCF retrieves the initial Request URI (which identifies the real destination of the session) is FFS.

NOTE: The I-CSCF does not add itself to the Record-Route header, as it has no need to remain in the signalling path once the session is established.

9. **100 Trying (S-CSCF to I-CSCF) – see example in Table 10.4.2-9**

   S-CSCF#2 responds to the INVITE request (8) with a 100 Trying provisional response.

Table 10.4.2-9: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

10. **Service Control**

S-CSCF#2 performs whatever service control logic is appropriate for this session attempt. Based on some service-specific criterion, S-CSCF#2 decides to redirect this session attempt to a new IM CN subsystem destination, at the URL sip:+1-212-555-3333@home.net;user=phone.

11. **INVITE (S-CSCF to I-CSCF) – see example in Table 10.4.2-11**

S-CSCF#2 performs an analysis of the destination address, and determines the new destination is served by the same network operator. S-CSCF#2 forwards the INVITE request directly to to I-CSCF#F (which may be different than I-CSCF#1 consulted earlier).

Table 10.4.2-11: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-3333@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP sip:scscf2.home.net, SIP/2.0/UDP sip:scscf1.home.net SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Request-URI: In the case where the Route header of the incoming INVITE request to S-CSCF contains a TEL-URL, it has to be translated to a globally routable SIP-URL before applying it as Request-URI of the outgoing INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: Need to verify that it is the Route header of the incoming INVITE request which may contain the TEL-URL, and not the Request-URI as indicated in the text of Tdoc N1-010353.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

12. **100 Trying (I-CSCF to S-CSCF) – see example in Table 10.4.2-12**

I-CSCF responds to the INVITE request (4) by sending a 100 Trying provisional response to S-CSCF#1.

Table 10.4.2-12: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

13. **Location Query**

I-CSCF (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

14. **Location Response**

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

15. **INVITE (I-CSCF to S-CSCF) – see example in Table 10.4.2-15**

I-CSCF forwards the INVITE request to the S-CSCF (S-CSCF#F) that will handle the session termination.

Table 10.4.2-15: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscff.home.net SIP/2.0
Via: SIP/2.0/UDP icscff.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net,
    SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which the INVITE is routed to S-CSCF, and the mechanism by which S-CSCF retrieves the initial Request URI (which identifies the real destination of the session) is FFS.

NOTE: The I-CSCF does not add itself to the Record-Route header, as it has no need to remain in the signalling path once the session is established.

16. **100 Trying (S-CSCF to I-CSCF) – see example in Table 10.4.2-16**

S-CSCF#2 responds to the INVITE request (8) with a 100 Trying provisional response.

Table 10.4.2-16: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscff.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net,
    SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

17. **Service Control**

S-CSCF#F performs whatever service control is appropriate for this termination session.

18. **INVITE (S-CSCF to MT) – see example in Table 10.4.2-18**

S-CSCF#2 forwards the INVITE request, as determined by the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 10.4.2-18: INVITE (S-CSCF to MT)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP2.0/UDP scscff.home.net, SIP/2.0/UDP icscff.home.net, SIP/2.0/UDP scscf2.home.net,
    SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-3333@home.net;user=phone
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

19. **100 Trying (MT to S-CSCF) – see example in Table 10.4.2-19**

S-CSCF#2 receives a 100 Trying provisional response to the INVITE request (11), as specified by the termination procedures.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 10.4.2-19: 100 Trying (MT to S-S#2)

```
SIP/2.0 100 Trying
Via: SIP2.0/UDP scscff.home.net, SIP/2.0/UDP icscff.home.net, SIP/2.0/UDP scscf2.home.net,
    SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
```

```
CSeq:
Content-length: 0
```

20. **183 Session Progress (MT to S-CSCF) – see example in Table 10.4.2-20**

    The media stream capabilities of the destination are returned along the signalling path, in a 183 Session Progress provisional response, as per the termination procedure.

NOTE: There are a number of different termination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 10.4.2-20: 183 Session Progress (MT to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP2.0/UDP scscff.home.net, SIP/2.0/UDP icscff.home.net, SIP/2.0/UDP scscf2.home.net,
    SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-3333>;privacy=off
Anonymity: Off
Require: 100rel
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=314159
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
RSeq: 9021
Content-Disposition: precondition
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::eee:fff:aaa:bbb
b=AS:64
t=907165275 0
m=audio 6544 RTP/AVP 97 3
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=qos:mandatory sendrecv confirm
```

21. **183 Session Progress (S-CSCF to I-CSCF) – see example in Table 10.4.2-21**

    S-CSCF#2 forwards the 183 Session Progress provisional response to I-CSCF.

Table 10.4.2-21: 183 Session Progress (S-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscff.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net,
    SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
```

22. **183 Session Progress (I-CSCF to S-CSCF) – see example in Table 10.4.2-22**

I-CSCF forwards the 183 Session Progress provisional response to S-CSCF#1.

Table 10.4.2-22: 183 Session Progress (I-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
```

23. **183 Session Progress (S-CSCF to I-CSCF) – see example in Table 10.4.2-23**

S-CSCF#2 forwards the 183 Session Progress provisional response to I-CSCF.

Table 10.4.2-23: 183 Session Progress (S-CSCF to I-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Remote-Party-ID: "John Smith" <tel:+1-212-555-3333>;privacy=off;screen=yes
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
```

24. **183 Session Progress (I-CSCF to S-CSCF) – see example in Table 10.4.2-24**

    I-CSCF forwards the 183 Session Progress provisional response to S-CSCF#1.

Table 10.4.2-24: 183 Session Progress (I-CSCF to S-CSCF)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
```

25. **183 Session Progress (S-CSCF to MO) – see example in Table 10.4.2-25**

    S-CSCF#1 forwards the 183 Session Progress to the originator, as per the originating procedure.

    NOTE: There are a number of different origination flows, and the table represents a typical example of what one of these flows may produce. In this case, MO#2 and MT#2 are assumed.

Table 10.4.2-25: 183 Session Progress (S-CSCF to MO)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
```

10.4.3   Session redirection initiated by S-CSCF to CS-domain

The S-CSCF in the scenario above may determine that the session is to be redirected to a CS-domain endpoint, or to the PSTN. It recognizes this situation by the redirected URL being a tel: URL.

For the simplest configuration (Mobile located in home service area (MO#2), initiating a session to a destination served by same network operator(S-S#2)), the handling of redirection to a tel: URL is shown in Figure 10.4.3-1. Other cases, which include roaming, PSTN origination, destinations served by other network operators, and firewalls, are handled in a similar manner.
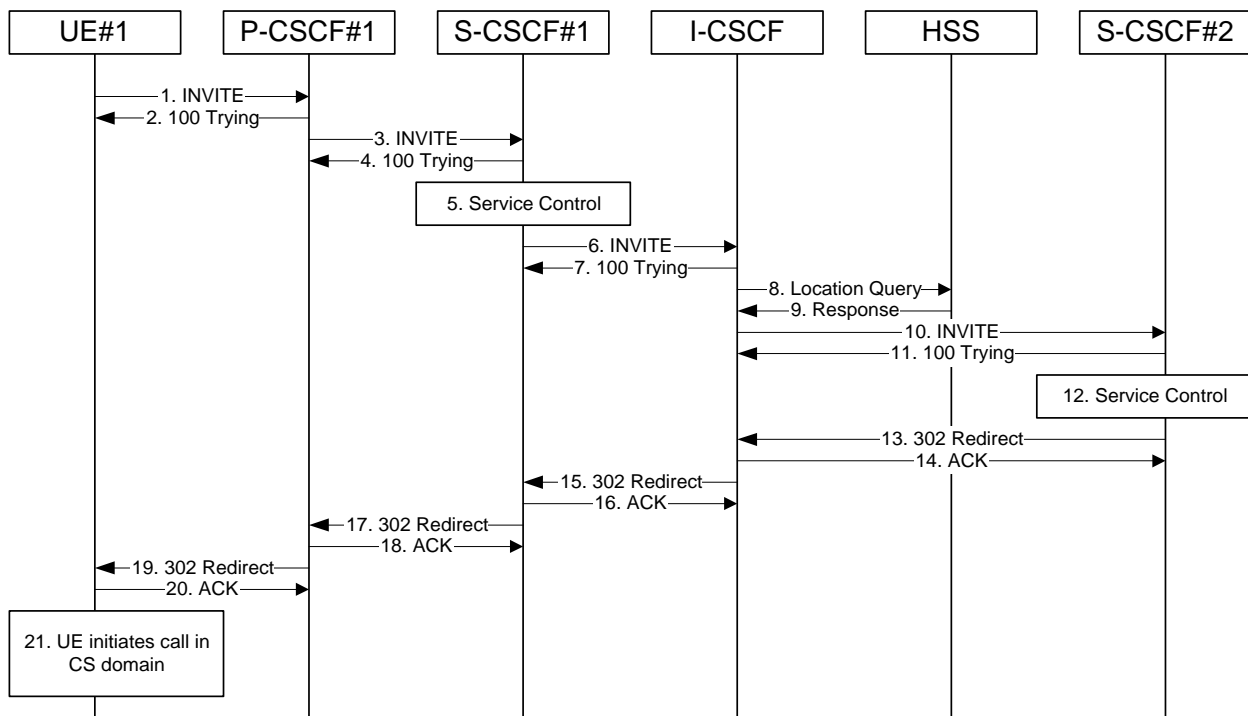
Figure 10.4.3-1: Session redirection initiated by S-CSCF to CS-Domain

Step-by-step processing is as follows:

1. **INVITE (UE to P-CSCF) – see example in Table 10.4.3-1**

   UE sends the INVITE request, containing an initial SDP, to the P-CSCF determined via the CSCF discovery mechanism.

   Editor's Note: Certain fields in the SDP carry no information. In particular the "o=", "s=" fields and "t=". These are, however, mandatory fields within SDP. Does 3GPP wish to define a non-standard version of SDP that removes these, and if so, how does this interwork with outside SIP networks that use standard SDP.

Table 10.4.3-1: INVITE (UE to P-CSCF)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

Request-URI: contains the keyed number from the user.

Via: contains the IP address or FQDN of the originating UE.

Remote-Party-ID: contains the public identity of the UE. The Display name is optional.

Proxy-Require: The sip privacy draft specifies that the usage of the Remote-Party-Id MUST be accompanied by a Proxy-Require header specifying "privacy" in all INVITE requests.

From:, To:, Call-ID: follow the recommendations of draft-ietf-sip-privacy-01, even though anonymity is not being requested for this session.

Cseq: a random starting number.

Contact: the IP address or FQDN of the originating UE.

2. **100 Trying (P-CSCF to UE) – see example in Table 10.4.3-2**

   P-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 10.4.3-2: 100 Trying (P-CSCF to UE)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.4.3-3**

   P-CSCF remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. This next hop is the S-CSCF within the home network.

   P-CSCF rewrites the Contact header, with a locally defined value that identifies the UE. P-CSCF adds itself to the Record-Route header, and adds a Via header.

   The INVITE request is forwarded to the S-CSCF.

Table 10.4.3-3: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Request-URI: the first component in the remembered Path header from Registration.

Route: contains the remaining elements from the Path header from Registration, with the initial Request-URI (received from the UE) appended as the final component.

Editor's Note: Need to represent the ability of the P-CSCF to take part in the negotiation of the media streams (by removing entries) in some manner.

4. **100 Trying (S-CSCF to P-CSCF) – see example in Table 10.4.3-4**

   S-CSCF responds to the INVITE request (3) with a 100 Trying provisional response.

Table 10.4.3-4: 100 Trying (S-CSCF to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

5. **Service Control**

   S-CSCF validates the service profile, and performs any origination service control required for this subscriber.

6. **INVITE (S-CSCF to I-CSCF) – see example in Table 10.4.3-6**

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

Editor's Note: Need to represent the ability of the S-CSCF to take part in the negotiation of the media streams (by removing entries) in some manner.

Editor's Note: Need for additional headers to transport e.g. Billing-Correlation-Identifier is FFS.

Table 10.4.3-6: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Request-URI: In the case where the Route header of the incoming INVITE request to S-CSCF contains a TEL-URL, it has to be translated to a globally routable SIP-URL before applying it as Request-URI of the outgoing INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: Need to verify that it is the Route header of the incoming INVITE request which may contain the TEL-URL, and not the Request-URI as indicated in the text of Tdoc N1-010353.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

7. **100 Trying (I-CSCF to S-CSCF) – see example in Table 10.4.3-7**

   S-CSCF receives a 100 Trying provisional response, as specified by the S-CSCF to S-CSCF procedures.

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 10.4.3-7: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

8. **Location Query**

   I-CSCF (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

9. **Location Response**

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

10. **INVITE (I-CSCF to S-CSCF) – see example in Table 10.4.3-10**

   I-CSCF forwards the INVITE request to the S-CSCF (S-CSCF#2) that will handle the session termination.

Table 10.4.3-10: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which the INVITE is routed to S-CSCF, and the mechanism by which S-CSCF retrieves the initial Request URI (which identifies the real destination of the session) is FFS.

NOTE: The I-CSCF does not add itself to the Record-Route header, as it has no need to remain in the signalling path once the session is established.

11. **100 Trying (S-CSCF to I-CSCF) – see example in Table 10.4.3-11**

   S-CSCF#2 responds to the INVITE request (8) with a 100 Trying provisional response.

Table 10.4.3-11: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

12. **Service Control**

S-CSCF#2 performs whatever service control logic is appropriate for this session attempt. Based on some service-specific criterion, S-CSCF#2 decides to redirect this session attempt to a CS-domain endpoint, at the URL tel:+1-212-555-3333.

13. **302 Redirect (S-CSCF to I-CSCF) – see example in Table 10.4.3-13**

S-CSCF#2 sends a 302 Redirect response to I-CSCF, containing the new destination.

Table 10.4.3-13: 302 Redirect (S-CSCF to I-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: tel:+1-212-555-3333
Content-length: 0
```

14. **ACK (I-CSCF to S-CSCF) – see example in Table 10.4.3-14**

I-CSCF acknowledges receipt of the 302 Redirect response (13) by sending an ACK request to S-CSCF#2.

Table 10.4.3-14: ACK (I-CSCF to S-CSCF)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

15. **302 Redirect (I-CSCF to S-CSCF) – see example in Table 10.4.3-15**

I-CSCF sends a 302 Redirect response to S-CSCF#1, containing the new destination.

Table 10.4.3-15: 302 Redirect (I-CSCF to S-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact:
Content-length: 0
```

16. **ACK (S-CSCF to I-CSCF) – see example in Table 10.4.3-16**

S-CSCF#1 acknowledges receipt of the 302 Redirect response (15) by sending an ACK request to I-CSCF.

Table 10.4.3-16: ACK (S-CSCF to I-CSCF)

```
ACK sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

17. **302 Redirect (S-CSCF to P-CSCF) – see example in Table 10.4.3-17**

S-CSCF#1 sends a 302 Redirect response to P-CSCF, containing the new destination.

Table 10.4.3-17: 302 Redirect (S-CSCF to P-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact:
Content-length: 0
```

18. **ACK (P-CSCF to S-CSCF) – see example in Table 10.4.3-18**

P-CSCF acknowledges receipt of the 302 Redirect response (17) by sending an ACK request to S-CSCF#1.

Table 10.4.3-18: ACK (P-CSCF to S-CSCF)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

19. **302 Redirect (P-CSCF to UE) – see example in Table 10.4.3-19**

P-CSCF sends a 302 Redirect response to UE, containing the new destination.

Table 10.4.3-19: 302 Redirect (P-CSCF to UE)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact:
Content-length: 0
```

20. **ACK (UE to P-CSCF) – see example in Table 10.4.3-20**

UE acknowledges receipt of the 302 Redirect response (19) by sending an ACK request to P-CSCF.

Table 10.4.3-20: ACK (UE to P-CSCF)

```
ACK sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
Cseq:
Content-length:
```

21. **UE initiates session in CS domain**

   UE initiates a session to the new destination given in the Contact header, using mechanisms of the CS domain.

10.4.4 Session redirection initiated by S-CSCF to general endpoint

The S-CSCF in the scenario above may determine that the session is to be redirected to an endpoint outside the IP MultiMedia System and outside the CS-domain. Examples of these destinations include web pages, email addresses, etc. It recognizes this situation by the redirected URL being other than a sip: or tel: URL.

For the simplest configuration (Mobile located in home service area (MO#2), initiating a session to a destination served by same network operator(S-S#2)), the handling of redirection to a general URL is shown in Figure 10.4.4-1. Other cases, which include roaming, PSTN origination, destinations served by other network operators, and firewalls, are handled in a similar manner.
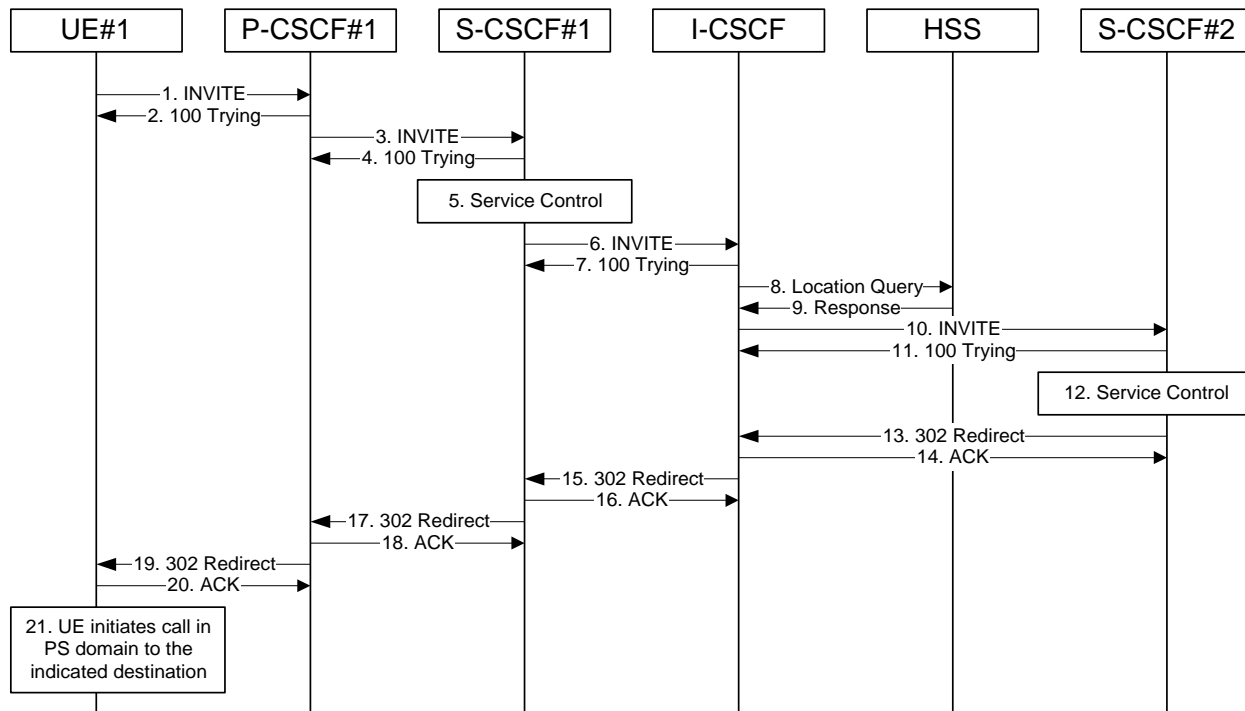
Figure 10.4.4-1: Session redirection initiated by S-CSCF to general endpoint

Step-by-step processing is as follows:

1. **INVITE (UE to P-CSCF) – see example in Table 10.4.4-1**

   UE sends the INVITE request, containing an initial SDP, to the P-CSCF determined via the CSCF discovery mechanism.

   Editor's Note: Certain fields in the SDP carry no information. In particular the "o=", "s=" fields and "t=". These are, however, mandatory fields within SDP. Does 3GPP wish to define a non-standard version of SDP that removes these, and if so, how does this interwork with outside SIP networks that use standard SDP.

Table 10.4.4-1: INVITE (UE to P-CSCF)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

Request-URI: contains the keyed number from the user.

Via: contains the IP address or FQDN of the originating UE.

Remote-Party-ID: contains the public identity of the UE. The Display name is optional.

Proxy-Require: The sip privacy draft specifies that the usage of the Remote-Party-Id MUST be accompanied by a Proxy-Require header specifying "privacy" in all INVITE requests.

From:, To:, Call-ID: follow the recommendations of draft-ietf-sip-privacy-01, even though anonymity is not being requested for this session.

Cseq: a random starting number.

Contact: the IP address or FQDN of the originating UE.

2. **100 Trying (P-CSCF to UE) – see example in Table 10.4.4-2**

P-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 10.4.4-2: 100 Trying (P-CSCF to UE)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.4.4-3**

P-CSCF remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. This next hop is the S-CSCF within the home network.

P-CSCF rewrites the Contact header, with a locally defined value that identifies the UE. P-CSCF adds itself to the Record-Route header, and adds a Via header.

The INVITE request is forwarded to the S-CSCF.

Table 10.4.4-3: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Request-URI: the first component in the remembered Path header from Registration.

Route: contains the remaining elements from the Path header from Registration, with the initial Request-URI (received from the UE) appended as the final component.

Editor's Note: Need to represent the ability of the P-CSCF to take part in the negotiation of the media streams (by removing entries) in some manner.

4. **100 Trying (S-CSCF to P-CSCF) – see example in Table 10.4.4-4**

   S-CSCF responds to the INVITE request (3) with a 100 Trying provisional response.

Table 10.4.4-4: 100 Trying (S-CSCF to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

5. **Service Control**

   S-CSCF validates the service profile, and performs any origination service control required for this subscriber.

6. **INVITE (S-CSCF to I-CSCF) – see example in Table 10.4.4-6**

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

Editor's Note: Need to represent the ability of the S-CSCF to take part in the negotiation of the media streams (by removing entries) in some manner.

Editor's Note: Need for additional headers to transport e.g. Billing-Correlation-Identifier is FFS.

Table 10.4.4-6: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Request-URI: In the case where the Route header of the incoming INVITE request to S-CSCF contains a TEL-URL, it has to be translated to a globally routable SIP-URL before applying it as Request-URI of the outgoing INVITE request. For this address translation the S-CSCF may use the services of an ENUM-DNS based database structure, or any other suitable translation database.

Editor's Note: Need to verify that it is the Route header of the incoming INVITE request which may contain the TEL-URL, and not the Request-URI as indicated in the text of Tdoc N1-010353.

Editor's Note: It remains to be clarified if the use of the word "may" in the above sentence, needs to be changed to "shall". 23.228v170 states that an S-CSCF shall support an ENUM DNS translation mechanism, so the above text needs to be aligned with Stage 2.

7. **100 Trying (I-CSCF to S-CSCF) – see example in Table 10.4.4-7**

   S-CSCF receives a 100 Trying provisional response, as specified by the S-CSCF to S-CSCF procedures.

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MT#2 are assumed.

Table 10.4.4-7: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

8. **Location Query**

   I-CSCF (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

9. **Location Response**

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

10. **INVITE (I-CSCF to S-CSCF) – see example in Table 10.4.4-10**

I-CSCF forwards the INVITE request to the S-CSCF (S-CSCF#2) that will handle the session termination.

Table 10.4.4-10: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:+1-212-555-2222@home2.net;user=phone
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which the INVITE is routed to S-CSCF, and the mechanism by which S-CSCF retrieves the initial Request URI (which identifies the real destination of the session) is FFS.

NOTE: The I-CSCF does not add itself to the Record-Route header, as it has no need to remain in the signalling path once the session is established.

11. **100 Trying (S-CSCF to I-CSCF) – see example in Table 10.4.4-11**

S-CSCF#2 responds to the INVITE request (8) with a 100 Trying provisional response.

Table 10.4.4-11: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

12. **Service Control**

S-CSCF#2 performs whatever service control logic is appropriate for this session attempt. Based on some service-specific criterion, S-CSCF#2 decides to redirect this session attempt to a PS-domain endpoint, at the URL mailto:alienblaster@home.net.

13. **302 Redirect (S-CSCF to I-CSCF) – see example in Table 10.4.4-13**

S-CSCF#2 sends a 302 Redirect response to I-CSCF, containing the new destination.

Table 10.4.4-13: 302 Redirect (S-CSCF to I-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: mailto:alienblaster@home.net
Content-length: 0
```

14. **ACK (I-CSCF to S-CSCF) – see example in Table 10.4.4-14**

   I-CSCF acknowledges receipt of the 302 Redirect response (13) by sending an ACK request to S-CSCF#2.

Table 10.4.4-14: ACK (I-CSCF to S-CSCF)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

15. **302 Redirect (I-CSCF to S-CSCF) – see example in Table 10.4.4-15**

   I-CSCF sends a 302 Redirect response to S-CSCF#1, containing the new destination.

Table 10.4.4-15: 302 Redirect (I-CSCF to S-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact:
Content-length: 0
```

16. **ACK (S-CSCF to I-CSCF) – see example in Table 10.4.4-16**

   S-CSCF#1 acknowledges receipt of the 302 Redirect response (15) by sending an ACK request to I-CSCF.

Table 10.4.4-16: ACK (S-CSCF to I-CSCF)

```
ACK sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

17. **302 Redirect (S-CSCF to P-CSCF) – see example in Table 10.4.4-17**

   S-CSCF#1 sends a 302 Redirect response to P-CSCF, containing the new destination.

Table 10.4.4-17: 302 Redirect (S-CSCF to P-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact:
Content-length: 0
```

18. **ACK (P-CSCF to S-CSCF) – see example in Table 10.4.4-18**

P-CSCF acknowledges receipt of the 302 Redirect response (17) by sending an ACK request to S-CSCF#1.

Table 10.4.4-18: ACK (P-CSCF to S-CSCF)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

19. **302 Redirect (P-CSCF to UE) – see example in Table 10.4.4-19**

P-CSCF sends a 302 Redirect response to UE, containing the new destination.

Table 10.4.4-19: 302 Redirect (P-CSCF to UE)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact:
Content-length: 0
```

20. **ACK (UE to P-CSCF) – see example in Table 10.4.4-20**

UE acknowledges receipt of the 302 Redirect response (19) by sending an ACK request to P-CSCF.

Table 10.4.4-20: ACK (UE to P-CSCF)

```
ACK sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
Cseq:
Content-length:
```

21. **UE initiates session in PS domain**

UE initiates a session to the new destination given in the Contact header, using mechanisms of the PS domain.

10.4.5 Session redirection initiated by P-CSCF

One of the functional elements in a basic session flow that may initiate a redirection is the P-CSCF of the destination subscriber. In handling of an incoming session setup attempt, the P-CSCF normally sends the INVITE request to the destination UE, and retransmits it as necessary until obtaining an acknowledgement indicating reception by the UE.

In cases when the destination subscriber is not currently reachable in the IM CN subsystem (due to such factors as roaming outside the service area or loss of battery, but the registration has not yet expired), the P-CSCF may initiate a redirection of the session. The P-CSCF informs the S-CSCF of this redirection, without specifying the new location; S-CSCF determines the new destination and performs according to subclauses 10.4.2, 10.4.3, or 10.4.4 above, based on the type of destination.

This is shown in Figure 10.4.5-1.
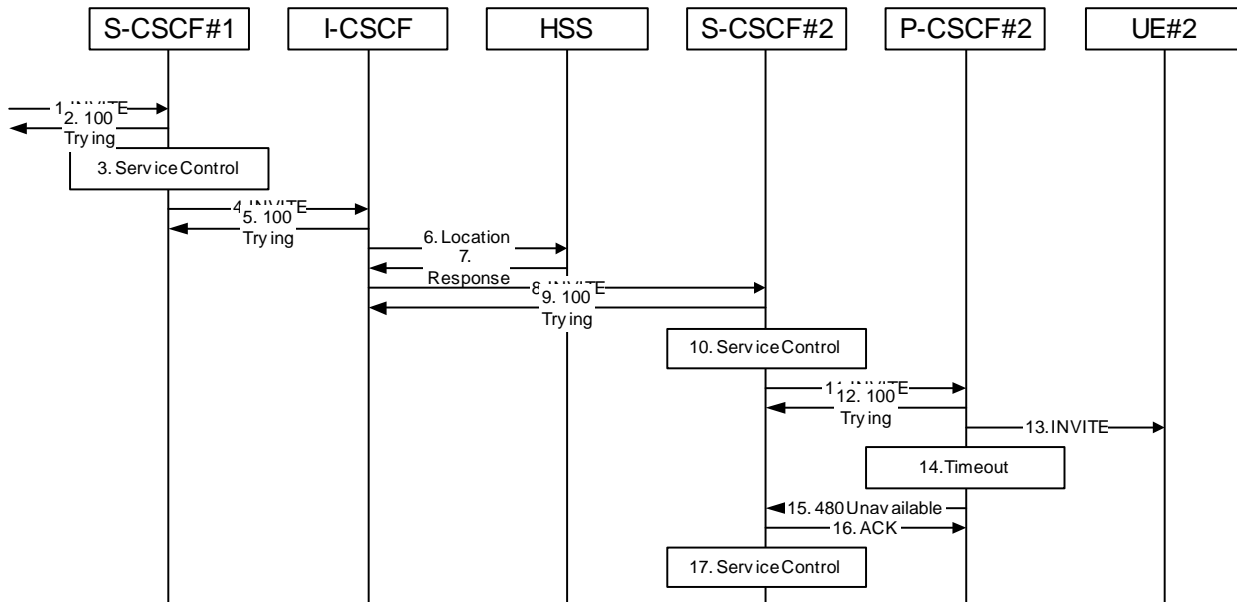
Figure 10.4.5-1: Session redirection initiated by P-CSCF

Beginning with step #8, the step-by-step processing is as follows:

8. **INVITE (I-CSCF to S-CSCF) – see example in Table 10.4.5-8**

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

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 10.4.5-8: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

9. **100 Trying (S-CSCF to I-CSCF) – see example in Table 10.4.5-9**

   S-CSCF responds to the INVITE request (8) with a 100 Trying provisional response.

Table 10.4.5-9: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

10. **Service Control**

    S-CSCF validates the service profile, and performs any termination service control required for this subscriber

11. **INVITE (S-CSCF to P-CSCF) – see example in Table 10.4.5-11**

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

Table 10.4.5-11: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Route: built from the registration information, (pcscf and UE contact name), followed by the initial Request-URI from the incoming INVITE request. The first component of the Route header, pcscf, is moved to the Request-URI of the request.

Via:, Record-Route: S-CSCF adds itself in the Record-Route and Via headers.

12. **100 Trying (P-CSCF to S-CSCF) – see example in Table 10.4.5-12**

   P-CSCF responds to the INVITE request (11) with a 100 Trying provisional response.

Table 10.4.5-12: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

13. **INVITE (P-CSCF to UE) – see example in Table 10.4.5-13**

   P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 10.4.5-13: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
Media-Authorization: 00200001001001017063663122e78797a2e6e6574000c02013942563330373200
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf.home.net, sip:scscf0.home.net,
 sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token

14. **Timeout**

P-CSCF never receives any response from UE#2, and assumes it is unreachable.

15. **480 Temporarily Unavailable (P-CSCF to S-CSCF) – see example in Table 10.4.5-15**

P-CSCF sends a 480 Temporarily Unavailable response to S-CSCF.

Table 10.4.5-15: 480 Temporarily Unavailable (P-CSCF to S-CSCF)

```
SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

16. **ACK (S-CSCF to P-CSCF) – see example in Table 10.4.5-16**

S-CSCF acknowledges receipt of the 480 Temporarily Unavailable response (15) by sending an ACK request to P-CSCF.

Table 10.4.5-16: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

17. **Service Control**

S-CSCF#2 determines the proper redirection action to take for this session, based on the subscriber profile and network operator policy.

- If the session is being redirected to a sip URL, then the information flow continues with step #11 of Subclause 10.4.2.

- If the session is being redirected to a tel URL, then the information flow continues with step #13 of Subclause 10.4.3.

If the session is being redirected to a general URL, then the information flow continues with step #13 of Subclause 10.4.4.

10.4.6 Session redirection initiated by UE

The next functional element in a basic session flow that may initiate a redirection is the UE of the destination subscriber. The UE may implement customer-specific feature processing, and base its decision to redirect this session on such things as identity of caller, current sessions in progress, other applications currently being accessed, etc. UE sends the SIP Redirect response to its P-CSCF, who forwards back along the signalling path to S-CSCF#1, who initiates a session to the new destination.

The service implemented by this information flow is typically "Session Forward Busy", "Session Forward Variable" or "Selective Session Forwarding".

This is shown in Figure 10.4.6-1:
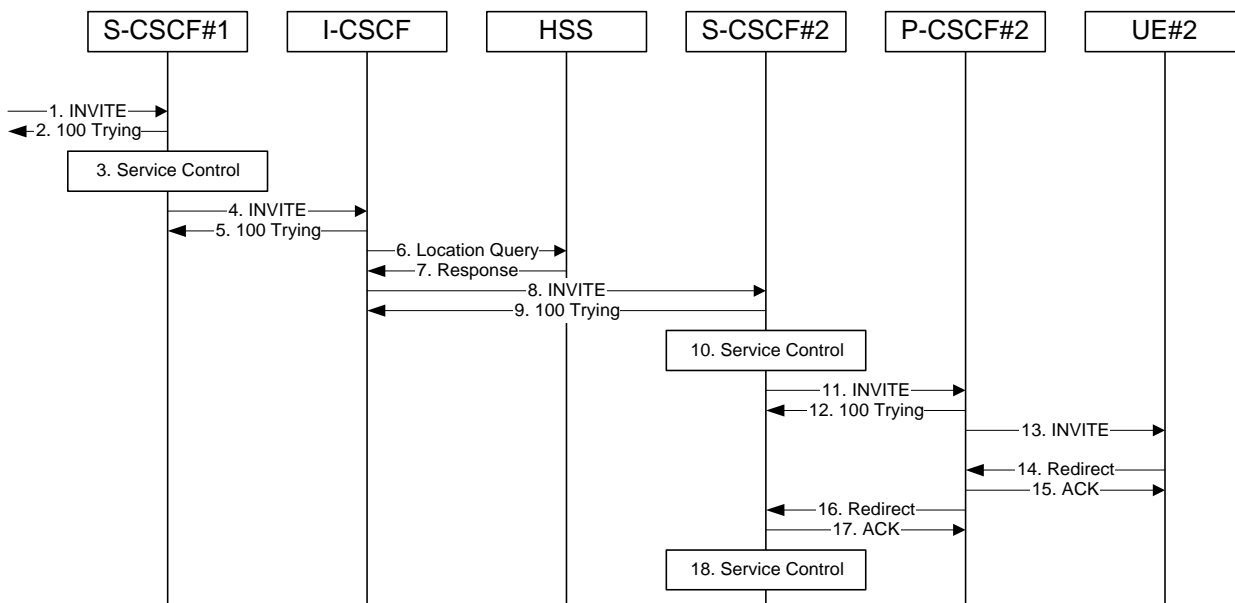
Figure 10.4.6-1: Session redirection initiated by UE

Beginning with step #8, the step-by-step processing is as follows:

8. **INVITE (I-CSCF to S-CSCF) – see example in Table 10.4.6-8**

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

    NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 10.4.6-8: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

9. **100 Trying (S-CSCF to I-CSCF) – see example in Table 10.4.6-9**

    S-CSCF responds to the INVITE request (8) with a 100 Trying provisional response.

Table 10.4.6-9: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

10. **Service Control**

    S-CSCF validates the service profile, and performs any termination service control required for this subscriber

11. **INVITE (S-CSCF to P-CSCF) – see example in Table 10.4.6-11**

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

Table 10.4.6-11: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Route: built from the registration information, (pcscf and UE contact name), followed by the initial Request-URI from the incoming INVITE request. The first component of the Route header, pcscf, is moved to the Request-URI of the request.

Via:, Record-Route: S-CSCF adds itself in the Record-Route and Via headers.

12. **100 Trying (P-CSCF to S-CSCF) – see example in Table 10.4.6-12**

P-CSCF responds to the INVITE request (11) with a 100 Trying provisional response.

Table 10.4.6-12: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

13. **INVITE (P-CSCF to UE) – see example in Table 10.4.6-13**

P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 10.4.6-13: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
Media-Authorization: 00200001001001001706366312e78797a2e6e6574000c02013942563330373200
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf.home.net, sip:scscf0.home.net,
    sip:%5b5555%3a%3aaaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token

14. **302 Redirect (UE to P-CSCF) – see example in Table 10.4.6-14**

UE sends a 302 Redirect response to UE, specifying a new destination.

Table 10.4.6-14: 302 Redirect (UE to P-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:+1-212-555-3333@home.net;user=phone
Content-length: 0
```

15. **ACK (P-CSCF to UE) – see example in Table 10.4.6-15**

S-CSCF acknowledges receipt of the 302 Redirect response (15) by sending an ACK request to P-CSCF.

Table 10.4.6-15: ACK (P-CSCF to UE)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

16. **302 Redirect (P-CSCF to S-CSCF) – see example in Table 10.4.6-16**

    P-CSCF sends a 302 Redirect response to S-CSCF, with the new destination.

Table 10.4.6-16: 302 Redirect (P-CSCF to S-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact:
Content-length: 0
```

17. **ACK (S-CSCF to P-CSCF) – see example in Table 10.4.6-17**

    S-CSCF acknowledges receipt of the 302 Redirect response (16) by sending an ACK request to P-CSCF.

Table 10.4.6-17: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

18. **Service Control**

    S-CSCF#2 determines the proper redirection action to take for this session, based on the subscriber profile and network operator policy.

    - If the session is being redirected to a sip URL, then the information flow continues with step #11 of Subclause 10.4.2.

    - If the session is being redirected to a tel URL, then the information flow continues with step #13 of Subclause 10.4.3.

    - If the session is being redirected to a general URL, then the information flow continues with step #13 of Subclause 10.4.4.

10.4.7 Session redirection initiated after bearer establishment

The UE of the destination subscriber may request the session be redirected after a customer-specified ringing interval. The UE may also implement customer-specific feature processing, and base its decision to redirect this session on such things as identity of caller, current sessions in progress, other applications currently being accessed, etc. UE sends the SIP Redirect response to its P-CSCF, who forwards back along the signaling path to the originating endpoint, who initiates a session to the new destination.

The service implemented by this information flow is typically "Session Forward No Answer".

Redirection to another I CN subsystem endpoint (e.g. a sip: URL) is shown in Figure 10.4.7-1. The figure starts at the point in the session establishment when the destination is known, resources have been reserved, and the destination subscriber is being alerted. If the desire for redirection was known earlier than this point, the procedures of Subclause 10.4.6 would be followed instead.
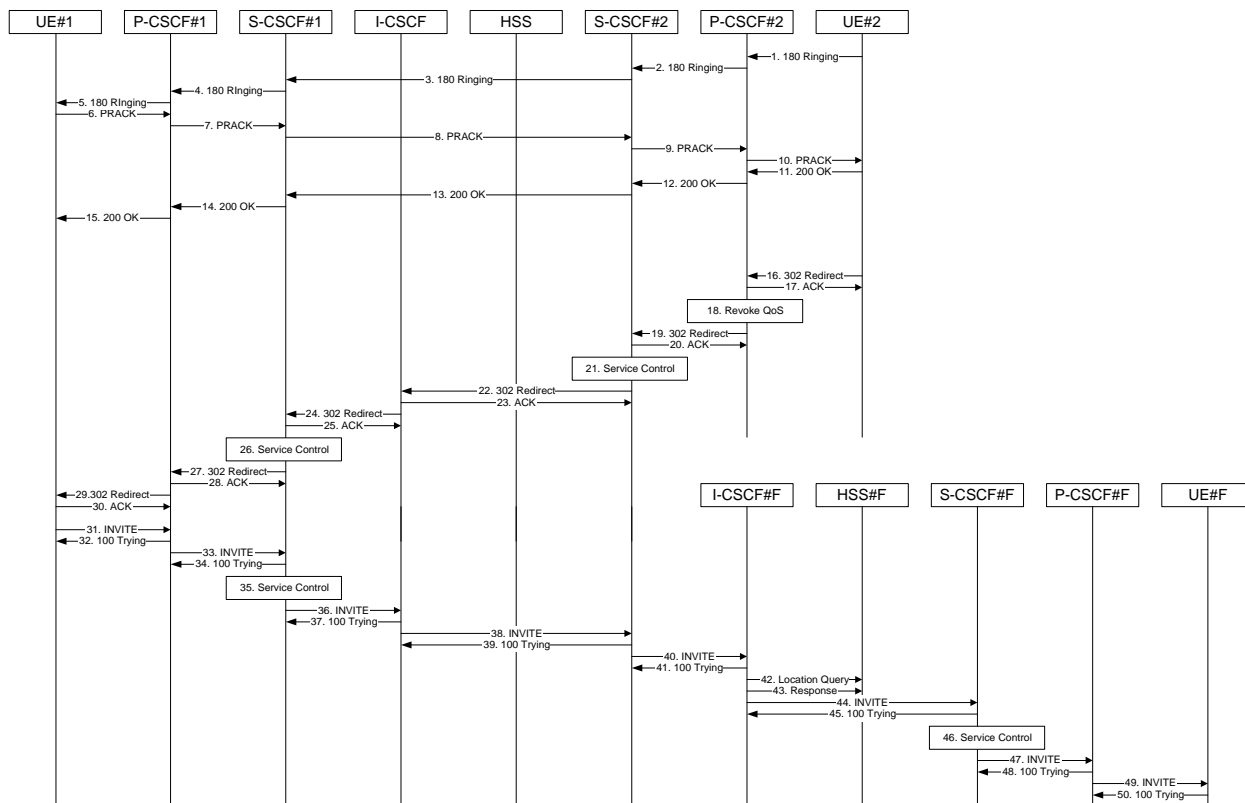
Figure 10.4.7-1: Session redirection after bearer establishment

Step-by-step processing is as follows:

1. **180 Ringing (UE to P-CSCF) – see example in Table 10.4.7-1**

    Depending on the type of codec change being performed, alerting may be required at the destination UE. If so, UE#2 sends a 180 Ringing provisional response to the originator, through P-CSCF#2.

Table 10.4.7-1: 180 Ringing (UE to P-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
Require: 100rel
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 INVITE
Contact: sip:[5555::eee:fff:aaa:bbb]
RSeq: 19
Content-length: 0
```

2. **180 Ringing (P-CSCF to S-CSCF) – see example in Table 10.4.7-2**

    P-CSCF#2 sends the 180 Ringing response to S-CSCF#2

Table 10.4.7-2: 180 Ringing (P-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

3. **180 Ringing (S-CSCF to S-CSCF) – see example in Table 10.4.7-3**

   S-CSCF#2 sends the 180 Ringing response to S-CSCF#1

Table 10.4.7-3: 180 Ringing (S-CSCF to S-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

4. **180 Ringing (S-CSCF to P-CSCF) – see example in Table 10.4.7-4**

   S-CSCF#1 sends the 180 Ringing response to P-CSCF#1

Table 10.4.7-4: 180 Ringing (S-CSCF to P-CSCF)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
Require:
From:
To:
Call-ID:
CSeq:
Contact:
RSeq:
Content-length:
```

5. **180 Ringing (P-CSCF to UE) – see example in Table 10.4.7-5**

   P-CSCF#1 sends the 180 Ringing response to UE#1

Table 10.4.7-5: 180 Ringing (P-CSCF to UE)

```
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token4@pcscf1.home.net
RSeq:
Content-length:
```

6. **PRACK (UE to P-CSCF) – see example in Table 10.4.7-6**

   UE#1 sends the PRACK request to UE#2, along the signalling path established by the INVITE request

Table 10.4.7-6: PRACK (UE to P-CSCF)

```
PRACK sip:token4@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 PRACK
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Rack: 19 131 INVITE
Content-length: 0
```

   Request-URI: takes the value of the Contact header of the 180 Ringing response.

   Via:, Contact: take the value of either the IP address or FQDN of the UE.

   From:, To:, Call-ID: copied from the 180 Ringing response so that they include any revised tag parameters.

   Cseq: takes a higher value than in the previous request.

7. **PRACK (P-CSCF to S-CSCF) – see example in Table 10.4.7-7**

   P-CSCF#1 sends the PRACK request to S-CSCF#1, along the signalling path established by the INVITE request

Table 10.4.7-7: PRACK (P-CSCF to S-CSCF)

```
PRACK sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Rack:
Content-length:
```

   Route: P-CSCF adds a Route header, with the saved value from the previous response. P-CSCF identifies the proper saved value by the Request-URI.

8. **PRACK (S-CSCF to S-CSCF) – see example in Table 10.4.7-8**

   S-CSCF#1 sends the PRACK request to S-CSCF#2, along the signalling path established by the INVITE request

Table 10.4.7-8: PRACK (S-CSCF to S-CSCF)

```
PRACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

9. **PRACK (S-CSCF to P-CSCF) – see example in Table 10.4.7-9**

S-CSCF#2 sends the PRACK request to P-CSCf#2, along the signalling path established by the INVITE request.

Table 10.4.7-9: PRACK (S-CSCF to P-CSCF)

```
PRACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Rack:
Content-length:
```

10. **PRACK (P-CSCF to UE) – see example in Table 10.4.7-10**

P-CSCF#2 sends the PRACK request to UE#2, along the signalling path established by the INVITE request

Table 10.4.7-10: PRACK (P-CSCF to UE)

```
PRACK sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token4
From:
To:
Call-ID:
Cseq:
Contact: token4@pcscf2.home.net
Rack:
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

11. **200 OK (UE to P-CSCF) – see example in Table 10.4.7-11**

UE#2 responds to the PRACK request (10) with a 200 OK response to P-CSCF#2

Table 10.4.7-11: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token4
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

12. **200 OK (P-CSCF to S-CSCF) – see example in Table 10.4.7-12**

P-CSCF#2 sends the 200 OK response to S-CSCF#2

Table 10.4.7-12: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

13. **200 OK (S-CSCF to S-CSCF) – see example in Table 10.4.7-13**

    S-CSCF#2 sends the 200 OK response to S-CSCF#1

Table 10.4.7-13: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

14. **200 OK (S-CSCF to P-CSCF) – see example in Table 10.4.7-14**

    S-CSCF#1 sends the 200 OK response to P-CSCF#1

Table 10.4.7-14: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

15. **200 OK (P-CSCF to UE) – see example in Table 10.4.7-15**

    P-CSCF#1 sends the 200 OK response to UE#1

Table 10.4.7-15: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token5@pcscf1.home.net
Content-Length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

16. **302 Redirect (UE to P-CSCF) – see example in Table 10.4.7-16**

Based on some service criterion, such as a timeout value, UE#2 decides to redirect this session request to another destination. UE#2 sends a 302 Redirect response to P-CSCF, containing the new destination. For this example, consider the new destination to be <sip:+1-212-555-3333@home.net;user=phone>.

Table 10.4.7-16: 302 Redirect (S-CSCF to I-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net,
    SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:+1-212-555-3333@home.net;user=phone
Content-length: 0
```

17. **ACK (P-CSCF to UE) – see example in Table 10.4.7-17**

    P-CSCF acknowledges receipt of the 302 Redirect response (16) by sending an ACK request to UE#2.

Table 10.4.7-17: ACK (I-CSCF to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

18. **Revoke QoS**

    P-CSCF revokes any authorization is had made for Quality of Service for this session

19. **302 Redirect (P-CSCF to S-CSCF) – see example in Table 10.4.7-19**

    P-CSCF#2 sends a 302 Redirect response to S-CSCF#2, containing the new destination.

Table 10.4.7-19: 302 Redirect (P-CSCF to S-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

20. **ACK (S-CSCF to P-CSCF) – see example in Table 10.4.7-20**

    S-CSCF acknowledges receipt of the 302 Redirect response (19) by sending an ACK request to P-CSCF#2.

Table 10.4.7-20: ACK (S-CSCF to P-CSCF)

```
ACK sip:pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

21. **Service Control**

S-CSCF#2 performs whatever service control is appropriate for this redirected session.

If UE#2 has not subscribed to a session redirection service, then S-CSCF#2 may change the error response to a 480 Temporarily Unavailable.

S-CSCF#2 generates a private URL containing the new destination, and places this new value as the Contact header in the response. Attempts to initiate a session to this destination shall be restricted to a short time period.

22. **302 Redirect (S-CSCF to I-CSCF) – see example in Table 10.4.7-22**

S-CSCF#2 sends a 302 Redirect response to I-CSCF, containing the updated destination.

Table 10.4.7-22: 302 Redirect (S-CSCF to I-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token(sip:+1-212-555-3333@home.net;user=phone)@scscf2.home.net
Content-length:
```

23. **ACK (I-CSCF to S-CSCF) – see example in Table 10.4.7-23**

I-CSCF acknowledges receipt of the 302 Redirect response (22) by sending an ACK request to S-CSCF#2.

Table 10.4.7-23: ACK (I-CSCF to S-CSCF)

```
ACK sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

24. **302 Redirect (I-CSCF to S-CSCF) – see example in Table 10.4.7-24**

I-CSCF may (based on operator preferences) update the new destination address, in order to hide the S-CSCF address and maintain configuration independence. If so, it generates a new private URL with its own hostname. I-CSCF sends a 302 Redirect response to S-CSCF#1, containing the new destination.

Table 10.4.7-24: 302 Redirect (I-CSCF to S-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token(sip:token(sip:+1-212-555-
3333@home.net;user=phone)@scscf2.home.net;user=private)@icscf.home.net;user=private
Content-length: 0
```

25. **ACK (S-CSCF to I-CSCF) – see example in Table 10.4.7-25**

S-CSCF#1 acknowledges receipt of the 302 Redirect response (24) by sending an ACK request to I-CSCF.

Table 10.4.7-25: ACK (S-CSCF to I-CSCF)

```
ACK sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

26. **Service Control**

S-CSCF#1 performs whatever service control is appropriate for this redirected session.

27. **302 Redirect (S-CSCF to P-CSCF) – see example in Table 10.4.7-27**

S-CSCF#1 sends a 302 Redirect response to P-CSCF, containing the new destination.

Table 10.4.7-27: 302 Redirect (S-CSCF to P-CSCF)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact:
Content-length: 0
```

28. **ACK (P-CSCF to S-CSCF) – see example in Table 10.4.7-28**

P-CSCF acknowledges receipt of the 302 Redirect response (27) by sending an ACK request to S-CSCF#1.

Table 10.4.7-28: ACK (P-CSCF to S-CSCF)

```
ACK sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Content-length:
```

29. **302 Redirect (P-CSCF to UE) – see example in Table 10.4.7-29**

P-CSCF sends a 302 Redirect response to UE, containing the new destination.

Table 10.4.7-29: 302 Redirect (P-CSCF to UE)

```
SIP/2.0 302 Redirect
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact:
Content-length: 0
```

30. **ACK (UE to P-CSCF) – see example in Table 10.4.7-30**

UE acknowledges receipt of the 302 Redirect response (29) by sending an ACK request to P-CSCF.

Table 10.4.7-30: ACK (UE to P-CSCF)

```
ACK sip:token6@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
Cseq:
Content-length:
```

31. **INVITE (UE to P-CSCF) – see example in Table 10.4.7-31**

UE sends the INVITE request, containing an initial SDP and the new destination, to the P-CSCF determined via the CSCF discovery mechanism.

Editor's Note: Certain fields in the SDP carry no information. In particular the "o=", "s=" fields and "t=". These are, however, mandatory fields within SDP. Does 3GPP wish to define a non-standard version of SDP that removes these, and if so, how does this interwork with outside SIP networks that use standard SDP.

Table 10.4.7-31: INVITE (UE to P-CSCF)

```
INVITE sip:token(sip:token(sip:+1-212-555-
   3333@home.net;user=phone)@scscf2.home.net;user=private)@icscf.home.net;user=private SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

Request-URI: contains the keyed number from the user.

Via: contains the IP address or FQDN of the originating UE.

Remote-Party-ID: contains the public identity of the UE. The Display name is optional.

Proxy-Require: The sip privacy draft specifies that the usage of the Remote-Party-Id MUST be accompanied by a Proxy-Require header specifying "privacy" in all INVITE requests.

From:, To:, Call-ID: follow the recommendations of draft-ietf-sip-privacy-01, even though anonymity is not being requested for this session.

Cseq: a random starting number.

Contact: the IP address or FQDN of the originating UE.

32. **100 Trying (P-CSCF to UE) – see example in Table 10.4.7-32**

P-CSCF responds to the INVITE request (31) with a 100 Trying provisional response.

Table 10.4.7-32: 100 Trying (P-CSCF to UE)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

33. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.4.7-33**

   P-CSCF remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. This next hop is the S-CSCF within the home network.

   P-CSCF rewrites the Contact header, with a locally defined value that identifies the UE. P-CSCF adds itself to the Record-Route header, and adds a Via header.

   The INVITE request is forwarded to the S-CSCF.

Table 10.4.7-33: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(sip:token(sip:+1-212-555-
    3333@home.net;user=phone)@scscf2.home.net;user=private)@icscf.home.net;user=private
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Request-URI:	the first component in the remembered Path header from Registration.
Route:	contains the remaining elements from the Path header from Registration, with the initial Request-URI (received from the UE) appended as the final component.
SDP:	The set of media flows described by the SDP may be reduced based on operator policy, or due to lack of authority of the subscriber to request such a media flow. Procedures are described in Subclause 10.3.5.

34. **100 Trying (S-CSCF to P-CSCF) – see example in Table 10.4.7-34**

   S-CSCF responds to the INVITE request (33) with a 100 Trying provisional response.

Table 10.4.7-34: 100 Trying (S-CSCF to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

35. **Service Control**

    S-CSCF validates the service profile, and performs any origination service control required for this subscriber.

36. **INVITE (S-CSCF to I-CSCF) – see example in Table 10.4.7-36**

    S-CSCF forwards the INVITE request to the I-CSCF specified in the destination URL.

    Editor's Note: Need for additional headers to transport e.g. Billing-Correlation-Identifier is FFS.

Table 10.4.7-36: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:token(sip:token(sip:+1-212-555-
    3333@home.net;user=phone)@scscf2.home.net;user=private)@icscf.home.net;user=private SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Request-URI:	This is the private URL obtained from the previous 302 Redirect response, which identifies the I-CSCF that must first translate the destination (then the S-CSCF that must further translate the destination). It was copied from the Route header of the incoming request to S-CSCF#1.
SDP:	The set of media flows described by the SDP may be reduced based on operator policy, or due to lack of authority of the subscriber to request such a media flow. Procedures are described in Subclause 10.3.1.

37. **100 Trying (I-CSCF to S-CSCF) – see example in Table 10.4.7-37**

    I-CSCF responds to the INVITE request (36) with a 100 Trying provisional response.

Table 10.4.7-37: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

38. **INVITE (I-CSCF to S-CSCF) – see example in Table 10.4.7-38**

I-CSCF translates the private portion of the URL, and determines the destination is S-CSCF2. I-CSCF forwards the INVITE request to the S-CSCF#2 that will further translate the destination.

Table 10.4.7-38: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:token(sip:+1-212-555-3333@home.net;user=phone)@scscf2.home.net;user=private SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

39. **100 Trying (S-CSCF to I-CSCF) – see example in Table 10.4.7-39**

S-CSCF#2 responds to the INVITE request (38) with a 100 Trying provisional response.

Table 10.4.7-39: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

40. **INVITE (S-CSCF to I-CSCF) – see example in Table 10.4.7-40**

S-CSCF translates the private portion of the URL, and determines the destination address. S-CSCF forwards the INVITE request to the I-CSCF#F, the entry point to the destination operator's network.

Table 10.4.7-40: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-3333@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

41. **100 Trying (I-CSCF to S-CSCF) – see example in Table 10.4.7-41**

   I-CSCF#F responds to the INVITE request (40) with a 100 Trying provisional response.

Table 10.4.7-41: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

The remainder of this session completes as shown in Clause 8.

100 Signalling flows for "regular" error handling

Editors Note: The purpose of this clause is to capture all the "regular" error handling scenarios. One example that falls into this class is Network Initiated Deregistration.

Editors Note: The Clause number "100" is temporary. It will be replaced with the appropriate clause number before this specification goes for approval, to enable this clause to be located towards the end of the Specification.

100.1 Registration: network initiated deregistration

Editor's Note: A figure detailing the message sequence needs to be added here.

Editor's Note: Further study is required to show how the P-CSCF is triggered to generate a NOTIFY request.

XX   SIP NOTIFY (P-CSCF to UE) – see example in Table 20.2.2-x

   The P-CSCF sends a NOTIFY to the UE in order to inform about the network initiated deregistration. The same Request URI, To, From, Call-ID are used as in the first NOTIFY. CSeq is incremented.

The contents of the Event header field (i.e. its parameters) are FFS, as this requires further input from stage 2. In the following the Event header field is only an example, to give some idea, how it may look like. Further possibilities are to use Warning and Retry-After header fields (instead of / in addition to Event header field parameters) for transferring the concerning information about the event.

Table 100.2.2-x SIP NOTIFY (P-CSCF to UE)

```
NOTIFY sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net
From: <sip:pcscf1.visited1.net>
To: <sip:sip:user1@home1.net>
Call-ID: uv45wx67yz89@pcscf1.visited1.net
CSeq: 2 NOTIFY
Expires: 7200
Event: org.3gpp.nwinitdereg; reason-code=999;
  reason-phrase="You have been deregistered from the network, please register again";
  registrar=sip:registrar.home_network.net
Content-Length: 0
```

XY   200 OK (UE to P-CSCF) – see example in Table 100.2.2-x

The UE stores the Call-ID, CSeq, etc. and generates a 200 OK response to the NOFIFY

Table 100.2.2-x SIP 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net
From:
To:
Call-ID:
CSeq:
Content-Length:
```

101   Signalling flows for "irregular" error handling

Editors Note: The purpose of this clause is to capture all the "irregular" error handling scenarios. An example is recovery/failure flows to deal with the event of a Network Element failure. The flows in this clause would typically refer to flows detailed in earlier clauses of this Specification.

Editors Note: The clause number "101" is temporary. It will be replaced with the appropriate clause number before this specification goes for approval, to enable this clause to be located towards the end of the Specification.

101.1   Error handling: Elements of signalling flows for provision of IP-connectivity network

101.2   Error Handling: Registration signalling

101.2.1   Re-registration – failure of re-registration

This information flow is a continuation of the information flow in subsubclause 7.2 "Registration Signalling: Re-Registration – User Currently Registered" after reception of information flow 4. This information flow shows the recovery after a failure of the S-CSCF that had been assigned to the subscriber in a previous registration.

Figure 101.2.1-1: Failure of previous S-CSCF during re-registration

Steps 1 through 4 are the same as the information flow in subclause 7.2 "Registration signalling: Re-registration – user currently registered"

5 **SIP REGISTER (I-CSCF to S-CSCF) – see example in Table 101.2.1-5**

This signalling flow forwards the SIP REGISTER request from the I-CSCF to the S-CSCF selected. The Request-URI is changed to the address of the S-CSCF.

I-CSCF adds a proper I-CSCF name to the Path header.

Table 101.2.1-5 SIP REGISTER request (I-CSCF to S-CSCF)

```
REGISTER sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf2.home1.net
Via: SIP/2.0/UDP pcscf1.visited1.net
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:icscf2.home1.net>
Path: <sip:pcscf1.visited1.net>
Proxy-require: path
Require: path
From: <sip:user_private1@home1.net>
To: <sip:user_public1@home1.net>
Contact: <sip:user1%40home1.net@pcscf.visited1.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 10 REGISTER
Expires: 7200
Content-Length: 0
```

6   **Timeout of Re-Register**

The I-CSCF times out, waiting for the response from the S-CSCF.

Editor's Note: The value of the timer in this particular instance is FFS.   Clearly the value of the timers in the P-CSCF and UE waiting for the response must be considered when choosing this value.

7   **Cx-Query (Optional)**

The I-CSCF informs the HSS that the S-CSCF for the subscriber is unreachable and requests information related to the required S-CSCF capabilities from the HSS, The HSS sends the capability information required for S-CSCF selection. The I-CSCF uses this information to select a suitable S-CSCF.

This step is optional. Depending on implementation, sufficient information may be available to the I-CSCF from Step 4, to allow the I-CSCF select an alternate S-CSCF. Alternative mechanisms (for example a CSCF management plane) would be used to enable the HSS learn of S-CSCF failure. In addition, the HSS will learn about the assignment of a new S-CSCF in Step 9.

8   **SIP REGISTER (I-CSCF to S-CSCF) – see example in Table 101.2.1-8**

This signalling flow forwards the SIP REGISTER request from the I-CSCF to the newly selected S-CSCF. The Request-URI is changed to the address of the new S-CSCF.

Table 101.2.1-8 SIP REGISTER request (I-CSCF to S-CSCF)

```
REGISTER sip: scscf2.home1.net SIP/2.0
Via:
Via:
Via:
Path:
Path:
Proxy-require:
Require:
From:
To:
Contact:
Call-ID:
CSeq:
Expires:
Content-Length:
```

The remaining steps (9-15) are the same as in the normal re-registration case (steps 6-12 in subclause 7.2)

101.2.2 Network initiated deregistration

101.3   Error handling: Session initiation

Editor's Note: Tdoc N1-010484 outlines a number of exception conditions, which should be expanded in future contributions and included under this subclause.

101.4   Error handling: Session termination

101.4.1 Network initiated session release – P-CSCF initiated

101.4.2 Network initiated session release – S-CSCF initiated

Annex A (informative): Documentation of preliminary material

7.1 Registration signalling: user not registered

This shows the registration signalling flow for the scenario when the user is not registered. For the purpose of this registration flow, the subscriber is considered to be roaming. In this signalling flow, the home network has network configuration hiding active.

Figure 7.1-1: UE roaming

1. **GPRS Attach / PDP Context Establishment (UE to GPRS)**

   This signalling flow is shown to indicate the GPRS 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 to GPRS/ DHCP)**

   This signalling flow is the procedure to discover the Proxy CSCF in the visited IM CN subsystem.

   The UE should be able to obtain the IP address of the P-CSCF during the PDP Context Activation procedure. At the PDP Context Activation time, the IP address of the P-CSCF shall be conveyed to the UE in the Activate PDP Context Accept message. The UEs that do not support DHCP shall use this P-CSCF discovery procedure.

   Additionally, the UEs that incorporate the DNS and DHCP client software that supports DHCP extensions (specified in draft-ietf-sip-dhcp-03.txt) may employ the DHCP mechanism to discover the P-CSCF in the visited IM CN subsystem. When allocating an IP address to the UE, the DHCP server may provide the UE with the fully-qualified domain name (FQDN) of the P-CSCF and the address of the DNS server in the visited IM CN

subsystem. Subsequently, the DNS client in the UE will utilise the provided FQDN and perform an address-record lookup (i.e. type A DNS access) to obtain the IP address of the P-CSCF in the visited IM CN subsystem.

NOTE: A UE may be roaming within the home network.

Editor's Note: IANA Considerations - Currently the IANA has not assigned an "DHCP option number" for the *SIP Servers DHCP Option* defined in the draft-ietf-sip-dhcp-03.txt. Therefore, the DHCP alternative can not be currently implemented.

Editor's Note: Second approach needs further study on the interactions with the restrictions on the Signalling PDP Context, TS 23.228 subclause 4.2.6.

3. **SIP REGISTER request (UE to P-CSCF) – see example in Table 7.1-3**

The purpose of this request is to register the user's SIP URI with a S-CSCF in the home network. This request is routed to the P-CSCF because it is the only SIP server known to the UE. In the following SIP request, 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@home1.net) and the host (terminal) address ([5555::aaa:bbb:ccc:ddd]) which was acquired during PDP context activation process.

Table 7.1-3 SIP REGISTER request (UE to P-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user1@home1.net>
To: <sip:user1@home1.net>
Contact: <Sip:[5555::aaa:bbb:ccc:ddd]>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 1 REGISTER
Expires: 7200
Allow-Events: org.3gpp.nwinitdereg
Content-Length: 0
```

Request-URI: The Request-URI (the URI that follows the method name, "REGISTER", in the first line) indicates the destination domain of this REGISTER request. The rules for routeing a SIP request describe how to use DNS to resolve this domain name ("home1.net") into an address or entry point into the home operator's network (the I-CSCF). This information is stored in the USIM.

Editor's note: One proposal is that the above URI forms an integral part of a global private identifier.

Via: IPv6 PDP address of the SIP session allocated during the PDP Context Activation process.

From: This indicates the SIP identity of the user (stored in the USIM) originating the REGISTER request. In SIP, this can be a third-party.

Editor's note: One proposal is: "This is a natural place for the private identifier or NAI for the subscriber. Forming a SIP URL from the NAI is a simple matter of prepending "sip:". For example, if the subscriber's NAI is 19725835472@operator.com, then the From: header would be sip:19725835472@operator.com." Alternatively it could be the SIP-URL of the party registering.

To: This indicates the public identifier being registered. This is the identity by which other parties know this subscriber. It is obtained from the USIM.

Editor's note: One proposed additional text: "In this case, this is the global SIP URL for the subscriber."

Contact: This indicates the point-of-presence for the subscriber – the IP address of the UE. This is the temporary point of contact for the subscriber that is being registered. Subsequent requests destined for this subscriber will be sent to this address. This information is stored in the P-CSCF.

Editor's note: It is for further study whether this information is stored in the HSS and the S-CSCF for the subscriber in order to support multiple registrations.

Call Id: Call Identifier for this Registration generated as per [3]

Cseq: Cseq for this Registration generated as per [3]

Allow-Events: Allow-Events is used to implicitly subscribe to the event "org.3gpp.nwinitdereg".

Editor's note: At this point, all that is missing from the REGISTER request is the visited operator's domain name. This piece of information is inserted into the Contact: header by the P-CSCF in the visited network upon receiving the REGISTER request from the UE. The P-CSCF takes the Contact: header indicating the IP address of the UE and replaces it with a new Contact: header indicating the address of the P-CSCF (including the visited operator's domain name). The IP address of the UE is stored in the P-CSCF for routing incoming requests to the appropriate UE.

For example, the UE's Contact: address ("sip:[5555::aaa:bbb:ccc:ddd]") is replaced by a Contact: address for the proxy ("sip:sean.olson%40home1.net@pcscf1.visited1.net"). This new Contact: address contains a specially encoded version of the subscriber's identity (To: header), plus the address of the P-CSCF.

If the proposed private identifier is not included in the From header, a mechanism is also needed to transfer this proposed private identifier.

Upon receiving this request the P-CSCF will set it's SIP registration timer for this UE to the Expires time in this request.

4. **DNS-Q**

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.

The P-CSCF sends the REGISTRATION request - after local processing - to the address indicated in the Request-URI. When forwarding the REGISTRATION request the P-CSCF needs to specify the protocol, port number and IP address of the I-CSCF server in the home network to which to send the REGISTRATION request. The P-CSCF tries to find this information by querying the DNS. Since the Request-URI does not specify the transport protocol the, P-CSCF selects the UDP.

Table 7.1-4a DNS Query (P-CSCF to DNS)

```
OPCODE=SQUERY
QNAME=_sip.udp.registrar.home_network.net, QCLASS=IN, QTYPE=SRV
```

The DNS records are retrieved according to RFC2782.

Table 7.1-4b DNS Query Response (DNS to P-CSCF)

```
OPCODE=SQUERY, RESPONSE, AA
QNAME=_sip.udp.registrar.home_network.net, QCLASS=IN, QTYPE=SRV

_sip._udp.registrar.home_network.net    0 IN SRV 1 10 5060 icscf2.home_network.com
                                        0 IN SRV 1  0 5060 icscf7.home_network.com

icscf2.home_network.com                 0 IN AAAA    5555::aba:dab:aaa:daa
icscf7.home_network.com                 0 IN AAAA    5555::a1a:b2b:c3c:d4d
```

In the Answer field of the query-response each I-CSCF is identified by its host domain name. The returned SRV Resource Records (RRs) are merged and ordered, and the selection technique (employing the Priority and Weight parameters returned in the RRs) as specified in RFC2782 is used to select the I-CSCF (i.e., the icscf2.home_network.com). Since the Additional Data field of the query-response also contains the IP address of the selected I-CSCF (i.e., 5555::aba:dab:aaa:daa), a new query to the DNS is not required.

Once the IP address of the I-CSCF is obtained, the P-CSCF forwards the REGISTRATION request to this IP address (i.e., 5555::aba:dab:aaa:daa) using the UDP protocol and port number 5060.

5. **SIP REGISTER request (P-CSCF to I-CSCF) – see example in Table 7.1-5**

Since this P-CSCF is a stateful proxy, 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%40home1.net@pcscf1.visited1.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 request for user1@home1.net, to the home registrar, using a contact address of user1%40home.net@pcscf1.visited1.net.

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

Table 7.1-5 SIP REGISTER request (P-CSCF to I-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:pcscf1.visited1.net>
Proxy-require: path
Require: path
From:
To:
Contact: <sip:user1%40home1.net@pcscf1.visited1.net>
Call-ID:
CSeq:
Expires:
Allow-Events:
Content-Length:
```

Path: This is the address of the P-CSCF and is included to inform the S-CSCF where to route terminating sessions.

Require, Proxy-Require: These headers are included to ensure that the recipient correctly handles the Path header. If the recipient does not support the path header, a response will be received with a status code of 420 and an Unsupported header indicating "path". Such a response indicates a misconfiguration of the routing tables and the request has been routed outside the IM CN subsystem.

6. **Cx-Query**

The I-CSCF requests information related to the required S-CSCF capabilities from the HSS. The HSS provides the I-CSCF with either the S-CSCF address for the subscriber (if the subscriber is currently registered) or the S-CSCF required capabilities (if the subscriber is not currently registered.) Since the subscriber is not registered in this case, the HSS returns the S-CSCF required capabilities and the I-CSCF uses this information to select a suitable S-CSCF.

7. **SIP REGISTER request (I-CSCF to S-CSCF) – see example in Table 7.1-7**

I-CSCF adds a proper I-CSCF name to the Path header.

This signalling flow forwards the SIP REGISTER from the I-CSCF to the S-CSCF selected. The Request-URI is changed to the address of the S-CSCF.

Table 7.1-7 SIP REGISTER request (I-CSCF to S-CSCF)

```
REGISTER sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Path: <sip:icscf2.home1.net>, <sip:pcscf1.visited1.net>
Proxy-require:
Require:
From:
To:
Contact:
Call-ID:
CSeq:
Expires:
Allow-Events:
Content-Length:
```

Path: The S-CSCF stores the contents of the Path headers and uses these addresses for routing mobile terminated sessions.

Upon receiving this request the S-CSCF will set it's SIP registration timer for this UE to the Expires time in this request.

8. **Cx-Location**

The S-CSCF shall send its location information to the HSS. The HSS stores the S-CSCF name for that subscriber. The HSS sends a response to the S-CSCF to acknowledge the sending of location information.

9. **Cx-Profile**

The S-CSCF shall send the subscriber's identity to the HSS in order to be able to download the subscriber profile to the S-CSCF. The HSS returns the subscriber's profile to the S-CSCF. The S-CSCF shall store the subscriber profile for that indicated user.

10. **SIP 200 OK response (S-CSCF to I-CSCF) – see example in Table 7.1-10**

The S-CSCF sends acknowledgment to the I-CSCF indicating that Registration was successful. This response will traverse the path that the REGISTER request took as described in the Via list.

Table 7.1-10 SIP 200 OK response (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Path: <sip:scscf1.home1.net>, <sip:icscf2.home1.net>, <sip:pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

Path: The S-CSCF inserts its own name to the front of the list.

11. **SIP 200 OK response (I-CSCF to P-CSCF) – see example in Table 7.1-11**

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

Table 7.1-11 SIP 200 OK response (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:token(scscf1.home1.net)>, <sip:icscf2.home1.net>, <sip:pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

12. **SIP 200 OK response (P-CSCF to UE) – see example in Table 7.1-12**

The P-CSCF removes its address from the Path header, reverses the order of the fields, saves the resulting Path header and associates it with the UE. The P-CSCF then removes the Path header from the 200 OK response. The P-CSCF then forwards acknowledgment from the I-CSCF to the UE indicating that Registration was successful.

Table 7.1-12 SIP 200 OK response (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

13. **SIP NOTIFY (S-CSCF to I-CSCF)**

14. **SIP NOTIFY (I-CSCF to UE)**

15. **SIP NOTIFY (P-CSCF to UE)**

Editor's Note: The content of messages 13 to 15 are FFS.

16. **SIP 200 OK (UE to P-CSCF)**

17. **SIP 200 OK (P-CSCF to I-CSCF)**

18. **SIP 200 OK (I-CSCF to S-CSCF)**

Editor's Note: The content of messages 14 to 16 are FFS.

7.2 Registration signalling: Re-registration – user currently registered

For the purpose of this re-registration flow, the subscriber is considered to be roaming. In this signalling flow, the home network has network configuration hiding active.

This flow assumes :

1. That the same PDP Context allocated during the initial registration scenario is still used for re-registration. For the case when the UE does not still have an active PDP context then PDP context procedures from subclause 7.1 must first be completed.

2. The DHCP procedure employed for P-CSCF discovery is not needed.

3. The S-CSCF selection procedure invoked by the I-CSCF is not needed.

Periodic application level re-registration is initiated by the UE either in response to the expiration of the existing registration information or in response to a change in the registration status of the UE. Re-registration follows the same path as described in subclause 7.1.

Figure 7.2-1: UE roaming

1. **SIP REGISTER request (UE to P-CSCF) – see example in Table 7.2-1**

   The registration expires in the UE. The UE re-registers by sending a new REGISTER request. This request is sent to the same P-CSCF with which the UE initially registered. The P-CSCF maintains the same binding between the User's SIP public address (user_public1@home1.net) and the host (terminal) address ([5555::aaa:bbb:ccc:ddd]) which it established during the original registration.

Table 7.2-1 SIP REGISTER (UE to P-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: <sip:user_private1@home1.net>
To: <sip:user_public1@home1.net>
Contact: <Sip:[5555::aaa:bbb:ccc:ddd]>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 7 REGISTER
Expires: 7200
Allow-Events: org.3gpp.nwinitdereg
Content-Length: 0
```

The header field usage is the same as for the initial registration scenario:

From: This indicates the private identity of the user (stored in the USIM) originating the REGISTER request.

Editor's note: The requirement for and impact on third party registration with this usage of the From header needs further study.

To: This indicates the target of the REGISTER request. The target is the public identity that is being registered. This is the identity by which other parties know this subscriber.

Contact: This indicates the point-of-presence for the subscriber – the IP address of the UE. This is the temporary identifier for the subscriber that is being registered. Subsequent requests destined for this subscriber will be sent to this address. This information is stored in the P-CSCF.

Editor's note: It is for further study whether this information is stored in the HSS and the S-CSCF for the subscriber in order to support multiple registrations.

Request-URI: The Request-URI (the URI that follows the method name, "REGISTER", in the first line) indicates the destination domain of this REGISTER request. The rules for routeing a SIP request describe how to use DNS to resolve this domain name ("home1.net") into an address or entry point into the home operator's network (the I-CSCF). This information is stored in the USIM.

Editor's note: At this point, all that is missing from the REGISTER request is the visited operator's domain name. This piece of information is inserted into the Contact: header by the P-CSCF in the visited network upon receiving the REGISTER request from the UE. The P-CSCF takes the Contact: header indicating the IP address of the UE and replaces it with a new Contact: header indicating the address of the P-CSCF (including the visited operator's domain name). The IP address of the UE is stored in the P-CSCF for routing incoming requests to the appropriate UE.

For example, the UE's Contact: address ("sip:[5555::aaa:bbb:ccc:ddd]") is replaced by a Contact: address for the proxy ("sip:sean.olson%40home1.net@pcscf1.visited1.net"). This new Contact: address contains a specially encoded version of the subscriber's identity (To: header), plus the address of the P-CSCF.

Upon receiving this request the P-CSCF will detect that it already has a registration record for this UE and will reset it's SIP registration timer for this UE to the Expires time in this request.

2. **DNS-Q**

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. The DNS provides the P-CSCF with an address of the I-CSCF in the home network. The P-CSCF must not use the I-CSCF address cached as a result of the previous registration.

3. **SIP REGISTER request (P-CSCF to I-CSCF) – see example in Table 7.2-3**

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

Table 7.2-3 SIP REGISTER request (P-CSCF to I-CSCF)

```
REGISTER sip:registrar.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:pcscf1.visited1.net>
Proxy-require: path
Require: path
From:
To:
Contact: <sip:user1%40home1.net@pcscf1.visited1.net>
Call-ID: CSeq:
Expires:

Content-Length:
```

Path: This is the address of the P-CSCF and is included to inform the S-CSCF where to route terminating sessions.

Require, Proxy-Require: These headers are included to ensure that the recipient correctly handles the Path header. If the recipient does not support the path header, a response will be received with a status code of 420 and an Unsupported header indicating "path". Such a response indicates a misconfiguration of the routing tables and the request has been routed outside the IM CN subsystem.

4. **Cx-Query**

The I-CSCF requests information related to the required S-CSCF capabilities from the HSS. The HSS shall determine that the user is currently registered, and send an indication of current S-CSCF to the I-CSCF. Hence, the S-CSCF selection procedure is not needed.

5. **SIP REGISTER request (I-CSCF to S-CSCF) – see example in Table 7.2-5**

This signalling flow forwards the SIP REGISTER request from the I-CSCF to the S-CSCF selected. The Request-URI is changed to the address of the S-CSCF.

I-CSCF adds a proper I-CSCF name to the Path header.

Table 7.2-5 SIP REGISTER request (I-CSCF to S-CSCF)

```
REGISTER sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Path: <sip:icscf2.home1.net>, <sip: pcscf1.visited1.net
Proxy-require:
Require:
From:
To:
Contact:
Call-ID:
CSeq:
Expires:
Allow-Events:
Content-Length:
```

Path: The S-CSCF stores the contents of the Path headers and uses these addresses for routing mobile terminated sessions.

Upon receiving this request the S-CSCF will detect that it already has a registration record for this UE and will reset it's SIP registration timer for this UE to the Expires time in this request.

6. **Cx-Location**

The S-CSCF shall send its location information to the HSS. The HSS stores the S-CSCF name for that subscriber. The HSS sends a response to the S-CSCF to acknowledge the sending of location information.

Editor's Note: Since the S-CSCF has detected this as a re-registration it may omit this step as an optimisation.

7. **Cx-Profile**

The S-CSCF shall send the subscriber's identity to the HSS in order to be able to download the subscriber profile to the S-CSCF. The HSS returns the subscriber's profile to the S-CSCF. The S-CSCF shall store the subscriber profile for that indicated user.

Editor's Note: Since the S-CSCF has detected this as a re-registration it may omit this step as an optimisation.

8. **SIP 200 OK response (S-CSCF to I-CSCF) – see example in Table 7.2-8**

The S-CSCF sends acknowledgment to the I-CSCF indicating that Registration was successful. This response will traverse the path that the REGISTER request took as described in the Via list.

Table 7.2-8 SIP 200 OK response (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Path: <sip: scscf2.home1.net>, <sip: icscf2.home1.net>, <sip: pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

Path: The S-CSCF inserts its own name to the front of the list.

9. **SIP 200 OK response (I-CSCF to P-CSCF) – see example in Table 7.2-9**

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

Table 7.2-9 SIP 200 OK response (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:token(scscf2.home1.net)>, <sip: icscf2.home1.net>, <sip: pcscf1.visited1.net>
From:
To:
Call-ID:
CSeq:
Expires:
Content-Length:
```

10. **SIP 200 OK response (P-CSCF to UE) – see example in Table 7.2-10**

   The P-CSCF removes its address from the Path header, reverses the order of the fields, saves the resulting Path header and associates it with the UE. The P-CSCF then removes the Path header from the 200 OK response. The P-CSCF then forwards acknowledgment from the I-CSCF to the UE indicating that Registration was successful.

Table 7.2-10 SIP 200 OK response (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Expires: Content-Length:
```

11. **SIP NOTIFY (S-CSCF to I-CSCF)**

12. **SIP NOTIFY (I-CSCF to UE)**

13. **SIP NOTIFY (P-CSCF to UE)**

   Editor's Note: The content of messages 11 to 13 are FFS.

14. **SIP 200 OK (UE to P-CSCF) – see example in Table 7.2-12**

15. **SIP 200 OK (P-CSCF to I-CSCF)**

16. **SIP 200 OK (I-CSCF to S-CSCF)**

   Editor's Note: The content of messages 14 to 16 are FFS.

Editor's note: The following material is a proposed change to the table of contents (reorganization).

8.2   Origination procedures

8.2.1   Introduction

8.2.2   (MO#1a) Mobile origination, roaming, without I-CSCF in home network providing configuration independence

8.2.2.1   Successful session establishment

8.2.2.2   Failure in termination procedure

8.2.2.3   Session abandoned, or resource failure

8.2.3   (MO#1b) Mobile origination, roaming, with I-CSCF in home network providing configuration independence

8.2.3.1   Successful session establishment

8.2.3.2   Failure in termination procedure

8.2.3.3   Session abandoned, or resource failure

8.2.4   (MO#2) Mobile origination, located in home network

8.2.4.1   Successful session establishment

8.2.4.2   Failure in termination procedure

8.2.4.3   Session abandoned, or resource failure

8.2.5   (PSTN-O) PSTN origination

8.2.5.1   Successful session establishment

8.2.5.2   Failure in termination procedure

8.2.5.3   Session abandoned, or resource failure

8.3   **Termination Procedures**

8.3.1   Introduction

8.3.2   (MT#1a) Mobile termination, roaming, without I-CSCF in home network providing configuration independence

8.3.2.1   Successful session establishment

8.3.2.2   UE-detected failure/resource failure

{text currently in 8.2.5 could be placed here}

8.3.2.3   Origination failure

8.3.2.4   Service failure

{text currently in 8.2.8 could be placed here}

8.3.3   (MT#1b) Mobile termination, roaming, with I-CSCF in home network providing configuration independence

8.3.3.1   Successful session establishment

8.3.3.2   UE-detected failure/resource failure

{text currently in 8.2.6 could be placed here}

8.3.3.3   Origination failure

8.3.3.4   Service failure

8.3.4   (MT#2) Mobile termination, located in home network

8.3.4.1   Successful session establishment

8.3.4.2   UE-detected failure/resource failure

{text currently in 8.2.7 could be placed here}

8.3.4.3 Origination failure

8.3.4.4 Service failure

8.3.5 (PSTN-T) PSTN termination

8.3.5.1 Successful session establishment

8.3.5.2 MGCF-detected failure/resource failure

8.3.5.3 Origination failure

8.4 Serving-CSCF (MGCF) to Serving-CSCF (MGCF) Procedures

8.4.2 (S-S#1a) Different network operators performing origination and termination, without firewall between networks

8.4.2.1 Successful session establishment

8.4.2.2 Termination failure

8.4.2.3 Origination failure

8.4.3 (S-S#1b) Different network operators performing origination and termination, with configuration hiding

8.4.3.1 Successful session establishment

8.4.3.2 Termination failure

8.4.3.3 Origination failure

8.4.4 (S-S#2) Single network operator performing origination and termination

8.4.4.1 Successful session establishment

8.4.4.2 Termination failure

8.4.4.3 Origination failure

8.4.5 (S-S#3) PSTN termination performed by home network of originator

8.4.5.1 Successful session establishment

8.4.5.2 Termination failure

8.4.5.3 Origination failure

8.4.6 (S-S#4) PSTN Termination performed by different operator than origination

8.4.6.1 Successful session establishment

8.4.6.2 Termination failure

8.4.6.3 Origination failure

Editor's Note: The following material is proposed text for handling failures in the session establishment corresponding to the previous proposed change in the table of contents.

8.2 Origination procedures

8.2.2 (MO#1a) Mobile origination, roaming, without I-CSCF in home network providing configuration independence

8.2.2.2 Failure in termination procedure

The roaming subscriber that initiated a session with procedure MO#1a had the attempt fail due to an error detected in the Termination procedure or in the Serving-CSCF-to-Serving-CSCF procedure. This could be due to, for example, destination busy (error code 486), destination service denied (error code 403), destination currently out of coverage (error code 480), or some other error.

Depending on the exact error that causes the session initiation failure, and when the error situation was detected, UE#1 could be at many different stages in the session establishment procedure. This is shown in Figure 8.2.2.2-1, below, as optional messages 7-33 that may appear in this error procedure.
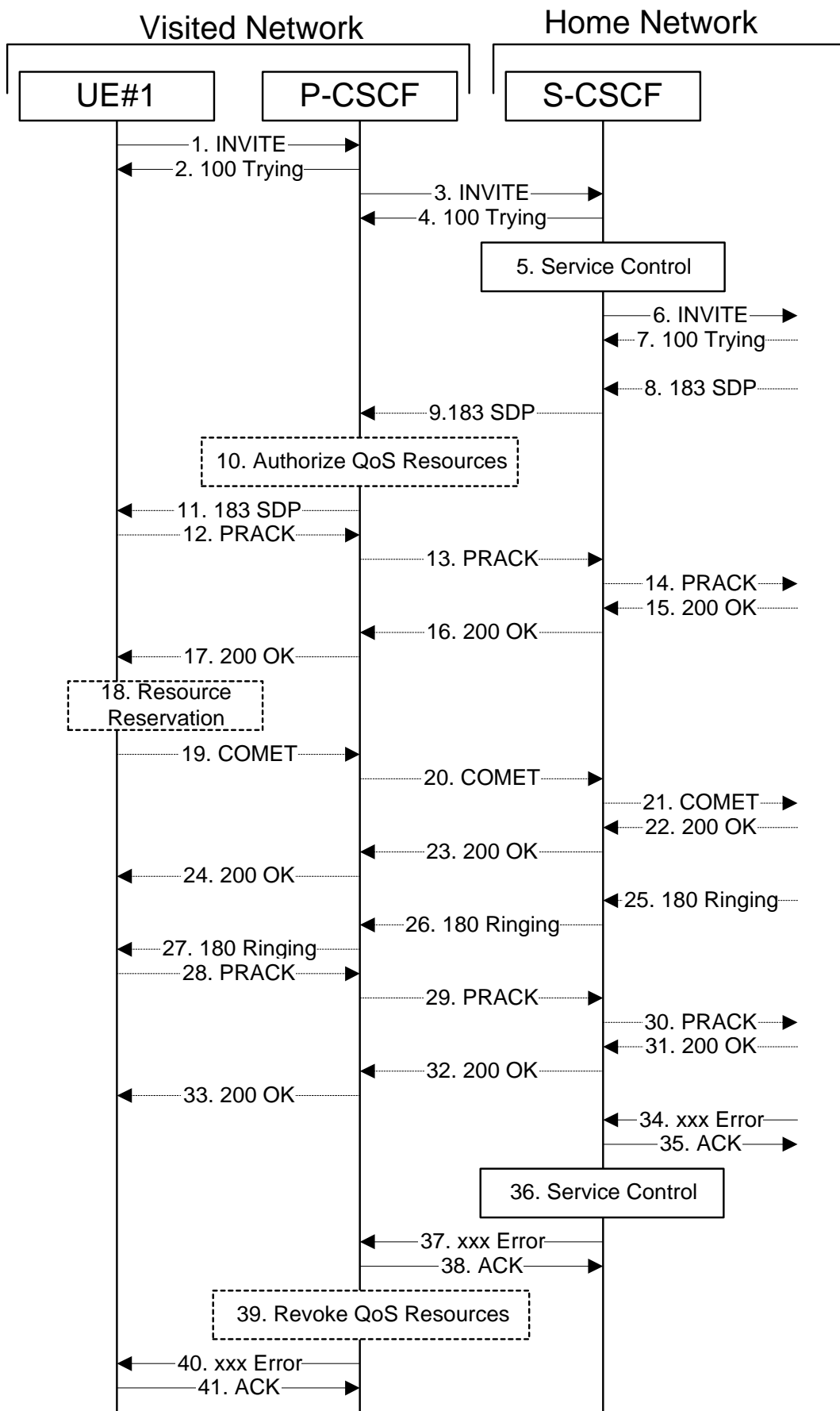
Figure 8.2.2.2-1: Failure in termination procedure

1-6. **INVITE (UE to P-CSCF) et seq**

UE#1 initiated a session, as described in Subclause 8.2.2.1.

7-33.**100 Trying (S-S to MO#1a) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.2.2.1.

34. **xxx Error (S-S to MO#1a) – see example in Table 8.2.2.2-34**

The termination procedure detected some error situation, and returned a SIP error response.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.2.2-34: 486 Busy Here (S-S to MO#1a)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

35. **ACK (MO#1a to S-S) – see example in Table 8.2.2.2-35**

Upon receive the 486 response from the S-S procedure, S-CSCF sends ACK.

Table 8.2.2.2-35: ACK (MO#1a to S-S)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

36. **Service Control**

S-CSCF performs whatever service control is appropriate for this failed session attempt.

37. **xxx Error (S-CSCF to P-CSCF) – see example in Table 8.2.2.2-37**

The S-CSCF returned a SIP error response to P-CSCF.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.2.2-37: 486 Busy Here (S-CSCF to P-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

38. **ACK (P-CSCF to S-CSCF) – see example in Table 8.2.2.2-38**

Upon receive the 486 response from the S-CSCF procedure, P-CSCF sends ACK.

Table 8.2.2.2-38: ACK (P-CSCF to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

39. **Revoke QoS authorization**

   P-CSCF removes the QoS authorization, if any, for this session.

40. **xxx Error (P-CSCF to UE) – see example in Table 8.2.2.2-40**

   The P-CSCF returned a SIP error response to UE.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.2.2-40: 486 Busy Here (P-CSCF to UE)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

41. **ACK (P-CSCF to S-CSCF) – see example in Table 8.2.2.2-41**

   Upon receive the 486 response from the P-CSCF, UE sends ACK.

Table 8.2.2.2-41: ACK (UE to P-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.2.2.3   Session abandoned, or resource failure

The roaming subscriber that initiated a session with procedure MO#1a either abandoned the attempt, or was unable to obtain the resources necessary for the session. The session flow diagram for this error handling is shown in Figure 8.2.2.3-1.

If the session is aborted due to failure to obtain resources, it will occur at step #18 in the session flow; steps 19-33 (marked as optional) will not be present. If the session is abandoned due to user command, it can happen at any point between steps 8-33.
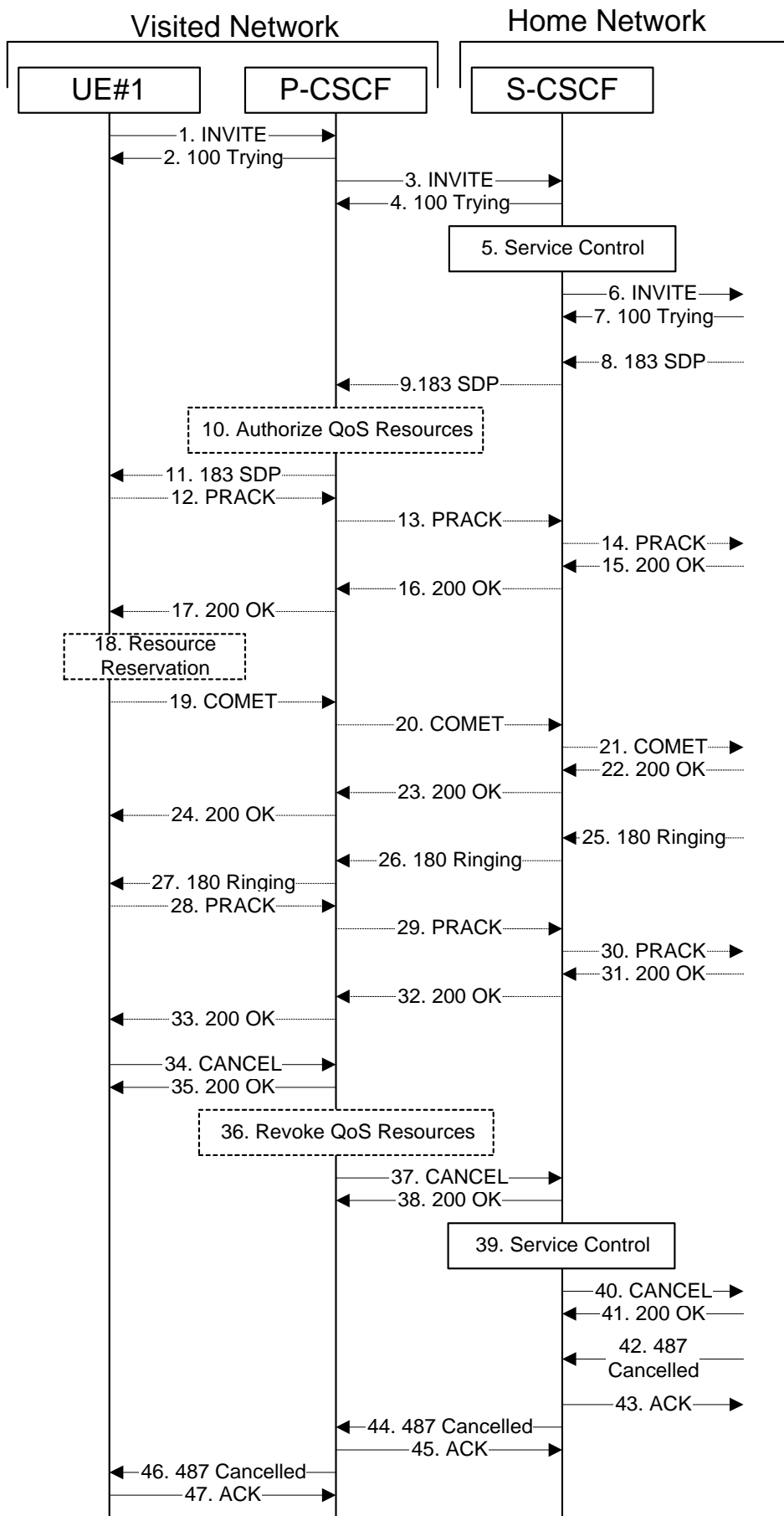
Figure 8.2.2.3-1: Session abandoned or resource failure

1-7. **INVITE (UE to P-CSCF) et seq**

UE#1 initiated a session, as described in Subclause 8.2.2.1.

8-33. **183 SDP (S-S to MO#1a) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.2.2.1.

34. **Cancel (UE to P-CSCF) – see example in Table 8.2.2.3-34**

The UE cancelled the original INVITE request.

Table 8.2.2.3-34: CANCEL (UE to P-CSCF)

```
CANCEL sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length:0
```

35. **200 OK (P-CSCF to UE) – see example in Table 8.2.2.3-35**

Upon receive the Cancel request from the UE, P-CSCF sends 200 OK.

Table 8.2.2.3-35: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

36. **Revoke QoS authorization**

P-CSCF removes the QoS authorization, if any, for this session.

37. **Cancel (P-CSCF to S-CSCF) – see example in Table** 8.2.2.3**-37**

The P-CSCF forwards the Cancel request to S-CSCF.

Table 8.2.2.3-37: Cancel (P-CSCF to S-CSCF)

```
CANCEL sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-length:
```

Editor's Note: Need to check whether P-CSCF should modify the Contact: header in the CANCEL.

Editor's Note: Need to check whether P-CSCF should insert a Route: header in the CANCEL

38. **200 OK (S-CSCF to P-CSCF) – see example in Table** 8.2.2.3**-38**

Upon receiving the Cancel request from the P-CSCF, S-CSCF sends 200 OK.

Table 8.2.2.3-38: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

39. **Service Control**

S-CSCF performs whatever service control is appropriate for this failed session attempt.

40. **Cancel (S-CSCF to S-S) – see example in Table 8.2.2.3-40**

The S-CSCF forwards the Cancel request to the appropriate Serving-CSCF-to-Serving-CSCF procedure.

Table 8.2.2.3-40: Cancel (S-CSCF to S-S)

```
CANCEL sip:+1-212-555-2222@home1.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

Editor's Note: Need to check whether a proxy inserts a Record-Route in a Cancel request.

41. **200 OK (S-S to S-CSCF) – see example in Table 8.2.2.3-41**

Upon receive the Cancel request from the S-CSCF, the next hop (whatever it is) sends 200 OK.

Table 8.2.2.3-41: 200 OK (S-S to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

42. **487 Cancelled (S-S to MO#1a) – see example in Table 8.2.2.3-41**

The termination procedure cancelled the request, and returned a SIP error response to the original INVITE request.

Table 8.2.2.3-41: 487 Cancelled (S-S to MO#1a)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Call-ID:
CSeq:
Content-length:0
```

43. **ACK (MO#1a to S-S) – see example in Table 8.2.2.3-43**

Upon receive the 487 response from the S-S procedure, S-CSCF sends ACK.

Table 8.2.2.3-43: ACK (MO#1a to S-S)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

44. **487 Cancelled (S-CSCF to P-CSCF) – see example in Table 8.2.2.3-44**

The S-CSCF returned the SIP error response to P-CSCF.

Table 8.2.2.3-44: 487 Cancelled (S-CSCF to P-CSCF)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
```

45. **ACK (P-CSCF to S-CSCF) – see example in Table 8.2.2.3-45**

Upon receive the 487 response from the S-CSCF, P-CSCF sends ACK.

Table 8.2.2.3-45: ACK (P-CSCF to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

46. **487 Cancelled (P-CSCF to UE) – see example in Table 8.2.2.3-46**

The P-CSCF returned a SIP error response to UE.

Table 8.2.2.3-46: 487 Cancelled (P-CSCF to UE)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

47. **ACK (UE to P-CSCF) – see example in Table 8.2.2.3-47**

Upon receive the 487 response from the P-CSCF, UE sends ACK.

Table 8.2.2.3-47: ACK (UE to P-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.2.3 (MO#1b) Mobile origination, roaming, with I-CSCF in home network providing configuration independence

8.2.3.2 Failure in termination procedure

The roaming subscriber that initiated a session with procedure MO#1b had the attempt fail due to an error detected in the Termination procedure or in the Serving-CSCF-to-Serving-CSCF procedure. This could be due to, for example, destination busy (error code 486), destination service denied (error code 403), destination currently out of coverage (error code 480), or some other error.

Depending on the exact error that causes the session initiation failure, and when the error situation was detected, UE#1 could be at many different stages in the session establishment procedure. This is shown in Figure 8.2.3.2-1, below, as optional messages 7-43 that may appear in this error procedure.
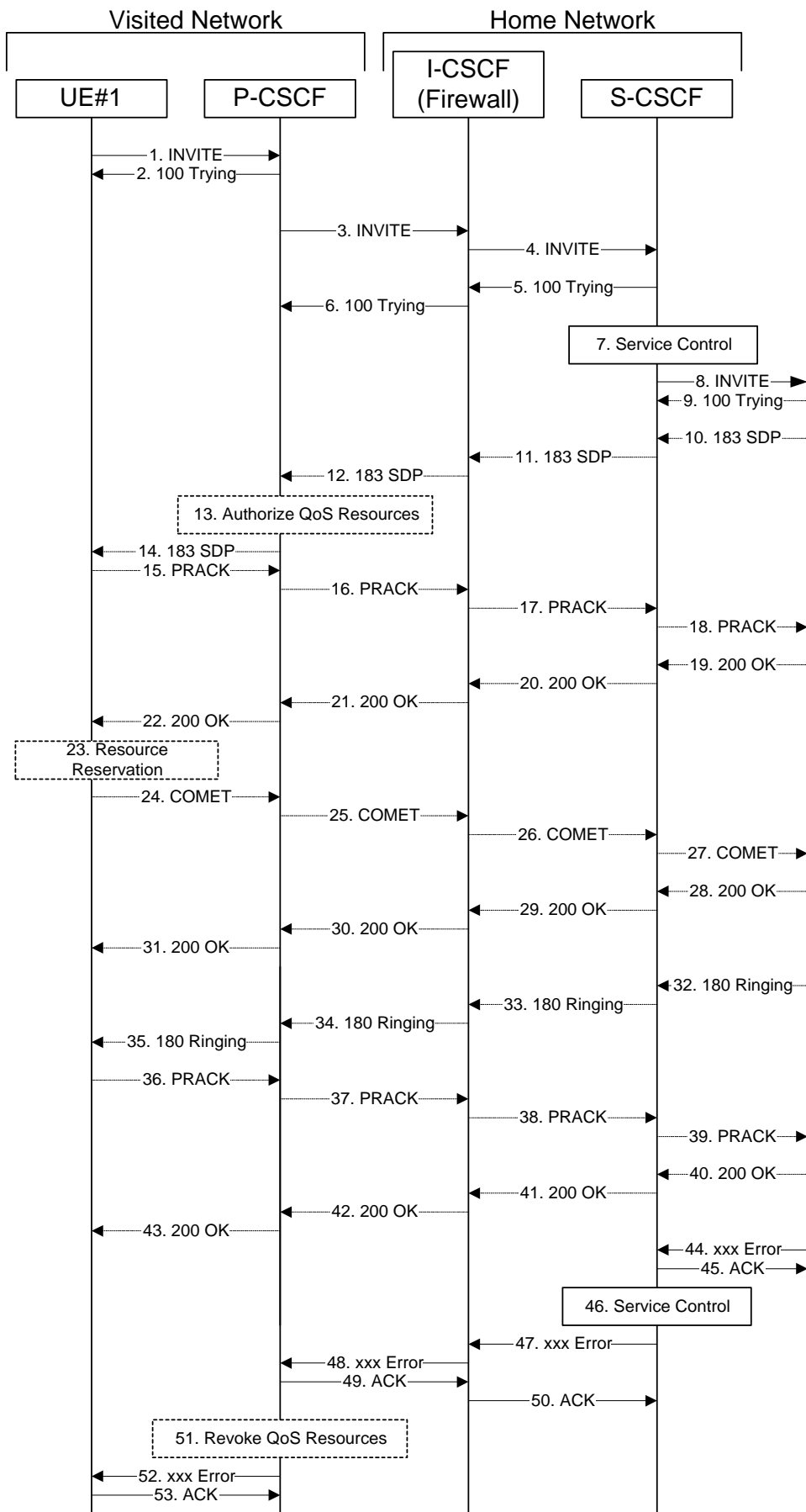
Figure 8.2.3.2-1: Failure in termination procedure

1-8. **INVITE (UE to P-CSCF) et seq**

UE#1 initiated a session, as described in Subclause 8.2.2.1.

9-43. **100 Trying (S-S to MO#1b) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.2.2.1.

44. **xxx Error (S-S to MO#1b) – see example in Table 8.2.3.2-44**

The termination procedure detected some error situation, and returned a SIP error response.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.3.2-34: 486 Busy Here (S-S to MO#1b)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Content-length:0
Retry-After:3600
```

45. **ACK (MO#1a to S-S) – see example in Table 8.2.3.2-45**

Upon receive the 486 response from the S-S procedure, S-CSCF sends ACK.

Table 8.2.3.2-45: ACK (MO#1a to S-S)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

46. **Service Control**

S-CSCF performs whatever service control is appropriate for this failed session attempt.

47. **xxx Error (S-CSCF to P-CSCF) – see example in Table 8.2.3.2-47**

The S-CSCF returned a SIP error response to I-CSCF.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.3.2-47: 486 Busy Here (S-CSCF to I-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

48. **xxx Error (I-CSCF to P-CSCF) – see example in Table 8.2.3.2-48**

The I-CSCF returned a SIP error response to P-CSCF.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.3.2-48: 486 Busy Here (S-CSCF to I-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

49. **ACK (P-CSCF to I-CSCF) – see example in Table 8.2.3.2-49**

Upon receive the 486 response from the I-CSCF, P-CSCF sends ACK.

Table 8.2.3.2-49: ACK (P-CSCF to I-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

50. **ACK (I-CSCF to S-CSCF) – see example in Table 8.2.3.2-50**

Upon receive the 486 response from the P-CSCF, I-CSCF sends ACK.

Table 8.2.3.2-50: ACK (I-CSCF to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP icscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

51. **Revoke QoS authorization**

P-CSCF removes the QoS authorization, if any, for this session.

52. **xxx Error (P-CSCF to UE) – see example in Table 8.2.3.2-52**

The P-CSCF returned a SIP error response to UE.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.3.2-52: 486 Busy Here (P-CSCF to UE)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

53. **ACK (UE to P-CSCF) – see example in Table 8.2.3.2-53**

Upon receive the 486 response from the P-CSCF, UE sends ACK.

Table 8.2.3.2-53: ACK (UE to P-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.2.3.3   Session abandoned, or resource failure

The roaming subscriber that initiated a session with procedure MO#1b either abandoned the attempt, or was unable to obtain the resources necessary for the session. The session flow diagram for this error handling is shown in Figure 8.2.3.3-1.

If the session is aborted due to failure to obtain resources, it will occur at step #23 in the session flow; steps 24-43 (marked as optional) will not be present. If the session is abandoned due to user command, it can happen at any point between steps 10-43.
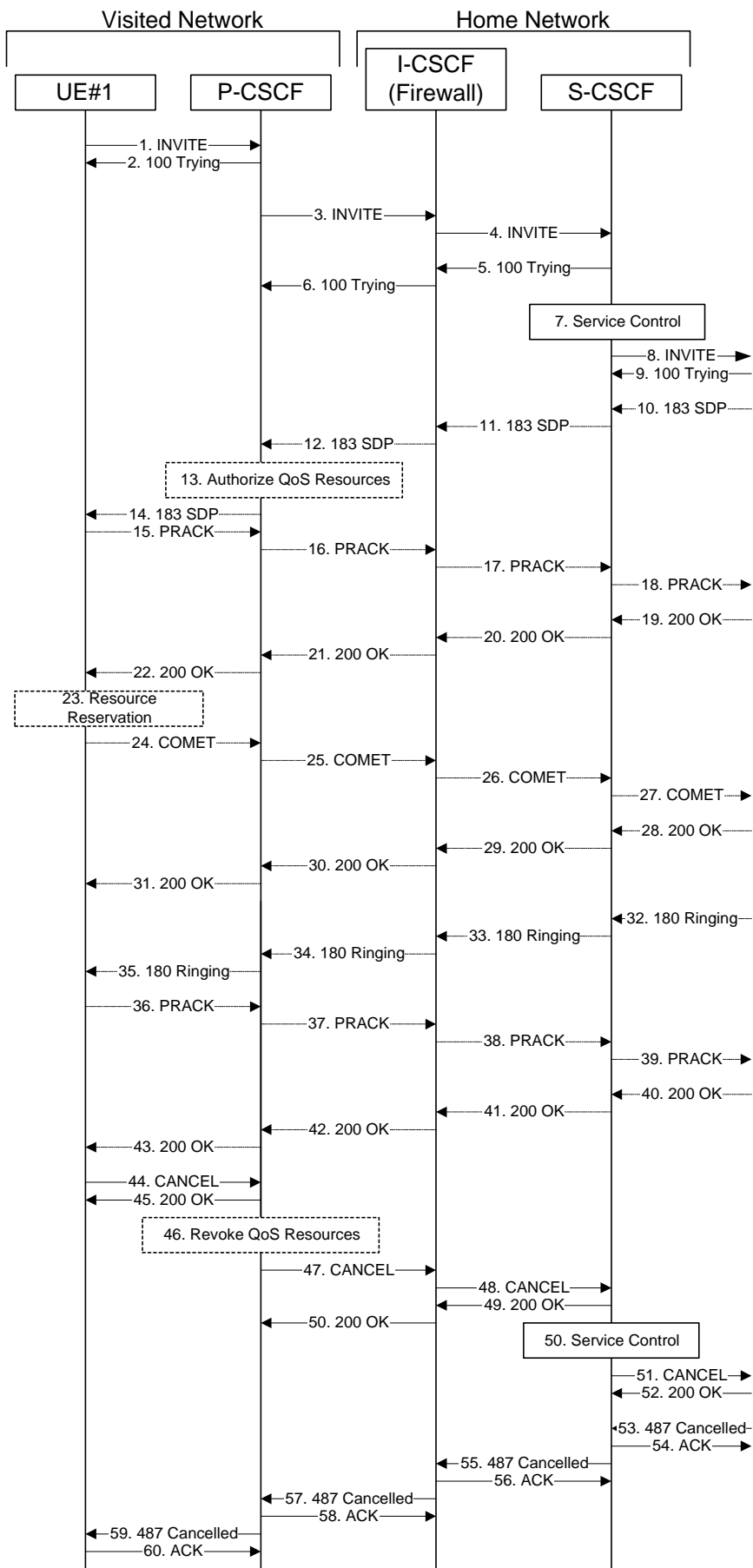
Figure 8.2.3.3-1: Session abandoned or resource failure

1-9. **INVITE (UE to P-CSCF) et seq**

UE#1 initiated a session, as described in Subclause 8.2.3.1.

10-43. **183 SDP (S-S to MO#1b) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.2.3.1.

44. **Cancel (UE to P-CSCF) – see example in Table 8.2.3.3-44**

The UE cancelled the original INVITE request.

Table 8.2.3.3-44: CANCEL (UE to P-CSCF)

```
CANCEL sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length:0
```

45. **200 OK (P-CSCF to UE) – see example in Table 8.2.3.3-45**

Upon receive the Cancel request from the UE, P-CSCF sends 200 OK.

Table 8.2.3.3-45: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

46. **Revoke QoS authorization**

P-CSCF removes the QoS authorization, if any, for this session.

47. **Cancel (P-CSCF to I-CSCF) – see example in Table 8.2.3.3-47**

The P-CSCF forwards the Cancel request to I-CSCF.

Table 8.2.3.3-47: Cancel (P-CSCF to I-CSCF)

```
CANCEL sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(scscf.home.net), sip:+1-212-555-2222@home.net;user=phone
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-length:
```

Editor's Note: Need to check whether P-CSCF should modify the Contact: header in the CANCEL.

Editor's Note: Need to check whether P-CSCF should insert a Route: header in the CANCEL

48. **Cancel (I-CSCF to S-CSCF) – see example in Table 8.2.3.3-48**

The I-CSCF forwards the Cancel request to S-CSCF.

Table 8.2.3.3-48: Cancel (I-CSCF to S-CSCF)

```
CANCEL sip:scscf.home.net SIP/2.0
Via: sip/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-length:
```

Editor's Note: Need to check whether P-CSCF should modify the Contact: header in the CANCEL.

Editor's Note: Need to check whether P-CSCF should insert a Route: header in the CANCEL

49. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.2.3.3-49**

Upon receiving the Cancel request from the P-CSCF, S-CSCF sends 200 OK.

Table 8.2.3.3-49: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: sip/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

50. **200 OK (I-CSCF to P-CSCF) – see example in Table 8.2.3.3-50**

Upon receiving the 200-OK response from the P-CSCF, I-CSCF sends 200 OK.

Table 8.2.3.3-50: 200 OK (I-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: sip/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

51. **Service Control**

S-CSCF performs whatever service control is appropriate for this failed session attempt.

52. **Cancel (S-CSCF to S-S) – see example in Table 8.2.3.3-52**

The S-CSCF forwards the Cancel request to the appropriate Serving-CSCF-to-Serving-CSCF procedure.

Table 8.2.3.3-52: Cancel (S-CSCF to S-S)

```
CANCEL sip:+1-212-555-2222@home1.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

Editor's Note: Need to check whether a proxy inserts a Record-Route in a Cancel request.

53. **200 OK (S-S to S-CSCF) – see example in Table 8.2.3.3-53**

Upon receive the Cancel request from the S-CSCF, the next hop (whatever it is) sends 200 OK.

Table 8.2.3.3-53: 200 OK (S-S to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

54. **487 Cancelled (S-S to MO#1b) – see example in Table 8.2.3.3-54**

   The termination procedure cancelled the request, and returned a SIP error response to the original INVITE request.

Table 8.2.3.3-54: 487 Cancelled (S-S to MO#1b)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Call-ID:
CSeq:
Content-length:0
```

55. **ACK (MO#1b to S-S) – see example in Table 8.2.3.3-55**

   Upon receive the 487 response from the S-S procedure, S-CSCF sends ACK.

Table 8.2.3.3-55: ACK (MO#1b to S-S)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

56. **487 Cancelled (S-CSCF to I-CSCF) – see example in Table 8.2.3.3-56**

   The S-CSCF returned the SIP error response to I-CSCF.

Table 8.2.3.3-56: 487 Cancelled (S-CSCF to I-CSCF)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
```

57. **487 Cancelled (I-CSCF to P-CSCF) – see example in Table 8.2.3.3-57**

   The I-CSCF returned the SIP error response to P-CSCF.

Table 8.2.3.3-57: 487 Cancelled (I-CSCF to P-CSCF)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
```

58. **ACK (P-CSCF to I-CSCF) – see example in Table 8.2.3.3-58**

Upon receive the 487 response from the S-CSCF, P-CSCF sends ACK.

Table 8.2.3.3-58: ACK (P-CSCF to I-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

59. **ACK (I-CSCF to S-CSCF) – see example in Table 8.2.3.3-59**

Upon receive the ACK from the P-CSCF, I-CSCF sends ACK.

Table 8.2.3.3-59: ACK (I-CSCF to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP pcscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

60. **487 Cancelled (P-CSCF to UE) – see example in Table 8.2.3.3-60**

The P-CSCF returned a SIP error response to UE.

Table 8.2.3.3-60: 487 Cancelled (P-CSCF to UE)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

61. **ACK (UE to P-CSCF) – see example in Table 8.2.3.3-61**

Upon receive the 487 response from the P-CSCF, UE sends ACK.

Table 8.2.3.3-61: ACK (UE to P-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.2.4 (MO#2) Mobile origination, located in home network

8.2.4.2 Failure in termination procedure

The roaming subscriber that initiated a session with procedure MO#2 had the attempt fail due to an error detected in the Termination procedure or in the Serving-CSCF-to-Serving-CSCF procedure. This could be due to, for example, destination busy (error code 486), destination service denied (error code 403), destination currently out of coverage (error code 480), or some other error.

Depending on the exact error that causes the session initiation failure, and when the error situation was detected, UE#1 could be at many different stages in the session establishment procedure. This is shown in Figure 8.2.4.2-1, below, as optional messages 7-33 that may appear in this error procedure.
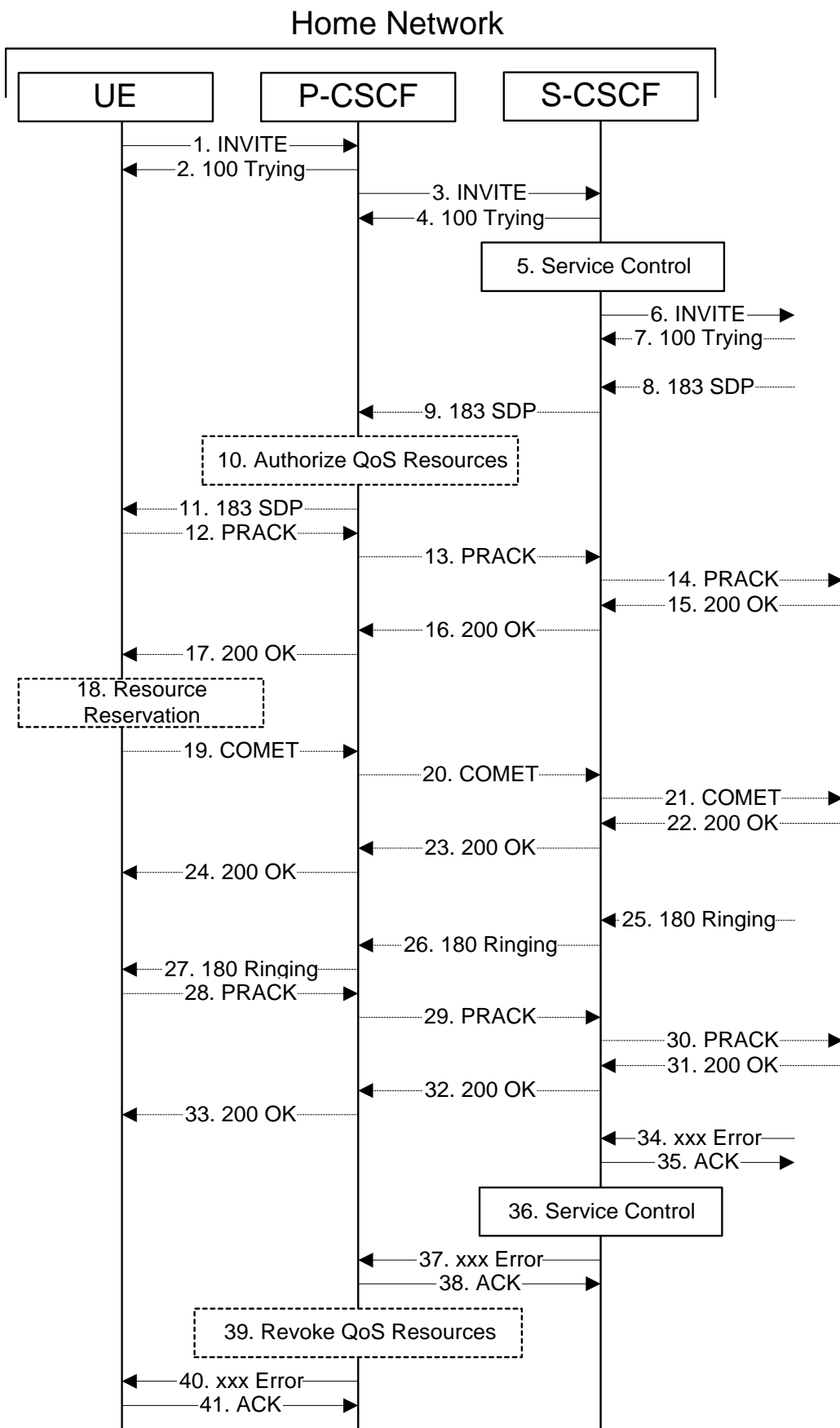
Home Network

Figure 8.2.4.2-1: Failure in termination procedure

1-6. **INVITE (UE to P-CSCF) et seq**

UE#1 initiated a session, as described in Subclause 8.2.4.1.

7-33. **100 Trying (S-S to MO#2) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.2.4.1.

34. **xxx Error (S-S to MO#2) – see example in Table 8.2.4.2-34**

The termination procedure detected some error situation, and returned a SIP error response.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.4.2-34: 486 Busy Here (S-S to MO#2)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:%5d5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Content-length:0
Retry-After:3600
```

35. **ACK (MO#2 to S-S) – see example in Table 8.2.4.2-35**

Upon receive the 486 response from the S-S procedure, S-CSCF sends ACK.

Table 8.2.4.2-35: ACK (MO#2 to S-S)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

36. **Service Control**

S-CSCF performs whatever service control is appropriate for this failed session attempt.

37. **xxx Error (S-CSCF to P-CSCF) – see example in Table 8.2.4.2-37**

The S-CSCF returned a SIP error response to P-CSCF.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.4.2-37: 486 Busy Here (S-CSCF to P-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

38. **ACK (P-CSCF to S-CSCF) – see example in Table 8.2.4.2-38**

Upon receive the 486 response from the S-CSCF procedure, P-CSCF sends ACK.

Table 8.2.4.2-38: ACK (P-CSCF to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

39. **Revoke QoS authorization**

    P-CSCF removes the QoS authorization, if any, for this session.

40. **xxx Error (P-CSCF to UE) – see example in Table 8.2.4.2-40**

    The P-CSCF returned a SIP error response to UE.

    NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.4.2-40: 486 Busy Here (P-CSCF to UE)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

41. **ACK (P-CSCF to S-CSCF) – see example in Table 8.2.4.2-41**

    Upon receive the 486 response from the P-CSCF, UE sends ACK.

Table 8.2.4.2-41: ACK (UE to P-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.2.4.3   Session abandoned, or resource failure

The roaming subscriber that initiated a session with procedure MO#2 either abandoned the attempt, or was unable to obtain the resources necessary for the session. The session flow diagram for this error handling is shown in Figure 8.2.4.3-1.

If the session is aborted due to failure to obtain resources, it will occur at step #18 in the session flow; steps 19-33 (marked as optional) will not be present. If the session is abandoned due to user command, it can happen at any point between steps 8-33.

Home Network
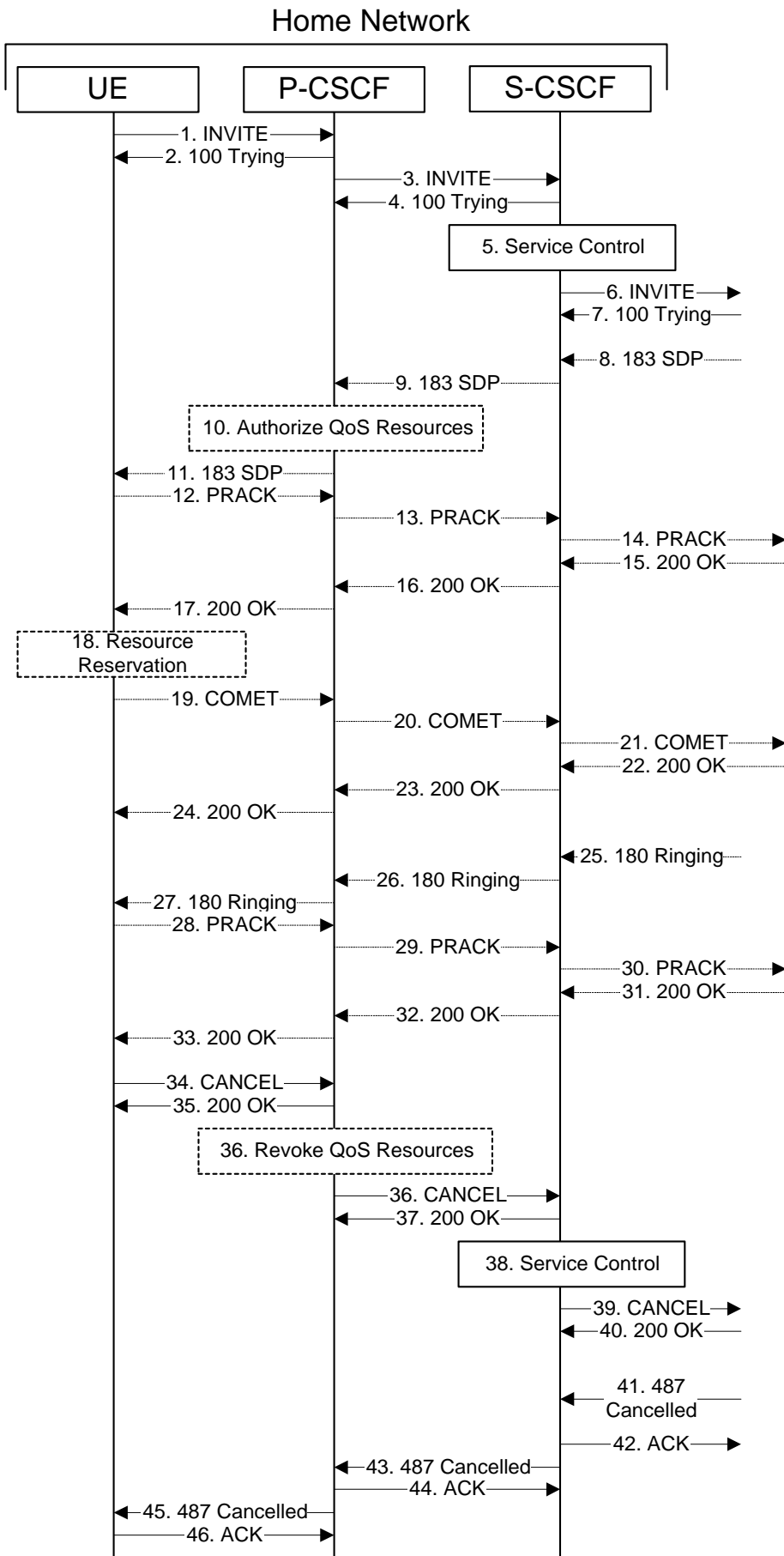
Figure 8.2.4.3-1: Session abandoned or resource failure

1-7. **INVITE (UE to P-CSCF) et seq**

UE#1 initiated a session, as described in Subclause 8.2.4.1.

8-33. **183 SDP (S-S to MO#2) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.2.4.1.

34. **Cancel (UE to P-CSCF) – see example in Table 8.2.4.3-34**

The UE cancelled the original INVITE request.

Table 8.2.4.3-34: CANCEL (UE to P-CSCF)

```
CANCEL sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length:0
```

35. **200 OK (P-CSCF to UE) – see example in Table 8.2.4.3-35**

Upon receive the Cancel request from the UE, P-CSCF sends 200 OK.

Table 8.2.4.3-35: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

36. **Revoke QoS authorization**

P-CSCF removes the QoS authorization, if any, for this session.

37. **Cancel (P-CSCF to S-CSCF) – see example in Table 8.2.4.3-37**

The P-CSCF forwards the Cancel request to S-CSCF.

Table 8.2.4.3-37: Cancel (P-CSCF to S-CSCF)

```
CANCEL sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-length:
```

Editor's Note: Need to check whether P-CSCF should modify the Contact: header in the CANCEL.

Editor's Note: Need to check whether P-CSCF should insert a Route: header in the CANCEL

38. **200 OK (S-CSCF to P-CSCF) – see example in Table 8.2.4.3-38**

Upon receiving the Cancel request from the P-CSCF, S-CSCF sends 200 OK.

Table 8.2.4.3-38: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

39. **Service Control**

   S-CSCF performs whatever service control is appropriate for this failed session attempt.

40. **Cancel (S-CSCF to S-S) – see example in Table 8.2.4.3-40**

   The S-CSCF forwards the Cancel request to the appropriate Serving-CSCF-to-Serving-CSCF procedure.

Table 8.2.4.3-40: Cancel (S-CSCF to S-S)

```
CANCEL sip:+1-212-555-2222@home1.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

Editor's Note: Need to check whether a proxy inserts a Record-Route in a Cancel request.

41. **200 OK (S-S to S-CSCF) – see example in Table 8.2.4.3-41**

   Upon receive the Cancel request from the S-CSCF, the next hop (whatever it is) sends 200 OK.

Table 8.2.4.3-41: 200 OK (S-S to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

42. **487 Cancelled (S-S to MO#2) – see example in Table 8.2.4.3-41**

   The termination procedure cancelled the request, and returned a SIP error response to the original INVITE request.

Table 8.2.4.3-41: 487 Cancelled (S-S to MO#2)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Call-ID:
CSeq:
Content-length:0
```

43. **ACK (MO#2 to S-S) – see example in Table 8.2.4.3-43**

   Upon receive the 487 response from the S-S procedure, S-CSCF sends ACK.

Table 8.2.4.3-43: ACK (MO#2 to S-S)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

44. **487 Cancelled (S-CSCF to P-CSCF) – see example in Table 8.2.4.3-44**

   The S-CSCF returned the SIP error response to P-CSCF.

Table 8.2.4.3-44: 487 Cancelled (S-CSCF to P-CSCF)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
```

45. **ACK (P-CSCF to S-CSCF) – see example in Table 8.2.4.3-45**

   Upon receive the 487 response from the S-CSCF, P-CSCF sends ACK.

Table 8.2.4.3-45: ACK (P-CSCF to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

46. **487 Cancelled (P-CSCF to UE) – see example in Table 8.2.4.3-46**

   The P-CSCF returned a SIP error response to UE.

Table 8.2.4.3-46: 487 Cancelled (P-CSCF to UE)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

47. **ACK (UE to P-CSCF) – see example in Table 8.2.4.3-47**

   Upon receive the 487 response from the P-CSCF, UE sends ACK.

Table 8.2.4.3-47: ACK (UE to P-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.2.5 (PSTN-O) PSTN origination

8.2.5.2 Failure in termination procedure

The PSTN subscriber that initiated a session with procedure PSTN-O had the attempt fail due to an error detected in the Termination procedure or in the Serving-CSCF-to-Serving-CSCF procedure. This could be due to, for example, destination busy (error code 486), destination service denied (error code 403), destination currently out of coverage (error code 480), or some other error.

Depending on the exact error that causes the session initiation failure, and when the error situation was detected, the originator could be at many different stages in the session establishment procedure. This is shown in Figure 8.2.5.2-1, below, as optional messages 5-17 that may appear in this error procedure.

Home Network
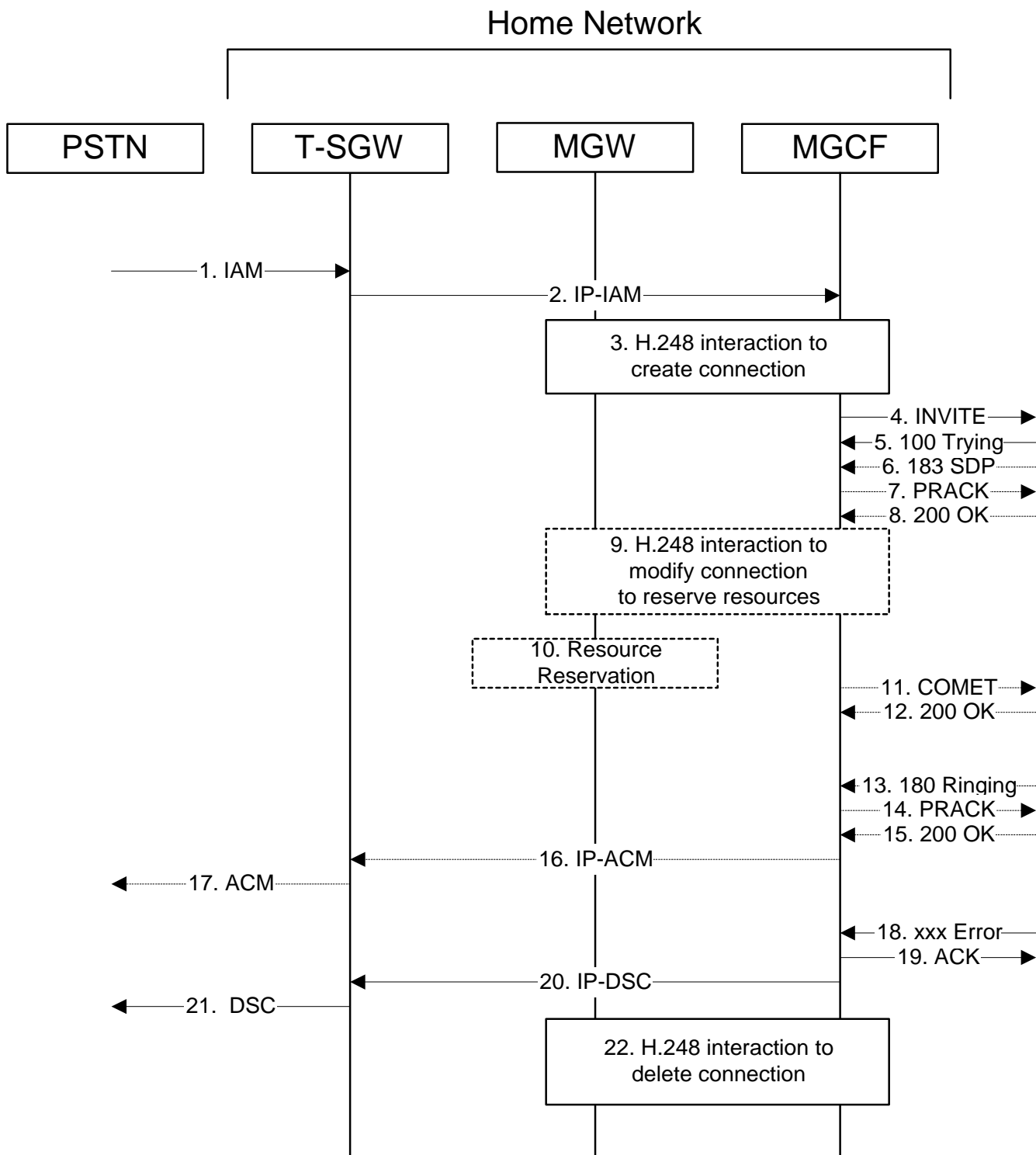
Figure 8.2.5.2-1: Failure in termination procedure

4. **INVITE (MGCF to S-S) et seq**

   The PSTN originator initiated a session, as described in Subclause 8.2.5.1.

5-17. **100 Trying (S-S to MO#1a) et seq**

   Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.2.5.1.

18. **xxx Error (S-S to PSTN-O) – see example in Table 8.2.5.2-18**

   The termination procedure detected some error situation, and returned a SIP error response.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily
Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.2.5.2-18: 486 Busy Here (S-S to PSTN-O)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP mgcf21.home.net
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Content-length:0
Retry-After:3600
```

19. **ACK (MO#1a to S-S) – see example in Table 8.2.5.2-19**

    Upon receive the 486 response from the S-S procedure, S-CSCF sends ACK.

Table 8.2.5.2-19: ACK (PSTN-O to S-S)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP mgcf21.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

20. **H.248 Interaction**

    MGCF initiates a H.248 interaction with MGW to delete the connection

8.2.5.3   Session abandoned, or resource failure

The PSTN subscriber that initiated a session with procedure PSTN-O either abandoned the attempt, or was unable to
obtain the resources necessary for the session. The session flow diagram for this error handling is shown in Figure
8.2.5.3-1.

If the session is aborted due to failure to obtain resources, it will occur at step #10 in the session flow; steps 11-17
(marked as optional) will not be present. If the session is abandoned due to user command, it can happen at any point
between steps 5-17.

Home Network
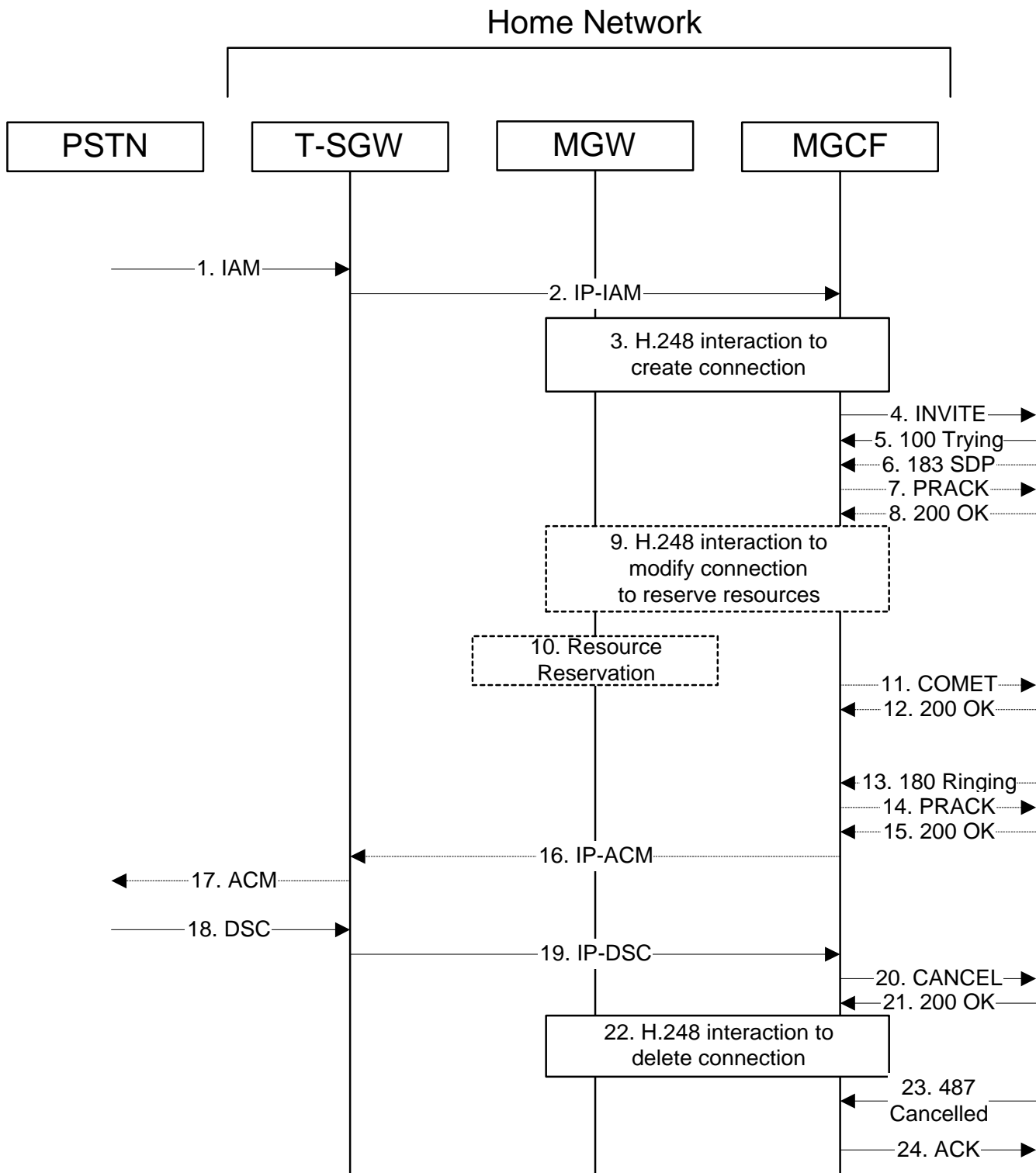
Figure 8.2.5.3-1: Session abandoned or resource failure

4. **INVITE (UE to S-S) et seq**

PSTN-O initiated a session, as described in Subclause 8.2.5.1.

5-15. **183 SDP (S-S to MO#1a) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.2.5.1.

20. **Cancel (UE to P-CSCF) – see example in Table 8.2.5.3-20**

   The PSTN cancelled the original INVITE request.

Table 8.2.5.3-20: CANCEL (PSTN-O to S-S)

```
CANCEL sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP mgcf21.home.net
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:017/4@mgcf21.home.net
Content-length:0
```

21. **200 OK (S-S to PSTN-O) – see example in Table 8.2.5.3-21**

   Upon receive the Cancel request from PSTN-O, the S-S procedure sends 200 OK.

Table 8.2.5.3-21: 200 OK (S-S to PSTN-O)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP mgcf21.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

22. **H.248 Interaction**

   MGCF initiates a H.248 interaction with MGW to delete the connection

23. **487 Error (S-S to PSTN-O) – see example in Table 8.2.5.3-23**

   The termination procedure processed the Cancel request, and returned a SIP error response.

Table 8.2.5.3-23: 487 Cancelled (S-S to PSTN-O)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP mgcf21.home.net
From:
To:
Contact:
Call-ID:
Cseq:
Content-length:0
Retry-After:3600
```

24. **ACK (PSTN-O to S-S) – see example in Table 8.2.5.3-24**

   Upon receive the 487 response from the S-S procedure, MGCF sends ACK.

Table 8.2.5.3-24: ACK (PSTN-O to S-S)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP mgcf21.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.3 Termination procedures

8.3.2 (MT#1a) Mobile termination, roaming, without I-CSCF in home network providing configuration independence

8.3.2.2 UE-detected failure/resource failure

The subscriber that initiated a session with one of the MO procedures had the attempt fail due to an error detected in the Termination procedure. This could be due to, for example, destination busy (error code 486), or some other error.

Depending on the exact error that causes the session initiation failure, and when the error situation was detected, MT#1a could be at many different stages in the session establishment procedure. This is shown in Figure 8.3.2.2-1, below, as optional messages 7-33 that may appear in this error procedure.

This clause also includes the procedures for the terminating UE to indicate a failure to allocate required resources for the session. This is detected in step #18 and reported with a 580-Precondition-Failure error response.

Figure 8.3.2.2-1: Failure in termination procedure

1-6. **INVITE (S-S to S-CSCF) et seq**

UE#1 initiated a session, as described in Subclause 8.3.2.1.

7-33. **100 Trying (S-CSCF to S-S) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.3.2.1.

34. **xxx Error (UE to P-CSCF) – see example in Table 8.3.2.2-34**

The termination procedure detected some error situation, and returned a SIP error response.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", "580 Precondition Failure", or others. For this example, "486 Busy" is shown.

Table 8.3.2.2-34: 486 Busy Here (UE to P-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:[5555::eee:fff:aaa:bbb]
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Content-length:0
Retry-After:3600
```

35. **ACK (P-CSCF to UE) – see example in Table 8.3.2.2-35**

Upon receive the 486 response from the UE, P-CSCF sends ACK.

Table 8.3.2.2-35: ACK (P-CSCF to UE)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

36. **Revoke QoS authorization**

P-CSCF removes the QoS authorization, if any, for this session.

37. **xxx Error (P-CSCF to S-CSCF) – see example in Table 8.3.2.2-37**

The P-CSCF returned a SIP error response to S-CSCF.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.3.2.2-37: 486 Busy Here (P-CSCF to S-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
   SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Content-length:0
Retry-After:3600
```

38. **ACK (S-CSCF to P-CSCF) – see example in Table 8.3.2.2-38**

Upon receive the 486 response from the P-CSCF procedure, S-CSCF sends ACK.

Table 8.3.2.2-38: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

39. **Service Control**

    S-CSCF performs whatever service control is appropriate for this failed session attempt.

40. **xxx Error (S-CSCF to S-S) – see example in Table 8.3.2.2-40**

    The S-CSCF returned a SIP error response to the appropriate S-S procedure.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.3.2.2-40: 486 Busy Here (S-CSCF to S-S)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

41. **ACK (S-S to S-CSCF) – see example in Table 8.3.2.2-41**

    Upon receive the 486 response from the S-CSCF, the S-S procedure sends ACK.

Table 8.3.2.2-41: ACK (S-S to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.3.2.3   Origination failure

After sending the initial INVITE for a multi-media session, the originating endpoint either abandoned the attempt or was unable to obtain the resources necessary for the session. The termination procedure is informed of this by a CANCEL request from the originator, which is shown in Figure 8.3.2.3-1.

If the session is aborted due to failure to obtain resources by the originator, it will occur prior to step #19; steps 19-33 (marked as optional) will not be present. If the session is abandoned due to user command, it can happen at any point between steps 8-33.
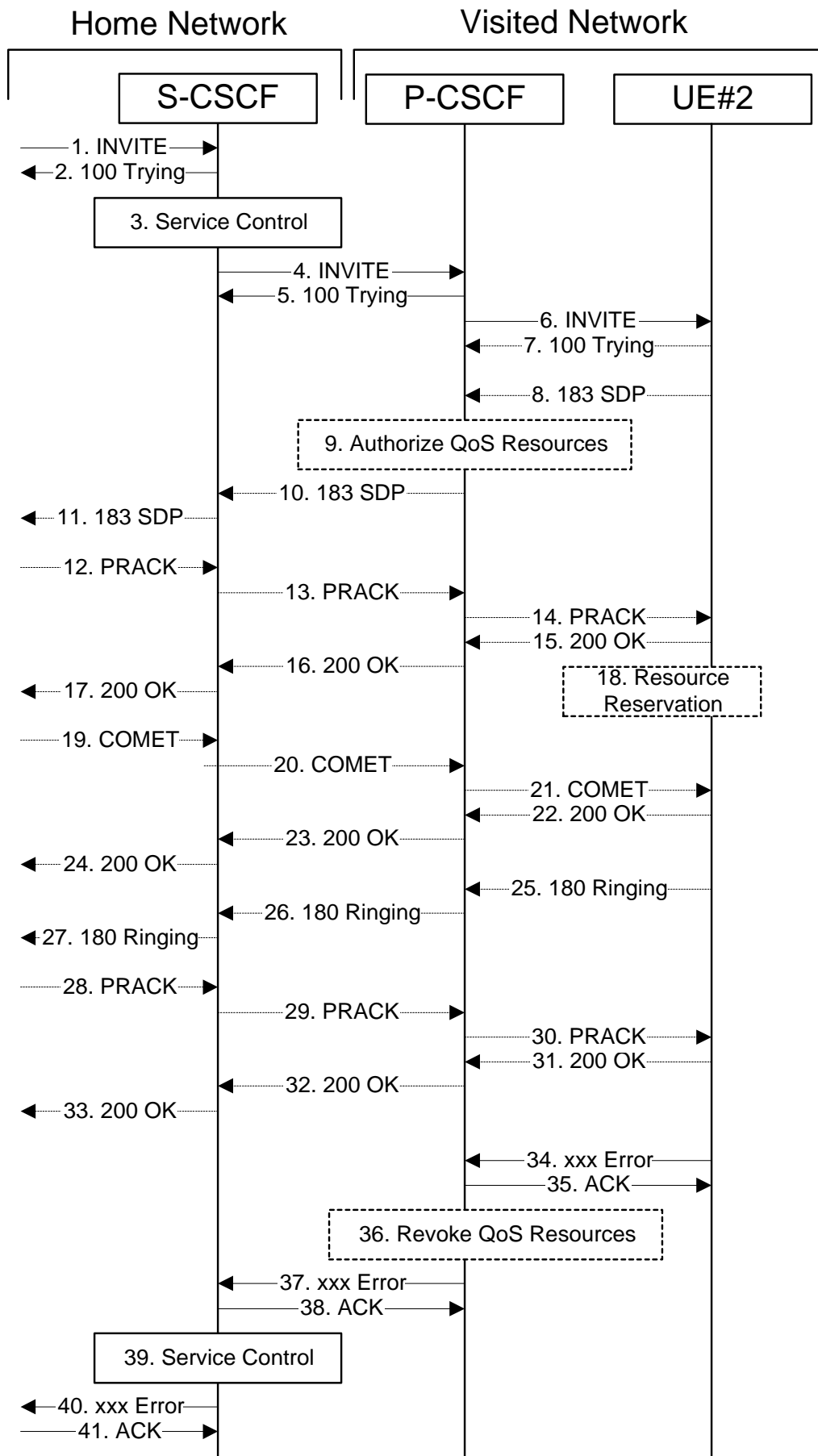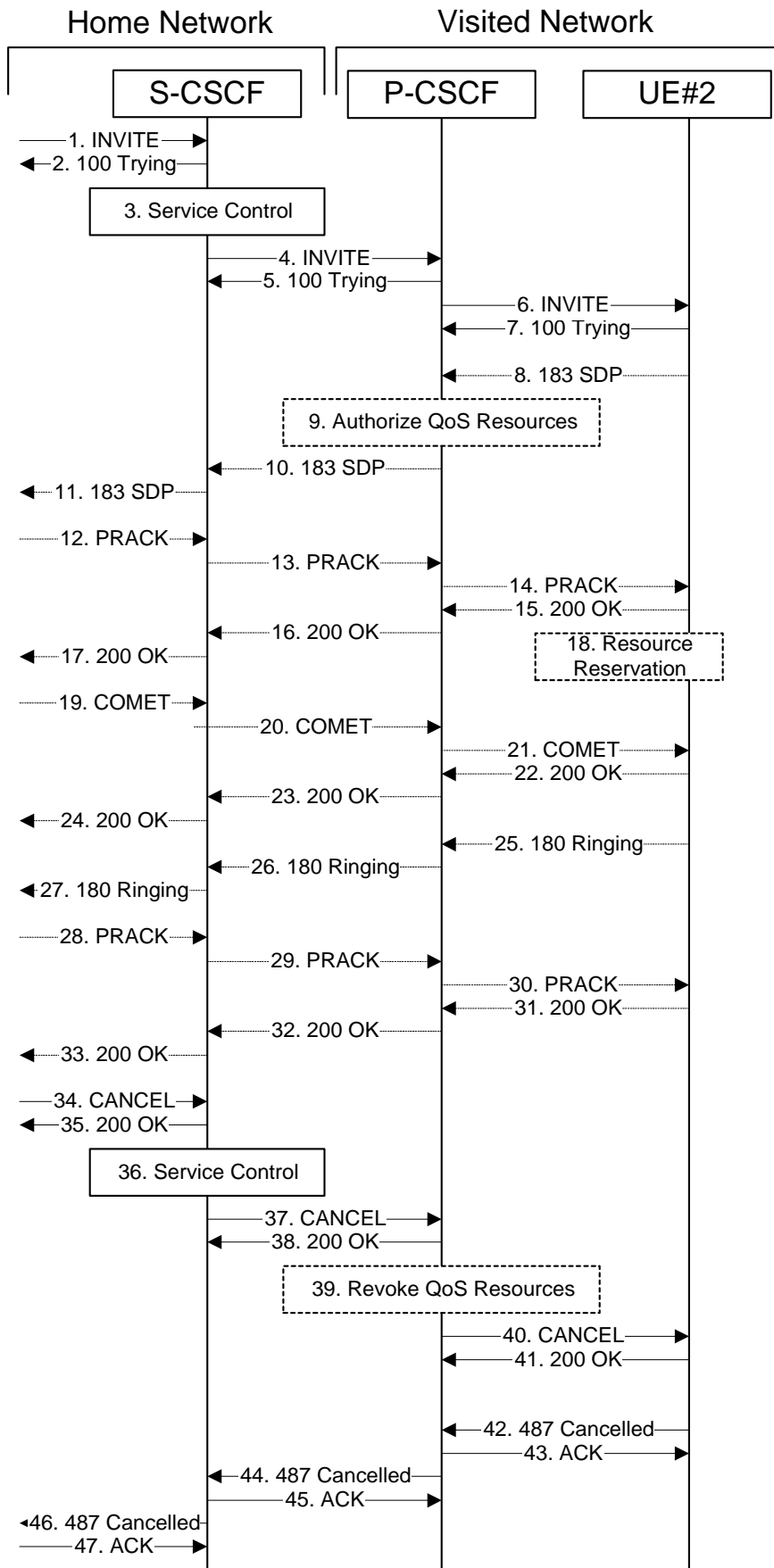
Figure 8.3.2.3-1: Failure in origination procedure

1-7. **INVITE (S-S to S-CSCF) et seq**

UE#1 initiated a session, as described in Subclause 8.3.2.1.

8-33. **183 SDP (UE to S-CSCF) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.3.2.1.

34. **Cancel (S-S to S-CSCF) – see example in Table 8.3.2.3-34**

The originator, through the S-S procedure, cancelled the original INVITE request.

Table 8.3.2.3-34: CANCEL (S-S to S-CSCF)

```
CANCEL sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length:0
```

35. **200 OK (S-CSCF to S-S) – see example in Table 8.3.2.3-35**

Upon receive the Cancel request from the S-S procedure, S-CSCF sends 200 OK.

Table 8.3.2.3-35: 200 OK (S-CSCF to S-S)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

36. **Service Control**

S-CSCF performs whatever service control is appropriate for this failed session attempt.

37. **Cancel (S-CSCF to P-CSCF) – see example in Table 8.3.2.3-37**

The S-CSCF forwards the Cancel request to P-CSCF.

Table 8.3.2.3-37: Cancel (S-CSCF to P-CSCF)

```
CANCEL sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-length:
```

38. **200 OK (P-CSCF to S-CSCF) – see example in Table 8.3.2.3-38**

Upon receiving the Cancel request from the S-CSCF, P-CSCF sends 200 OK.

Table 8.3.2.3-38: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

39. **Revoke QoS authorization**

P-CSCF removes the QoS authorization, if any, for this session.

40. **Cancel (P-CSCF to UE) – see example in Table 8.3.2.3-40**

The P-CSCF forwards the Cancel request to the UE.

Table 8.3.2.3-40: Cancel (P-CSCF to UE)

```
CANCEL sip:+1-212-555-2222@home1.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

41. **200 OK (UE to P-CSCF) – see example in Table 8.3.2.3-41**

Upon receive the Cancel request from the P-CSCF, the UE sends 200 OK.

Table 8.3.2.3-41: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
From:
To:
Call-ID:
CSeq:
Content-length:0
```

42. **487 Cancelled (UE to P-CSCF) – see example in Table 8.3.2.3-42**

The termination procedure performed the cancel operation, and returned a SIP error response to the initial INVITE request.

Table 8.3.2.3-34: 487 Cancelled (UE to P-CSCF)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:[5555::eee:fff:aaa:bbb]
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Content-length:0
Retry-After:3600
```

43. **ACK (P-CSCF to UE) – see example in Table 8.3.2.3-43**

Upon receive the 487 response from the UE, P-CSCF sends ACK.

Table 8.3.2.3-43: ACK (P-CSCF to UE)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

44. **487 Cancelled (P-CSCF to S-CSCF) – see example in Table 8.3.2.3-44**

The P-CSCF returned a SIP error response to S-CSCF.

Table 8.3.2.3-44: 487 Cancelled (P-CSCF to S-CSCF)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Content-length:0
Retry-After:3600
```

45. **ACK (S-CSCF to P-CSCF) – see example in Table 8.3.2.3-45**

Upon receive the 487 response from the P-CSCF procedure, S-CSCF sends ACK.

Table 8.3.2.3-45: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

46. **xxx Error (S-CSCF to S-S) – see example in Table 8.3.2.3-46**

The S-CSCF returned a SIP error response to the appropriate S-S procedure.

Table 8.3.2.3-40: 487 Cancelled (S-CSCF to S-S)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

47. **ACK (S-S to S-CSCF) – see example in Table 8.3.2.3-47**

Upon receive the 487 response from the S-CSCF, the S-S procedure sends ACK.

Table 8.3.2.3-47: ACK (S-S to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.3.2.4 Service failure

This termination 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. During registration, the home network allocates a S-CSCF in the home network, 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.

Figure 8.3.8 – MT#2a

Procedure MT#2a is as follows:

1. **INVITE (S-S to MT#2a) – see example in Table 8.3.8-1**

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

   NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.8-1: INVITE (S-S to MT#2A)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (MT#2a to S-S) – see example in Table 8.3.8-2**

   S-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.3.8-2: 100 Trying (MT#2A to S-S)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. **Service Control**

   S-CSCF validates the service profile, and performs any termination service control required for this subscriber.

   S-CSCF examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

4. **INVITE (S-CSCF to P-CSCF) – see example in Table 8.3.8-4**

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

Table 8.3.8-4: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Route: built from the registration information, (pcscf and UE contact name), followed by the initial Request-URI from the incoming INVITE request. The first component of the Route header, pcscf, is moved to the Request-URI of the request.

Via:, Record-Route: S-CSCF adds itself in the Record-Route and Via headers.

5. **100 Trying (P-CSCF to S-CSCF) – see example in Table 8.3.8-5**

   P-CSCF responds to the INVITE request (4) with a 100 Trying provisional response.

Table 8.3.8-5: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **INVITE (P-CSCF to UE) – see example in Table 8.3.8-6**

   P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.8-6: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
Supported:
Remote-Party-ID:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.home.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf.home.net, sip:scscf0.home.net,
    sip:%5b5555%3a%3aaaaa%3abbbb%3accc%3addd%5d@pcscf0.home.net
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

7. **486 Busy Here (UE to P-CSCF) – see example in Table 8.3.8-7**

   UE is contacted successfully but it is currently not willing or able to take additional sessions. The response MAY indicate a better time to session in the Retry-After header.

Table 8.3.8-7: 486 Busy Here (UE to P-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:[5555::eee:fff:aaa:bbb]
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

8. **ACK (P-CSCF to UE) – see example in Table 8.3.8-8**

   Upon receive the 486 response from the UE, P-CSCF sends ACK back to the UE.

Table 8.3.8-8: ACK (P-CSCF to UE)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

9. **486 Busy Here (P-CSCF to S-CSCF) – see example in Table 8.3.8-9**

P-CSCF forwards the 486 response to the S-CSCF.

Table 8.3.8-9: 486 Busy Here (P-CSCF to S-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=4321
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

10. **ACK (S-CSCF to P-CSCF) – see example in Table 8.3.8-10**

S-CSCF sends ACK to the P-CSCF.

Table 8.3.8-10: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

11. **Service Control**

The S-CSCF validates the service profile and performs any service control required for this subscriber.

12. **486 Busy Here (MT#2a to S-S) – see example in Table 8.3.8-12**

S-CSCF forwards the 486 BusyHere response to the originator, per the S-CSCF to S-CSCF procedure.

Table 8.3.8-12: 486 Busy Here (MT#2a to S-S)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=9999
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

13. **ACK (S-S to MT#2a) – see example in Table 8.3.8-13**

The S-CSCF of calling party responds to the 486 Busy Here response with an ACK request that is sent to S-CSCF via the S-CSCF to S-CSCF procedure.

Table 8.3.8-13: ACK (S-S to MT#1c)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.3.3 (MT#1b) Mobile termination, roaming, with I-CSCF in home network providing configuration independence

8.3.3.2 UE-detected failure/resource failure

8.3.3.3 Origination failure

8.3.3.4 Service failure

Failure of the session establishment due to service restrictions, as determined by S-CSCF, is identical to the MT#1a case. Refer to subclause 8.3.2.4 for the procedures.

8.3.4 (MT#2) Mobile termination, located in home network

8.3.4.2 UE-detected failure/resource failure

8.3.4.3 Origination failure

8.3.4.4 Service failure

Failure of the session establishment due to service restrictions, as determined by S-CSCF, is identical to the MT#1a case. Refer to subclause 8.3.2.4 for the procedures.

8.3.5 (PSTN-T) PSTN termination

8.3.5.2 MGCF-detected failure/resource failure

8.3.5.3 Origination failure

Editor's Note: It is not clear how the following text (8.3.9) should be included in the previous re-organization of 8.3. Should this text replace 8.3.2.2 above or should it be placed in a new 8.3.2.4 subclause?

8.3.9 Mobile termination, roaming, without I-CSCF in home network providing configuration independence, terminal is out of radio coverage

This termination procedure applies to roaming subscribers when the home network operator does not desire to keep its internal configuration hidden from the visited network. The UE is located in a visited network, and determines the P-CSCF via the CSCF discovery procedure. During registration, the home network allocates the S-CSCF. The terminal is out of the radio coverage when caller starts the session setup. Two assumptions are made here: caller is automated servers not directly controlled by a human being; there is no mechanism to inform the CSCF about the failure of the radio link.
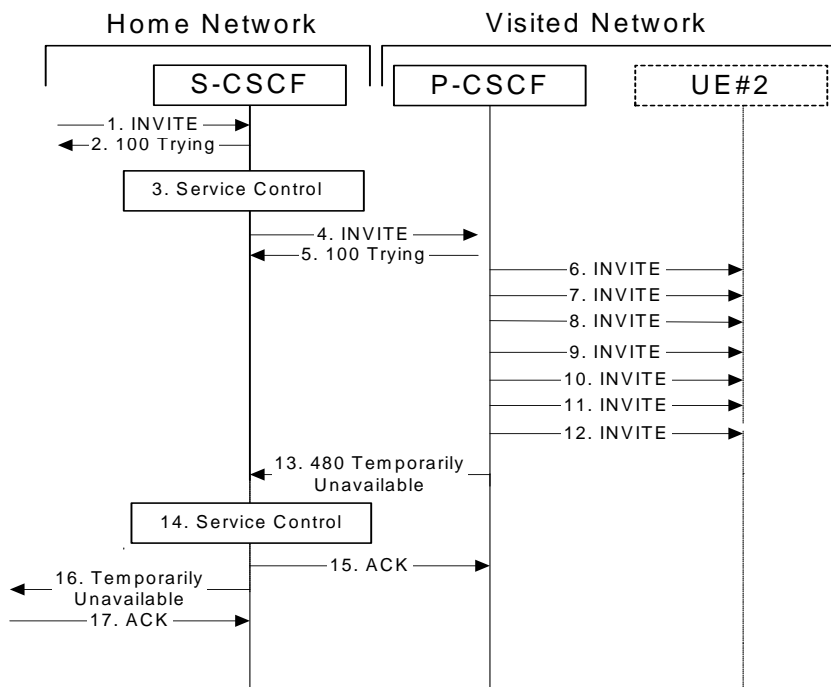
Figure 8.3.9-1: Mobile termination, roaming, without I-CSCF in home network providing configuration independence, terminal is out of radio coverage

1. **INVITE (S-S to MT#1a) – see example in Table 8.3.9-1**

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

NOTE: There are a number of different S-CSCF to S-CSCF flows, and the table represents a typical example of what one of these flows may produce. In this case, S-S#2 and MO#2 are assumed.

Table 8.3.9-1: INVITE (S-S to MT#1a)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf0.home.net
Supported: 100rel
Remote-Party-ID: John Doe <tel:+1-212-555-1111>
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

2. **100 Trying (MT#1a to S-S) – see example in Table 8.3.9-2**

    S-CSCF responds to the INVITE request (1) with a 100 Trying provisional response.

Table 8.3.9-2: 100 Trying (MT#1a to S-S)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

3. Service Control

S-CSCF validates the service profile, and performs any termination service control required for this subscriber

4. INVITE (S-CSCF to P-CSCF) – see example in Table 8.3.9-4

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

Table 8.3.9-4: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf.home.net, sip:scscf0.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Route: built from the registration information, (pcscf and UE contact name), followed by the initial Request-URI from the incoming INVITE request. The first component of the Route header, pcscf, is moved to the Request-URI of the request.

Via:, Record-Route: S-CSCF adds itself

5. 100 Trying (P-CSCF to S-CSCF) – see example in Table 8.3.9-5

P-CSCF responds to the INVITE request (4) with a 100 Trying provisional response.

Table 8.3.9-5: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

6. **INVITE (P-CSCF to UE) – see example in Table 8.3.9-6**

P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.9-6: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Media-Authorization: 31S14621
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.visited.net
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the Route header is:

```
Route: sip:scscf.home.net, sip:scscf0.home.net,
    sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token.

7. **Retransmission INVITE (P-CSCF to UE)**

8. **Retransmission INVITE (P-CSCF to UE)**

9. **Retransmission INVITE (P-CSCF to UE)**

10. **Retransmission INVITE (P-CSCF to UE)**

11. **Retransmission INVITE (P-CSCF to UE)**

12. **Retransmission INVITE (P-CSCF to UE)**

13. **480 Temporarily Unavailable (P-CSCF to S-CSCF)**

    P-CSCF forwards the 480 Temporarily Unavailable final reponse to the S-CSCF.

Table 8.3.9-13: 480 Temporarily Unavailable (P-CSCF to S-CSCF)

```
SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=4321
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.visited.net
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

14. **Service Control**

    S-CSCF validates the service profile, and performs any termination service control required for this subscriber

15. **ACK (S-CSCF to P-CSCF)**

    S-CSCF sends ACK to the P-CSCF.

Table 8.3.9-15: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

16. **480 Temporarily Unavailable (S-CSCF to S-S)**

    S-CSCF forwards the 480 reponse to the S-S procedure.

Table 8.3.9-16: 480 Temporarily Unavailable (S-CSCF to S-S)

```
SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf.visited.net
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

17. **ACK (S-S to S-CSCF)**

    The S-CSCF of calling party responds to the 480 Temporarily Unavailable response with an ACK request that is sent to S-CSCF via the S-CSCF to S-CSCF procedure.

Table 8.3.9-17: ACK (S-S to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.4 Serving-CSCF (MGCF) to Serving-CSCF (MGCF) Procedures

8.4.2 (S-S#1a) Different network operators performing origination and termination, without firewall between networks

8.4.2.2 Termination failure

The subscriber that originated a session with one of the MO procedures had the attempt fail due to an error detected in the termination procedure. This could be due to, for example, destination busy (error code 486), resource failure (error code 580), or some other error.

Depending on the exact error that causes the session initiation failure, and when the error situation was detected, the Serving-CSCF-to-Serving-CSCF procedure could be at many different stages in the session establishment procedure. This is shown in Figure 8.4.2.2-1, below, as optional messages 12-38 that may appear in this error procedure.
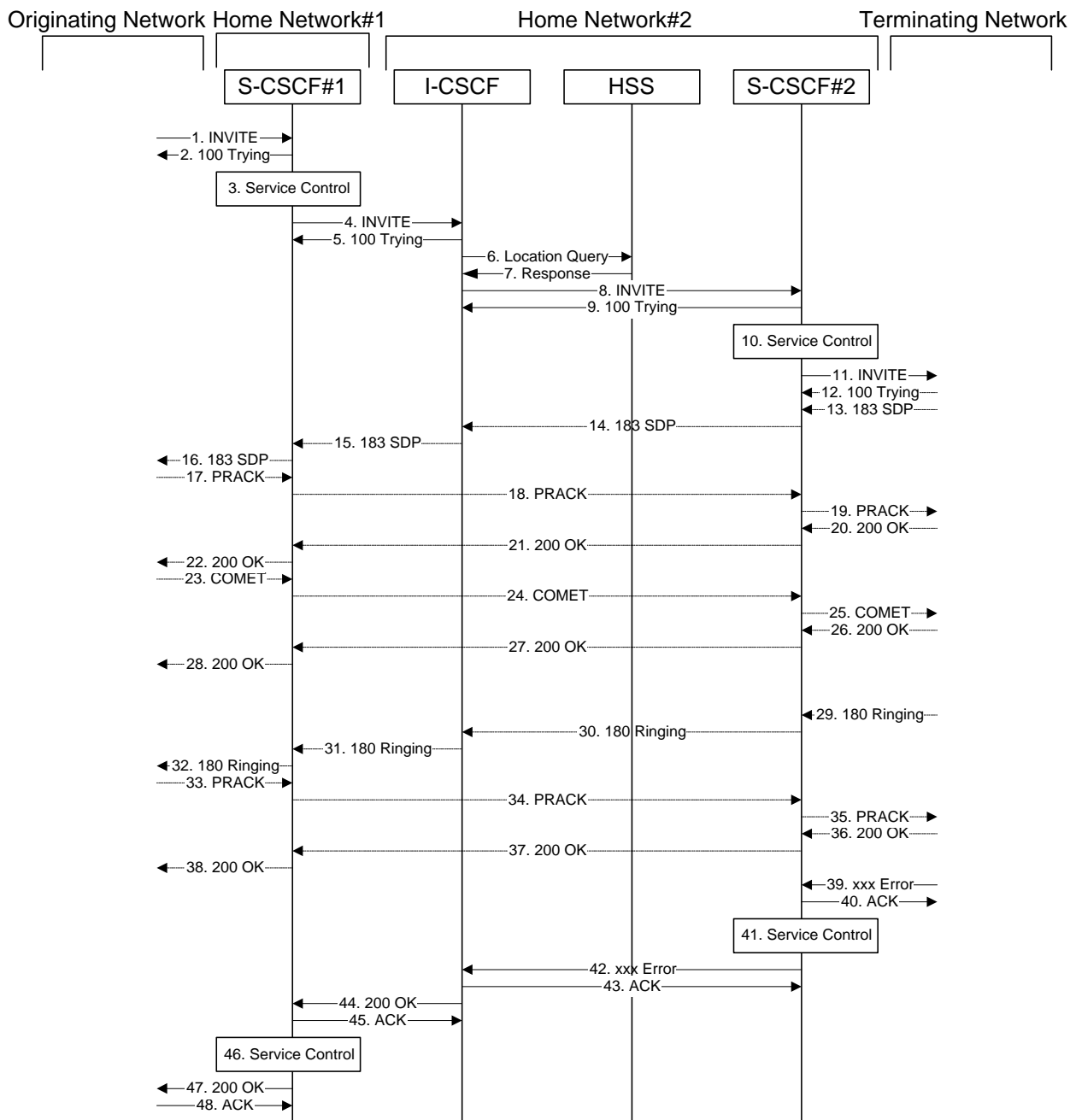
Figure 8.4.2.2-1: Failure in termination procedure

1-11. **INVITE (MO to S-CSCF) et seq**

A subscriber of the originating network initiated a session, as described in Subclause 8.4.2.1.

12-38. **100 Trying (MT to S-CSCF) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.4.2.1.

39. **xxx Error (MT to S-CSCF) – see example in Table 8.4.2.2-39**

The termination procedure detected some error situation, and returned a SIP error response.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", "580 Precondition Failure", or others. For this example, "486 Busy" is shown.

Table 8.4.2.2-39: 486 Busy Here (MT to S-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:[5555::eee:fff:aaa:bbb]
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Content-length:0
Retry-After:3600
```

40. **ACK (S-CSCF to MT) – see example in Table 8.4.2.2-40**

Upon receive the 486 response from the MT procedure, S-CSCF sends ACK.

Table 8.4.2.2-40: ACK (S-CSCF to MT)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

41. **Service Control**

S-CSCF performs whatever service control is appropriate for this failed session attempt.

42. **xxx Error (S-CSCF to I-CSCF) – see example in Table 8.4.2.2-42**

The S-CSCF returned a SIP error response to I-CSCF.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.4.2.2-42: 486 Busy Here (S-CSCF to I-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Contact: sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Content-length:0
Retry-After:3600
```

43. **ACK (I-CSCF to S-CSCF) – see example in Table 8.4.2.2-43**

Upon receive the 486 response from the S-CSCF procedure, I-CSCF sends ACK.

Table 8.4.2.2-43: ACK (S-CSCF to P-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

44. **xxx Error (I-CSCF to S-CSCF) – see example in Table 8.4.2.2-44**

The I-CSCF returned a SIP error response to S-CSCF.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.4.2.2-44: 486 Busy Here (I-CSCF to S-CSCF)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Content-length:0
Retry-After:3600
```

45. **ACK (S-CSCF to I-CSCF) – see example in Table 8.4.2.2-45**

Upon receive the 486 response from the S-CSCF procedure, I-CSCF sends ACK.

Table 8.4.2.2-45: ACK (S-CSCF to I-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

46. **Service Control**

S-CSCF performs whatever service control is appropriate for this failed session attempt.

47. **xxx Error (S-CSCF to MO) – see example in Table 8.4.2.2-47**

The S-CSCF returned a SIP error response to the appropriate MO procedure.

NOTE: The error response may be, for example, "486 Busy", "403 Service Denied", "480 Temporarily Unavailable", or others. For this example, "486 Busy" is shown.

Table 8.4.2.2-47: 486 Busy Here (S-CSCF to MO)

```
SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

48. **ACK (MO to S-CSCF) – see example in Table 8.4.2.2-48**

Upon receiving the 486 response from the S-CSCF, the MO procedure sends ACK.

Table 8.4.2.2-48: ACK (MO to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.4.2.3   Origination failure

The subscriber that initiated a session with one of the MO procedures either abandoned the attempt, or was unable to obtain the resources necessary for the session. The session flow diagram for this error handling is shown in Figure 8.4.2.3-1.

If the session is aborted due to failure to obtain resources, it will occur at step #23 in the session flow; steps 23-38 (marked as optional) will not be present. If the session is abandoned due to user command, it can happen at any point between steps 13-38.

Originating Network Home Network#1 Home Network#2 Terminating Network

S-CSCF#1 I-CSCF HSS S-CSCF#2

1. INVITE
2. 100 Trying
3. Service Control
4. INVITE
5. 100 Trying
6. Location Query
7. Response
8. INVITE
9. 100 Trying
10. Service Control
11. INVITE
12. 100 Trying
13. 183 SDP
14. 183 SDP
15. 183 SDP
16. 183 SDP
17. PRACK
18. PRACK
19. PRACK
20. 200 OK
21. 200 OK
22. 200 OK
23. COMET
24. COMET
25. COMET
26. 200 OK
27. 200 OK
28. 200 OK
29. 180 Ringing
30. 180 Ringing
31. 180 Ringing
32. 180 Ringing
33. PRACK
34. PRACK
35. PRACK
36. 200 OK
37. 200 OK
38. 200 OK
39. CANCEL
40. 200 OK
41. Service Control
42. CANCEL
43. 200 OK
44. CANCEL
45. 200 OK
46. Service Control
47. CANCEL
48. 200 OK
49. 487 Cancelled
50. ACK
51. 487 Cancelled
52. ACK
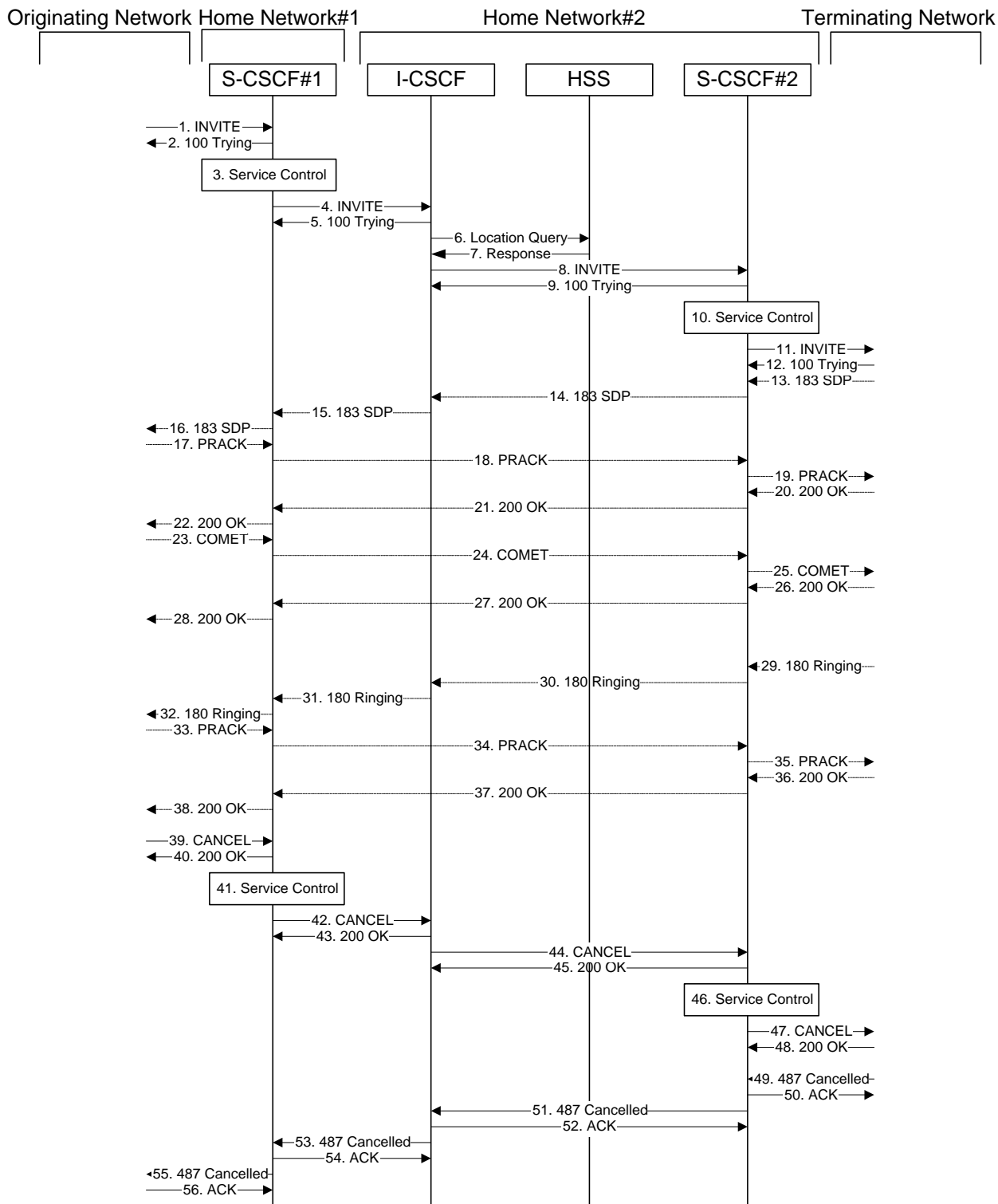53. 487 Cancelled
54. ACK
55. 487 Cancelled
56. ACK

Figure 8.4.2.3-1: Failure in origination procedure

1-12. **INVITE (MO to S-CSCF) et seq**

UE#1 initiated a session, as described in Subclause 8.4.2.1.

13-38. **183 SDP (MT to S-CSCF) et seq**

Session initiation possibly continued, prior to detection of a failure condition, as described in Subclause 8.4.2.1.

39. **Cancel (MO to S-CSCF) – see example in Table 8.4.2.3-39**

The originator, through the MO procedure, cancelled the original INVITE request.

Table 8.4.2.3-39: CANCEL (MO to S-CSCF)

```
CANCEL sip:scscf.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home2.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length:0
```

Editor's Note: Use of Route header in CANCEL is FFS.

40. **200 OK (S-CSCF to S-S) – see example in Table 8.4.2.3-40**

Upon receive the Cancel request from the MO procedure, S-CSCF sends 200 OK.

Table 8.4.2.3-40: 200 OK (S-CSCF to MO)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

41. **Service Control**

S-CSCF performs whatever service control is appropriate for this failed session attempt.

42. **Cancel (S-CSCF to I-CSCF) – see example in Table 8.4.2.3-42**

The S-CSCF forwards the Cancel request to I-CSCF.

Table 8.4.2.3-42: Cancel (S-CSCF to I-CSCF)

```
CANCEL sip:+1-212-555-2222@home2.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-length:
```

43. **200 OK (I-CSCF to S-CSCF) – see example in Table 8.4.2.3-43**

Upon receiving the Cancel request from the S-CSCF, P-CSCF sends 200 OK.

Table 8.4.2.3-43: 200 OK (I-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

44. **Cancel (I-CSCF to S-CSCF) – see example in Table 8.4.2.3-44**

The I-CSCF forwards the Cancel request to S-CSCF.

Table 8.4.2.3-44: Cancel (I-CSCF to S-CSCF)

```
CANCEL sip:scscf.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-length:
```

Editor's Note: Use of Route header in CANCEL is FFS.

45. **200 OK (S-CSCF to I-CSCF) – see example in Table 8.4.2.3-45**

Upon receiving the Cancel request from the I-CSCF, S-CSCF sends 200 OK.

Table 8.4.2.3-45: 200 OK (S-CSCF to I-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf.home.net, SIP/2.0/UDP scscf0.home.net, SIP/2.0/UDP pcscf0.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length:0
```

46. **Service Control**

S-CSCF performs whatever service control is appropriate for this failed session attempt.

47. **Cancel (S-CSCF to MT) – see example in Table 8.4.2.3-47**

The P-CSCF forwards the Cancel request to the appropriate MT procedure.

Table 8.4.2.3-47: Cancel (S-CSCF to MT)

```
CANCEL sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.home2.net SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Route: sip:+1-212-555-2222@home2.net;user=phone
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

Editor's Note: Use of Route header in CANCEL is FFS.

48. **200 OK (MT to S-CSCF) – see example in Table 8.4.2.3-48**

Upon receive the Cancel request from the S-CSCF, the MT procedure sends 200 OK.

Table 8.4.2.3-48: 200 OK (MT to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
From:
To:
Call-ID:
CSeq:
Content-length:0
```

49. **487 Cancelled (MT to S-CSCF) – see example in Table 8.4.2.3-49**

The termination procedure detected some error situation, and returned a SIP error response.

Table 8.4.2.3-49: 487 Cancelled (MT to S-CSCF)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP scscf.home2.net, SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net,
    SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>; tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost; tag=1234
Contact: sip:[5555::eee:fff:aaa:bbb]
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Content-length:0
Retry-After:3600
```

50. **ACK (S-CSCF to MT) – see example in Table 8.4.2.3-50**

Upon receive the 487 response from the MT procedure, S-CSCF sends ACK.

Table 8.4.2.3-50: ACK (S-CSCF to MT)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

51. **487 Cancelled (S-CSCF to I-CSCF) – see example in Table 8.4.2.3-51**

The S-CSCF returned a SIP error response to I-CSCF.

Table 8.4.2.3-51: 487 Cancelled (S-CSCF to I-CSCF)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP icscf.home2.net, SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Content-length:0
Retry-After:3600
```

52. **ACK (I-CSCF to S-CSCF) – see example in Table 8.4.2.3-52**

Upon receive the 487 response from the S-CSCF procedure, I-CSCF sends ACK.

Table 8.4.2.3-52: ACK (I-CSCF to S-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

53. **487 Cancelled (I-CSCF to S-CSCF) – see example in Table 8.4.2.3-53**

The I-CSCF returned a SIP error response to S-CSCF.

Table 8.4.2.3-53: 487 Cancelled (I-CSCF to S-CSCF)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP scscf.home1.net, SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
Content-length:0
Retry-After:3600
```

54. **ACK (S-CSCF to I-CSCF) – see example in Table 8.4.2.3-54**

Upon receive the 487 response from the S-CSCF procedure, I-CSCF sends ACK.

Table 8.4.2.3-54: ACK (S-CSCF to I-CSCF)

```
ACK sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net SIP/2.0
Via: SIP/2.0/UDP scscf.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

55. **487 Cancelled (S-CSCF to MO) – see example in Table 8.4.2.3-55**

The S-CSCF returned a SIP error response to the appropriate MO procedure.

Table 8.4.2.3-55: 487 Cancelled (S-CSCF to MO)

```
SIP/2.0 487 Cancelled
Via: SIP/2.0/UDP pcscf.home1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
From:
To:
Contact:
Call-ID:
CSeq:
Content-length:0
Retry-After:3600
```

56. **ACK (MO to S-CSCF) – see example in Table 8.4.2.3-56**

Upon receive the 487 response from the S-CSCF, the MO procedure sends ACK.

Table 8.4.2.3-56: ACK (MO to S-CSCF)

```
ACK sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscf0.home.net
From:
To:
Call-ID:
CSeq:
Content-length:0
```

8.4.3 (S-S#1b) Different network operators performing origination and termination, with configuration hiding

8.4.3.2 Termination failure

8.4.3.3 Origination failure

8.4.4 (S-S#2) Single network operator performing origination and termination

8.4.4.2 Termination failure

8.4.4.3 Origination failure

8.4.5 (S-S#3) PSTN termination performed by home network of originator

8.4.5.2 Termination failure

8.4.5.3 Origination failure

8.4.6 (S-S#4) PSTN termination performed by different operator than origination

8.4.6.2 Termination failure

8.4.6.3 Origination failure

Editor's Note: The following is modified text regarding the storage of state at the P-CSCF and S-CSCF. This text applies to the following: subclause 8.2.2 in steps 3, 5, 11, 35; subclause 8.2.3 in steps 3, 7, 14, 45; subclause 8.2.4 steps 3, 5, 11, 35; subclause 8.3.2 in steps 3, 6; subclause 8.3.3 in steps 3, 8; and subclause 8.3.4 in steps 3, 6.

8.2.2 (MO#1a) Mobile origination, roaming, without I-CSCF in home network providing configuration independence

3. **INVITE (P-CSCF to S-CSCF) – see example in Table 8.2.2-3**

P-CSCF remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. This next hop is the S-CSCF within the home network.

P-CSCF rewrites the Contact header, with a locally defined value that identifies the UE. P-CSCF adds itself to the Record-Route header, and adds a Via header.

P-CSCF#1 examines the media parameters, and removes any choices that the network operator decides based on local policy, not to allow on the network.

For this example, assume the network operator disallows H261 video encoding.

Editor's Note: Need to describe the procedures for a P-CSCF/S-CSCF to follow when it doesn't understand a media line in the SDP. It is clear that it should not remove such lines, as that destroys the ability to create new services; however the CSCF needs sufficient information about the unknown media stream to perform the authorization.

P-CSCF stores the following information about this session, for use in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B;
 seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 127 INVITE
Contact(original): sip:[5555::aaa:bbb:ccc:ddd]
Contact(modified): sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
```

The INVITE request is forwarded to the S-CSCF.

Table 8.2.2-3: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv

a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Request-URI: is the first component in the remembered Path header from Registration.

Route: contains the remaining elements from the Path header from Registration, with the initial Request-URI (received from the UE) appended as the final component.

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the video media streams no longer list code 98 (H261).

5. **Service Control**

S-CSCF validates the service profile, and performs any origination service control required for this subscriber.

S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

For this example, assume the subscriber is not allowed video.

S-CSCF stores the following information about this session, for use in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
```

11. **183 Session Progress (P-CSCF to UE) – see example in Table 8.2.2-11**

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the information for this session is:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 127 INVITE
Contact(original): sip:[5555::aaa:bbb:ccc:ddd]
Contact(modified): sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Route: sip:scscf.home.net, sip:scscf2.home.net,
 sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
```

P-CSCF forwards the 183 Session Progress response to the originating endpoint.

Table 8.2.2-11: 183 Session Progress (P-CSCF to UE)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Media-Authorization: 0020000100100101706366312e78797a2e6e6574000c02013942563330373200
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token1@pcscf1.visited.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=a=
a=
a=
a=
m=
```

Contact: a locally unique token to identify the saved routing information.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf1.xyz.net" with credentials "9BV3072". "00" at the end of the authorixation token is required to pad to a multiple of 4 bytes.

35. **Service Control**

S-CSCF performs whatever service control is appropriate for the completed call

S-CSCF stores the following information about this session, for use in providing enhanced services or in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq(2dest): 127 INVITE
CSeq(2orig): none
Route(2dest): sip:scscf2.home.net,sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Route(2orig): sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
```

8.2.3   (MO#1b) Mobile origination, roaming, with I-CSCF in home network providing configuration independence

3. **INVITE (P-CSCF to I-CSCF) – see example in Table 8.2.3-3**

P-CSCF#1 examines the media parameters, and removes any choices that the network operator decides based on local policy, not to allow on the network.

For this example, assume the network operator disallows H261 video encoding.

P-CSCF stores the following information about this session, for use in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
```

```
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 127 INVITE
Contact(original): sip:[5555::aaa:bbb:ccc:ddd]
Contact(modified): sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
```

The INVITE request is forwarded through this I-CSCF to the S-CSCF.

Table 8.2.3-3: INVITE (P-CSCF to I-CSCF)

```
INVITE sip:icscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(scscf.home.net), sip:+1-212-555-2222@home.net;user=phone
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecva=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Route: P-CSCF remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. The initial Request-URI is appended to the Route header.

Contact: a locally defined value that identifies the UE.

Request-URI: the first component in the remembered Path header from Registration.

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the video media streams no longer list code 98 (H261).

7. **Service Control**

S-CSCF validates the service profile, and performs any origination service control required for this subscriber.

S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

For this example, assume the subscriber is not allowed video.

S-CSCF stores the following information about this session, for use in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
```

14. **183 Session Progress (P-CSCF to UE) – see example in Table 8.2.3-14**

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the information for this session is:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 127 INVITE
Contact(original): sip:[5555::aaa:bbb:ccc:ddd]
Contact(modified): sip:%5b5555%3a%3aaaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
Route: sip:icscf.home.net,
    sip:Token(sip:scscf.home.net, sip:scscf2.home.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
```

P-CSCF forwards the 183 Session Progress response to the originating endpoint

Table 8.2.3-14: 183 Session Progress (P-CSCF to UE)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Media-Authorization: 00200001001001017063663312e78797a2e6e6574000c02013942563330373200
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token1@pcscf1.visited.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
m=
```

Contact: a locally unique token to identify the saved routing information.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf1.xyz.net" with credentials "9BV3072". "00" at the end of the authorixation token is required to pad to a multiple of 4 bytes.

45. **Service Control**

S-CSCF performs whatever service control is appropriate for the completed call

S-CSCF stores the following information about this session, for use in providing enhanced services or in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq(2dest): 127 INVITE
CSeq(2orig): none
Route(2dest): sip:scscf2.home.net,sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Route(2orig): sip:icscf.home.net,
sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited.net
```

8.2.4 (MO#2) Mobile origination, located in home network

3. **INVITE (P-CSCF to S-CSCF) – see example in Table 8.2.4-3**

P-CSCF remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. This next hop is the S-CSCF within the home network.

P-CSCF rewrites the Contact header, with a locally defined value that identifies the UE. P-CSCF adds itself to the Record-Route header, and adds a Via header.

P-CSCF#1 examines the media parameters, and removes any choices that the network operator decides based on local policy, not to allow on the network.

For this example, assume the network operator disallows H261 video encoding.

P-CSCF stores the following information about this session, for use in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 127 INVITE
Contact(original): sip:[5555::aaa:bbb:ccc:ddd]
Contact(modified): sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
```

The INVITE request is forwarded to the S-CSCF.

Table 8.2.4-3: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home1.net;user=phone
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=video 3400 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=video 3402 RTP/AVP 99
a=qos:mandatory sendrecv
a=rtpmap:99:MPV
m=audio 3456 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 3458 RTP/AVP 97 96 0 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
```

Request-URI: the first component in the remembered Path header from Registration.

Route: contains the remaining elements from the Path header from Registration, with the initial Request-URI (received from the UE) appended as the final component.

SDP   The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the video media streams no longer list code 98 (H261).

5. **Service Control**

    S-CSCF validates the service profile, and performs any origination service control required for this subscriber.

    S-CSCF#1 examines the media parameters, and removes any choices that the subscriber does not have authority to request.

    For this example, assume the subscriber is not allowed video.

    S-CSCF stores the following information about this session, for use in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
```

11. **183 Session Progress (P-CSCF to UE) – see example in Table 8.2.4-11**

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the information for this session is:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 127 INVITE
Contact(original): sip:[5555::aaa:bbb:ccc:ddd]
Contact(modified): sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Route: sip:scscf.home.net, sip:scscf2.home.net,
 sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
```

P-CSCF forwards the 183 Session Progress response to the originating endpoint.

Table 8.2.4-11: 183 Session Progress (P-CSCF to UE)

```
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Media-Authorization: 0020000100100101706366312e78797a2e6e6574000c02013942563330373200
Remote-Party-ID:
Anonymity:
Require:
From:
To:
Call-ID:
CSeq:
Contact: sip:token1@pcscf1.home.net
RSeq:
Content-Disposition:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
m=
m=
a=
a=
a=
a=
m=
```

Contact: a locally unique token to identify the saved routing information.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf1.xyz.net" with credentials "9BV3072". "00" at the end of the authorixation token is required to pad to a multiple of 4 bytes.

35. **Service Control**

S-CSCF performs whatever service control is appropriate for the completed call

S-CSCF stores the following information about this session, for use in providing enhanced services or in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq(2dest): 127 INVITE
CSeq(2orig): none
Route(2dest): sip:scscf2.home.net,sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Route(2orig): sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
```

8.3.2 (MT#1a) Mobile termination, roaming, without I-CSCF in home network providing configuration independence

3. **Service Control**

S-CSCF validates the service profile, and performs any termination service control required for this subscriber

S-CSCF examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

S-CSCF stores the following information about this session, for use in providing enhanced services or in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq(2dest): 127 INVITE
CSeq(2orig): none
Route(2orig): sip:scscf0.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Route(2dest): sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
```

6. **INVITE (P-CSCF to UE) – see example in Table 8.3.2-6**

P-CSCF examines the media parameters, and removes any that the network operator decides, based on local policy, not to allow on the network.

For this example, assume the network operator does not allow 64 kb/s audio, so the PCMU codec is removed.

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the information for this session is:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq(2dest): 127 INVITE
CSeq(2orig): none
Route(2dest): sip:[5555::eee:fff:aaa:bbb]
Route(2orig): sip:scscf.home.net, sip:scscf0.home.net,
sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.2-6: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.visited.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the first audio stream no longer contains codec "0" (PCMU), which removes it from the negotiation.

8.3.3 (MT#1b) Mobile termination, roaming, with I-CSCF in home network providing configuration independence

3. **Service Control**

S-CSCF validates the service profile, and performs any termination service control required for this subscriber

S-CSCF examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

S-CSCF stores the following information about this session, for use in providing enhanced services or in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq(2dest): 127 INVITE
CSeq(2orig): none
Route(2orig): sip:scscf0.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

```
Route(2dest): sip:icscf.home.net,
sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
```

8. **INVITE (P-CSCF to UE) – see example in Table 8.3.3-8**

P-CSCF examines the media parameters, and removes any that the network operator decides, based on local policy, not to allow on the network.

For this example, assume the network operator does not allow 64 kb/s audio, so the PCMU codec is removed.

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the information for this session is:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq(2dest): 127 INVITE
CSeq(2orig): none
Route(2dest): sip:[5555::eee:fff:aaa:bbb]
Route(2orig): sip:icscf.home.net,
sip:Token(sip:scscf.home.net, sip:scscf0.home.net),
sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.3-8: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.visited.net;branch=token1
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.visited.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the first audio stream no longer contains codec "0" (PCMU), which removes it from the negotiation.

8.3.4 (MT#2) Mobile termination, located in home network

3. **Service Control**

S-CSCF validates the service profile, and performs any termination service control required for this subscriber

S-CSCF examines the media parameters, and removes any choices that the destination subscriber does not have authority to request.

For this example, assume the destination subscriber is not allowed stereo, so only a single audio stream is permitted.

S-CSCF stores the following information about this session, for use in providing enhanced services or in possible error recovery actions:

```
Request-URI: sip:+1-212-555-2222@home.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq(2dest): 127 INVITE
CSeq(2orig): none
Route(2orig): sip:scscf0.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
Route(2dest): sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf.visited.net
```

6. **INVITE (P-CSCF to UE) – see example in Table 8.3.4-6**

P-CSCF examines the media parameters, and removes any that the network operator decides, based on local policy, not to allow on the network.

For this example, assume the network operator does not allow 64 kb/s audio, so the PCMU codec is removed.

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE. The saved value of the information for this session is:

```
Request-URI: sip:+1-212-555-2222@home1.net;user=phone
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;
tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq(2dest): 127 INVITE
CSeq(2orig): none
Route(2dest): sip:[5555::eee:fff:aaa:bbb]
Route(2orig): sip:scscf.home.net, sip:scscf0.home.net,
sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf0.home.net
```

P-CSCF determines the UE address from the value of the Request-URI (which was previously returned by P-CSCF as a contact header value in the registration procedure), and forwards the INVITE request to the UE.

Table 8.3.4-6: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-2222@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscf.home.net;branch=token1
Media-Authorization: 0020000100100101706366322e78797a2e6e6574000c020133315331343363231
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token1@pcscf.home.net
Content-Type:
Content-length:

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=video 0 RTP/AVP 99
m=video 0 RTP/AVP 99
m=audio 3456 RTP/AVP 97 96 15
a=qos:mandatory sendrecv
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
m=audio 0 RTP/AVP 97 96 0 15
```

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saves values. It inserts this as a branch value on its Via header.

Media-Authorization: a P-CSCF generated authorization token. This particular example shows a Policy-Element generated by "pcf2.xyz.net" with credentials "31S14621".

SDP The SDP contains the restricted set of codecs allowed by the network operator. The "m=" lines for the first audio stream no longer contains codec "0" (PCMU), which removes it from the negotiation.

Editor's Note: The following text proposes session release procedures to be placed in clause 9.

9 Signalling flows for session termination

9.1 Mobile terminal initiated session release

9.1.1 Mobile terminal initiated session release (without I-CSCF providing configuration independence)

Figure 9-1 shows a mobile terminal initiated IM CN subsystem application (SIP) session release. It is assumed that the session is active and that the bearer was established directly between the two visited networks (the visited networks could be the Home network in either or both cases).
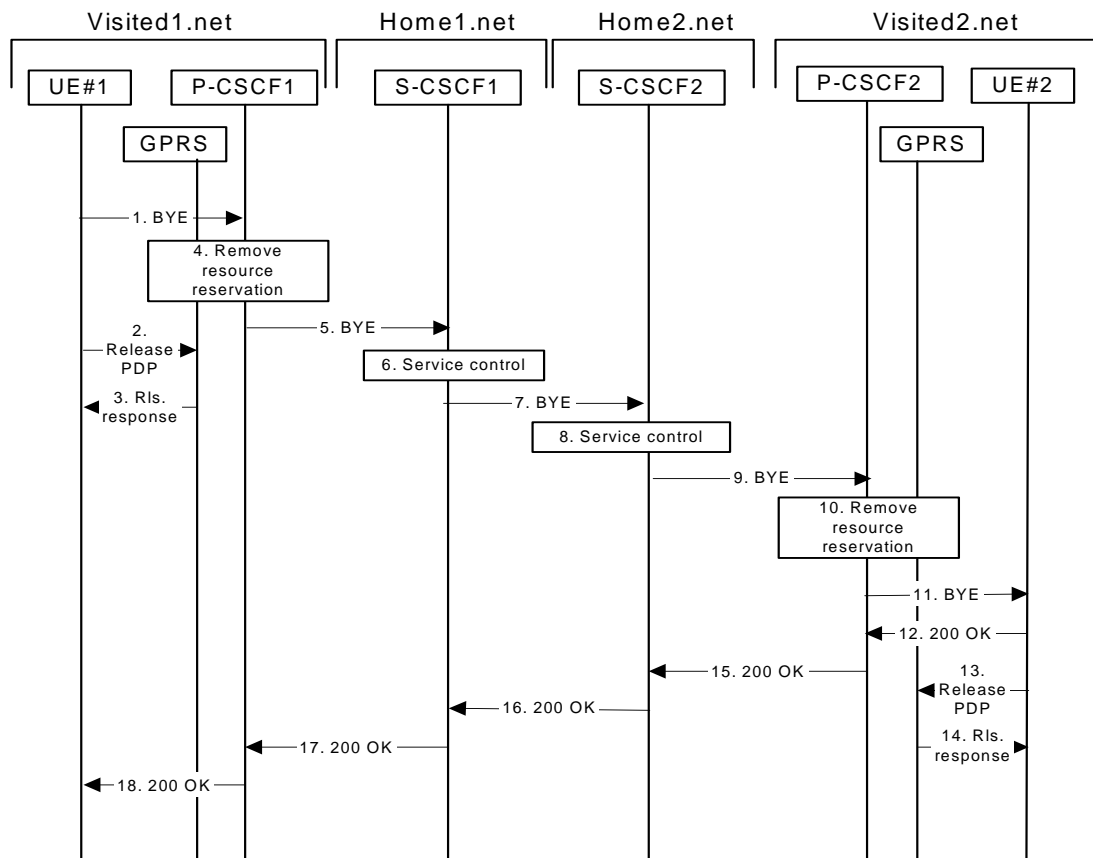
Figure 9-1: Mobile initiated session release

1 **SIP BYE (UE (P-CSCF) - see example in Table 9.1.1-1**

One mobile party hangs up, which generates a SIP BYE request from the UE to the P-CSCF.

Table 9.1.1-1: Example contents of SIP BYE request

```
BYE sip:token10@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 153 BYE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Length: 0
```

The request-URI takes the value of the Contact header of the previously received response.

The Via header and the contact header take the value of either the IP address or the FQDN of the originating UE.

The example contents of the From header, the To header and Call-ID header are used to identify the session being cleared, and therefore are identical to those of the previously received response for that session, so that they include any tag parameters.

The content of the Cseq header must have a higher sequence number than the previous transaction. Here it is assumed that a Cseq value no greater than 152 has been previously used.

2 **Release PDP**

Steps 2 and 3 may take place before or after Step 1 and in parallel with Step 4. The UE initiates the release of the bearer PDP context. The GPRS subsystem releases the PDP context. The IP network resources that had were reserved for the message receive path to the mobile for this session are now released. This is initiated from the GGSN. If RSVP was used to allocated resources, then the appropriate release messages for that protocol would invoked here.

3 **Rls. Response**

The GPRS subsystem responds to the UE.

4 **Remove resource reservation**

The P-CSCF removes the authorization for resources that had previously been issued for this endpoint for this session. This step will also result in a release indication to the GPRS subsystem to confirm that the IP bearers associated with the session have been deleted.

5 **SIP BYE (P-CSCF (S-CSCF) see example in Table 9.1.1-5**

The P-CSCF sends a SIP BYE request to the S-CSCF of the releasing party.

Table 9.1.1-5: Example contents of the SIP BYE request

```
BYE sip:scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home2.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length: 0
```

6 **Service control**

The S-CSCF performs whatever service control procedures are appropriate for this ending session.

7 **SIP BYE (S-CSCF (S-CSCF) see example in Table 9.1.1-7**

The SIP BYE request is sent from the S-CSCF to the S-CSCF of the network of the other party.

Table 9.1.1-7: Example contents of the SIP BYE request

```
BYE sip:scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home2.net
Record-Route: sip:scscf1.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length:
```

8 **Service control**

The S-CSCF performs whatever service control procedures are appropriate for this ending session.

9 **SIP BYE (S-CSCF (P-CSCF) - see example in Table 9.1.1-9**

The SIP BYE request is forwarded directly to the P-CSCF.

Table 9.1.1-9: Example contents of the SIP BYE request

```
BYE sip:pcscf2.visited2.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length: 0
```

10 **Remove resource reservation**

The P-CSCF removes the authorisation for resources that had previously been issued for this endpoint for this session. This step also results in a release indication to the GPRS subsystem to confirm that the IP bearers associated with the UE#2 session have been deleted.

11 **SIP BYE (P-CSCF (UE) - see example in Table 9.1.1-11**

The P-CSCF forwards the SIP BYE request on to the UE.

Table 9.1.1-11: Example contents of the SIP BYE request

```
BYE sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token25
From:
To:
Call-ID:
CSeq:
Contact: token25@pcscf2.visited2.net
Content-Length: 0
```

12 **SIP 200 OK (UE (P-CSCF) - see example in Table 9.1.1-12**

The mobile responds with a SIP 200 OK response, which is sent back to the P-CSCF.

Table 9.1.1-12: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token25
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

13 **Release PDP**

Steps 14 and 15 may be done in parallel with step 13. The Mobile initiates the release of the bearer PDP context.

14 **Rls response**

The GPRS subsystem releases the PDP context. The IP network resources that had were reserved for the message receive path to the mobile for this session are now released. This is initiated from the GGSN. If RSVP was used to allocated resources, then the appropriate release messages for that protocol would invoked here.

15 **SIP 200 OK (P-CSCF (S-CSCF) - see example in Table 9.1.1-15**

The P-CSCF sends a SIP 200 OK response to the S-CSCF directly.

Table 9.1.1-15: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

16 **SIP 200 OK (S-CSCF (S-CSCF) - see example in Table 9.1.1-16**

The S-CSCF of the other party forwards the SIP 200 OK response to its local S-CSCF.

Table 9.1.1-16: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

17 **SIP 200 OK (S-CSCF (P-CSCF) - see example in Table 9.1.1-17**

The S-CSCF of the releasing party forwards the SIP 200 OK response to the P-CSCF of the releasing party. The example contents of the SIP 200 OK response are shown in table 9-9.

Table 9-9: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

18 **SIP 200 OK (P-CSCF (UE) - see example in Table 9.1.1-18**

The P-CSCF of the releasing party forwards the SIP 200 OK response to the UE.

Table 9.1.1-18: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

9.1.2 Mobile terminal initiated session release (with I-CSCF providing configuration independence)

Figure 9-2 shows a mobile terminal initiated IM CN subsystem application (SIP) session release. It is assumed that the session is active and that the bearer was established directly between the two visited networks (the visited networks could be the home network in either or both cases).

NOTE: For the pupuses of the description of the I-CSCF in figure 9-2 and in the associated text, it is assumed that the party that established the session initiated the clearing. For clearing in the reverse direction, there is a slight change in the optionality of the I-CSCFs between the S-CSCFs. This is as described for session establishment.

Editor's Note: The figure in contribution N1-010884 (SIP Release Flows) could not be saved or copied due to a Word Error. This figure will be reproduced later.

Figure 9-2: Mobile initiated session release

1 **SIP BYE (UE (P-CSCF) - see example in Table 9.1.2-1**

One mobile party hangs up, which generates a SIP BYE request from the UE to the P-CSCF.

Table 9.1.2-1: Example contents of SIP BYE request

```
BYE sip:token10@pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>
To: <sip:B36(SHA-1(555-2222; time=36123E5B; seq=73))@localhost>;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
CSeq: 153 BYE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-Length: 0
```

The request-URI takes the value of the Contact header of the previously received response.

The Via header and the contact header take the value of either the IP address or the FQDN of the originating UE.

The example contents of the From header, the To header and Call-ID header are used to identify the session being cleared, and therefore are identical to those of the previously received response for that session, so that they include any tag parameters.

The content of the Cseq header must have a higher sequence number than the previous transaction. Here it is assumed that a Cseq value no greater than 152 has been previously used.

2 **Release PDP**

Steps 2 and 3 may take place before or after Step 1 and in parallel with Step 4. The UE initiates the release of the bearer PDP context. The GPRS subsystem releases the PDP context. The IP network resources that had were reserved for the message receive path to the mobile for this session are now released. This is initiated from the GGSN. If RSVP was used to allocated resources, then the appropriate release messages for that protocol would invoked here.

3 **Rls. Response**

The GPRS subsystem responds to the UE.

4 **Remove resource reservation**

The P-CSCF removes the authorization for resources that had previously been issued for this endpoint for this session. This step will also result in a release indication to the GPRS subsystem to confirm that the IP bearers associated with the session have been deleted.

5 **SIP BYE (P-CSCF - I-CSCF) - see example in Table 9.1.2-5**

The P-CSCF sends a SIP BYE request to the Firewall I-CSCF hiding the S-CSCF of the releasing party.

Table 9.1.2-5: Example contents of the SIP BYE request

```
BYE sip:icscf1a.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:Token(sip:scscf1.home1.net, sip:scscf2.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.visited1.net
Content-Length: 0
```

6   **SIP BYE (I-CSCF (S-CSCF) - see example in Table 9.1.2-6**

The Firewall I-CSCF sends a SIP BYE request to the S-CSCF of the releasing party.

Table 9.1.2-6: Example contents of the SIP BYE request

```
BYE sip:scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:icscf.home.net
Route: sip:scscf2.home2.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length: 0
```

7   **Service control**

The S-CSCF performs whatever service control procedures are appropriate for this ending session.

8   **SIP BYE (S-CSCF (I-CSCF) - see example in Table 9.1.2-8**

The SIP BYE request is sent from the S-CSCF to the Firewall I-CSCF.

Table 9.1.2-8: Example contents of the SIP BYE request

```
BYE sip:icscf1b.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:icscf2b.home2.net, sip:Token(sip:scscf2.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf1.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length: 0
```

9   **SIP BYE (I-CSCF (I-CSCF) - see example in Table 9.1.2-9**

The SIP BYE request is sent from the Firewall I-CSCF to the I-CSCF of the network of the other party.

Table 9.1.2-9: Example contents of the SIP BYE request

```
BYE sip:icscf2b.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:Token(sip:scscf2.home2.net),
    sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf1b.home1.net, sip:Token(sip:scscf1.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length: 0
```

10 **SIP BYE (I-CSCF (S-CSCF) - see example in Table 9.1.2-10**

The SIP BYE request is forwarded from the I-CSCF that was used to determine the location of S-CSCF of the other party to the S-CSCF of the the other party.

Table 9.1.2-10: Example contents of the SIP BYE request

```
BYE sip:scscf2.home2.net SIP/2.0
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:icscf2b.home2.net, sip:icscf1b.home1.net, sip:Token(sip:scscf1.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length: 0
```

11 **Service control**

The S-CSCF performs whatever service control procedures are appropriate for this ending session.

12 **SIP BYE (S-CSCF (I-CSCF) - see example in Table 9.1.2-12**

The SIP BYE request is forwarded to a Firewall I-CSCF.

Table 9.1.2-12: Example contents of the SIP BYE request

```
BYE sip:icscf2a.home2.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length: 0
```

13 **SIP BYE (I-CSCF (P-CSCF) - see example in Table 9.1.2-13**

The Firewall I-CSCF forwards the SIP BYE request to the P-CSCF.

Table 9.1.2-13: Example contents of the SIP BYE request

```
BYE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.visited2.net SIP/2.0
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2b.home2.net,
    SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]))
```

```
Record-Route: sip:icscf2a.home2.net, sip:Token(sip:scscf2.home2.net, sip:scscf1.home1.net)
From:
To:
Call-ID:
CSeq:
Contact:
Content-Length: 0
```

14 **Remove resource reservation**

The P-CSCF removes the authorisation for resources that had previously been issued for this endpoint for this session. This step also results in a release indication to the GPRS subsystem to confirm that the IP bearers associated with the UE#2 session have been deleted.

15 **SIP BYE (P-CSCF (UE) - see example in Table 9.1.2-15**

The P-CSCF forwards the SIP BYE request on to the UE.

Table 9.1.2-15: Example contents of the SIP BYE request

```
BYE sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token25
From:
To:
Call-ID:
CSeq:
Contact: token26@pcscf2.visited2.net
Content-Length: 0
```

16 **SIP 200 OK (UE (P-CSCF) - see example in Table 9.1.2-16**

The mobile responds with a SIP 200 OK response, which is sent back to the P-CSCF.

Table 9.1.2-16: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.visited2.net;branch=token26
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

17 **Release PDP**

Steps 14 and 15 may be done in parallel with step 13. The Mobile initiates the release of the bearer PDP context.

18 **Rls response**

The GPRS subsystem releases the PDP context. The IP network resources that had were reserved for the message receive path to the mobile for this session are now released. This is initiated from the GGSN. If RSVP was used to allocated resources, then the appropriate release messages for that protocol would invoked here.

19 **SIP 200 OK (P-CSCF (I-CSCF) - see example in Table 9.1.2-19**

The P-CSCF sends a SIP 200 OK response to the Firewall I-CSCF.

Table 9.1.2-19: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2a.home2.net, SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2b.home2.net,
    SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]))
Record-Route: sip:icscf2a.home2.net, sip:Token(sip:scscf2.home2.net, sip:scscf1.home1.net)
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

20 **SIP 200 OK (I-CSCF (S-CSCF) - see example in Table 9.1.2-20**

The Firewall I-CSCF sends a SIP 200 OK response to the S-CSCF.

Table 9.1.2-20: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home2.net, SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP icscf1b.home1.net,
    SIP/2.0/UDP Token(SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP
    pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf2a.home2.net, sip:scscf2.home2.net, sip:scscf1.home1.net
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

21 **SIP 200 OK (S-CSCF (I-CSCF) - see example in Table 9.1.2-21**

The S-CSCF of the other party forwards the SIP 200 OK response to its selecting I-CSCF.

Table 9.1.2-21: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf2b.home2.net, SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP
    scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:icscf2b.home2.net, sip:icscf1b.home1.net, sip:Token(sip:scscf1.home1.net)
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

22 **SIP 200 OK (I-CSCF (I-CSCF) - see example in Table 9.1.2-22**

The selecting I-CSCF forwards the SIP 200 OK response to the Firewall I-CSCF.

Table 9.1.2-22: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1b.home1.net, SIP/2.0/UDP Token(SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP
    icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd])
Record-Route: sip:Token(sip:icscf1b.home1.net), sip:scscf1.home1.net
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

23 **SIP 200 OK (I-CSCF (S-CSCF) - see example in Table 9.1.2-23**

The Firewall I-CSCF forwards the SIP 200 OK response to the S-CSCF.

Table 9.1.2-23: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home1.net, SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf2.home2.net), sip:icscf.home2.net, sip:icscf.home1.net,
sip:scscf.home1.net
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

24 **SIP 200 OK (S-CSCF (I-CSCF) - see example in Table 9.1.2-24**

The S-CSCF of the releasing party forwards the SIP 200 OK response to the Firewall I-CSCF.

Table 9.1.2-24: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP icscf1a.home1.net, SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home2.net, sip:scscf1.home1.net, sip:icscf1a.home1.net
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

25 **SIP 200 OK (I-CSCF (P-CSCF) - see example in Table 9.1.2-25**

The Firewall I-CSCF forwards the SIP 200 OK response to the P-CSCF of the releasing party.

Table 9.1.2-25: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:Token(sip:scscf1.home1.net, sip:scscf2.home2.net), sip:icscf2.home2.net
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

26 **SIP OK (P-CSCF to UE) - see example in Table 9.1.2-26**

The P-CSCF of the releasing party forwards the SIP 200 OK response to the UE.

Table 9.1.2-26: Example contents of the SIP 200 OK response

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

Editor's Note: The following text proposes session transfer procedures to be placed in clause 10.5.

10.5 Session transfer procedures

This clause gives information flows for the procedures for performing session transfers. Subclause 10.5.1 gives the procedures for a transfer that initiates a new session (i.e. to a new destination not previously involved in the session).

Subclause 10.5.2 gives the procedures for a transfer that replaces an existing session (i.e. to a destination that was previously involved in the session).

10.5.1 Session transfer initiating a new session

An IP multi-media session already exists between UE#1 and UE#2. UE#2 desires UE#1 to initiate a new session to a new destination, UE#F, and terminate the existing session. The procedures for this transfer are shown in Figure 10.5.1-1.
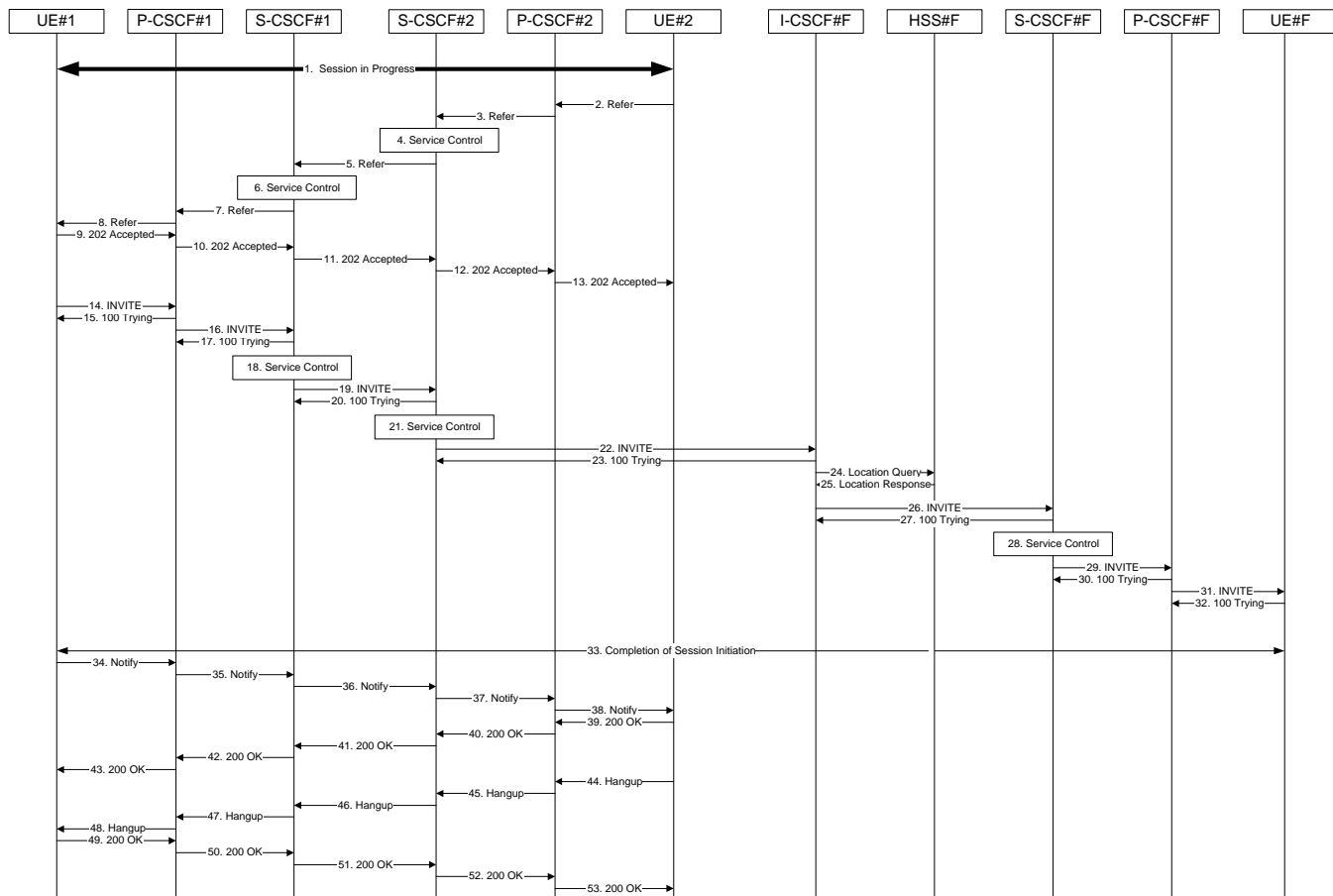
Figure 10.5.1-1 – Session transfer initiating a new session

1. **Session in Progress**

   UE#1 initiates a multi-media session with UE#2. As a result, the state information stored at P-CSCF#2 is shown in Table 10.5.1-1

Table 10.5.1-1: State Information

```
Request-URI: sip:token6@pcscf2.home.net

From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost

Route: sip:scscf2.home.net, sip:scscf1.home.net,
    sip:%5b5555%3a%3aaaaa%3abbbb%3accc%3addd%5d@pcscf1.home.net
```

2. **REFER (UE to P-CSCF) – see example in Table 10.5.1-2**

UE#2 sends a Refer request to its proxy, P-CSCF#2.

Table 10.5.1-2: REFER (UE to P-CSCF)

```
REFER sip:token6@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
To: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 REFER
Contact: sip:[5555::eee:fff:aaa:bbb]
Refer-To: tel:+1-212-555-3333
Refer-By: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Content-length: 0
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

Editor's Note: Use of Remote-Party-ID in REFER is FFS.

Editor's Note: The proper value for the Refer-By header is FFS.

3. **REFER (P-CSCF to S-CSCF) – see example in Table 10.5.1-3**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#2 forwards the Refer request to S-CSCF#2.

Table 10.5.1-3: REFER (P-CSCF to S-CSCF)

```
REFER sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:scscf1.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

4. **Service Control**

5. **REFER (S-CSCF to S-CSCF) – see example in Table 10.5.1-5**

In order to maintain the expectation of privacy of the identity of the new destination, S-CSCF#2 converts the "Refer-To" header into a private URL. S-CSCF#2 forwards the Refer request to S-CSCF#1.

Table 10.5.1-5: REFER (S-CSCF to S-CSCF)

```
REFER sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Refer-To: sip:token(tel:+1-212-555-3333)@scscf2.home.net;private
Refer-By:
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Content-length:
```

6. **Service Control**

7. **REFER (S-CSCF to P-CSCF) – see example in Table 10.5.1-7**

   S-CSCF#1 forwards the Refer request to P-CSCF#1.

Table 10.5.1-7: REFER (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UEP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
        SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

8. **REFER (P-CSCF to UE) – see example in Table 10.5.1-8**

   P-CSCF#1 forwards the Refer request to UE#1.

Table 10.5.1-8: REFER (P-CSCF to UE)

```
REFER sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

9. **202-Accepted (UE to P-CSCF) – see example in Table 10.5.1-9**

UE#2 acknowledges receipt of the Refer request (8) with a 202-Accepted final response, sent to P-CSCF#1.

Table 10.5.1-8: 202 Accepted (UE to P-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

10. **202-Accepted (P-CSCF to S-CSCF) – see example in Table 10.5.1-10**

P-CSCF#1 forwards the 202 Accepted final response to S-CSCF#1.

Table 10.5.1-10: 202 Accepted (P-CSCF to S-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UEP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

11. **202-Accepted (S-CSCF to S-CSCF) – see example in Table 10.5.1-11**

S-CSCF#1 forwards the 202 Accepted final response to S-CSCF#2.

Table 10.5.1-11: 202 Accepted (S-CSCF to S-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

12. **202-Accepted (S-CSCF to P-CSCF) – see example in Table 10.5.1-12**

S-CSCF#2 forwards the 202 Accepted final response to P-CSCF#2.

Table 10.5.1-12: 202 Accepted (S-CSCF to P-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

13. **202-Accepted (P-CSCF to UE) – see example in Table 10.5.1-13**

    P-CSCF#2 forwards the 202 Accepted final response to UE#2.

Table 10.5.1-13: 202 Accepted (P-CSCF to UE)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF removes the Record-Route header

14. **INVITE (UE to P-CSCF) – see example in Table 10.5.1-14**

    UE#1 initiates an INVITE request based on the Refer-To header URL in the REFER request. The INVITE is sent from the UE to P-CSCF#1.

Table 10.5.1-14: INVITE (UE to P-CSCF)

```
INVITE sip:token(tel:+1-212-555-3333)@scscf2.home.net;private SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=74))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=75))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Refer-By: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

15. **100 Trying (P-CSCF to UE) – see example in Table 10.5.1-15**

    P-CSCF#1 responds to the INVITE request (14) with a 100 Trying provisional response.

Table 10.5.1-15: 100 Trying (P-CSCF to UE)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

16. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.5.1-16**

    P-CSCF#1 remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. The next hop is the S-CSCF serving this UE. P-CSCF rewrites the Contact header with a locally defined value that identifies the UE. P-CSCF adds itself to the Via header.

Table 10.5.1-16: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(tel:+1-212-555-3333)@scscf2.home.net;private
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

17. **100 Trying (S-CSCF to P-CSCF) – see example in Table 10.5.1-17**

S-CSCF#1 responds to the INVITE request (16) with a 100 Trying provisional response.

Table 10.5.1-17: 100 Trying (S-CSCF to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

18. **Service Control**

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

19. **INVITE (S-CSCF to S-CSCF) – see example in Table 10.5.1-19**

S-CSCF#1 performs an analysis of the destination address, which is a private URL generated by S-CSCF#2. Since it is a destination within the same operator's network, S-CSCF#1 forwards the INVITE request directly to S-CSCF#2.

Table 10.5.1-19: INVITE (S-CSCF to S-CSCF)

```
INVITE sip:token(tel:+1-212-555-3333)@scscf2.home.net;private SIP/2.0
Via: SIP/2.0/UDP sip:scscf1.home.net SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

20. **100 Trying (S-CSCF to S-CSCF) – see example in Table 10.5.1-20**

    S-CSCF#2 responds to the INVITE request (19) by sending a 100 Trying provisional response to S-CSCF#1.

Table 10.5.1-20: 100 Trying (S-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

21. **Service Control**

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

22. **INVITE (S-CSCF to I-CSCF) – see example in Table 10.5.1-22**

    S-CSCF#2 determines the destination address from the private URL contained in the INVITE request. Based on information in that URL, and information saved from step #4 above (implementation decision), S-CSCF#2 verifies the validity of the transfer request, and that it is within a short time delay from the REFER request.

    S-CSCF#2 performs an analysis of the destination address, and determines the network operator to whom the destination subscriber belongs. Since (for this example) the forwarding network operator does not desire to keep their internal configuration hidden, S-CSCF#2 forwards the INVITE request directly to I-CSCF#F.

Table 10.5.1-22: INVITE (S-CSCF to I-CSCF)

```
INVITE sip:+1-212-555-3333@home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes;party=transferor
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Editor's Note: Use of "party=transferor" in a separate Remote-Party-ID header is FFS.

23. **100 Trying (I-CSCF to S-CSCF) – see example in Table 10.5.1-23**

    I-CSCF#F responds to the INVITE request (22) by sending a 100 Trying provisional response to S-CSCF#2.

Table 10.5.1-23: 100 Trying (I-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

24. **Location Query**

    I-CSCF (at the border of the terminating subscriber's network) queries the HSS for current location information. It will send "Cx-location-query" to the HSS to obtain the location information for the destination.

25. **Location Response**

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

26. **INVITE (I-CSCF to S-CSCF) – see example in Table 10.5.1-26**

    I-CSCF#F forwards the INVITE request to the S-CSCF (S-CSCF#F) that will handle the session termination.

Table 10.5.1-26: INVITE (I-CSCF to S-CSCF)

```
INVITE sip:scscff.home.net SIP/2.0
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:+1-212-555-2222@home.net;user=phone
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

NOTE: The I-CSCF does not add itself to the Record-Route header, as it has no need to remain in the signalling path once the session is established.

27. **100 Trying (S-CSCF to I-CSCF) – see example in Table 10.5.1-27**

S-CSCF#F responds to the INVITE request (26) with a 100 Trying provisional response.

Table 10.5.1-27: 100 Trying (S-CSCF to I-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net,
    SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

28. **Service Control**

S-CSCF#F performs whatever service control logic is appropriate for this session attempt.

29. **INVITE (S-CSCF to P-CSCF) – see example in Table 10.5.1-29**

S-CSCF#F remembers (from the registration procedure) the next hop CSCF for this UE. It forwards the INVITE request to P-CSCF#F.

Table 10.5.1-29: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:+1-212-555-3333@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP scscff.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net,
    SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

30. **100 Trying (P-CSCF to S-CSCF) – see example in Table 10.5.1-30**

P-CSCF#F responds to the INVITE request (29) by sending a 100 Trying provisional response to S-CSCF#F.

Table 10.5.1-12: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net,
    SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

31. **INVITE (P-CSCF to UE) – see example in Table 10.5.1-31**

P-CSCF forwards the INVITE request to the UE.

Table 10.5.1-31: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-3333@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscff.home.net;branch=token6
Supported:
Remote-Party-ID:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token6@pcscff.home.net
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

32. **100 Trying (UE to P-CSCF) – see example in Table 10.5.1-32**

UE#F may optionally send a 100 Trying provisional response to P-CSCF.

Table 10.5.1-32: 100 Trying (UE to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net;branch=token6
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

33. **Completion of Session Initiation**

UE#1 and UE#F complete the session initiation, as shown in the MO, S-S, and MT procedures.

34. **NOTIFY (UE to P-CSCF) – see example in Table 10.5.1-34**

When the session with UE#F has been successfully established, UE#1 sends a Notify request to its proxy, P-CSCF#1.

Table 10.5.1-34: Notify (UE to P-CSCF)

```
NOTIFY sip:token7@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 NOTIFY
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Event: refer
Content-Type: application/sip
Content-length: (…)

200 OK
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

35. **Notify (P-CSCF to S-CSCF) – see example in Table 10.5.1-35**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#1 forwards the Notify request to S-CSCF#1.

Table 10.5.1-35: Notify (P-CSCF to S-CSCF)

```
NOTIFY sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Event:
Content-Type:
Content-length:

200 OK
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

36. **Notify (S-CSCF to S-CSCF) – see example in Table 10.5.1-36**

S-CSCF#1 forwards the Notify request to S-CSCF#2.

Table 10.5.1-36: Notify (S-CSCF to S-CSCF)

```
NOTIFY sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Event:
Content-Type:
Content-length:

200 OK
```

37. **Notify (S-CSCF to P-CSCF) – see example in Table 10.5.1-37**

S-CSCF#2 forwards the Notify request to P-CSCF#2.

Table 10.5.1-37: Notify (S-CSCF to P-CSCF)

```
NOTIFY sip%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Event:
Content-Type:
Content-length:

200 OK
```

38. **Notify (P-CSCF to UE) – see example in Table 10.5.1-38**

    P-CSCF#2 forwards the Notify request to UE#2.

Table 10.5.1-38: Notify (P-CSCF to UE)

```
NOTIFY sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Event:
Content-Type:
Content-length:

200 OK
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

39. **200-OK (UE to P-CSCF) – see example in Table 10.5.1-39**

    UE#2 acknowledges receipt of the Notify request (38) with a 200-OK final response, sent to P-CSCF#2.

Table 10.5.1-39: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
Content-length: 0
```

40. **200-OK (P-CSCF to S-CSCF) – see example in Table 10.5.1-40**

    P-CSCF#2 forwards the 200 OK final response to S-CSCF#2.

Table 10.5.1-40: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

41. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.5.1-41**

S-CSCF#2 forwards the 200 OK final response to S-CSCF#1.

Table 10.5.1-41: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

42. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.5.1-42**

S-CSCF#1 forwards the 200 OK final response to P-CSCF#1.

Table 10.5.1-42: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

43. **200-OK (P-CSCF to UE) – see example in Table 10.5.1-43**

P-CSCF#1 forwards the 200 OK final response to UE#1.

Table 10.5.1-43: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf1.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

44. **BYE (UE to P-CSCF) – see example in Table 10.5.1-44**

Upon receiving the notification of successful refer operation (38), UE#2 terminates the session with UE#1..

Table 10.5.1-34: Bye (UE to P-CSCF)

```
BYE sip:token6@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
To: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 BYE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

Request-URI: contains the value of the Contact header from the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response. Since this request is being initiated by the destination, the From and To are reversed.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

45. **Bye (P-CSCF to S-CSCF) – see example in Table 10.5.1-45**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#2 forwards the Notify request to S-CSCF#2.

Table 10.5.1-45: Bye (P-CSCF to S-CSCF)

```
BYE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:scscf1.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf1.home.net
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

46. **Bye (S-CSCF to S-CSCF) – see example in Table 10.5.1-46**

S-CSCF#2 forwards the Bye request to S-CSCF#1.

Table 10.5.1-46: Bye (S-CSCF to S-CSCF)

```
BYE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

47. **Bye (S-CSCF to P-CSCF) – see example in Table 10.5.1-47**

   S-CSCF#1 forwards the Bye request to P-CSCF#1.

Table 10.5.1-47: Bye (S-CSCF to P-CSCF)

```
BYE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

48. **Bye (P-CSCF to UE) – see example in Table 10.5.1-48**

   P-CSCF#2 forwards the Bye request to UE#2.

Table 10.5.1-48: Bye (P-CSCF to UE)

```
BYE sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net;branch=token9
From:
To:
Call-ID:
Cseq:
Contact: token9@pcscf1.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

49. **200-OK (UE to P-CSCF) – see example in Table 10.5.1-49**

   UE#2 acknowledges receipt of the Bye request (48) with a 200-OK final response, sent to P-CSCF#1.

Table 10.5.1-49: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net;branch=token9
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

50. **200-OK (P-CSCF to S-CSCF) – see example in Table 10.5.1-50**

    P-CSCF#1 forwards the 200 OK final response to S-CSCF#1.

Table 10.5.1-50: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf2.home.net
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

51. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.5.1-51**

    S-CSCF#1 forwards the 200 OK final response to S-CSCF#2.

Table 10.5.1-51: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

52. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.5.1-52**

    S-CSCF#2 forwards the 200 OK final response to P-CSCF#2.

Table 10.5.1-52: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

53. **200-OK (P-CSCF to UE) – see example in Table 10.5.1-53**

    P-CSCF#2 forwards the 200 OK final response to UE#2.

Table 10.5.1-53: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf2.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

10.5.2 Session transfer replacing an existing session

An IP multi-media session already exists between UE#1 and UE#2, and an IP multi-media session already exists between UE#2 and UE#F. UE#2 desires UE#1 to initiate a new session to destination UE#F, and terminate the existing sessions. The procedures for this transfer are shown in Figure 10.5.2-1.
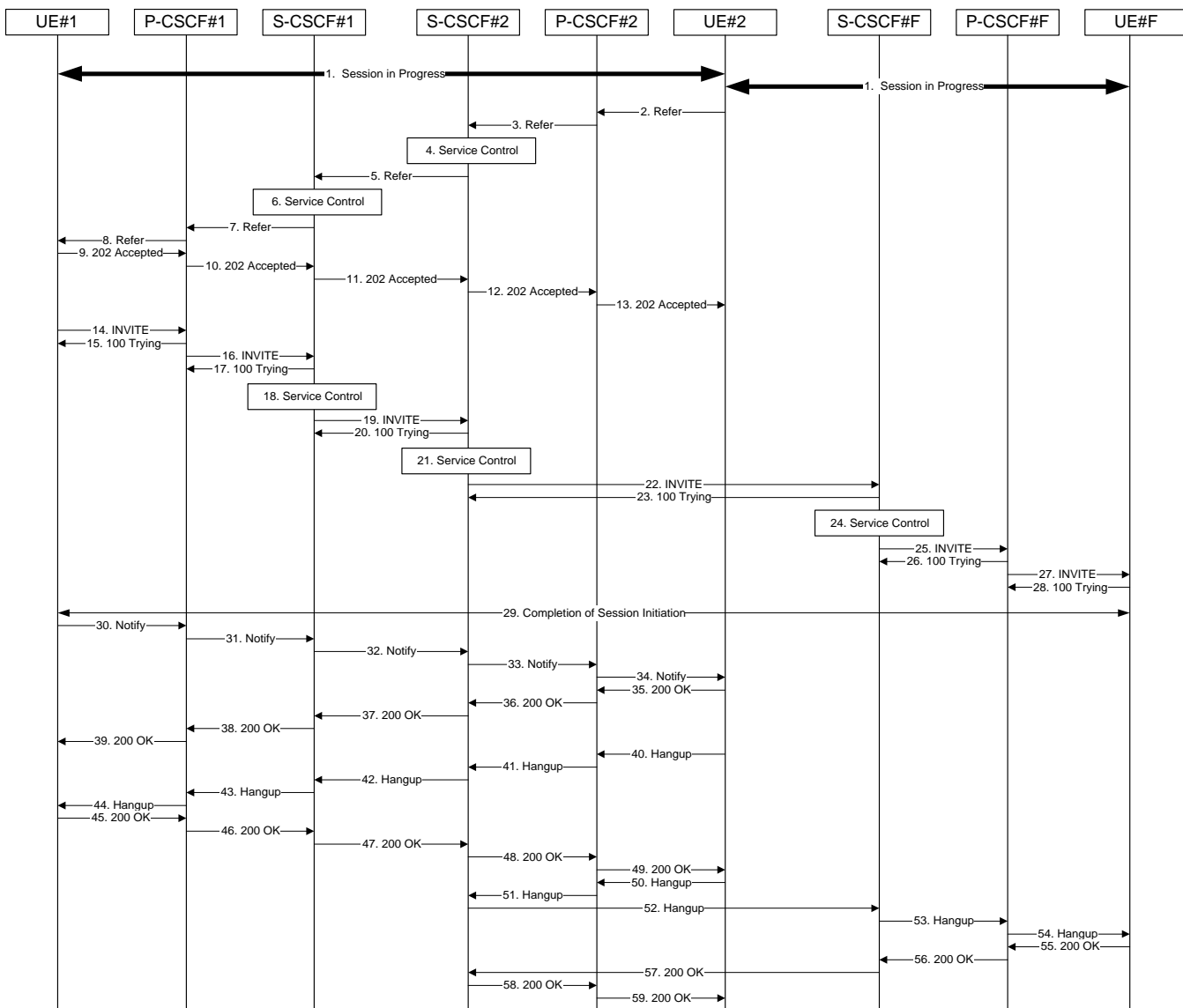
Figure 10.5.2-1 – Session Transfer replacing an existing session

1. Sessions in Progress

   UE#1 initiates a multi-media session with UE#2. As a result, the state information stored at P-CSCF#2 is shown in Table 10.5.2-1a

Table 10.5.2-1a: State Information

```
Request-URI: sip:token6@pcscf2.home.net

From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost

Route: sip:scscf2.home.net, sip:scscf1.home.net,
sip:%5b5555%3a%3aaaaa%3abbbb%3accc%3addd%5d@pcscf1.home.net
```

   UE#2 initiates a multi-media session with UE#F. As a result, the state information stored at P-CSCF#2 is shown in Table 10.5.2-1b

Table 10.5.2-1b: State Information

```
Request-URI: sip:token3@pcscf2.home.net

From: sip:B36(SHA-1(+1-212-555-2222; time=36123F05; seq=31))@localhost;tag=171828
To: sip:B36(SHA-1(+1-212-555-3333; time=36123F05; seq=32))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=31))@localhost

Route: sip:scscf2.home.net, sip:scscff.home.net,
sip:%5b5555%3a%3aaaaa%3abbbb%3accc%3addd%5d@pcscff.home.net
```

UE#2 has placed both of these sessions on hold.

2. **REFER (UE to P-CSCF) – see example in Table 10.5.2-2**

UE#2 sends a Refer request to its proxy, P-CSCF#2.

Table 10.5.2-2: REFER (UE to P-CSCF)

```
REFER sip:token6@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
To: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 REFER
Contact: sip:[5555::eee:fff:aaa:bbb]
Refer-To: tel:+1-212-555-3333 ? Call-ID=B36(SHA-1(555-1111;time=36123E5B;seq=31))@localhost
Refer-By: sip:B36(SHA-1(+1-212-555-2222; time=36123F05; seq=31))@localhost;tag=171828
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off
Content-length: 0
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

Editor's Note: Use of Remote-Party-ID in REFER is FFS.

Editor's Note: The proper value for the Refer-By header is FFS. The value of the From header of the session to be replaced seems most appropriate.

3. **REFER (P-CSCF to S-CSCF) – see example in Table 10.5.2-3**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#2 forwards the Refer request to S-CSCF#2.

Table 10.5.2-3: REFER (P-CSCF to S-CSCF)

```
REFER sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:scscf1.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

4. **Service Control**

5. **REFER (S-CSCF to S-CSCF) – see example in Table 10.5.2-5**

In order to maintain the expectation of privacy of the identity of the new destination, S-CSCF#2 converts the "Refer-To" header into a private URL. S-CSCF#2 forwards the Refer request to S-CSCF#1.

NOTE: If the network operator desired configuration independence, the REFER would be routed through an I-CSCF before leaving the operator's network. For example, see configuration S-S#1b. That I-CSCF would convert the private URL into one that specified the I-CSCF as the hostname.

Table 10.5.2-5: REFER (S-CSCF to S-CSCF)

```
REFER sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Refer-To: sip:token(tel:+1-212-555-3333)@scscf2.home.net;private ? Call-ID=B36(SHA-1(555-
   1111;time=36123E5B;seq=31))@localhost
Refer-By:
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes
Content-length:
```

6. **Service Control**

7. **REFER (S-CSCF to P-CSCF) – see example in Table 10.5.2-7**

S-CSCF#1 forwards the Refer request to P-CSCF#1.

Table 10.5.2-7: REFER (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aaaaa%3abbb%3accc%3addd%5d@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UEP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

8. **REFER (P-CSCF to UE) – see example in Table 10.5.2-8**

   P-CSCF#1 forwards the Refer request to UE#1.

Table 10.5.2-8: REFER (P-CSCF to UE)

```
REFER sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Refer-To:
Refer-By:
Remote-Party-ID:
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

9. **202-Accepted (UE to P-CSCF) – see example in Table 10.5.2-9**

   UE#2 acknowledges receipt of the Refer request (8) with a 202-Accepted final response, sent to P-CSCF#1.

Table 10.5.2-8: 202 Accepted (UE to P-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

10. **202-Accepted (P-CSCF to S-CSCF) – see example in Table 10.5.2-10**

   P-CSCF#1 forwards the 202 Accepted final response to S-CSCF#1.

Table 10.5.2-10: 202 Accepted (P-CSCF to S-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UEP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

11. **202-Accepted (S-CSCF to S-CSCF) – see example in Table 10.5.2-11**

S-CSCF#1 forwards the 202 Accepted final response to S-CSCF#2.

Table 10.5.2-11: 202 Accepted (S-CSCF to S-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

12. **202-Accepted (S-CSCF to P-CSCF) – see example in Table 10.5.2-12**

S-CSCF#2 forwards the 202 Accepted final response to P-CSCF#2.

Table 10.5.2-12: 202 Accepted (S-CSCF to P-CSCF)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Content-length:
```

13. **202-Accepted (P-CSCF to UE) – see example in Table 10.5.2-13**

P-CSCF#2 forwards the 202 Accepted final response to UE#2.

Table 10.5.2-13: 202 Accepted (P-CSCF to UE)

```
SIP/2.0 202 Accepted
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From:
To:
Call-ID:
CSeq:
Content-length:
```

P-CSCF removes the Record-Route header

14. **INVITE (UE to P-CSCF) – see example in Table 10.5.2-14**

UE#1 initiates an INVITE request based on the Refer-To header URL in the REFER request. The INVITE is sent from the UE to P-CSCF#1.

Table 10.5.2-14: INVITE (UE to P-CSCF)

```
INVITE sip:token(tel:+1-212-555-3333)@scscf2.home.net;private SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Supported: 100rel
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off
Proxy-Require: privacy
Anonymity: Off
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=74))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=75))@localhost
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=31))@localhost
Cseq: 127 INVITE
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Refer-By: sip:B36(SHA-1(+1-212-555-2222; time=36123F05; seq=31))@localhost;tag=171828
Content-Type: application/sdp
Content-length: (…)

v=0
o=- 2987933615 2987933615 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c= IN IP6 5555::aaa:bbb:ccc:ddd
b=AS:64
t=907165275 0
m=audio 3456 RTP/AVP 97 3 96
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
a=rtpmap:96 G726-32/8000
a=qos:mandatory sendrecv
```

Call-ID: value taken from the URL parameter attached to the Refer-To header

15. **100 Trying (P-CSCF to UE) – see example in Table 10.5.2-15**

P-CSCF#1 responds to the INVITE request (14) with a 100 Trying provisional response.

Table 10.5.2-15: 100 Trying (P-CSCF to UE)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

16. **INVITE (P-CSCF to S-CSCF) – see example in Table 10.5.2-16**

P-CSCF#1 remembers (from the registration procedure) the request routing for this UE. This becomes a Route header in the request. The next hop is the S-CSCF serving this UE. P-CSCF rewrites the Contact header with a locally defined value that identifies the UE. P-CSCF adds itself to the Via header.

Table 10.5.2-16: INVITE (P-CSCF to S-CSCF)

```
INVITE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:token(tel:+1-212-555-3333)@scscf2.home.net;private
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

17. **100 Trying (S-CSCF to P-CSCF) – see example in Table 10.5.2-17**

S-CSCF#1 responds to the INVITE request (16) with a 100 Trying provisional response.

Table 10.5.2-17: 100 Trying (S-CSCF to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

18. **Service Control**

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

19. **INVITE (S-CSCF to S-CSCF) – see example in Table 10.5.2-19**

S-CSCF#1 performs an analysis of the destination address, which is a private URL generated by S-CSCF#2. Since it is a destination within the same operator's network, S-CSCF#1 forwards the INVITE request directly to S-CSCF#2.

Table 10.5.2-19: INVITE (S-CSCF to S-CSCF)

```
INVITE sip:token(tel:+1-212-555-3333)@scscf2.home.net;private SIP/2.0
Via: SIP/2.0/UDP sip:scscf1.home.net SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP
    [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf1.home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

20. **100 Trying (S-CSCF to S-CSCF) – see example in Table 10.5.2-20**

   S-CSCF#2 responds to the INVITE request (19) by sending a 100 Trying provisional response to S-CSCF#1.

Table 10.5.2-20: 100 Trying (S-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

21. **Service Control**

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

22. **INVITE (S-CSCF to S-CSCF) – see example in Table 10.5.2-22**

   S-CSCF#2 determines the destination address from the private URL contained in the INVITE request. Based on information in that URL, and information saved from step #4 above (implementation decision), S-CSCF#2 verifies the validity of the transfer request, and that it is within a short time delay from the REFER request.

   S-CSCF#2 builds a Route header based on stored state information for the Refer'd session (as determined by the Call-ID and Refer-By values in step #4 above).

   S-CSCF#2 forwards the INVITE request to S-CSCF#F.

Table 10.5.2-22: INVITE (S-CSCF to S-CSCF)

```
INVITE sip:scscff.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net, sip:+1-212-555-3333@home.net
Supported:
Remote-Party-ID: "John Doe" <tel:+1-212-555-1111>;privacy=off;screen=yes
Remote-Party-ID: "John Smith" <tel:+1-212-555-2222>;privacy=off;screen=yes;party=transferor
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Editor's Note: Use of "party=transferor" in a separate Remote-Party-ID header is FFS.

23. **100 Trying (S-CSCF to S-CSCF) – see example in Table 10.5.2-23**

    S-CSCF#F responds to the INVITE request (22) by sending a 100 Trying provisional response to S-CSCF#2.

Table 10.5.2-23: 100 Trying (S-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

24. **Service Control**

    S-CSCF#F performs whatever service control logic is appropriate for this session attempt.

25. **INVITE (S-CSCF to P-CSCF) – see example in Table 10.5.2-25**

    S-CSCF#F uses the Route header value to determine the next hop CSCF for this UE. It forwards the INVITE request to P-CSCF#F.

Table 10.5.2-25: INVITE (S-CSCF to P-CSCF)

```
INVITE sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net SIP/2.0
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net,
    SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net, sip:scscf1.home.net
Route: sip:+1-212-555-3333@home.net
Supported:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact:
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

26. **100 Trying (P-CSCF to S-CSCF) – see example in Table 10.5.2-26**

    P-CSCF#F responds to the INVITE request (25) by sending a 100 Trying provisional response to S-CSCF#F.

Table 10.5.2-26: 100 Trying (P-CSCF to S-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP icscf.home.net, SIP/2.0/UDP scscf2.home.net,
    SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Content-length: 0
```

27. **INVITE (P-CSCF to UE) – see example in Table 10.5.2-27**

    P-CSCF forwards the INVITE request to the UE.

Table 10.5.2-27: INVITE (P-CSCF to UE)

```
INVITE sip:+1-212-555-3333@home.net;user=phone SIP/2.0
Via: SIP/2.0/UDP pcscff.home.net;branch=token6
Supported:
Remote-Party-ID:
Remote-Party-ID:
Proxy-Require:
Anonymity:
From:
To:
Call-ID:
Cseq:
Contact: token6@pcscff.home.net
Refer-By:
Content-Type:
Content-length:

v=
o=
s=
c=
b=
t=
m=
a=
a=
a=
a=
```

Editor's Note: The mechanism by which UE#F knows it is to replace an existing session with this new session is FFS. A match of the Call-ID to an existing session, and the Refer-By to either From or To of the same session should be sufficient.

28. **100 Trying (UE to P-CSCF) – see example in Table 10.5.2-28**

UE#F may optionally send a 100 Trying provisional response to P-CSCF.

Table 10.5.2-28: 100 Trying (UE to P-CSCF)

```
SIP/2.0 100 Trying
Via: SIP/2.0/UDP pcscf1.home.net;branch=token6
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

29. **Completion of Session Initiation**

UE#1 and UE#F complete the session initiation, as shown in the MO, S-S, and MT procedures.

30. **NOTIFY (UE to P-CSCF) – see example in Table 10.5.2-30**

When the session with UE#F has been successfully established, UE#1 sends a Notify request to its proxy, P-CSCF#1. The session leg identification for this Notify is taken from that used in the Refer, earlier.

Table 10.5.2-30: Notify (UE to P-CSCF)

```
NOTIFY sip:token7@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
To: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 130 NOTIFY
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Event: refer
Content-Type: application/sip
Content-length: (…)

200 OK
```

Request-URI: contains the value of the Contact header from the 200-OK response to the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

31. **Notify (P-CSCF to S-CSCF) – see example in Table 10.5.2-31**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#1 forwards the Notify request to S-CSCF#1.

Table 10.5.2-31: Notify (P-CSCF to S-CSCF)

```
NOTIFY sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:scscf2.home.net, sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Event:
Content-Type:
Content-length:

200 OK
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

32. **Notify (S-CSCF to S-CSCF) – see example in Table 10.5.2-32**

S-CSCF#1 forwards the Notify request to S-CSCF#2.

Table 10.5.2-32: Notify (S-CSCF to S-CSCF)

```
NOTIFY sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Route: sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net
Record-Route: sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Event:
Content-Type:
Content-length:

200 OK
```

33. **Notify (S-CSCF to P-CSCF) – see example in Table 10.5.2-33**

    S-CSCF#2 forwards the Notify request to P-CSCF#2.

Table 10.5.2-33: Notify (S-CSCF to P-CSCF)

```
NOTIFY sip:%5b5555%3a%3aeee%3afff%3aaaaa%3abbb%5d@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Event:
Content-Type:
Content-length:

200 OK
```

34. **Notify (P-CSCF to UE) – see example in Table 10.5.2-34**

    P-CSCF#2 forwards the Notify request to UE#2.

Table 10.5.2-34: Notify (P-CSCF to UE)

```
NOTIFY sip:[5555::eee:fff:aaa:bbb] SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
Cseq:
Contact: token3@pcscf2.home.net
Event:
Content-Type:
Content-length:

200 OK
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

35. **200-OK (UE to P-CSCF) – see example in Table 10.5.2-35**

UE#2 acknowledges receipt of the Notify request (34) with a 200-OK final response, sent to P-CSCF#2.

Table 10.5.2-35: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net;branch=token3
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::eee:fff:aaa:bbb]
Content-length: 0
```

36. **200-OK (P-CSCF to S-CSCF) – see example in Table 10.5.2-36**

P-CSCF#2 forwards the 200 OK final response to S-CSCF#2.

Table 10.5.2-36: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net,
    SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route: sip:scscf2.home.net, sip:scscf1.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

37. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.5.2-37**

S-CSCF#2 forwards the 200 OK final response to S-CSCF#1.

Table 10.5.2-37: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

38. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.5.2-38**

S-CSCF#1 forwards the 200 OK final response to P-CSCF#1.

Table 10.5.2-38: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net, SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

39. **200-OK (P-CSCF to UE) – see example in Table 10.5.2-39**

P-CSCF#1 forwards the 200 OK final response to UE#1.

Table 10.5.2-39: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf1.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

40. **BYE (UE to P-CSCF) – see example in Table 10.5.2-40**

Upon receiving the notification of successful refer operation (34), UE#2 terminates the session with UE#1.

Table 10.5.2-40: Bye (UE to P-CSCF)

```
BYE sip:token6@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
To: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 BYE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

Request-URI: contains the value of the Contact header from the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response. Since this request is being initiated by the destination, the From and To are reversed.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

41. **Bye (P-CSCF to S-CSCF) – see example in Table 10.5.2-41**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#2 forwards the Notify request to S-CSCF#2.

Table 10.5.2-41: Bye (P-CSCF to S-CSCF)

```
BYE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:scscf1.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf1.home.net
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

42. **Bye (S-CSCF to S-CSCF) – see example in Table 10.5.2-42**

S-CSCF#2 forwards the Bye request to S-CSCF#1.

Table 10.5.2-42: Bye (S-CSCF to S-CSCF)

```
BYE sip:scscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf1.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

43. **Bye (S-CSCF to P-CSCF) – see example in Table 10.5.2-43**

S-CSCF#1 forwards the Bye request to P-CSCF#1.

Table 10.5.2-43: Bye (S-CSCF to P-CSCF)

```
BYE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf1.home.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

44. **Bye (P-CSCF to UE) – see example in Table 10.5.2-44**

P-CSCF#2 forwards the Bye request to UE#2.

Table 10.5.2-44: Bye (P-CSCF to UE)

```
BYE sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf1.home.net;branch=token9
From:
To:
Call-ID:
Cseq:
Contact: token9@pcscf1.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

45. **200-OK (UE to P-CSCF) – see example in Table 10.5.2-45**

UE#2 acknowledges receipt of the Bye request (44) with a 200-OK final response, sent to P-CSCF#1.

Table 10.5.2-45: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home.net;branch=token9
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

46. **200-OK (P-CSCF to S-CSCF) – see example in Table 10.5.2-46**

P-CSCF#1 forwards the 200 OK final response to S-CSCF#1.

Table 10.5.2-46: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscf1.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscf2.home.net
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

47. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.5.2-47**

S-CSCF#1 forwards the 200 OK final response to S-CSCF#2.

Table 10.5.2-547: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

48. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.5.2-48**

S-CSCF#2 forwards the 200 OK final response to P-CSCF#2.

Table 10.5.2-48: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

49. **200-OK (P-CSCF to UE) – see example in Table 10.5.2-49**

P-CSCF#2 forwards the 200 OK final response to UE#2.

Table 10.5.2-49: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf2.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

50. **BYE (UE to P-CSCF) – see example in Table 10.5.2-50**

Upon receiving the notification of successful refer operation (38), UE#2 terminates the session with UE#F.

Table 10.5.2-50: Bye (UE to P-CSCF)

```
BYE sip:token4@pcscf2.home.net SIP/2.0
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From: sip:B36(SHA-1(+1-212-555-2222; time=36123E5B; seq=73))@localhost;tag=314159
To: "Alien Blaster" <sip:B36(SHA-1(+1-212-555-1111; time=36123E5B; seq=72))@localhost>;tag=171828
Call-ID: B36(SHA-1(555-1111;time=36123E5B;seq=72))@localhost
Cseq: 131 BYE
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

Request-URI: contains the value of the Contact header from the initial INVITE.

Via: contains the IP address or FQDN of the originating UE.

From:, To:, Call-ID: contain the values previously used to establish the session, including the tag value from the response. Since this request is being initiated by the destination, the From and To are reversed.

Cseq: next higher sequential value.

Contact: the IP address or FQDN of the originating UE.

51. **Bye (P-CSCF to S-CSCF) – see example in Table 10.5.2-51**

P-CSCF adds a Route header, with the saved value from the previous 200-OK response. P-CSCF identifies the proper saved value by the Request-URI.

P-CSCF#2 forwards the Notify request to S-CSCF#2.

Table 10.5.2-51: Bye (P-CSCF to S-CSCF)

```
BYE sip:scscf2.home.net SIP/2.0
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:scscff.home.net, sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net
From:
To:
Call-ID:
Cseq:
Contact: sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscf2.home.net
Content-length:
```

Request-URI: the first component of the saved Route header.

Route: saved from the 200-OK response to the initial INVITE (with first element moved to Request-URI).

Contact: a locally defined value that identifies the UE.

52. **Bye (S-CSCF to S-CSCF) – see example in Table 10.5.2-52**

S-CSCF#2 forwards the Bye request to S-CSCF#F.

Table 10.5.2-52: Bye (S-CSCF to S-CSCF)

```
BYE sip:scscff.home.net SIP/2.0
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Route: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net
Record-Route: sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

53. **Bye (S-CSCF to P-CSCF) – see example in Table 10.5.2-53**

S-CSCF#F forwards the Bye request to P-CSCF#F.

Table 10.5.2-53: Bye (S-CSCF to P-CSCF)

```
BYE sip:%5b5555%3a%3aeee%3afff%3aaaa%3abbb%5d@pcscff.home.net SIP/2.0
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
Cseq:
Contact:
Content-length:
```

54. **Bye (P-CSCF to UE) – see example in Table 10.5.2-54**

P-CSCF#F forwards the Bye request to UE#F.

Table 10.5.2-54: Bye (P-CSCF to UE)

```
BYE sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscff.home.net;branch=token9
From:
To:
Call-ID:
Cseq:
Contact: token9@pcscff.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information.

Via: P-CSCF removes the Via headers, and generates a locally unique token to identify the saved values. It inserts this as a branch value on its Via header.

55. **200-OK (UE to P-CSCF) – see example in Table 10.5.2-55**

UE#F acknowledges receipt of the Bye request (58) with a 200-OK final response, sent to P-CSCF#F.

Table 10.5.2-55: 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscff.home.net;branch=token9
From:
To:
Call-ID:
CSeq:
Contact: sip:[5555::aaa:bbb:ccc:ddd]
Content-length: 0
```

56. **200-OK (P-CSCF to S-CSCF) – see example in Table 10.5.2-56**

P-CSCF#F forwards the 200 OK final response to S-CSCF#F.

Table 10.5.2-56: 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscff.home.net, SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net,
    SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route: sip:scscff.home.net, sip:scscf2.home.net
From:
To:
Call-ID:
CSeq:
Contact: sip:%5b5555%3a%3aaaa%3abbb%3accc%3addd%5d@pcscff.home.net
Content-length:
```

P-CSCF restores the Via headers and Record-Route headers from the branch value in its Via.

Contact: a locally defined value that identifies the UE.

57. **200-OK (S-CSCF to S-CSCF) – see example in Table 10.5.2-57**

S-CSCF#F forwards the 200 OK final response to S-CSCF#2.

Table 10.5.2-57: 200 OK (S-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf2.home.net, SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

58. **200-OK (S-CSCF to P-CSCF) – see example in Table 10.5.2-58**

S-CSCF#2 forwards the 200 OK final response to P-CSCF#2.

Table 10.5.2-58: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf2.home.net, SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
Record-Route:
From:
To:
Call-ID:
CSeq:
Contact:
Content-length:
```

59. **200-OK (P-CSCF to UE) – see example in Table 10.5.2-59**

P-CSCF#2 forwards the 200 OK final response to UE#2.

Table 10.5.2-59: 200 OK (P-CSCF to UE)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP [5555::eee:fff:aaa:bbb]
From:
To:
Call-ID:
CSeq:
Contact: sip:token2@pcscf2.home.net
Content-length:
```

P-CSCF removes the Record-Route and Contact headers, calculates the proper Route header to add to future requests, and saves that information without passing it to UE.

Contact: a locally unique token to identify the saved routing information

100 Signalling flows for "regular" error handling

Editors Note: The purpose of this clause is to capture all the "regular" error handling scenarios. One example that falls into this class is Network Initiated Deregistration.

Editors Note: The Clause number "100" is temporary. It will be replaced with the appropriate clause number before this specification goes for approval, to enable this clause to be located towards the end of the Specification.

100.1 Registration: network initiated deregistration
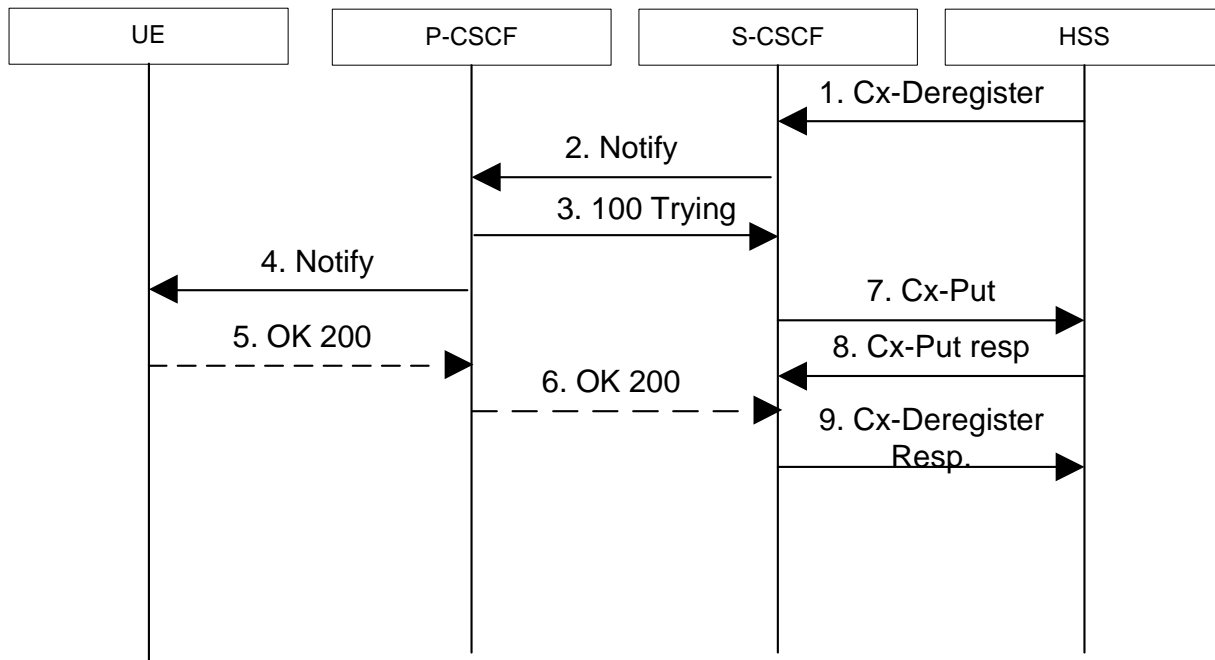
1. **Cx-Deregister**

   HSS initiates the de-registration, sending a Cx-Deregister (subscriber identity)

2. **SIP NOTIFY (S-CSCF to P-CSCF) – see example in Table 100.1-2**

   If the UE has subscribed to the network initiated deregistration event, the S-CSCF sends a NOTIFY to the P-CSCF in order to inform about the network initiated deregistration. The same To, From, Call-ID are used as in the first NOTIFY. CSeq is incremented.

Editor's Note: The contents of the Event header field (i.e. its parameters) are FFS, as this requires further input from stage 2. In the following the Event header field is only an example, to give some idea, how it may look like. Further possibilities are to use Warning and Retry-After header fields (instead of / in addition to Event header field parameters) for transferring the concerning information about the event.

Table 100.1-2 SIP NOTIFY (S-CSCF to P-CSCF)

```
NOTIFY sip: pcscf1.visited1.net SIP/2.0
Via: SIP/2.0/UDP scscf1.visited1.net
From: <sip:sip:user1@home1.net>
To: <sip:sip:user1@home1.net>
Call-ID: 12345@ue_a.home_a.com
CSeq: 2 NOTIFY
Expires: 7200
Event: org.3gpp.nwinitdereg; reason-code=999;
  reason-phrase="You have been deregistered from the network, please register again";
  registrar=sip:registrar.home_network.net
Content-Length: 0
```

> **Call-ID:** contains the value of the Call-ID from the SUBSCRIBE request.

3. 100 Trying (P-CSCF to S-CSCF) – see example in Table 100.1-3

> P-CSCF responds to the Notify request with a 100 Trying provisional response.

Table 100.1-3 100 Trying (P-CSCF to S-CSCF)

```
sip/2.0 100 trying
Via: SIP/2.0/UDP scscf1.home1.net
From:
To:
Call-ID:
CSeq:
Content-Length: 0
```

4. SIP NOTIFY (P-CSCF to UE) – see example in Table 100.1-4

> The P-CSCF sends a NOTIFY to the UE in order to inform about the network initiated deregistration.

Table 100.1-4 SIP NOTIFY (P-CSCF to UE)

```
NOTIFY sip:[5555::aaa:bbb:ccc:ddd] SIP/2.0
Via: SIP/2.0/UDP pcscf1.visited1.net, SIP/2.0/UDP scscf1.visited1.net
From:
To:
Call-ID:
CSeq:
Expires:
Event:
Content-Length:
```

5. 200 OK (UE to P-CSCF) – see example in Table 100.1-5

> The UE stores the Call-ID, CSeq, etc. and, when possible, generates a 200 OK response to the NOFIFY request.

Table 100.1-4 SIP 200 OK (UE to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.visited1.net
From:
To:
Call-ID:
CSeq:
Content-Length:
```

6. 200 OK (P-CSCF to S-CSCF) – see example in Table 100.1-6

> The P-CSCF sends a 200 OK response to the S-CSCF and updates its internal database to remove the UE from being registered. The P-CSCF does not need to wait for the 200 OK response from the UE.

Table 100.1-6 SIP 200 OK (P-CSCF to S-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP scscf1.visited1.net, SIP/2.0/UDP pcscf1.visited1.net
From:
To:
Call-ID:
CSeq:
Content-Length:
```

7. Cx-Put

The S-CSCF sends an update to the HSS to remove itself as the registered S-CSCF for this UE.

Note: Steps 7 may be performed right after receiving the 100 Trying response from the P-CSCF. S-CSCF shall not wait for the 200 OK response from the UE.

8. Cx-Put response

The HSS confirms the update.

9. Cx-Deregister response

The S-CSCF returns a response to the entity that initiated the process.

101 Signalling flows for "irregular" error handling

Editors Note: The purpose of this clause is to capture all the "irregular" error handling scenarios. An example is recovery/failure flows to deal with the event of a Network Element failure. The flows in this clause would typically refer to flows detailed in earlier clauses of this Specification.

Editors Note: The clause number "101" is temporary. It will be replaced with the appropriate clause number before this specification goes for approval, to enable this clause to be located towards the end of the Specification.
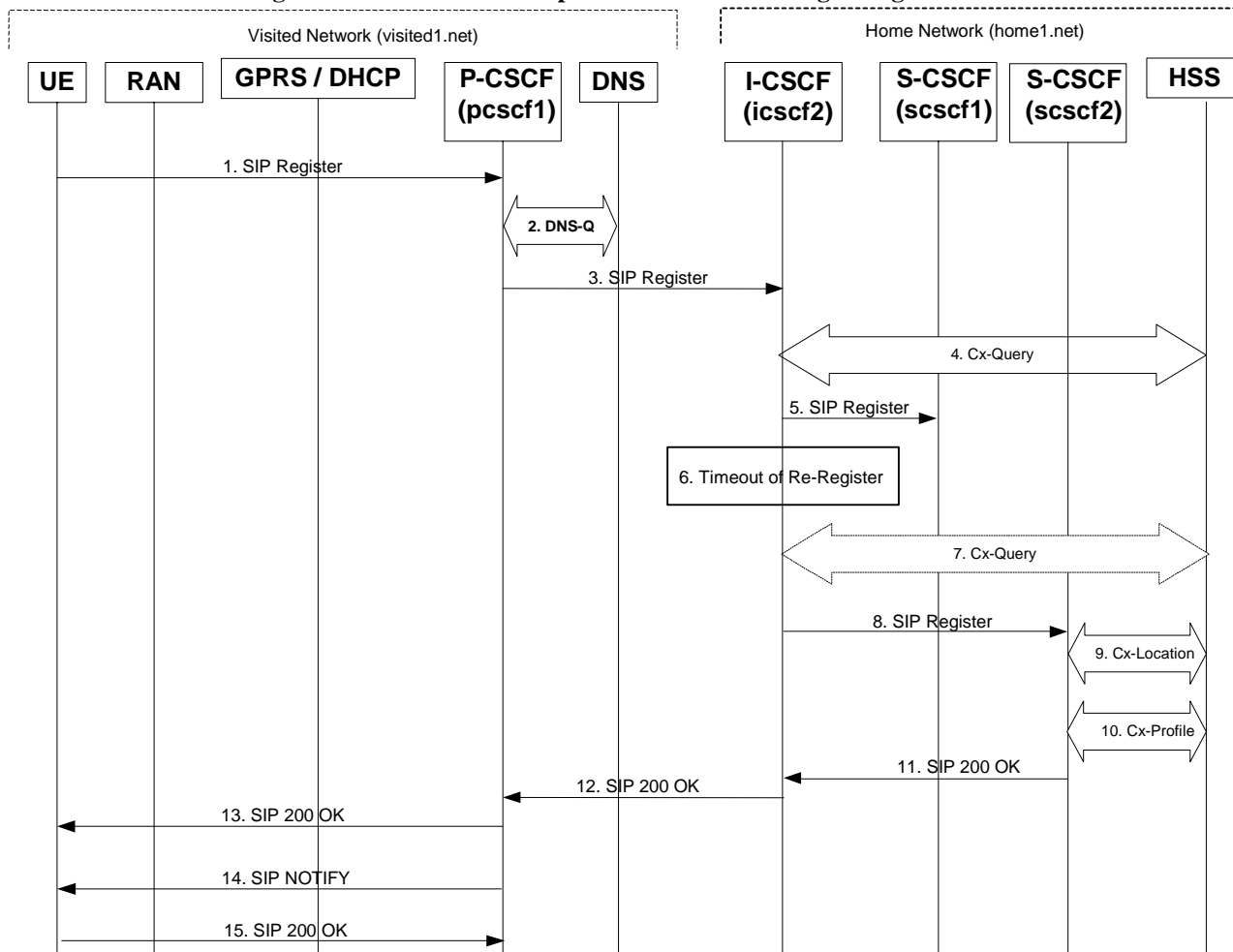
101.1 Error handling: Elements of signalling flows for provision of IP-connectivity network

101.2 Error Handling: Registration Signalling

101.2.1 Re-registration – Failure of re-registration

This information flow is a continuation of the information flow in subclause 7.2 "Registration Signalling: Re-Registration – User Currently Registered" after reception of information flow 4. This information flow shows the recovery after a failure of the S-CSCF that had been assigned to the subscriber in a previous registration.

Figure 101.2.1-1: Failure of previous S-CSCF during re-registration

Steps 1 through 4 are the same as the information flow in subclause 7.2 "Registration Signalling: Re-Registration – User Currently Registered"

5   **SIP REGISTER (I-CSCF to S-CSCF) – see example in Table 101.2.1-5**

This signalling flow forwards the SIP REGISTER request from the I-CSCF to the S-CSCF selected. The Request-URI is changed to the address of the S-CSCF.

I-CSCF adds a proper I-CSCF name to the Path header.

Table 101.2.1-5 SIP REGISTER request (I-CSCF to S-CSCF)

```
REGISTER sip: scscf1.home1.net SIP/2.0
Via: SIP/2.0/UDP icscf2.home1.net
Via: SIP/2.0/UDP pcscf1.visited1.net
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd]
Path: <sip:icscf2.home1.net>
Path: <sip:pcscf1.visited1.net>
Proxy-require: path
Require: path
From: <sip:user_private1@home1.net>
To: <sip:user_public1@home1.net>
Contact: <sip:user1%40home1.net@pcscf.visited1.net>
Call-ID: 123456789@[5555::aaa:bbb:ccc:ddd]
CSeq: 10 REGISTER
Expires: 7200
Content-Length: 0
```

6 **Timeout of Re-Register**

The I-CSCF times out, waiting for the response from the S-CSCF.

Editor's Note: The value of the timer in this particular instance is FFS. Clearly the value of the timers in the P-CSCF and UE waiting for the response must be considered when choosing this value.

Editor's note: "Whether it is appropriate or not to send the Register request to S-CSCF2 when I-CSCF times out waiting for a response from S-CSCF1 is FFS. While doing a new HSS query or performing a new S-CSCF selection the UE might time out and resend the Register request.

If this step is found to be not a problem for the UE, then the issue of having one subscriber registered to only one S-CSCF must be clarified."

7 **Cx-Query (Optional)**

The I-CSCF informs the HSS that the S-CSCF for the subscriber is unreachable and requests information related to the required S-CSCF capabilities from the HSS, The HSS sends the capability information required for S-CSCF selection. The I-CSCF uses this information to select a suitable S-CSCF.

This step is optional. Depending on implementation, sufficient information may be available to the I-CSCF from Step 4, to allow the I-CSCF select an alternate S-CSCF. Alternative mechanisms (for example a CSCF management plane) would be used to enable the HSS learn of S-CSCF failure. In addition, the HSS will learn about the assignment of a new S-CSCF in Step 9.

8 **SIP REGISTER (I-CSCF to S-CSCF) – see example in Table 101.2.1-8**

This signalling flow forwards the SIP REGISTER request from the I-CSCF to the newly selected S-CSCF. The Request-URI is changed to the address of the new S-CSCF.

Table 101.2.1-8 SIP REGISTER request (I-CSCF to S-CSCF)

```
REGISTER sip: scscf2.home1.net SIP/2.0
Via:
Via:
Via:
Path:
Path:
Proxy-require:
Require:
From:
To:
Contact:
Call-ID:
CSeq:
Expires:
Content-Length:
```

The remaining steps (9-15) are the same as in the normal re-registration case (steps 6-12 in subclause 7.2)

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		0.0.1
19/10/00		N1-001114			Version 0.0.2 Revisions identified by CN1 SIP ad-hoc #1	0.0.1	0.0.2
08/11/00		N1-001189			Version 0.0.3 Changes to new 3GPP template and styles	0.0.2	0.0.3
22/11/00		N1-001378			Version 0.1.0 Inclusion of baseline material from CN #14 REGISTER and INVITE flows	0.0.3	0.1.0
29/11/00					Version 0.2.0 Material added to UE generated REGISTER request	0.1.0	0.2.0
21/02/01		N1-010371			Version 0.3.0 Inclusion of contributions and updates from CN1/SA2 SIP#3. Tdocs include N1-010247, N1-010248, N1-010255, N1-010258, N1-010259, N1-010263, N1-010265, N1-010267, N1-010274.	0.2.0	0.3.0
09/03/01		N1-010505			Version 0.4.0 Inclusion of contributions and updates from CN1#16. Tdocs include N1-010352 (NJ-010123), N1-010353 (NJ-010124), N1-010416 (NJ-010142), N1-010425 (NJ-010143), N1-010483, N1-010484.	0.3.0	0.4.0
17/04/01		N1-010756			Version 0.5.0 Inclusion of contributions and updates from CN1/SA2 SIP#4. Tdocs include N1-010573, N1-010552, N1-010531, N1-010572, N1-010571, N1-010570, N1-010569, N1-010529, N1-010515, N1-010514, N1-010513, N1-010511, N1-010510, N1-010580, N1-010564, N1-010563, N1-010562, N1-010549, N1-010538	0.4.0	0.5.0
21/05/01		None			Version 1.0.0 Inclusion of contributions and updates from CN1#17. Version for presentation for Information at CN#12 Tdocs include N1-010702, N1-010703, N1-010705, N1-010706, N1-010707, N1-010747, N1-010849, N1-010874, N1-010875, N1-010883, N1-010884, N1-010885, N1-010891, N1-010895, N1-010898, N1-010900	0.5.0	1.0.0
22/06/01					Version 1.1.0. Completion of two contributions (i.e. N1-010849, N1-010898) from CN1#17, plus numerous formatting changes.	1.0.0	1.1.0