3GPP TSG-CN WG3 Meeting #28
San Diego, USA, 19th - 23rd May 2003.

N3-030195

Source: **CN3**

Title: **All LSs sent from CN3 since TSG CN#19 meeting**

Agenda item: **6.3.1**

Document for: **INFORMATION**

Introduction:

This document contains **4 approved** LSs sent from **TSG CN WG3**, and are forwarded to TSG CN Plenary meeting for information only.

Tdoc #	Tdoc Title	LS to	LS cc	Attachment
N3-030452	Response LS on Radio Access Bearer for PS conversational testing	SA4	CN	-
N3-030461	LS to SA2 on SIP signalling interworking between IM CN subsystem entities and SIP network entities external to the IN CN subsystem	SA2	CN1	N3-030460
N3-030413	LS on IMS Session Hold and Resume stage 2 and 3 descriptions	SA2, CN1, SA5	-	N3-030189
N3-030414	LS on Handling of SIP redirect messages (3xx responses)	SA2	-	-

3GPP TSG-CN WG3 Meeting #28 ***N3-030413***
San Diego, USA, 19th - 23rd May 2003.

Title: **LS on IMS Session Hold and Resume stage 2 and 3 descriptions**
Release: **Rel-6**

Source: **CN3**
To: **SA2, CN1, SA5**

Contact Person:
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Attachments: N3-030189 [Discussion document on IMS Session Hold and Resume stage 2 and 3 descriptions]

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1. Overall Description:

TSG CN3 is responsible for the specification of interworking between the IM CN and CS networks. CN3 investigated SA2 and CN1 specifications on handling of session hold and resume requests originating from the IMS.

CN3 feels that the existing SA2 and CN1 specifications don't clarify all aspects of hold and resume request and SA2 and CN1 guidance would be required.

CN3 found the following issues during the discussion of hold and resume:

1. There is no service description of the hold and resume service in the IMS that would describe e.g. the charging implications.
2. The call flows showing the handling of hold and resume from the IMS side at the MGCF/IM-MGW in TS 23.228 and TS 24.228 do not depict any hold and resume messages on the CS side, although they depict a PSTN box. Thus they seem to imply that the hold and resume service originating in the IMS is terminated at the MGCF/IM-MGW, rather than interworked with the corresponding service at the CS side.
3. CN3 could not agree if suspension of media sending at the IM-MGW (and the needed H.248 procedures) is really required as gating is also done in the GGSN. There were also concerns during the discussion of the service about why to act upon the receipt of the hold request at the Mn interface side if the subscriber is still charged due to the use of the CS side resources. In a pure CS-CS call no action is done in user plane for a hold resume sequence. In a pure IMS-IMS call, the gating function in the GGSN is understood by CN3 to be optional and therefore no interaction with user plane seems to be required.
4. CN3 assumes that putting media on hold is done using the SDP "inactive" attribute, although RFC 3264 describes the possibility that an UA puts "sendrecv" media on hold by making them "sendonly" and refrains from sending, as the later possibility can not be discriminated within the network from making media streams unidirectional.

The attached discussion document contains the SA2 and CN1 specification excerpts that were discussed during the meeting.

2. Actions:

To SA2 group.

ACTION: CN3 kindly asks SA2 to state whether the H.248 interactions between the MGCF and MGW are needed as shown in the information flow in TS 23.228. Furthermore, clarification is required about whether signalling messages need to be sent towards the CS NW represented by a PSTN box in the information flow or whether CS NW involvement is not needed in this case.

To CN1 group.

ACTION: CN3 kindly asks CN1 to confirm which SIP request message and SDP attribute is used to express the hold and resume. The message flow examples show that UPDATE is used but discussions have revealed that the usage of INVITE mechanism would be also possible. Furthermore, clarification is required on which SDP attributes are considered as a trigger condition for a possible hold message. Are ‘a=inactive’ and ‘a=sendonly’ both trigger conditions, or only ‘a=inactive’?

To SA5 group.

ACTION: CN3 kindly asks SA5 to point to/create a charging model of the hold and resume service for the IMS-CS interworking case.

3. Date of Next CN3 Meetings:

CN3#29	25^{th} - 29^{th} August 2003	Sophia Antipolis, France.
CN3#30	27^{th} - 31^{th} October 2003	

3GPP TSG-CN WG3 Ad Hoc Meeting on IMS CS Interworking **Tdoc N3-030189**
Sophia Antipolis, France, 1st – 2nd April 2003.

Source:	**Nokia**
Title:	**Discussion document on IMS Session Hold and Resume stage 2 and 3 descriptions**
Agenda item:	
Document for:	**INFORMATION**

Discussion

The CN3#27 meeting rejected a CR for the draft Rel-6 TS 29.163 on IMS session hold and resume, document N3-030053. The grounds for the rejection were that there is only a stage 1 service description but no stage 2 or stage 3 requirements/descriptions for IMS session hold and resume in 3GPP specifications.

A further study after the CN3#27 meeting has shown that

- There is a stage 2 description in Rel-5 TS 23.228, in section 5.11.1.2 “Mobile initiated Hold and Resume of a Mobile-PSTN Session”. (An excerpt below)
- There is a stage 3 description for SIP in Rel-5 TS 24.228, in section 10.1.3. (An excerpt below)

Based on the above mentioned and below cited stage 2 and 3 descriptions, it is proposed that the CN3 ad hoc meeting re-evaluates the hold and resume contribution, re-issued in document N3-030190.

Excerpts from 23.228, v.5.7.0:

5.11.1 Session Hold and Resume Procedures

This section gives information flows for the procedures for placing sessions on hold that were previously established by the mechanisms of sections 5.4, 5.5, 5.6, and 5.7, 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 shall be possible to place a subset of the media streams on hold while maintaining the others.

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 section.

…

5.11.1.2 Mobile-initiated Hold and Resume of a Mobile-PSTN Session

An IMS 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 as its GGSN is located, 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 section 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.

The procedures for placing a media stream on hold, and later resuming the media stream, are as shown in the following information flow:

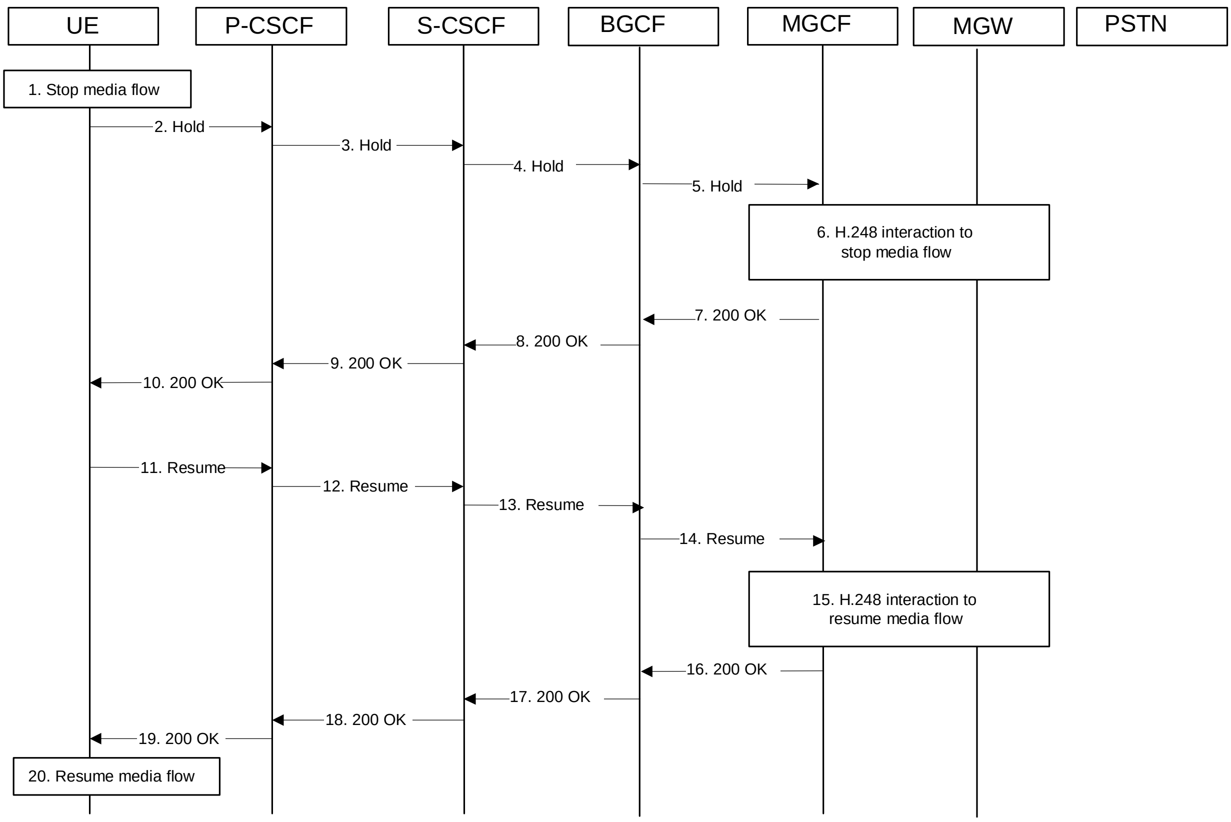

Figure 5.29: Mobile to PSTN session hold and resume

Information flow procedures are as follows:

1. UE detects a request from the user 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. UE sends a Hold message to its proxy, P-CSCF.
3. P-CSCF forwards the Hold message to S-CSCF.
4. S-CSCF forwards the Hold message to BGCF.
5. BGCF forwards the Hold message to MGCF.
6. 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.
7. MGCF acknowledges receipt of the Hold message with a 200-OK final response, send to BGCF.
8. BGCF forwards the 200-OK to the S-CSCF.
9. S-CSCF forwards the 200 OK final response to P-CSCF.
10. P-CSCF forwards the 200 OK final response to UE.
11. UE detects a request from the user to resume the media stream previously placed on hold. UE sends a Resume message to its proxy, P-CSCF.
12. P-CSCF forwards the Resume message to S-CSCF.
13. S-CSCF forwards the Resume message to BGCF.
14. BGCF forwards the Resume message to MGCF.
15. MGCF initiates a H.248 interaction with MGW instructing it to resume sending the media stream.
16. MGCF acknowledges receipt of the Resume message with a 200-OK final response, sent to BGCF.
17. BGCF forwards the 200 OK final response to the S-CSCF.
18. S-CSCF forwards the 200 OK final response to P-CSCF.
19. P-CSCF forwards the 200 OK final response to UE.
20. UE resumes sending the media stream to the remote endpoint.

An excerpt from 24.228, v.5.3.0:

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. 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 CS-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 signalling flows such as the ones described in this clause. Procedures at the I-CSCFs are identical to those described for the BYE, PRACK, and UPDATE requests and responses described in other clauses.

As this flow does not require a user interaction at the remote end, it is realized with an UPDATE request.

The procedures for placing a media stream on hold, and later resuming the media stream, are as shown in figure 10.1.3-1:

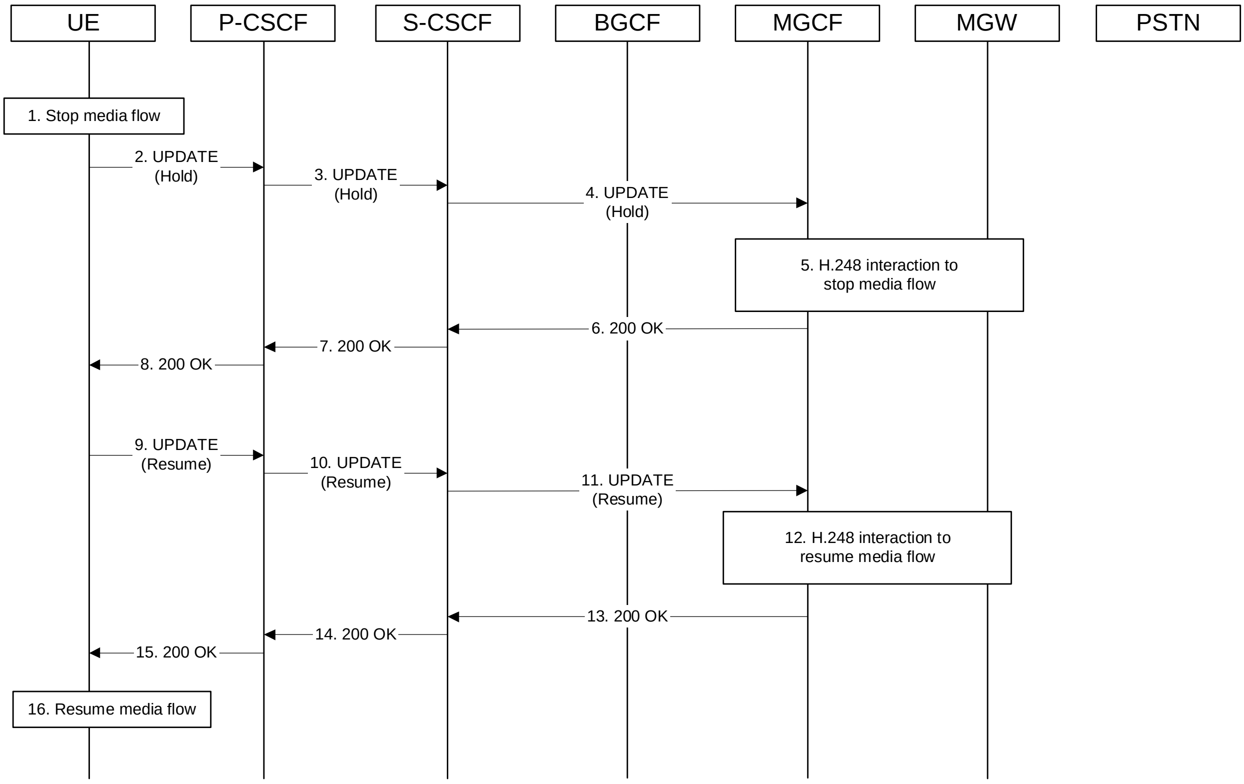

Figure 10.1.3-1: Mobile to PSTN session hold and resume

Signalling 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. **UPDATE (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: UPDATE (Hold) (UE to P-CSCF)

```
UPDATE sip:mgcf1.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
Max-Forwards: 70
From: sip:user1_public1@home1.net; tag=171828
To: tel:+1-212-555-2222;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 130 UPDATE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615  2987933616 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=audio 3456 RTP/AVP 97
b=AS:25.4
a=inactive
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 The sendrecv media stream is placed on hold by changing it to inactive media stream, and no media is sent to the far end.

3. **UPDATE (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 forwards the Hold request to S-CSCF.

Table 10.1.3-3: UPDATE (Hold) (P-CSCF to S-CSCF)

```
UPDATE sip:mgcf1.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home1.net;branch=z9hG4bK431h23.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
Max-Forwards: 69
Route: sip:scscf1.home1.net;lr
P-Charging-Vector: icid-value=a834bc192a44; icid-generated-at=[5555::e9e:d8d:c7c:b6b]
From:
To:
Call-ID:
Cseq:
Content-Type:
Content-Length:

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

Route: Saved from the 200 (OK) response to the initial INVITE.

P-Charging-Vector: The P-CSCF inserts this header and populates the icid parameters with a unique value and the IP address of the P-CSCF.

4. **UPDATE (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: UPDATE (Hold) (S-CSCF to MGCF)

```
UPDATE sip:mgcf1.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
   pcscf1.home1.net;branch=z9hG4bK431h23.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
Max-Forwards: 68
P-Charging-Vector:
From:
To:
Call-ID:
Cseq:
Content-Type:
Content-Length:

v=
o=
s=
c=
t=
m=
b=
a=
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 scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
   pcscf1.home1.net;branch=z9hG4bK431h23.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
From:
To:
Call-ID:
CSeq:
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615  2987933616 IN IP6 5555::eee:fff:aaa:bbb
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=audio 3456 RTP/AVP 97
b=AS:25.4
a=inactive
a=rtpmap:97 AMR
a=fmtp:97 mode-set=0,2,5,7; maxframes=2
```

SDP: Since the media stream was offered as inactive, it is marked as inactive in the response.

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 pcscf1.home1.net;branch=z9hG4bK431h23.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
From:
To:
Call-ID:
CSeq:
Content-Type:
Content-Length:

v=
o=
s=
c=
t=
m=
b=
a=
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];branch=z9hG4bKnashds7
From:
To:
Call-ID:
CSeq:
Content-Type:
Content-Length:

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

9. **UPDATE (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: UPDATE (Resume) (UE to P-CSCF)

```
UPDATE sip:mgcf1.home1.net SIP/2.0
Via: SIP/2.0/UDP [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
Max-Forwards: 70
From: sip:user1_public1@home1.net; tag=171828
To: tel:+1-212-555-2222;tag=314159
Call-ID: cb03a0s09a2sdfglkj490333
Cseq: 131 UPDATE
Content-Type: application/sdp
Content-Length: (…)

v=0
o=- 2987933615 2987933617 IN IP6 5555::aaa:bbb:ccc:ddd
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=audio 3456 RTP/AVP 97
b=AS:25.4
a=sendrecv
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.

SDP Same SDP as negotiated during the session setup, restores the sendrecv media stream.

10. **UPDATE (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 forwards the Resume request to S-CSCF.

Table 10.1.3-10: UPDATE(Resume) (P-CSCF to S-CSCF)

```
UPDATE sip:mgcf1.home1.net SIP/2.0
Via: SIP/2.0/UDP pcscf1.home1.net;branch=z9hG4bK431h23.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
Max-Forwards: 69
Route: sip:scscf1.home1.net;lr
P-Charging-Vector: icid-value=a834bc192a45; icid-generated-at=[5555::e9e:d8d:c7c:b6b]
From:
To:
Call-ID:
Cseq:
Content-Type:
Content-Length:

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

Route: Saved from the 200 (OK) response to the initial INVITE.

P-Charging-Vector: The P-CSCF inserts this header and populates the icid parameters with a unique value and the IP address of the P-CSCF.

11. **UPDATE(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: UPDATE(Resume) (S-CSCF to MGCF)

```
UPDATE sip:mgcf1.home1.net SIP/2.0
Via: SIP/2.0/UDP scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
   pcscf1.home1.net;branch=z9hG4bK431h23.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
Max-Forwards: 68
P-Charging-Vector:
From:
To:
Call-ID:
Cseq:
Content-Type:
Content-Length:

v=
o=
s=
c=
t=
m=
b=
a=
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 scscf1.home1.net;branch=z9hG4bK332b23.1, SIP/2.0/UDP
   pcscf1.home1.net;branch=z9hG4bK431h23.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
From:
To:
Call-ID:
CSeq:
Content-Type:
Content-Length:

v=0
o=- 2987933615  2987933617 IN IP6 5555::eee:fff:aaa:bbb
s=-
c=IN IP6 5555::aaa:bbb:ccc:ddd
t=907165275 0
m=audio 6402 RTP/AVP 97
b=AS:25.4
a=sendrecv
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**

Table 10.1.3-14: 200 OK (S-CSCF to P-CSCF)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP pcscf1.home1.net;branch=z9hG4bK431h23.1, SIP/2.0/UDP
   [5555::aaa:bbb:ccc:ddd];branch=z9hG4bKnashds7
From:
To:
Call-ID:
CSeq:
Content-Type:
Content-Length:

v=
o=
s=
c=
t=
m=
b=
a=
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];branch=z9hG4bKnashds7
From:
To:
Call-ID:
CSeq:
Content-Type:
Content-Length:

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

16. **Resume Media Flow**

UE resumes sending the media stream to the remote endpoint.

3GPP TSG-CN WG3 Meeting #28 ***N3-030452***
San Diego, USA, 19th - 23rd May 2003.

Title:	**Response LS on Radio Access Bearer for PS conversational testing**
Response to:	**LS (N3-030376/NP-030125/S4-030260) on Radio Access Bearer for PS conversational testing**
Release:	**Rel-5**
Source:	**CN3**
To:	**SA4**
Cc:	**CN**
Attachments:	None

Contact Person:
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1. Overall Description:

CN3 is pleased to answer some of the questions SA4 raised in their LS:

- *Is this example RAB the only one available for that type of service?*
- *If the previous statement is not right, could you provide us with the right and most suitable RAB parameters knowing the service we want to set (as described in the overall description)?*

CN3´s answer: CN3 would like to propose to take IPv6 transport also into consideration, as this IP version is mandated for the IP multimedia subsystem.

- *Is it the understanding of RAN that the end to end delay is the sum of the 2 transfer delays plus the CN delay? Are there more delays to be taken into account?*

CN3´s answer: This assumption depends upon the considered scenario. The assumption is valid only for a mobile to mobile call using 3GPP PS access on both call legs, e.g. through the IM CN subsystem. However, a mobile user may also be interconnected to a user that is using an other access to the network.
For instance, for Rel-6 CN3 is specifying the interworking between the IM CN subsystem and CS networks, such as a PSTN or a 3GPP CS domain. Here, the PS transfer delay plus a transfer delay for the IP multimedia subsystem core network, plus either a transfer delay in the PSTN (which may or may not include international connections) or a transfer delay for the 3GPP CS domain applies.
An other scenario that could be considered is the interconnection between a 3GPP PS user and a user that is directly interconnected to the Internet.

2. Date of Next CN3 Meetings:

CN3 #29	25th - 29th August 2003	Sophia Antipolis, France.
CN3 #30	27th - 31th October 2003	t.b.a

3GPP TSG-CN WG3 Meeting #28 **_N3-030414_**
San Diego, USA, 19th - 23rd May 2003.

Title:	**LS on Handling of SIP redirect messages (3xx responses)**
Response to:	
Release:	Rel-6
Work Item:	IMS CCR IWCS
Source:	CN3
To:	SA2
Cc:	

Contact Person:

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Attachments: NONE

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1. Overall Description:

CN3 is considering the specification of the handling of SIP redirection (3xx) responses for the IMS CS network interworking in TS 29.163 for Rel-6. CN3 would like to ask for guidance from SA2 on the issue, especially which scenario to standardize and how to handle the charging.

CN3 understands that there are various alternatives or options to handle the 3xx responses within this interworking scenario, with possibly different problems to be solved, e.g. what kind of a charging scenario to apply.

CN3 thinks that at least the following alternatives or options to handle the SIP redirection (3xx) responses should be considered:

1. Call release procedures are started with a cause code indicating that interworking is not possible.
2. A new INVITE is constructed using the URI received in the Contact header.
3. If the CS side network makes it possible, call re-routing is done (first) in the CS network.

2. Actions:

To SA2 group.

ACTION: CN3 asks SA2 group to give advice on the preferred scenario to handle the SIP redirection (3xx) interworking with CS networks and on the charging scenario to be used.

3. Date of Next CN3 Meetings:

CN3#29 25th - 29th August 2003 Sophia Antipolis, France.

3GPP TR 29.962 V2.0.0 (2003-05)

Technical Report

3rd Generation Partnership Project;
Technical Specification Group Core Network
Signalling interworking between the 3GPP profile of the Session Initiation Protocol (SIP) and non-3GPP SIP usage
(Release 6)

Keywords

Core Network, SIP Interworking

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Contents

Foreword

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

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

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

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

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

1 Scope

The present document investigates the SIP signalling interworking between entities of IM CN Subsystems behaving as specified in the 3GPP profile of SIP and SDP in TS 24.229 [1], with related call flow examples in TS 24.228 [2], and SIP network entities external to the IM CN subsystems, which may not adhere to the 3GPP profile of SIP and SDP.

The present document assumes that GPRS access and service based local policy using the Go interface is applied.

Non-GPRS access to IMS may have implications on the TR, which are not yet discussed.

The considered SIP network entities external to the IM CN Subsystems may feature different SIP capabilities, such as the support of arbitrary SIP options.

The document focuses on scenarios where the non-3GPP UA does not support one or more of the following SIP extensions:

Preconditions: “Integration of Resource Management and SIP” RFC 3312 [5]

Update: “The Session Initiation Protocol UPDATE Method”, RFC 3311 [7]

100rel: “Reliability of Provisional Responses in SIP”, RFC 3262 [6]

The present document focuses on the preconditions, the update and 100rel extensions because only these extensions imply interworking issues since they require the end-to-end cooperation of both UAs.

Security interworking may also have implications on the TR, which are not yet discussed.

The present document does not make any a-priory assumptions where a possible interworking is performed within the IM CN Subsystem. Any SIP network entity within the IM CN Subsystem may take part in the interworking. The network entities that may become involved in a certain interworking topic are identified for each of these topics separately.

The present document features a discussion of topics, where an interworking is possibly required. Aspects of the 3GPP profile of SIP and SDP, which obviously do not require any interworking, are not discussed. An assessment of the impact and probability of occurrence of the discussed scenarios is also provided.

Problems due to network elements within the IM CN Subsystem, which do not or only partly satisfy the 3GPP profile of SIP and SDP, in particular non 3GPP compliant SIP UAs, are out of scope of the present document.

The present document is dedicated exclusively to issues inherent in the SIP and SDP signalling. Related topics in a wider sense, such as Ipv6 to Ipv4 address translation or user plane transcoding are out of scope.

For brevity, in what follows the above SIP extensions are only mentioned if a SIP UA does not make use them. Otherwise, it is understood that the UA makes use of them.

2 References

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

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TS 24.229: “IP multimedia Call Control Protocol based on SIP and SDP”

[2] 3GPP TS 24.228: ”Signalling flows for the IP multimedia call control based on SIP and SDP”

[3] 3GPP TS 23.228: “IP Multimedia (IM) Subsystem - Stage 2”

[4] IETF RFC 3261: “SIP: Session Initiation Protocol”

[5] IETF RFC 3312: “Integration of Resource Management and SIP”

[6] IETF RFC 3262: “Reliability of Provisional Responses in SIP”

[7] IETF RFC 3311: “The Session Initiation Protocol UPDATE Method”

[8] IETF RFC 3264: “An Offer/Answer Model with SDP”

[9] 3GPP TS 29.208: “End to end Quality of Service (QoS) signalling flows”

[10] 3GPP TS 32.225: “Charging Management: Charging Data Description for the IP Multimedia Subsystem”

3 Definitions, symbols and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in TS 24.229 [1] and RFC 3261 [4] and the following apply.

The 3GPP profile of SIP: The specification of the usage of SIP within 3GPP networks in TS 24.229 [1].

SIP-preconditions extension: The SIP and SDP “precondition” extensions, as defined in RFC 3312 [5]

SIP update extension: The SIP “update” extension, including the SIP “UPDATE” method, as defined in RFC 3311 [7]

SIP 100rel extension: The SIP “100rel” extension, including the SIP “PRACK” method, as defined in RFC 3262 [6]

Not making use of the SIP 100rel extension: the UA is either supporting the SIP 100rel extension but not willing to use it, or not supporting it at all.

Not making use of the SIP update extension: the UA is either supporting the SIP update extension but not willing to use it, or not supporting it at all.

Not making use of the SIP precondition extension: the UA is either supporting the SIP precondition extension but not willing to use it, or not supporting it at all.

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in TS 24.229 [1] and RFC 3261 [4] apply.

4 Session Setup from Calling 3GPP UA towards Called non-3GPP UA

Each topic is contained in an own subsection with the structure defined in Annex A.

The following scenarios are not considered, since they are not compliant with RFC 3312 [5], Section 11:

- Session Setup towards non-3GPP UA not making use of the SIP 100rel extension.
- Session Setup towards non-3GPP UA not making use of the SIP update extension.
- Session Setup towards non-3GPP UA not making use of the SIP 100rel extension and the SIP update extension.

In the SIP signalling flows of the different scenarios, a UA that supports the SIP preconditions extension shall also support the SIP 100rel extension and the SIP Update extension therefore it includes the "precondition" tag in the Require or in the Supported header, the "100rel" tag in the Supported header and the "Update" tag in the Allow header.

4.1 Session Setup towards a non-3GPP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

4.1.1 Description of interworking issue

Since the originating 3GPP UA requires the SIP precondition extension in the SIP INVITE request, the call will fail.

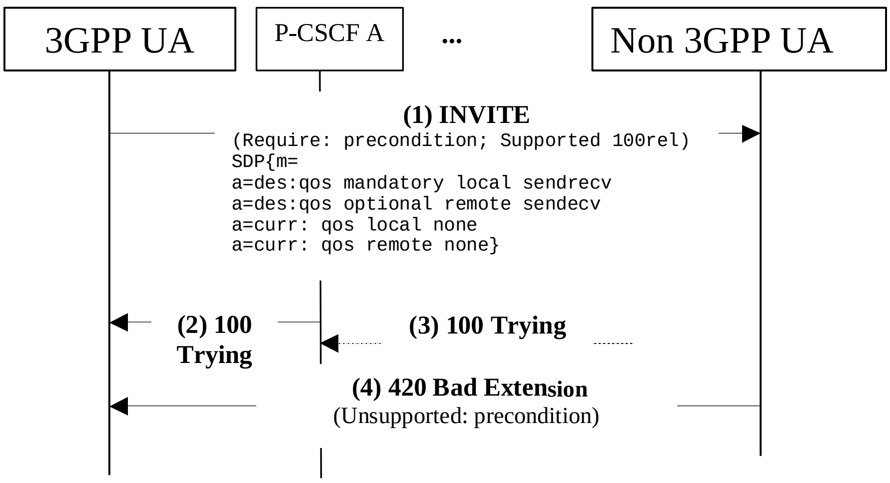

Figure 4.1.1/1: Session Setup towards a non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension.

4.1.2 Proposed Resolution B2BUA

A B2BUA is used.

Insertion of B2BUA

A B2BUA is permanently inserted at connections between the IMS and a given external network. This B2BUA handles all SIP signalling, including session attempts, subscriptions, instant messaging, etc, including signalling where the flows may forward without B2BUA intervention.

In the ideal case, the originating S-CSCF should insert the B2BUA for all the SIP signalling attempts when the destination network is outside 3GPP. However, the originating S-CSCF does not have any means, according to 3GPP TS 24.229, to decide when the call is destined for a 3GPP network or not. As a consequence, the only solution we can provide is for the originating S-CSCF to statically insert the B2BUA for all the signalling that it is leaving the home network.

New functionality is required in the S-CSCF to decide by routing criteria if a call leaves the home network.

The B2BUA becomes active only when receiving a 420 (Bad Extension) response with the "Unsupported" header field including the "preconditions" tag from the non-3GPP UA, as depicted in Figure 4.1.2/1. Note however that this behaviour is a hack to the protocol SIP, because an entity should decide its behaviour (proxy or UA) prior to forwarding any request or generating any response. Among other things, population and interpretation of certain headers (such a Contact, Proxy-Require, Require, etc.) will depend on the behaviour of the entity. Therefore, it is not possible for a entity to start behaving as a SIP proxy, and upon the reception of a response, change its behaviour to a B2BUA.

The B2BUA shall store the SDP offer in initial INVITE requests for all calls until receiving a provisional or final response from the Non-3GPP UA.

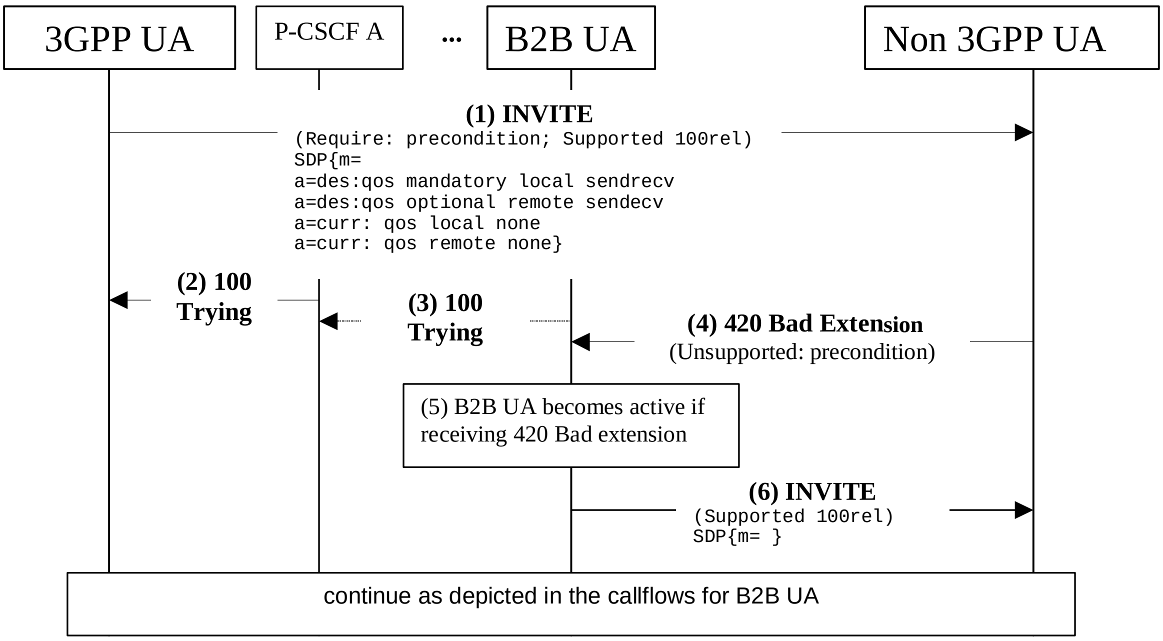

Figure 4.1.2/1: Activation of static B2BUA connecting 3GPP UA to non-3GPP UA not making use of the SIP preconditions extension

Functionality of B2BUA

The B2BUA shall apply the following rules:

1. The B2BUA shall behave as a SIP UA according to RFC 3261 and the SIP 100rel and update extensions on both call legs.

2. On the non-IMS side, the B2BUA shall also comply to the SIP 100rel and update extensions.

3. On the IMS side, the B2BUA shall also comply to and apply the SIP 100rel, update and precondition extensions.

4. The B2BUA shall forward SIP requests arriving at one call leg to the call leg at the other side, with the exceptions below.

5. The B2BUA shall forward SIP provisional responses arriving at one call leg to the call leg at the other side, provided that a transaction at the other call leg is open and with the exceptions below.

6. The B2BUA shall forward SIP final responses arriving at one call leg to the call leg at the other side, with the exceptions below.

7. The B2BUA shall not require the SIP preconditions extension on the non-IMS side.

8. The B2BUA shall insert preconditions in SDP sent from non-3GPP UA to 3GPP UA.

9. The B2BUA should remove preconditions in SDP sent from 3GPP UA to non-3GPP UA

10. The B2BUA shall not forward PRACK requests and 200 (OK) responses for the PRACK request.

11. The B2BUA shall delay forwarding a 200 (OK) response for an INVITE request from the non-IMS side to the IMS side until the mandatory preconditions are met on the IMS side.

12. The B2BUA shall handle subsequent SDP offers on the IMS side in an INVITE transaction locally, if only the preconditions are modified

13. If the B2BUA receives a subsequent SDP offers on the IMS side with modified media, it shall suspend the transaction on the IMS side and forward this SDP offer to a re-INVITE request on the non-IMS Side. The B2BUA shall forward the SDP answer received in the re-INVITE request on the non-IMS side to the appropriate message according to the rules for the transport of SDP offer answer pairs in RFC 3261 and continue with the transaction on the IMS side.

14. The B2BUA shall forward an SDP answer within the 200 (OK) response for the INVITE request of the original INVITE request from the non-IMS side to a provisional response on the IMS side.

15. For a re-INVITE request from the Non-IMS side to the IMS side, the B2BUA shall apply the rules in Section 5.1.2.

The B2BUA relies requests and responses as indicated by the red dotted arrows in the figures below.

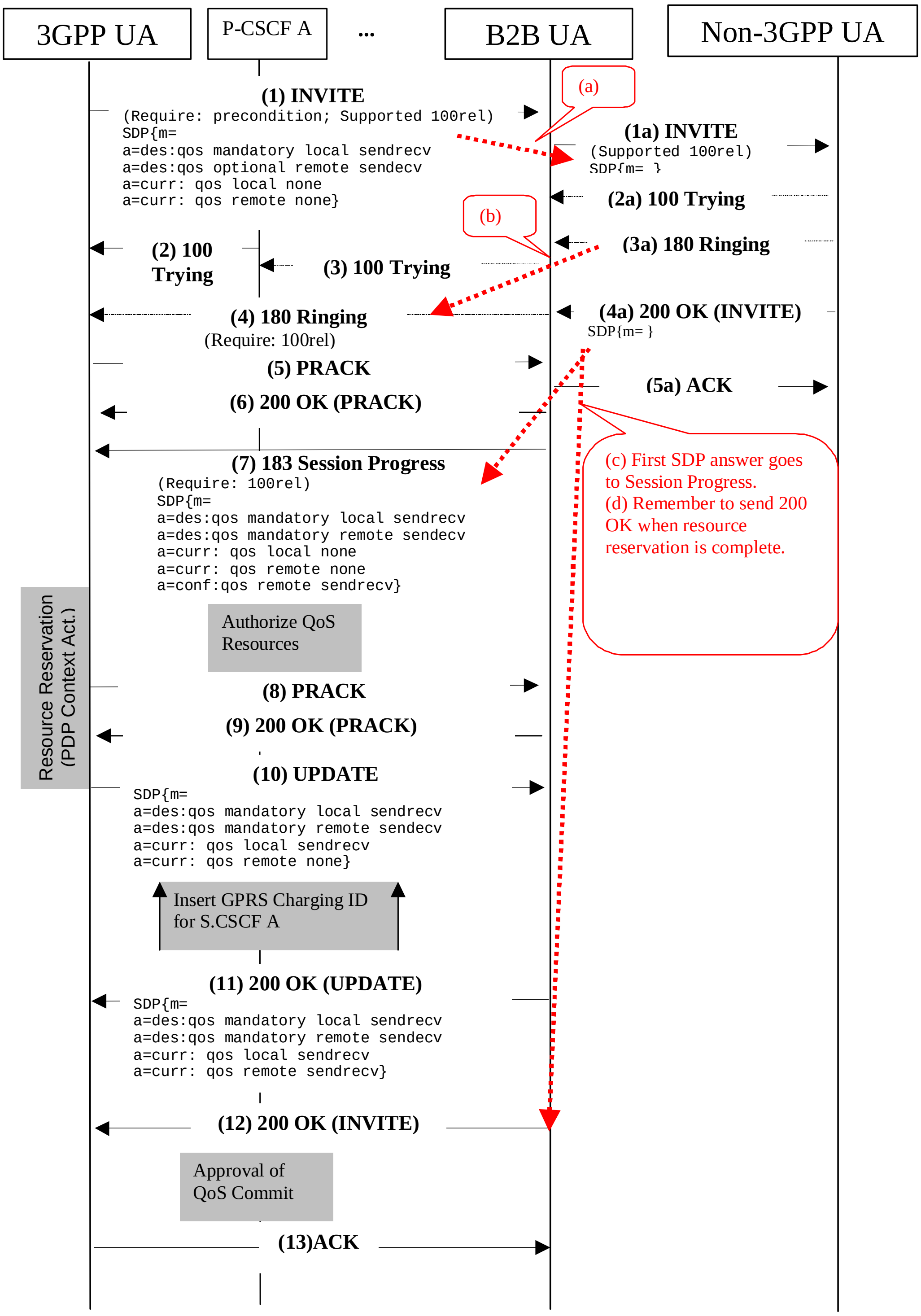

Figure 4.1.2/2: Functionality of B2BUA connecting an originating 3GPP UA to a terminating non-3GPP UA not making use of the SIP preconditions extension, the SIP update extension and the SIP 100rel extension. The originating UA sends no second SDP offer

There may be re-transmissions of the INVITE (1) by the 3GPP UA, which should be forwarded transparently by the B2BUA, as indicated in interaction (a).

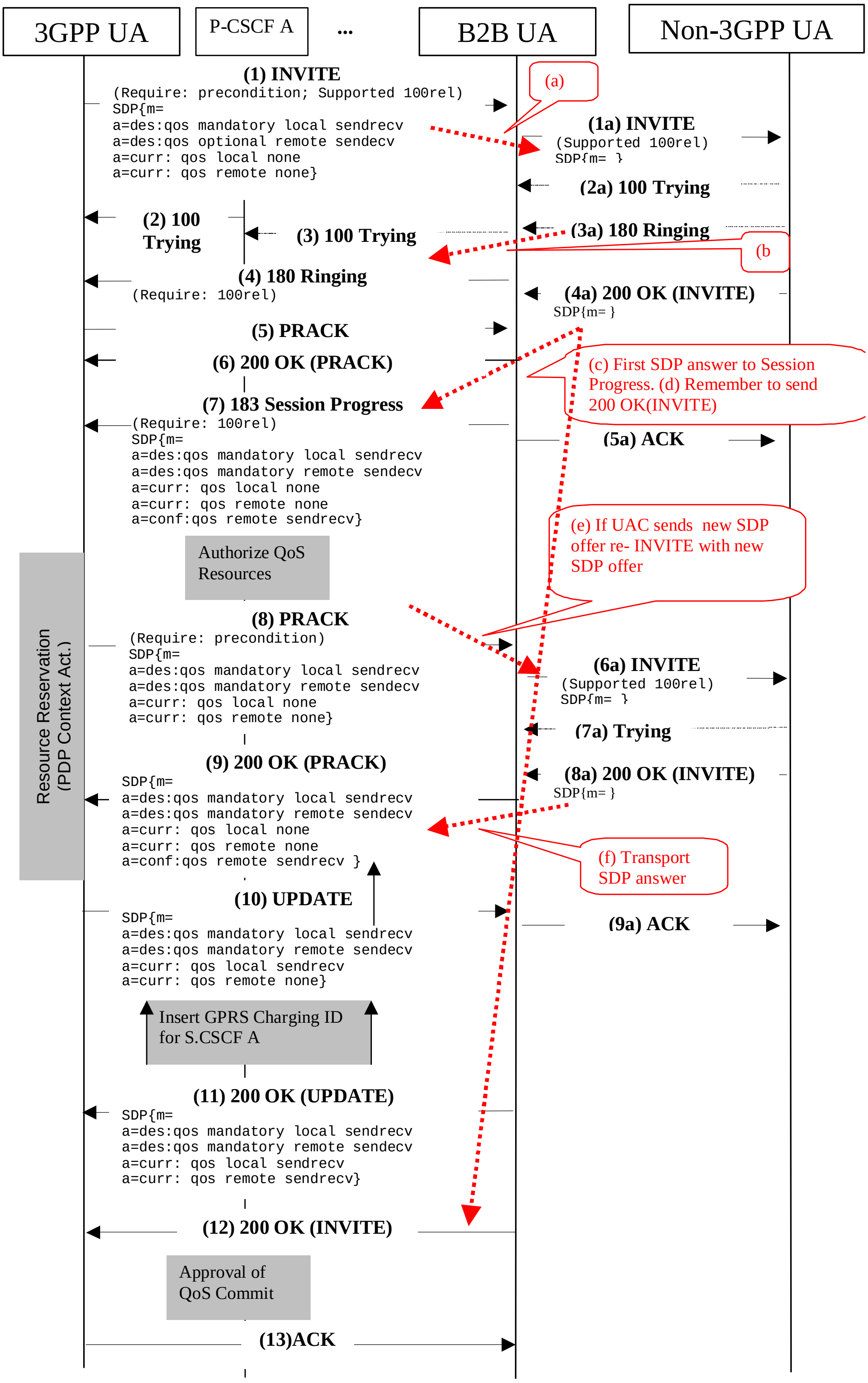

Figure 4.1.2/3: Functionality of B2BUA connecting an originating 3GPP UA to a terminating non-3GPP UA not making use of the SIP preconditions extension, the SIP update extension and the SIP 100rel extension. The originating UA sends second SDP offer

Implications of the above solution are discussed in Section 6.1.

4.1.3 Proposed Resolution Modified end-to-end Call Flow

The following changes need to be introduced in 3GPP specifications:

1. (e.g. in TS 24.229) The originating 3GPP UA should (not shall) require preconditions in an initial INVITE request. The originating 3GPP UA may (re-)INVITE an external UA without requiring preconditions (but only indicating the support for it), e.g. if receiving a 420 (Bad Extension) response including an "Unsupported" header field with the value of "precondition". In this case, in order to avoid the non-3GPP UA to send media to the 3GPP UA, the 3GPP UA shall set the media to “inactive” when generating an SDP offer. The 3GPP UA may send a re-INVITE activating the media by setting them to “send”, ”recv”, or “sendrecv” in SDP once the local resource reservation is complete.

2. (e.g. in TS 24.229) The terminating non-3GPP UA may send provisional responses without requiring the 100rel extension. The terminating non-3GPP UA may also send a 200 (OK) response for an INVITE request before the 3GPP UA has complete the resource reservation, but will not send media, because it was requested in the SDP offer (media was inactive) by the 3GPP UA. The 3GPP UA may send a re-INVITE activating the media by setting them to “send”, ”recv”, or “sendrecv” in SDP once the local resource reservation is complete.

3. (e.g. in TS 29.207 and 29.208) The P-CSCF(PDF) shall approve the QoS Commit when receiving a 200 (OK) response for an INVITE request only, if media streams are active (“send”, ”recv”, or “sendrecv” in SDP). To avoid fraud, P-CSCF(PDF) should not open gates for inactive media streams.

4. (e.g. in SA5 specifications): P-CSCF and S-CSCF shall notify the charging subsystem of a session establishment only when receiving a 200 (OK) response for an INVITErequest and media streams are active (“send”, ”recv”, or “sendrecv” in SDP).

5. (e.g. in TS 24.229): GPRS Charging ID is transported in INVITE request .

The following additional change can be introduced to avoid error situations when interworking with external SIP UAs not supporting the “inactive” SDP attribute

6. (e.g. TS 29.207 and 29.208): P-CSCF and S-CSCF shall treat media in a SDP answer as “inactive” with respect to the rules above, ignoring any other setting, if the media were set to “inactive” in the SDP offer.

Rules for the session setup from a non-3GPP UA towards a 3GPP UA are listed in Section 5.3.1

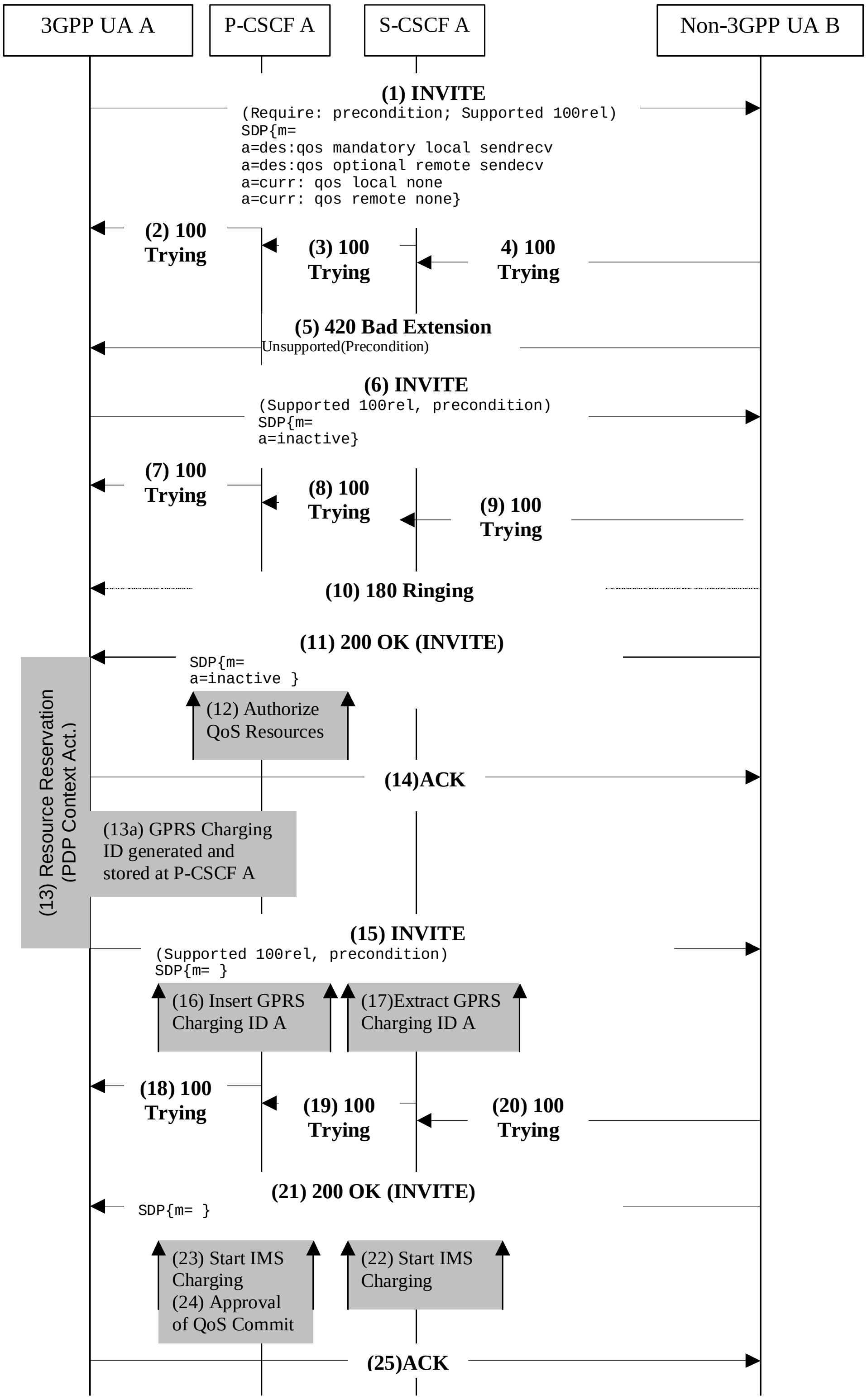

Figure 4.1.3/1: Using re-INVITE to connect an originating 3GPP UA to a terminating non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension.

Implications of the above solution are discussed in Section 6.2.

4.2 Session Setup towards a non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension

4.2.1 Description of interworking issue

The call fails, as detailed in Section 4.1.1.

4.2.2 Proposed resolution B2BUA

Insertion of B2BUA

The Insertion of the B2BUA is detailed in Section 4.1.2.

Functionality of B2BUA

The B2BUA applies the rules detailed in Section 4.1.2.

The B2BUA relies requests and responses as indicated by the red dotted arrows in the figures below.

The terminating UA may also send no 183 (Session progress) response and include the SDP answer in the 200 (OK) response for the INVITE request instead. This case is discussed in Section 4.1.2.

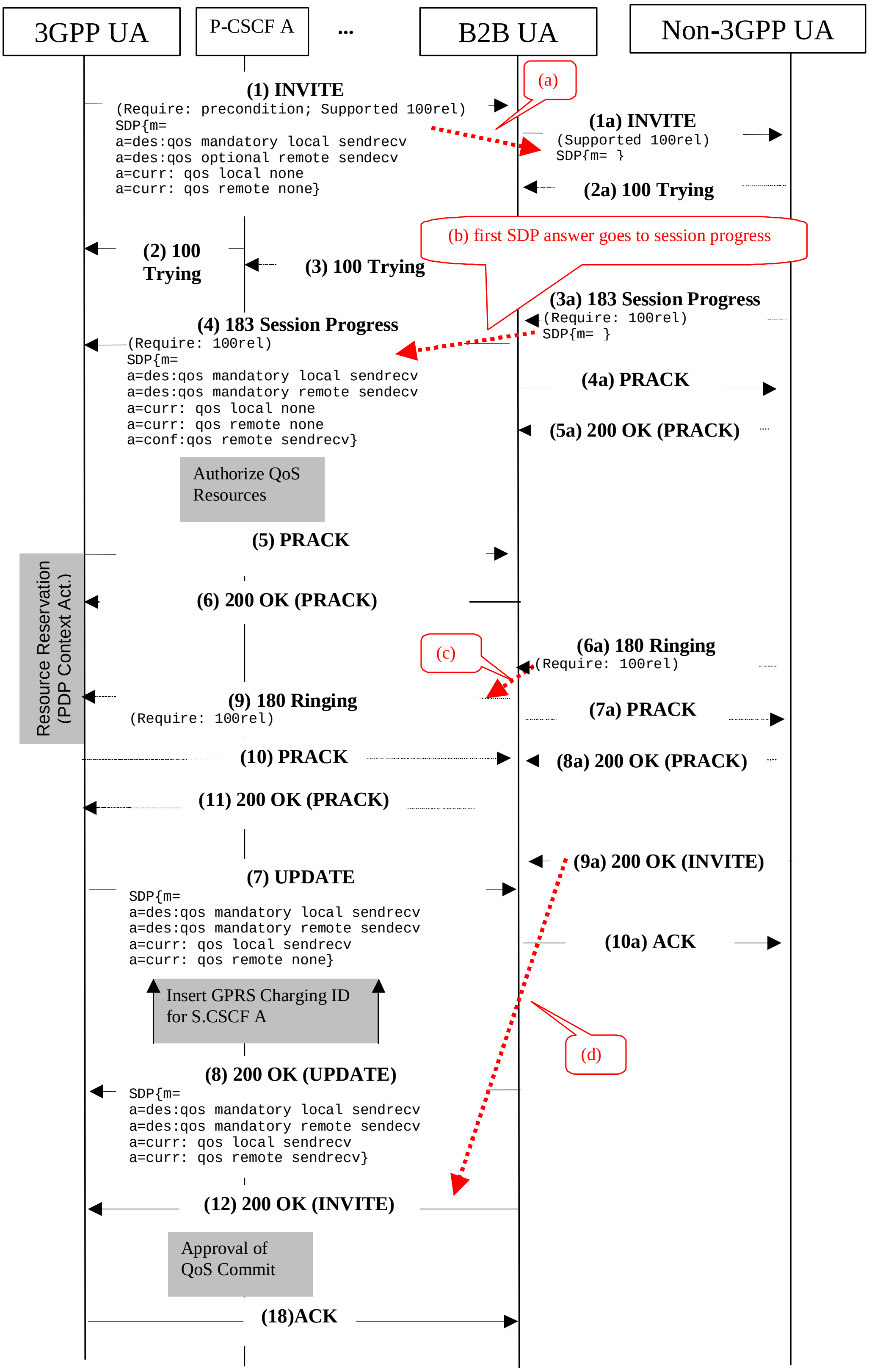

Figure 4.2.2/1: Functionality of B2BUA connecting 3GPP UA to non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension. The originating UA includes SDP answer in 183 "Session Progress". The terminating UA sends no second SDP offer.

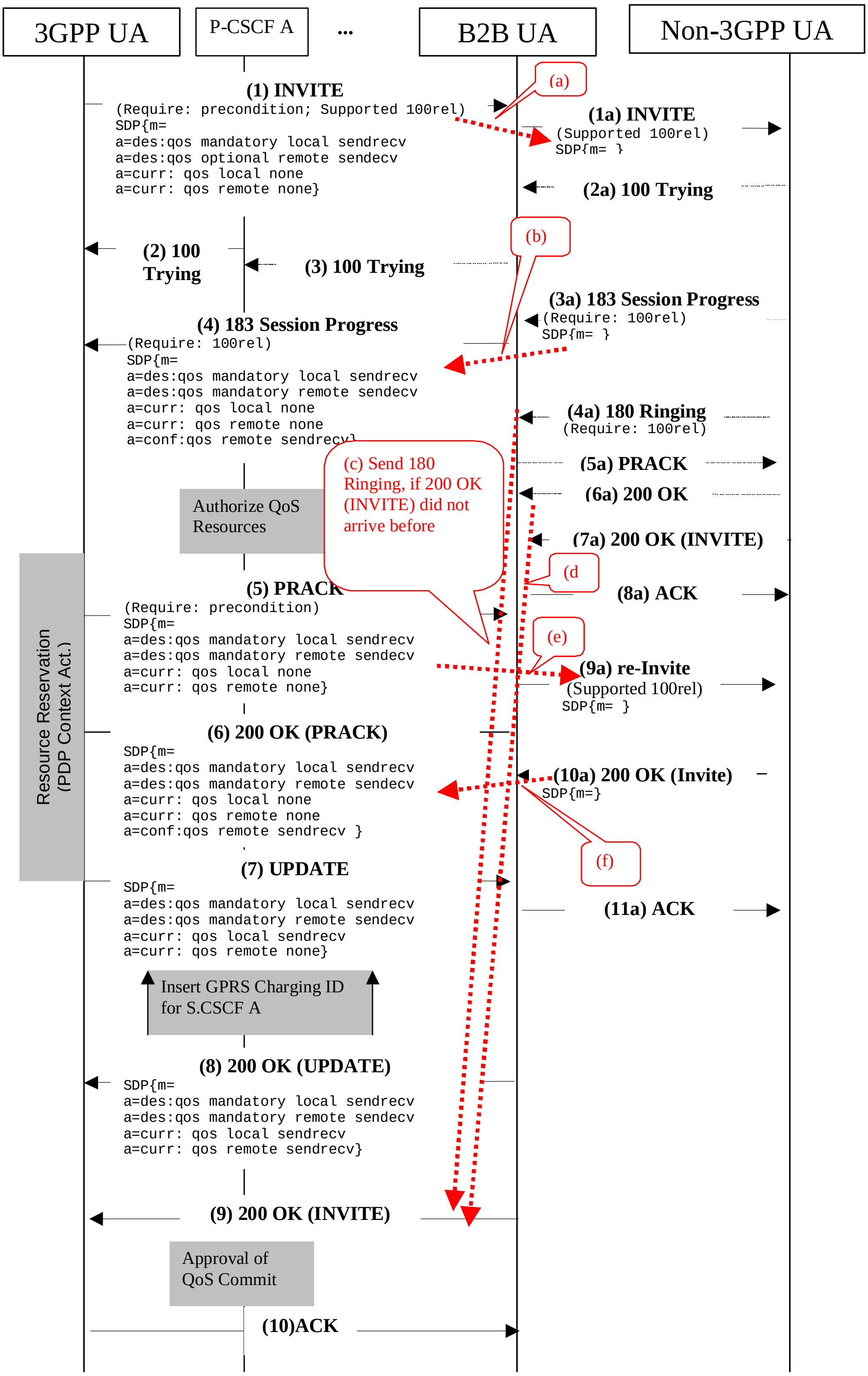

Figure 4.2.2/2: Functionality of B2BUA connecting 3GPP UA to non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension. The terminating UA includes SDP answer in 183 "Session Progress". The originating UA sends second SDP offer.

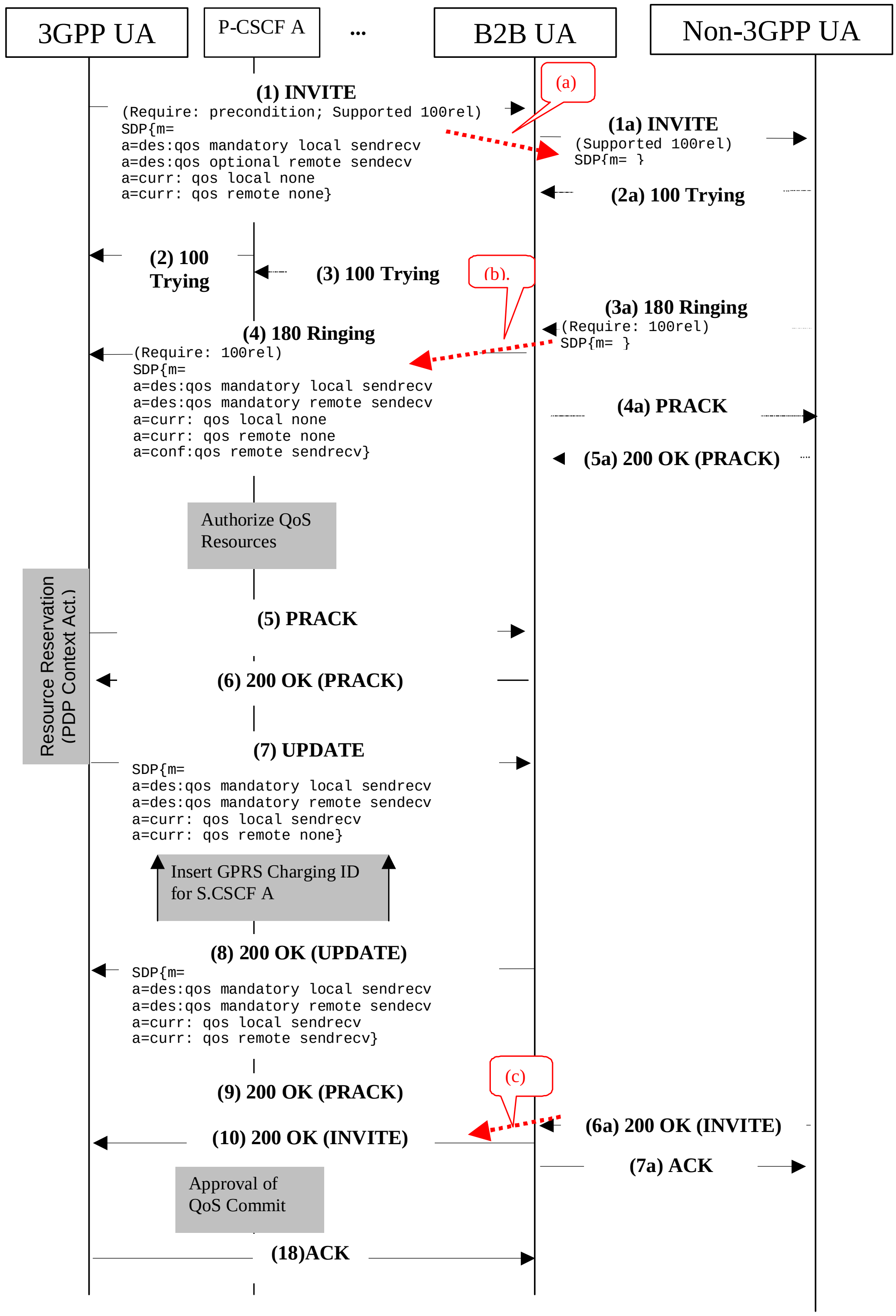

Figure 4.2.2/3: Functionality of B2BUA connecting 3GPP UA to non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension. The terminating UA includes SDP answer in 180 "Ringing". The originating UA sends no second SDP offer.

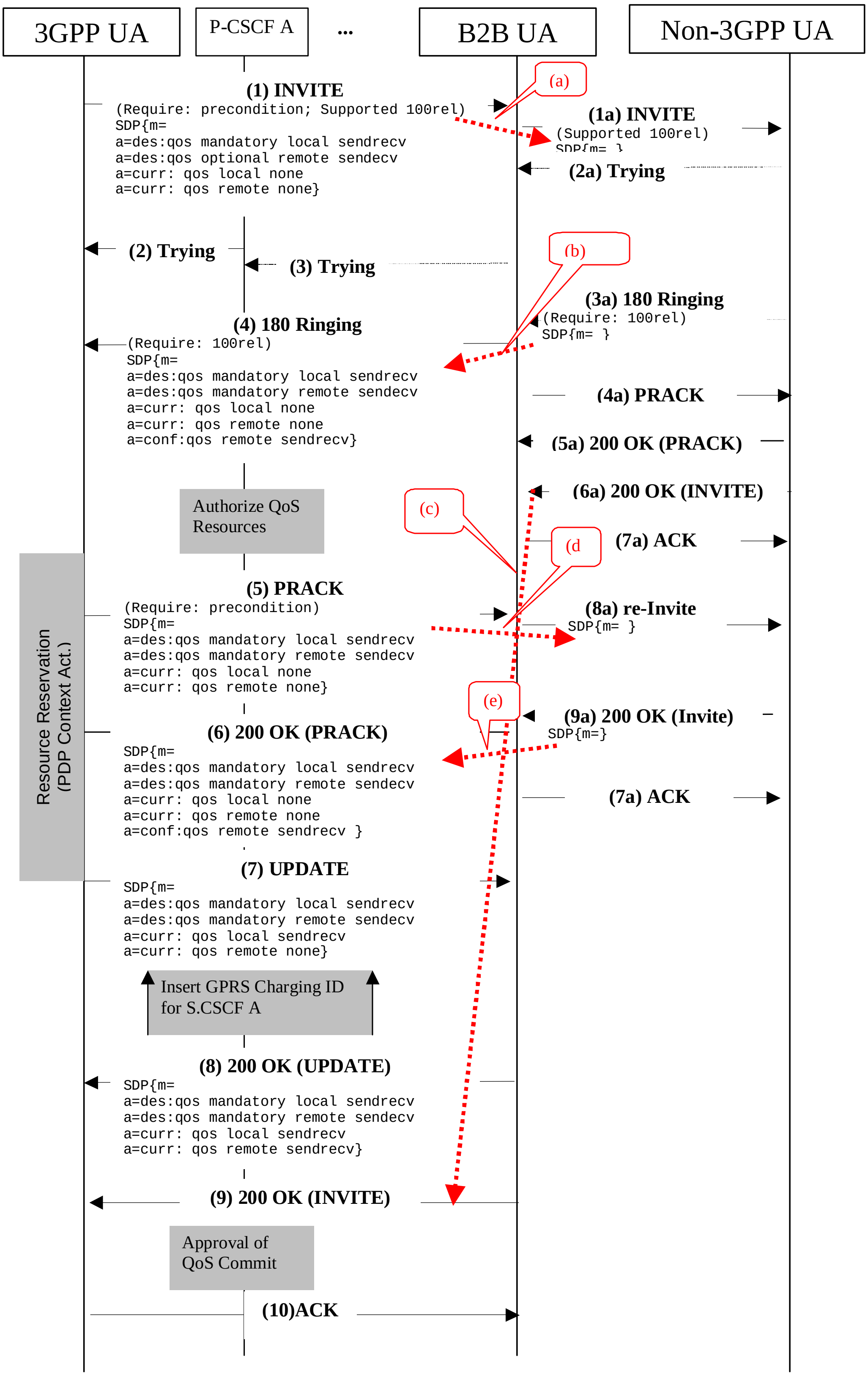

Figure 4.2.2/4: Functionality of B2BUA connecting an originating 3GPP UA to terminating non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension. The terminating UA includes SDP answer in 180 "Ringing". The terminating UA sends second SDP offer.

Implications of the above solution are detailed in Section 6.1.

4.2.3 Proposed Resolution Modified end-to-end Call Flow

The changes in 3GPP specifications described in Section 4.1.3 are applied.

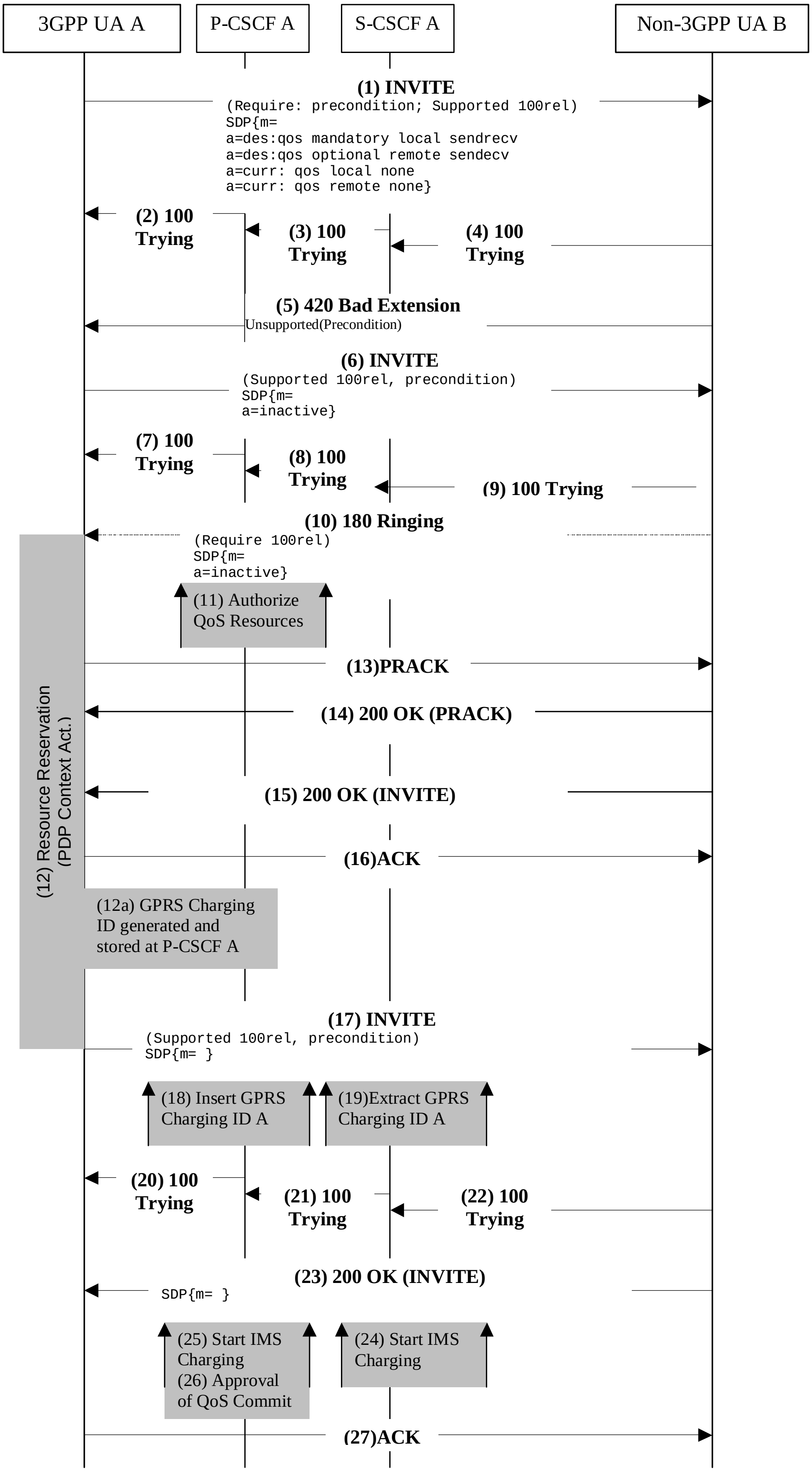

Figure 4.2.3/1: Using re-INVITE to connect an originating 3GPP UA to a terminating non-3GPP UA not making use of the SIP preconditions extension.

Implications of the above solution are detailed in Section 6.2.

4.3 Session Setup towards non-3GPP UA not making use of the SIP preconditions extension

4.3.1 Description of interworking issue

As the session attempt requires the support of the preconditions extension, and as the non-3GPP UA does not make use of such extension, the session attempt fails, as detailed in Section 4.1.1.

Within 3GPP, the SIP update extension is required to indicate the called party that the calling party has setup the bearers, and therefore, the terminating party can alert the user.

4.3.2 Proposed Resolution B2BUA

A B2BUA is used.

Insertion of B2BUA

How the B2BUA is inserted is discussed within Section 4.1.2.

Functionality of B2BUA

The functionality of the B2BUA is as discussed in Section 4.1.2.

The B2BUA shall forward additional UPDATE requests, which are not related to the precondition extension, and related provisional acknowledgement (PRACK) requests.

Implications of the above solution

General implications of the functionality of the B2BUA are discussed in Section 6.1.

Future use of UPDATE is not predictable. Thus, harmful effects may not be ruled out.

4.3.3 Proposed Resolution Modified end-to-end Call Flow

A 3GPP UA initiates a regular session attempt. This attempt requires the usage of preconditions. The non-3GPP UA returns a 420 response indicating that it does not support the preconditions extension. The 3GPP UA initiates a second INVITE attempt, in this case without requiring the usage of the preconditions extension. The session attempt shall contain an SDP offer which sets the media streams in "inactive" to avoid receive media at this time. When the 3GPP UA gets all the bearers setup, it re-INVITEs the calling party resuming the "inactive" media. This indicates the non-3GPP UA that the 3GPP UA is ready to receive media.

The changes in 3GPP specifications described in Section 4.1.3. are applied.

The resulting call flow is similar to Figure 4.2.3/1, possibly with additional SIP UPDATE requests.

Implications of the above solution are detailed in Section 6.2.

5 Session Setup from Calling non-3GPP UA towards Called 3GPP UA

Each topic is contained in an own subsection with the structure defined in Annex A.

The following scenarios are not considered, since they are not compliant with RFC 3312 [5], Section 11:

- Session Setup from non-3GPP UA not making use of the SIP 100rel extension.
- Session Setup from non-3GPP UA not making use of the SIP update extension.
- Session Setup from non-3GPP UA not making use of the SIP update extension and the SIP 100rel extension.

5.1 Session Setup from non-3GPP SIP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

5.1.1 Description of interworking issue

Since the terminating 3GPP UA mandates the support of the SIP precondition extension in the SIP INVITE request, the call will be aborted.

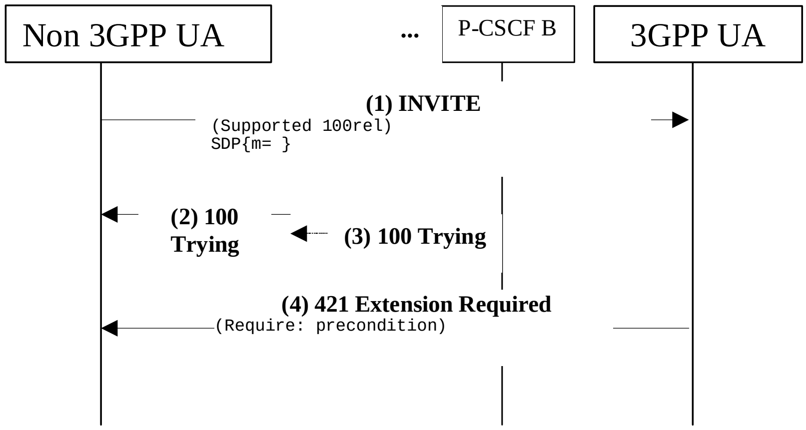

Figure 5.1.1/1: Session Setup from non-3GPP SIP UA not making use of the SIP preconditions extension and the SIP update extension, to 3GPP UA.

5.1.2 Proposed Resolution B2BUA

A B2BUA is used.

<u>Insertion of B2BUA</u>

A B2BUA is permanently inserted at connection between the home operator and any other network. This B2BUA handles all calls, including calls where the call flows may be forwarded without modification.

The B2BUA shall be inserted in the border of the home network for all session attempts entering the home network.

To implement this, new functionality is required, e.g. a suitable DNS lookup mechanism or a decision based on routing criteria in the I-CSCF.

The B2BUA becomes active only when receiving an INVITE request without an indication of the support or requirement of the 100rel extension from the Non-3GPP UA, as depicted in Figure 5.1.2/1. Otherwise, the B2BUA forwards all SIP requests and responses received at one side to the other side. Note that even a 3GPP to 3GPP session attempts could potentially bypass the B2BUA, it is not possible to distinguish the origin of the session. As a consequence of it, the B2BUA has to be permanently inserted for all session attempts.

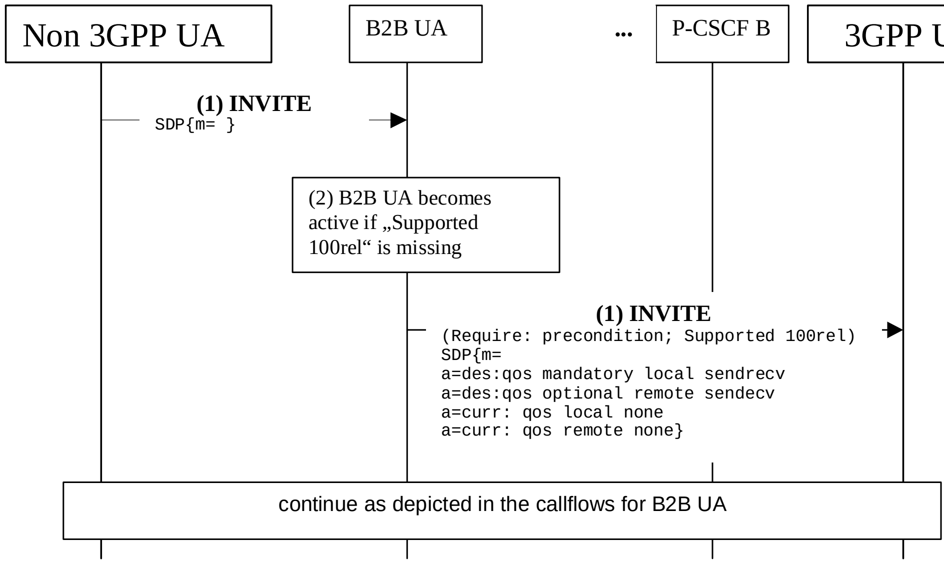

Figure 5.1.2/1: Activation of static B2B connecting Non-3GPP SIP UA not indicating support of the SIP preconditions extension to 3GPP UA.

Functionality of B2BUA

The B2BUA shall apply the following rules:

1. The B2BUA shall behave as a SIP UA according to RFC 3261 and the SIP 100rel and update extensions on both call legs.
2. On the non-IMS side, the B2BUA shall also comply with the SIP 100rel and update extensions.
3. On the IMS side, the B2BUA shall also comply to and apply the SIP 100rel, update and precondition extensions.
4. The B2BUA shall forward SIP requests arriving at one call leg to the call leg at the other side, with the exceptions below.
5. The B2BUA shall forward SIP provisional responses arriving at one call leg to the call leg at the other side, provided that a transaction at the other call leg is open and with the exceptions below.
6. The B2BUA shall forward SIP final responses arriving at one call leg to the call leg at the other side, with the exceptions below.
7. The B2BUA shall not require the SIP preconditions extension on the non-IMS side.
8. The B2BUA shall insert preconditions in SDP sent from non-3GPP UA to 3GPP UA.
9. The B2BUA should remove preconditions in SDP sent from 3GPP UA to non-3GPP UA
10. The B2BUA shall not forward PRACK requests and 200 (OK) responses for the PRACK request.
11. The B2BUA shall inspect an INVITE request from the non-IMS side to determine if the support of the 100rel extension is indicated.
12. If the 100rel extension is not supported on the non-IMS side, and the B2BUA receives an SDP offer in a provisional response from the IMS side, the B2BUA shall send the SDP offer in a 200 (OK) response for an INVITE request at the non-IMS side. The B2BUA shall then forward the SDP answer received in the ACK request from the non-IMS side to the PRACK request for the provisional response on the IMS-side.
13. If the 100rel extension is not supported on the non-IMS side, and the B2BUA receives an SDP answer in a provisional response from the IMS side, the B2BUA shall send the SDP answer in a 200 (OK) response for an INVITE request response at the non-IMS side.
14. For a re-INVITE request from the IMS side to the Non-IMS side, the B2BUA shall apply the rules in Section 4.1.2.

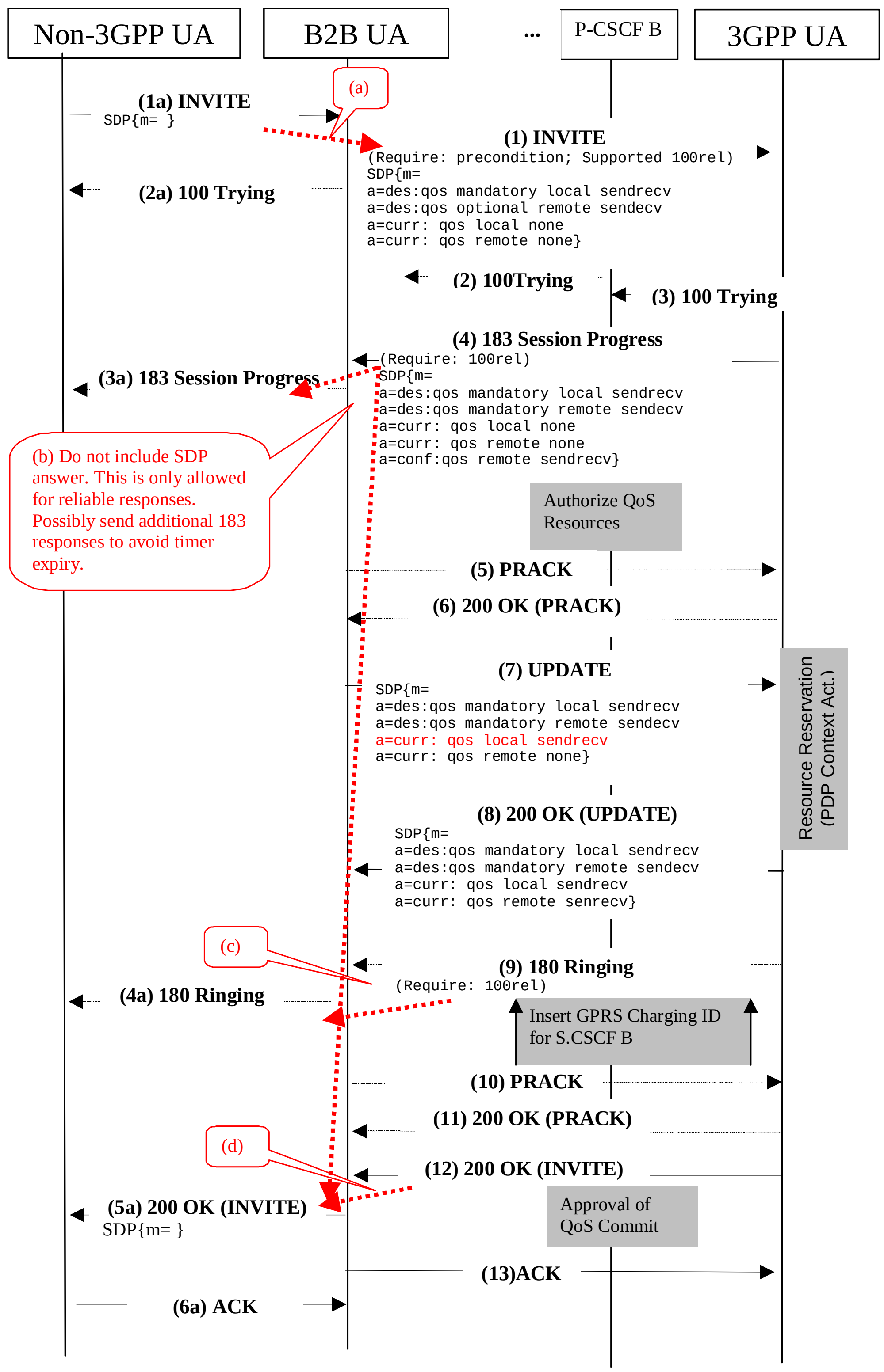

Figure 5.1.2/2: Functionality of B2BUA connecting an originating non-3GPP SIP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to a terminating 3GPP UA. SDP offer in INVITE request.

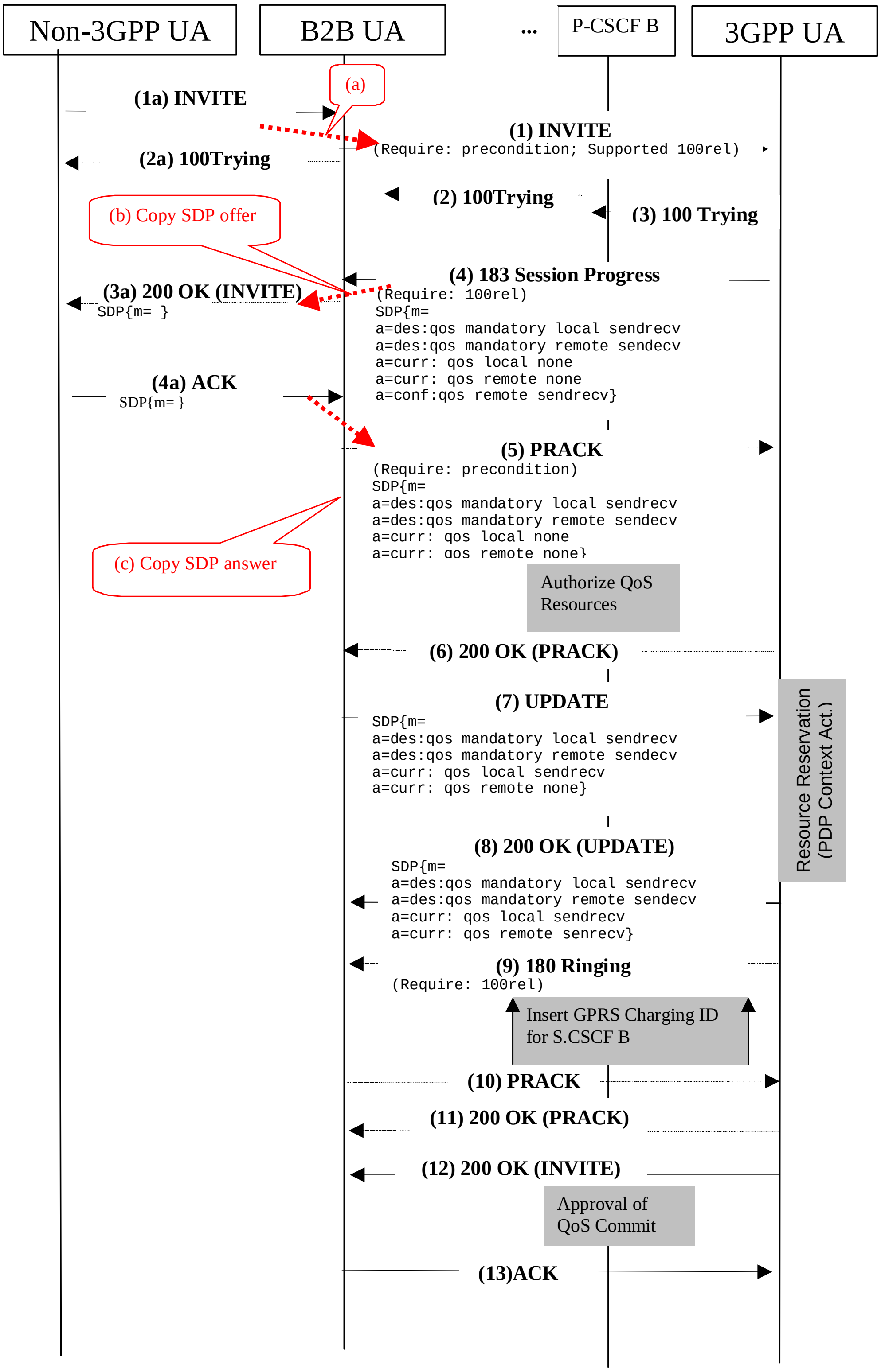

Figure 5.1.2/3: Functionality of B2BUA connecting an originating non-3GPP SIP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to a terminating 3GPP UA. SDP offer in OK response.

Implications of the above Solution are discussed in Section 6.1.

5.1.3 Proposed Resolution Modified end-to-end Call Flow

The following changes need to be introduced in 3GPP specifications:

1. (e.g. in TS 24.229) IF a 3GPP UA receives an INVITE request without the support for preconditions, and if the media in the SDP offer requires the 3GPP UA to reserve resources, the 3GPP UA shall put the media on inactive when answering the INVITE request. This avoids the non 3GPP UA to start sending media when receiving the final response to the request. The 3GPP UA shall start reserving resources and send a re-INVITE to resume the inactive media whenever it is ready with the resource reservation.

2. (e.g. in TS 24.229) If a 3GPP UA receives an INVITE request without the support for preconditions and without any SDP payload, the 3GPP UA shall put the media on inactive in the SDP offer when answering the INVITE request. When an ACK is received with an SDP answer, and the media in it requires the 3GPP UA to reserve resources, the 3GPP UA shall start reserving resources and send a re-INVITE to resume the inactive media whenever it is ready with the resource reservation.

3. (e.g. in TS 29.207 and 29.208) The P-CSCF(PDF) shall approve the QoS Commit when receiving a 200 (OK) response for an INVITE request only, if media streams are active ("send", "recv", or "sendrecv" in SDP). To avoid fraud, P-CSCF(PDF) should not open gates for inactive media streams.

4. (e.g. in SA5 specifications): P-CSCF and S-CSCF shall notify the charging subsystem of a session establishment only when receiving a 200 (OK) response for an INVITErequest and media streams are active ("send", "recv", or "sendrecv" in SDP).

5. (e.g. in TS 24.229): GPRS Charging ID is transported in the re-INVITE request .

The following additional change can be introduced to avoid error situations when interworking with external SIP UAs not supporting the "inactive" SDP attribute

6. (e.g. TS 29.207 and 29.208): P-CSCF and S-CSCF shall treat media in a SDP answer as "inactive" with respect to the rules above, ignoring any other setting, if the media were set to "inactive" in the SDP offer.

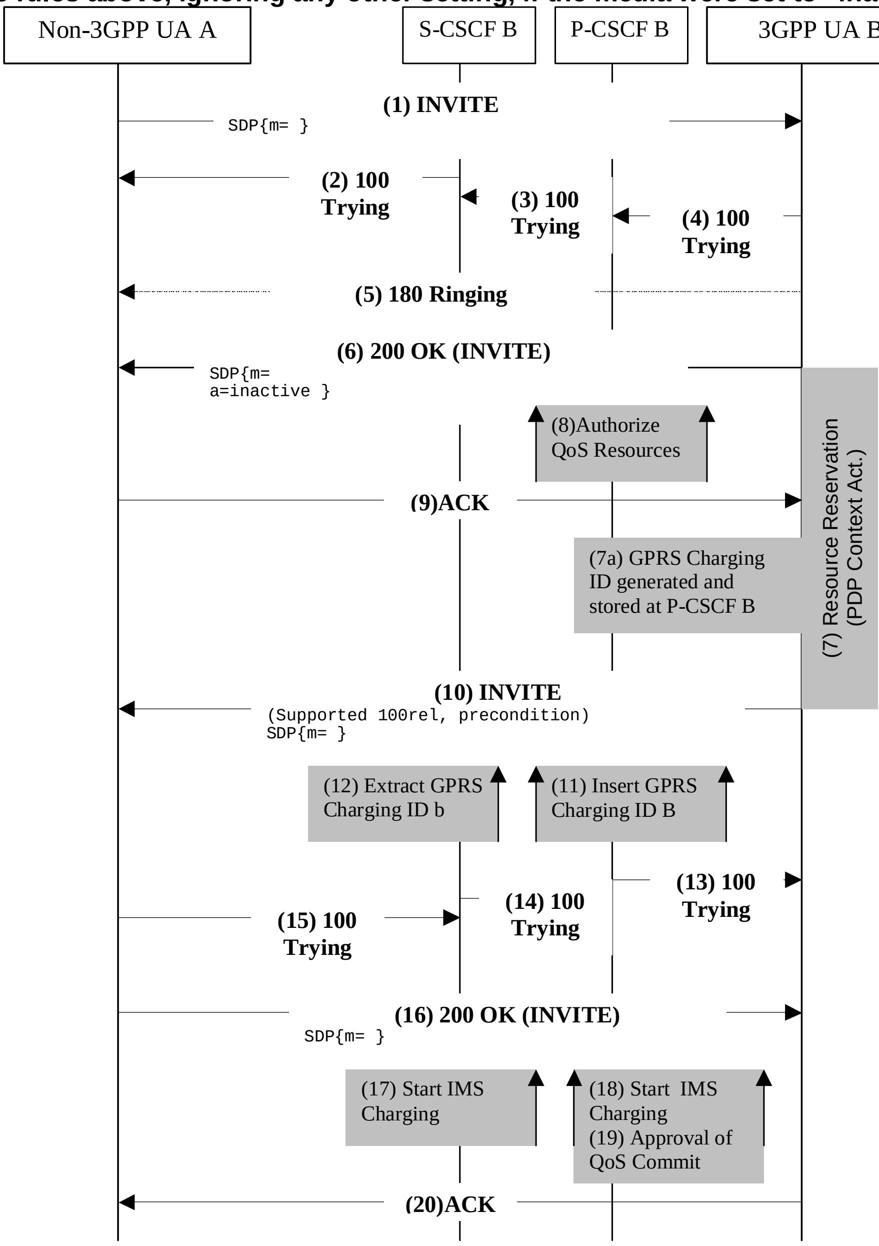

Figure 5.1.3/1: Using re-INVITE to connect an originating non-3GPP UA not making use of the SIP preconditions extension, the SIP update extension and the SIP 100rel extension to a terminating 3GPP UA. The INVITE request contains SDP.

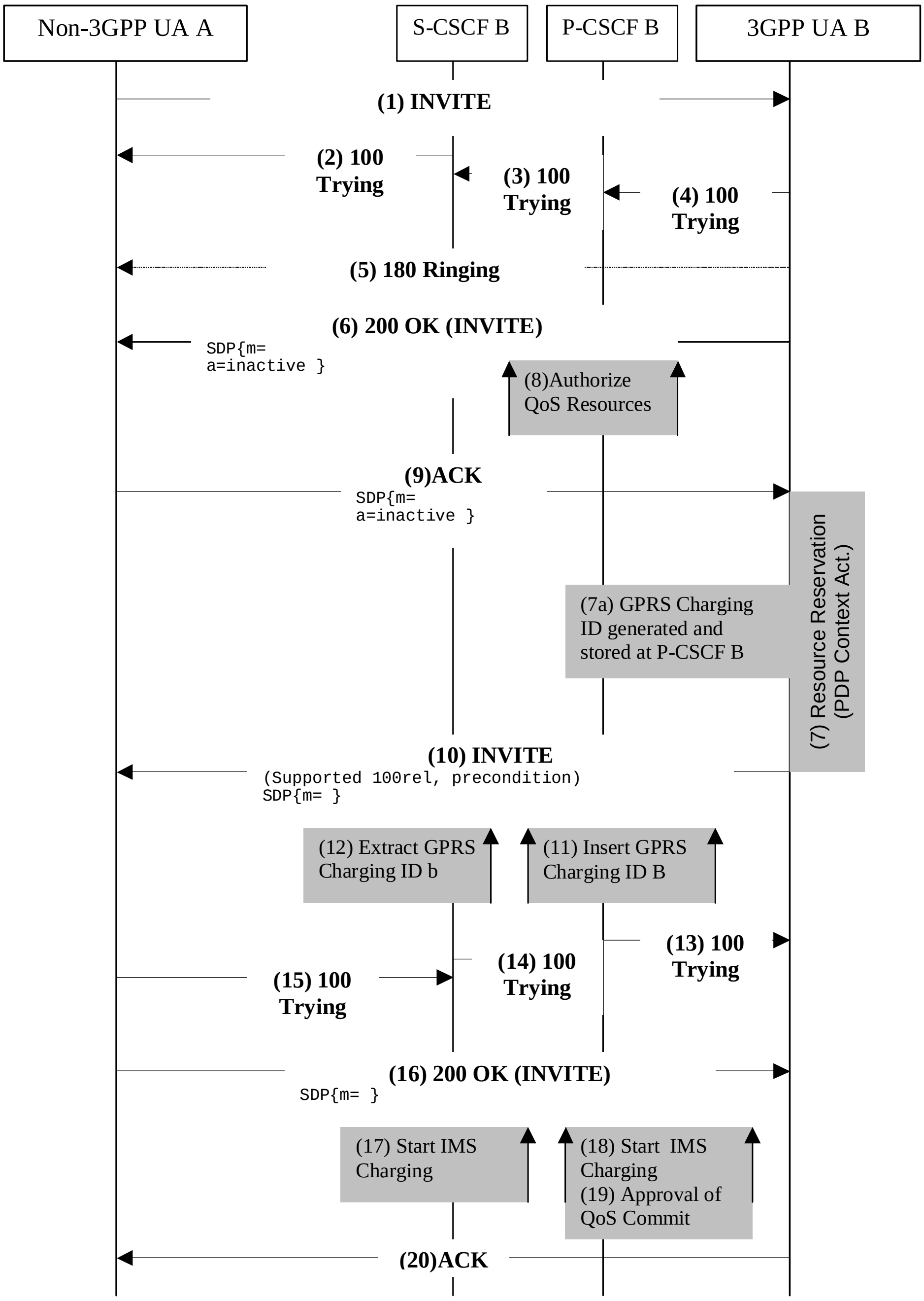

Figure 5.1.3/2: Using re-INVITE to connect an originating non-3GPP UA not making use of the SIP preconditions extension, the SIP update extension and the SIP 100rel extension to a terminating 3GPP UA. The INVITE request contains no SDP.

Implications of the above Solution are discussed in Section 6.2.

5.2 Session Setup from non-3GPP SIP UA not making use of the SIP preconditions extension and the SIP update extension

5.2.1 Description of interworking issue

The call fails, as detailed in Section 5.1.1.

5.2.2 Proposed resolution B2BUA

A B2BUA is used.

Insertion of B2BUA

A B2BUA is permanently inserted at connection between IMS and a given external network. This B2BUA handles all calls, including calls where the call flows may be forwarded without modification.

The B2BUA shall be inserted in the border of the home network for all session attempts entering the home network. Note that, even a 3GPP to 3GPP session attempt could potentially bypass the B2BUA, it is not possible to distinguish the origin of the session. As a consequence of it, the B2BUA has to be permanently inserted for all session attempts.

The B2BUA becomes active only when receiving an INVITE request without an indication of the support or requirement of the preconditions extension from the Non-3GPP UA, as depicted in Figure 5.2.2/1. Otherwise, the B2BUA forwards all SIP requests and responses received at one side to the other side. Note however that this behaviour is a hack to the protocol SIP, because an entity should decide its behaviour (proxy or UA) prior to forwarding any request. Among other things, population of certain headers (such a Contact, Proxy-Require, Require, etc.) will depend on the behaviour of the entity. Therefore, it is not possible for a entity to start behaving as a SIP proxy, and upon the reception of a response, change its behaviour to a B2BUA.

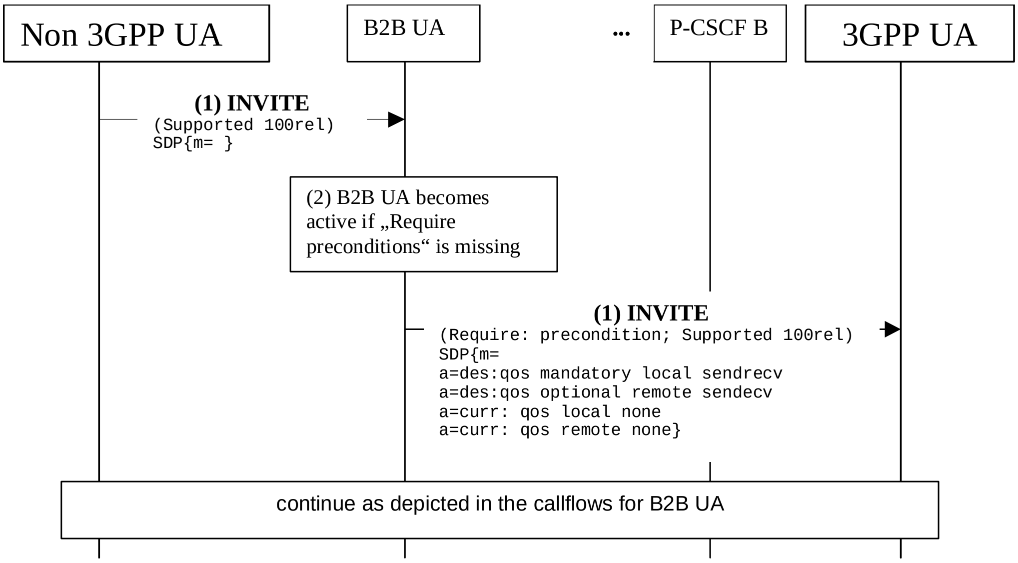

Figure 5.2.2/1: Activation of static B2BUA connecting Non-3GPP UA not indicating support of the SIP preconditions extension to 3GPP UA.

Functionality of B2BUA

The B2BUA shall apply the rules detailed in Section 5.1.2.

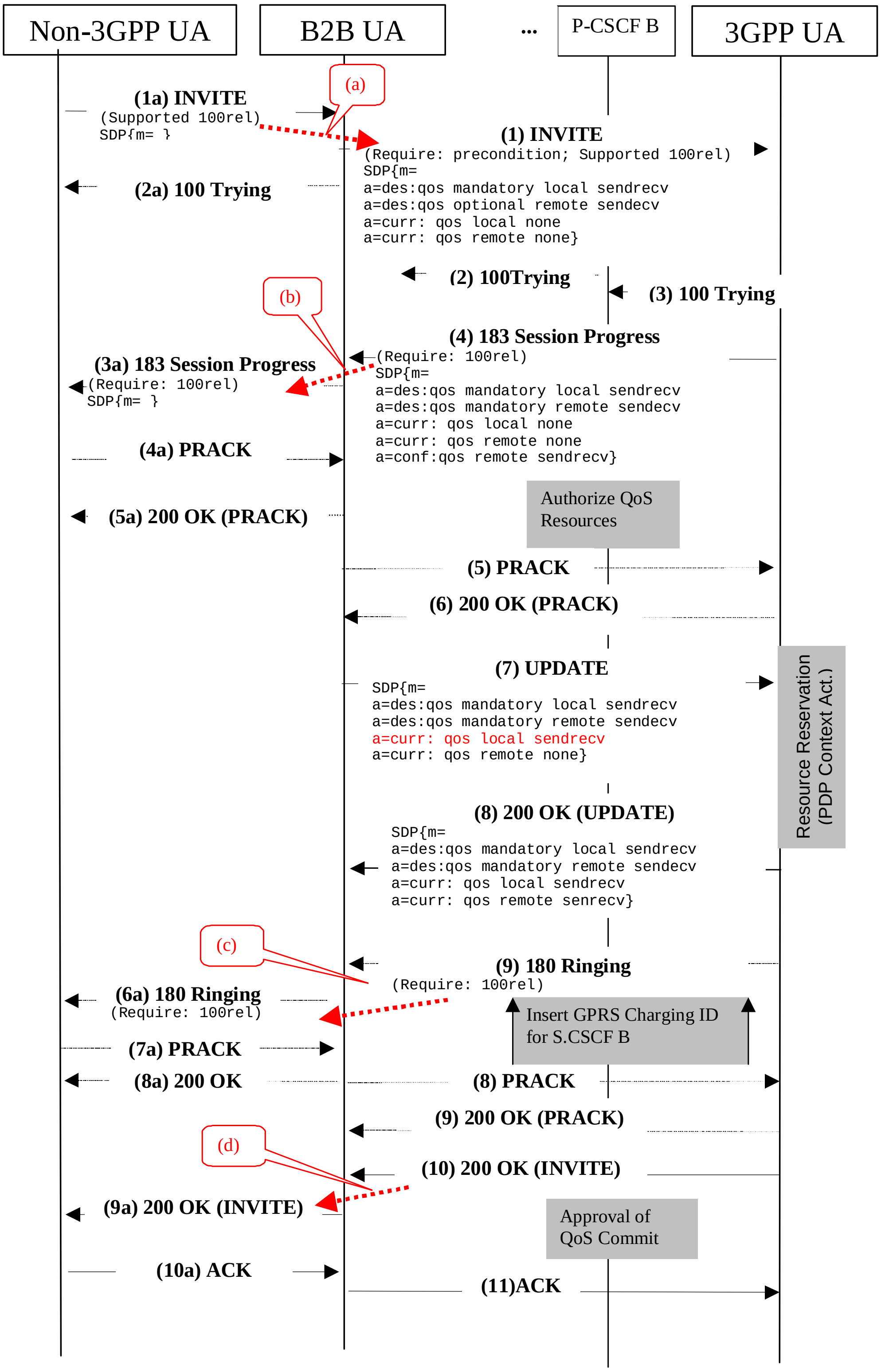

Figure 5.2.2/2: Functionality of B2BUA connecting an originating non-3GPP SIP UA not making use of the SIP preconditions extension and the SIP update extension, to a terminating 3GPP UA.

Implications of the above solution are discussed in Section 6.1.

5.2.3 Proposed Resolution Modified end-to-end Call Flow

Currently, 3GPP TS 24.229 mandates the usage of the SIP preconditions extension for incoming session attempts to a 3GPP UA. The usage of preconditions is intended to provide extra capabilities to the originating side of the session, but it is not really required at the terminating side. The restriction in 3GPP TS 24.229 to support preconditions at the terminating side does not have any effect on terminating session attempts. Therefore, it is proposed to remove such restriction.

Providing that the restriction is gone, the 3GPP UA will accept sessions even in the case there is no requirement to support preconditions from the non-3GPP UA. In this case, the 3GPP UA will answer with a 183 Session Progress response (as in normal circumstances), and will setup bearers. When the bearers are setup, the 3GPP UA will alert the user and generate a 180 Ringing response. When the user accepts the session attempt, the 3GPP UA will answer the INVITE with a 200 OK response.

The restriction to mandate the usage of the SIP preconditions extension for terminating session attempts is removed from TS 24.229.

Furthermore, the 3GPP UA shall not require the SIP preconditions extension in subsequent re-INVITE requests within the same dialogue.

Existing Charging mechanisms and Go functionality is not affected by these modifications.

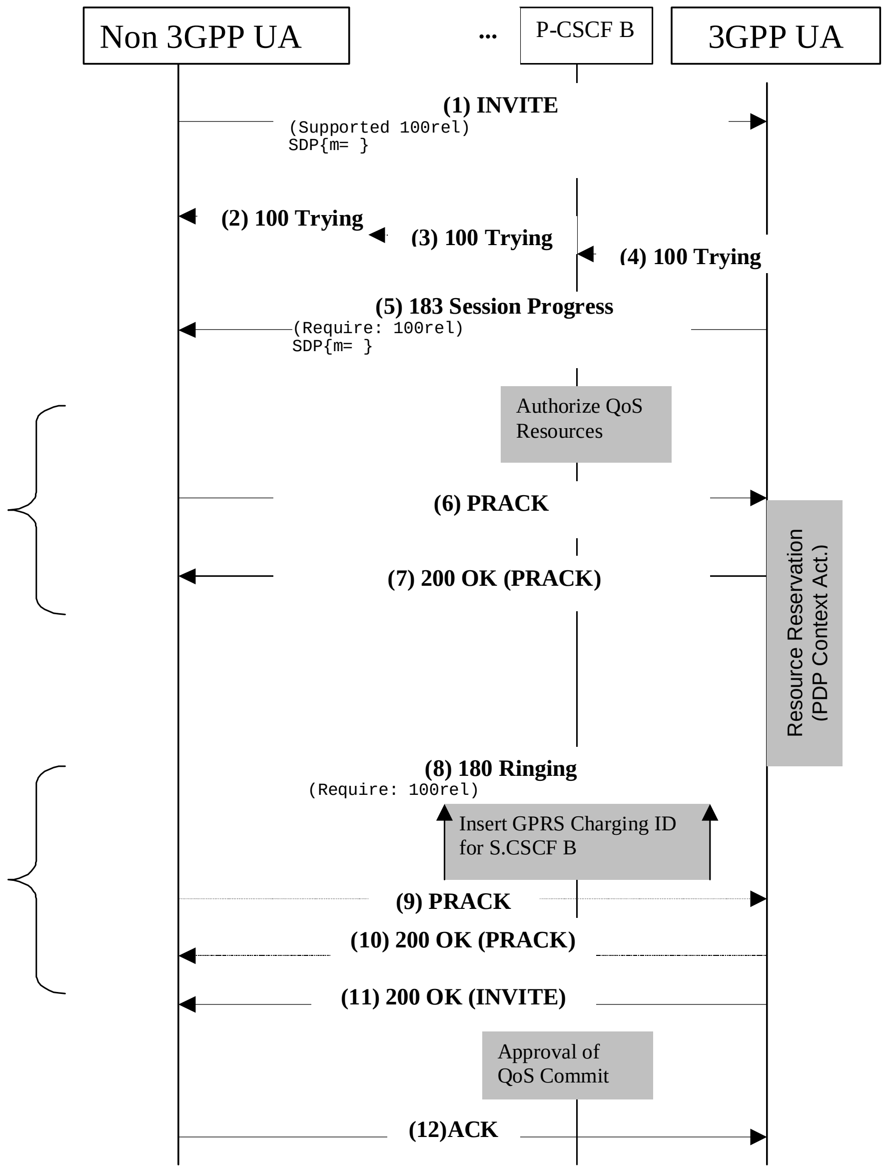

Figure 5.2.3/1: Modified end-to-end call flow for Non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension, to 3GPP UA. SDP offer in INVITE request.

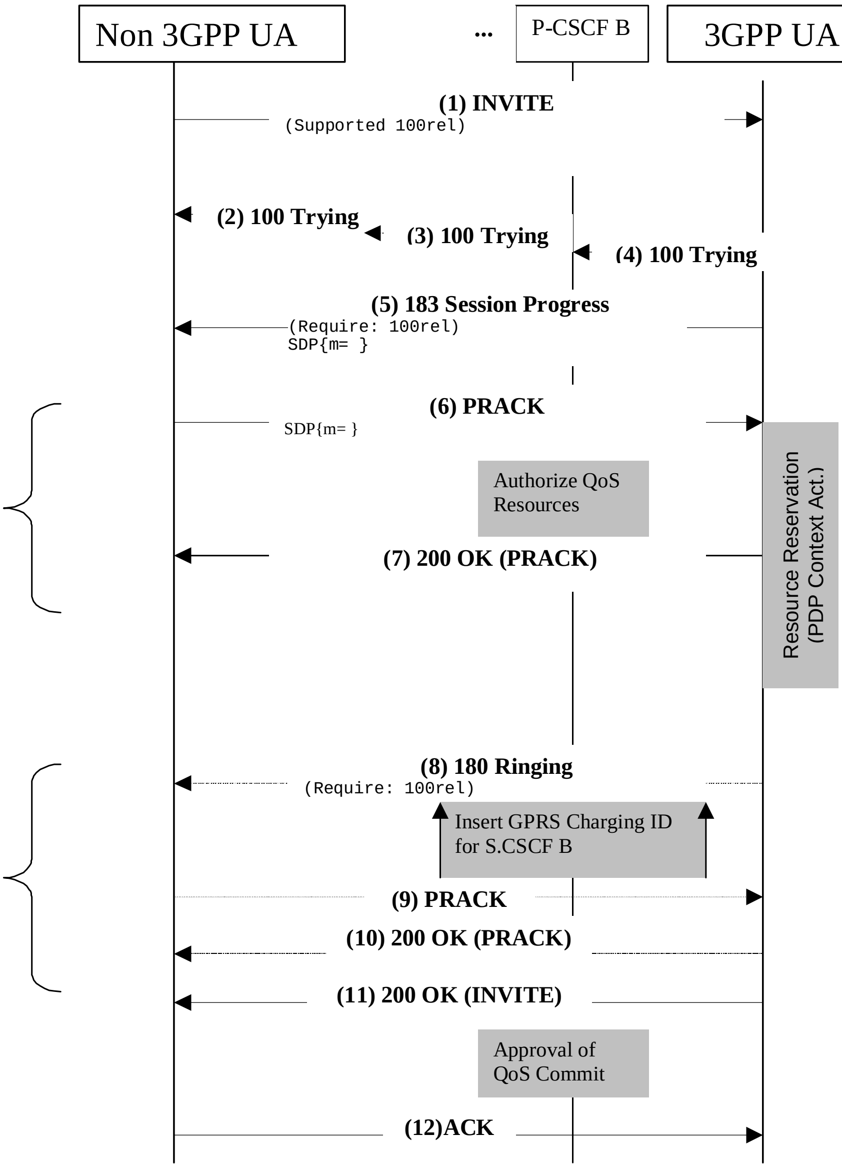

Figure 5.2.3/2: Modified end-to-end call flow for Non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension, to 3GPP UA. No SDP offer in INVITE request.

Impacts of the above solution

No modifications or extra functionality compared to Rel.5 required.

No disadvantages have been identified.

5.3 Session Setup from non-3GPP UA not making use of the SIP preconditions extension

5.3.1 Description of interworking issue

As the 3GPP UA requires the usage of the preconditions extension, and as the non-3GPP UA does not support such extensions, the session attempt fails, as detailed in Section 5.1.1.

Within 3GPP, the SIP update extension is required to convey SDP offer/answer with SDP attributes defined in the SIP precondition extension with the help of the SIP UPDATE method.

A non-3GPP UA supporting the SIP update extension may use features of this extension for purposes not related to the SIP precondition extension. As a result, various extra messages may be inserted into the call flow.

5.3.2 Proposed Resolution B2BUA

A B2BUA is used.

How the B2BUA is inserted is discussed within Section 5.2.2.

Functionality of B2BUA

The functionality of the B2BUA is as discussed in Section 5.2.2.

The B2BUA shall forward additional UPDATE requests, which are not related to the precondition extension, and related provisional acknowledgement (PRACK) requests.

Impacts of the above solution.

General impacts of the B2BUA are discussed in Section 6.1.

The originating and the terminating UA may send UPDATE requests at various places within the call flow. Those requests may include additional SDP offers. Due to the large number of possibilities, such call flows can not be depicted. The dialog state is not altered by UPDATE requests, and thus these requests probably do not have harmful side effects.

Future use of UPDATE is not predictable. Thus, harmful effects may not be ruled out.

5.3.3 Proposed Resolution Modified end-to-end Call Flow

The restriction in 3GPP TS 24.229 to mandate the usage of preconditions at the terminating side does not have any effect on terminating session attempts. Therefore, it is proposed to remove this restriction, as detailed in Section 5.2.3.

The resulting call flows are similar to the flows in Section 5.2.3, possibly with additional UPDATE requests inserted.

Impacts of the above solution

No modifications or extra functionality compared to Rel.5 required.

No disadvantages have been identified.

6 Implications of the Proposed Solutions

Editor's Note: This section shall summarise the findings within the corresponding subsections within Sections 4 and 5.

6.1 B2BUA

The functionality and implementation of the B2BUA is complicated and will have to include the following features:

- Storage of SDP.
- Own timer supervision of dialogues on the call leg at the 3GPP side and on the call leg at the non 3GPP side.

Additional processing load and additional delay may result.

The compatibility with future SIP extensions may be limited by the need to update the B2BUA. This may limit the network’s ability to deploy new IP multimedia applications. Lack of signalling transparency may restrict the compatibility with future extensions for all session attempts, subscriptions, and instant messaging.

The B2BUA is automatically in the signalling path for all communications. For a session setup from a calling non-3GPP UA towards called 3GPP UA, the B2BUA may be activated unnecessarily, if the Non-3GPP UA supports the

100rel extension, but fails to indicate this in the INVITE request. RFC 3262 [6] recommends that a UAC supporting the 100rel extension indicates this capability in the INVITE request, but does not mandate the UAC to do so. The B2BUA may also be activated unnecessarily, if the Non-3GPP UA supports the precondition extension, but fails to indicate this in the INVITE request.

For a session setup from a calling 3GPP UA towards called non-3GPP UA, the 3GPP user perception suffers if the non-3GPP UA does not answer the call immediately, but does not send a 180 (Ringing) response. Moreover, the non-3GPP UA may suffer clipping.

Media may be sent from the non-3GPP side to the 3GGP side while at the 3GPP side the session establishment is not completed.

The B2BUA has no means to guarantee that the QoS requirements are met in the non-3GPP side.

Trying to specify the behaviour of a B2BUA in a deterministic way seems to be complicated. In particular, the change of the B2BUA behaviour when it discovers that the non-3GPP side does not support mandated SIP extensions may not be aligned with IETF principles.

This solution does not involve interworking between two different nodes, therefore, it can be applied at the discretion of the network operator, with or without any standardisation effort.

6.2 Modified end-to-end Call Flow

Only relatively minor changes to the 3GPP specifications are required. Charging procedures and QoS authorization procedures are impacted.

This solution does not require updates in the network to allow the usage of future SIP extension, provided both endpoints support those extensions.

Changes have to be performed in various network entities.

Annex A: Interworking topic template

x Topic Name

x.1 Description of interworking issue

Editor's Note: This section contains the technical description of the possible interworking topic. This section also details capabilities, or the lack of capabilities, of the SIP client outside the 3GPP network, which are relevant to make the considered topic applicable. This section also contains a flow diagram illustrating the technical description of the possible interworking topic, such as

- User interaction (call setup time, delay etc)
- Charging and Billing Implications (no charging etc)
- SIP Media authorisation (Interaction with Go Interface for token validation)
- SIP Media allocation (Interaction with Go Interface for "Gating" service)
- Fraudulent opportunities and security risks
- Network operator control (e.g. unable to cut calls)
- Network resource management/coordination allocation; (incorrect tear down resulting in hanging calls etc)
- Probability of occurrence

x.2 Proposed Resolution *yy*

Description

Editor's Note: This section details the suggestion. The involved 3GPP network entities are identified.

Implications of the above solution

Editor's Note: This section list possible advantages of this suggestion compared to competing suggestions.

Editor's Note: This section list possible disadvantages of this suggestion compared to competing suggestions.

Annex B: Mechanisms allowing optional Additions within SIP

Excerpts from RFC 3261

8.1 UAC Behavior

...

8.1.1.9 Supported and Require. If the UAC supports extensions to SIP that can be applied by the server to the response, the UAC SHOULD include a Supported header field in the request listing the option tags (Section 19.2) for those extensions. The option tags listed MUST only refer to extensions defined in standards-track RFCs. This is to prevent servers from insisting that clients implement non-standard, vendor-defined features in order to receive service. Extensions defined by experimental and informational RFCs are explicitly excluded from usage with the Supported header field in a request, since they too are often used to document vendor-defined extensions. If the UAC wishes to insist that a UAS understand an extension that the UAC will apply to the request in order to process the request, it MUST insert a Require header field into the request listing the option tag for that extension. If the UAC wishes to apply an extension to the request and insist that any proxies that are traversed understand that extension, it MUST insert a Proxy-Require header field into the request listing the option tag for that extension. As with the Supported header field, the option tags in the Require and Proxy-Require header fields MUST only refer to extensions defined in standards-track RFCs.

...

8.1.3.2 Unrecognized Responses. A UAC MUST treat any final response it does not recognize as being equivalent to the x00 response code of that class, and MUST be able to process the x00 response code for all classes. For example, if a UAC receives an unrecognized response code of 431, it can safely assume that there was something wrong with its request and treat the response as if it had received a 400 (Bad Request) response code. A UAC MUST treat any provisional response different than 100 that it does not recognize as 183 (Session Progress). A UAC MUST be able to process 100 and 183 responses.

...

8.1.3.5 Processing 4xx Responses Certain 4xx response codes require specific UA processing, independent of the method.

...

If a 420 (Bad Extension) response is received (Section 21.4.15), the request contained a Require or Proxy-Require header field listing an option-tag for a feature not supported by a proxy or UAS. The UAC SHOULD retry the request, this time omitting any extensions listed in the Unsupported header field in the response. In all of the above cases, the request is retried by creating a new request with the appropriate modifications. This new request SHOULD have the same value of the Call-ID, To, and From of the previous request, but the CSeq should contain a new sequence number that is one higher than the previous.

...

8.2 UAS Behavior

...

8.2.1 Method Inspection
Once a request is authenticated (or authentication is skipped), the UAS MUST inspect the method of the request. If the UAS recognizes but does not support the method of a request, it MUST generate a 405 (Method Not Allowed) response. Procedures for generating responses are described in Section 8.2.6. The UAS MUST also add an Allow header field to the 405 (Method Not Allowed) response. The Allow header field MUST list the set of methods supported by the UAS generating the message. The Allow header field is presented in Section 20.5. If the method is one supported by the server, processing continues.

8.2.2 Header Inspection
If a UAS does not understand a header field in a request (that is, the header field is not defined in this specification or in any supported extension), the server MUST ignore that header field and continue processing the message. A UAS SHOULD ignore any malformed header fields that are not necessary for processing requests.
...

8.2.2.3 Require Assuming the UAS decides that it is the proper element to process the request, it ex-amines the Require header field, if present. The Require header field is used by a UAC to tell a UAS about SIP extensions that the UAC expects the UAS to support in order to process the request properly. Its format is described in Section 20.32. If a UAS does not understand an option-tag listed in a Require header field, it MUST respond by generating a response with status code 420 (Bad Extension). The UAS MUST add an Unsupported header field, and list in it those options it does not understand amongst those in the Require header field of the request. Note that Require and Proxy-Require MUST NOT be used in a SIP CANCEL request, or in an ACK request sent for a non-2xx response. These header fields MUST be ignored if they are present in these requests. An ACK request for a 2xx response MUST contain only those Require and Proxy-Require values that were present in the initial request.
...

8.2.4 Applying Extensions
A UAS that wishes to apply some extension when generating the response MUST NOT do so unless support for that extension is indicated in the Supported header field in the request. If the desired extension is not supported, the server SHOULD rely only on baseline SIP and any other extensions supported by the client. In rare circumstances, where the server cannot process the request without the extension, the server MAY send a 421 (Extension Required) response. This response indicates that the proper response cannot be generated without support of a specific extension. The needed extension(s) MUST be included in a Require header field in the response. This behavior is NOT RECOMMENDED, as it will generally break interoperability.
Any extensions applied to a non-421 response MUST be listed in a Require header field included in the response. Of course, the server MUST NOT apply extensions not listed in the Supported header field in the request. As a result of this, the Require header field in a response will only ever contain option tags defined in standards-track RFCs.
...

20 Header Fields

...

20.5 Allow

The Allow header field lists the set of methods supported by the UA generating the message. All methods, including ACK and CANCEL, understood by the UA MUST be included in the list of methods in the Allow header field, when present. The absence of an Allow header field MUST NOT be interpreted to mean that the UA sending the message supports no methods. Rather, it implies that the UA is not providing any information on what methods it supports. Supplying an Allow header field in responses to methods other than OPTIONS reduces the number of messages needed. Example:

```
Allow: INVITE, ACK, OPTIONS, CANCEL, BYE
```

...

20.29 Proxy-Require

The Proxy-Require header field is used to indicate proxy-sensitive features that must be supported by the proxy. See Section 20.32 for more details on the mechanics of this message and a usage example. Example:

```
Proxy-Require: foo
```

...

20.32 Require

The Require header field is used by UACs to tell UASs about options that the UAC expects the UAS to support in order to process the request. Although an optional header field, the Require MUST NOT be ignored if it is present The Require header field contains a list of option tags, described in Section 19.2. Each option tag defines a SIP extension that MUST be understood to process the request. Frequently, this is used to indicate that a specific set of extension header fields need to be understood. A UAC compliant to this specification MUST only include option tags corresponding to standards-track RFCs. Example:

```
Require: 100rel
```

...

20.37 Supported

The Supported header field enumerates all the extensions supported by the UAC or UAS.

The Supported header field contains a list of option tags, described in Section 19.2, that are understood by the UAC or UAS. A UA compliant to this specification MUST only include option tags corresponding to standards-track RFCs. If empty, it means that no extensions are supported.

Example:

```
Supported: 100rel
```

...

21 Response Codes

...

21.4.15 420 Bad Extension

The server did not understand the protocol extension specified in a Proxy-Require (Section 20.29) or Require (Section 20.32) header field. The server MUST include a list of the unsupported extensions in an Unsupported header field in the response. UAC processing of this response is described in Section 8.1.3.5.

21.4.16 421 Extension Required

The UAS needs a particular extension to process the request, but this extension is not listed in a Supported header field in the request. Responses with this status code MUST contain a Require header field listing the required extensions.

Annex C: Impacts of Session Setup Call flows where SIP extensions mandated by 3GPP are not applied.

According to TS 24.229, a 3GPP UA shall use the SIP 100rel, update, and precondition extensions. The 3GPP UA shall abort the call set-up in manners detailed in the main part of this TR, if the non-3GPP UA is not making use of these extensions.

This annex aims to explain why TS 24.229 introduces these restrictions. It details the consequences, if a 3GPP UA would not behave according to TS 24.229 and would not apply some or all of the above SIP extensions.

C.1 Impacts of Session Setup Callflows from Calling 3GPP UA

C.1.1 Session Setup towards non-3GPP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

C.1.1.1 Description of interworking issue

Since the calling 3GPP UA, requires the SIP precondition extension in the SIP INVITE request, the call will be aborted.

After this failure, the calling rogue 3GPP UA may decide to invite the non-3GPP UA not requiring the SIP precondition extension. In what follows, the consequences of this behaviour are discussed.

According to RFC3261 [4], Section 13.2.1, "If the initial (SDP) offer is an INVITE request, the answer MUST be in a reliable non-failure message from UAS back to UAC which is correlated to that INVITE." Since the non-3GPP UE does not support the 100rel extension, provisional responses, such as "183 Session progress" and "180 Ringing", cannot be send reliably, and UE B must include the SDP answer in the 200 OK message.

Thus, resource reservation at the rogue calling 3GPP UA and resource authorisation at P-CSCF will be triggered by this message.

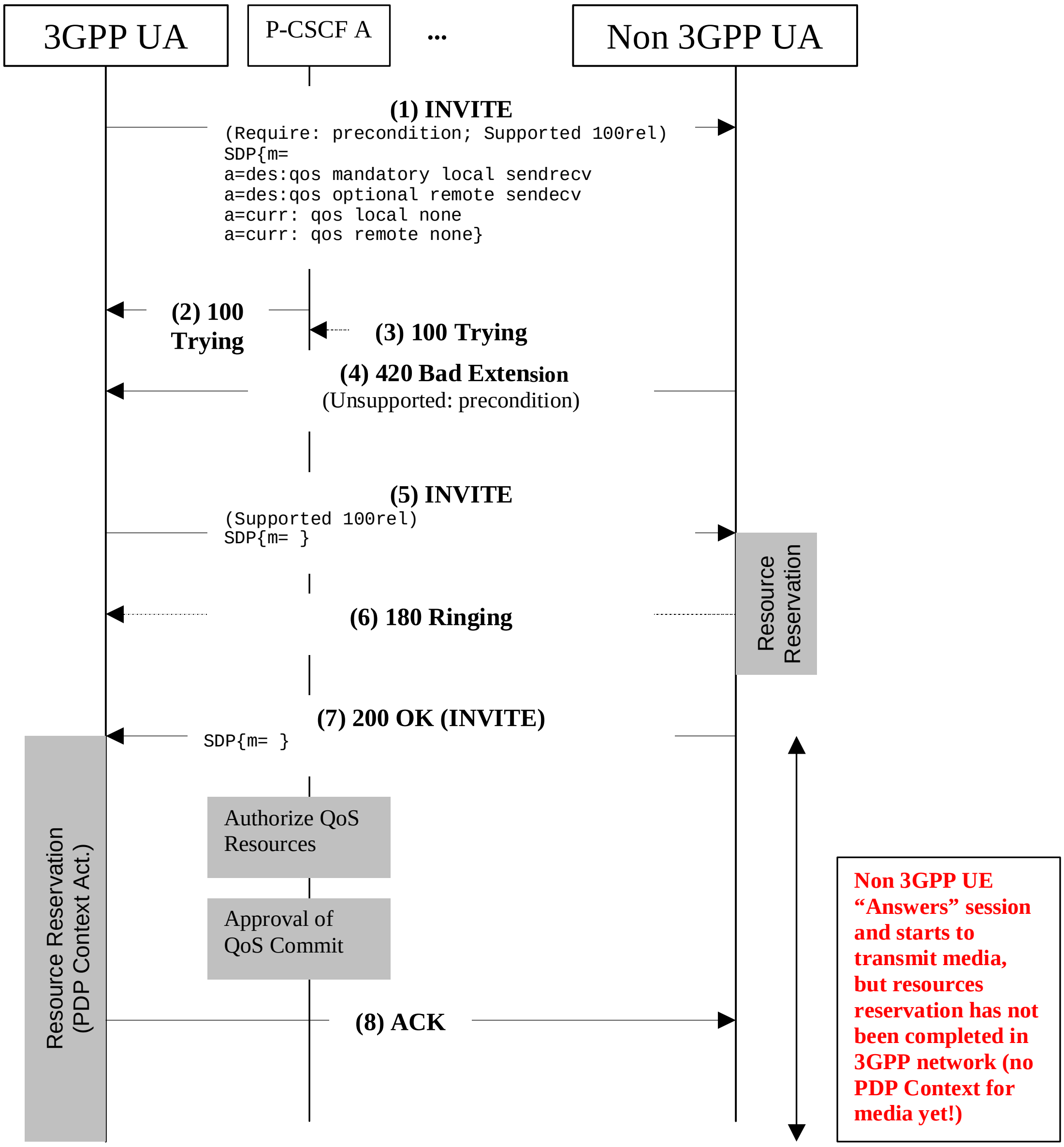

Figure C.1.1.1/1: Session Setup towards non-3GPP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension.

(5) INVITE

The 3GPP UE sends the "INVITE" message to the non-3GPP UA. This includes the "SUPPORTED: 100Rel" line which indicates that the 3GPP UE supports the "Reliability of Provisional Responses" extension.

(6) 180 Ringing

The non-3GPP UA **may optionally** send the "180 Ringing" message to the 3GPP UE. As the non-3GPP UA does **not** support the "100Rel" SIP extension, then there is no mention of the "100Rel" extension in the response back to the 3GPP UE.

(7) 200 OK (Answer)

The non-3GPP UA sends the "200 OK" message to the 3GPP UE to indicate that the called party has answered. As the non-3GPP UA has the "media" RTP port and IP addresses (from the initial INVITE), then it starts to transmit "media" packets (i.e. Speech) to the 3GPP UE.

The 3GPP UE cannot send or recieve "media" until the Resource Reservation (PDP Context Setup) phase has ended.

(8) ACK

The 3GPP UE sends the "ACK" message to the non-3GPP UA to acknowldege the 200 OK "final response" message.

C.1.1.2 Impacts of Identified interworking issue

User interaction

Due to the fact that the call can be "answered" before the media channel is established, the user would experience a delay upon answer of the call. The user experience would be very poor, as users expect to be able to hear/speak to the other party immediately once the call is answered.

Charging and Billing Implications

IMS Charging fails because the Update message is not available to transport GPRS-Charging ID from P-CSCF A to S-CSCF A. There is no other message to replace the UPDATE request for this purpose, which is certain to be sent after the interaction at the Go interface, which delivers the GPRS-Charging-ID to the P-CSCF.

Moreover, if this problem is solved, the Charging (triggered by the 200 OK(INVITE) response) may commence before the resources are available.

SIP Media authorisation

The P-CSCF would have to authorise QoS in the PDF and provide a token, which would be sent to the 3GPP UE at the earliest possible time, i.e. in the 200 OK message

SIP Media allocation

The "Approval of QoS Commit" procedure ("open gate") would have to occur at the same time as the bearer authorisation. In normal operation, the 200 OK(INVITE) message would be the trigger to send the "COPS" DEC message on the Go from the PDF to the GGSN to open the Gate for the media. However, here it also triggers the "PDP Context activation" procedure for the media, and as such bearer authorisation via the Go is also requested. This may cause unstable conditions in the P-CSCF(PDF).

Fraudulent and security risks

A user might invoke this scenario with the purpose to avoid charging.

C.1.2 Session Setup towards non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension

C.1.2.1 Description of interworking issue

Since the 3GPP UA, requires the SIP precondition extension in the SIP INVITE request, the call will be aborted.

After this failure, the calling rogue 3GPP UA may decide to invite the called non-3GPP not requiring the SIP precondition extension. In what follows, the consequences of this behaviour are discussed.

As outlined in Annex D, Note 7, the "183 Session Progress" provisional response may be omitted, if the rogue 3GPP UA does not require SIP preconditions. The use of the "180 Ringing" provisional response also is optional. If both are omitted, the flow diagram and discussion in Section C.1.1 applies. Severe IMS Charging implications have been identified.

Here, it shall be assumed that both the "183 Session Progress" provisional response and the "180 Ringing" provisional response are used.

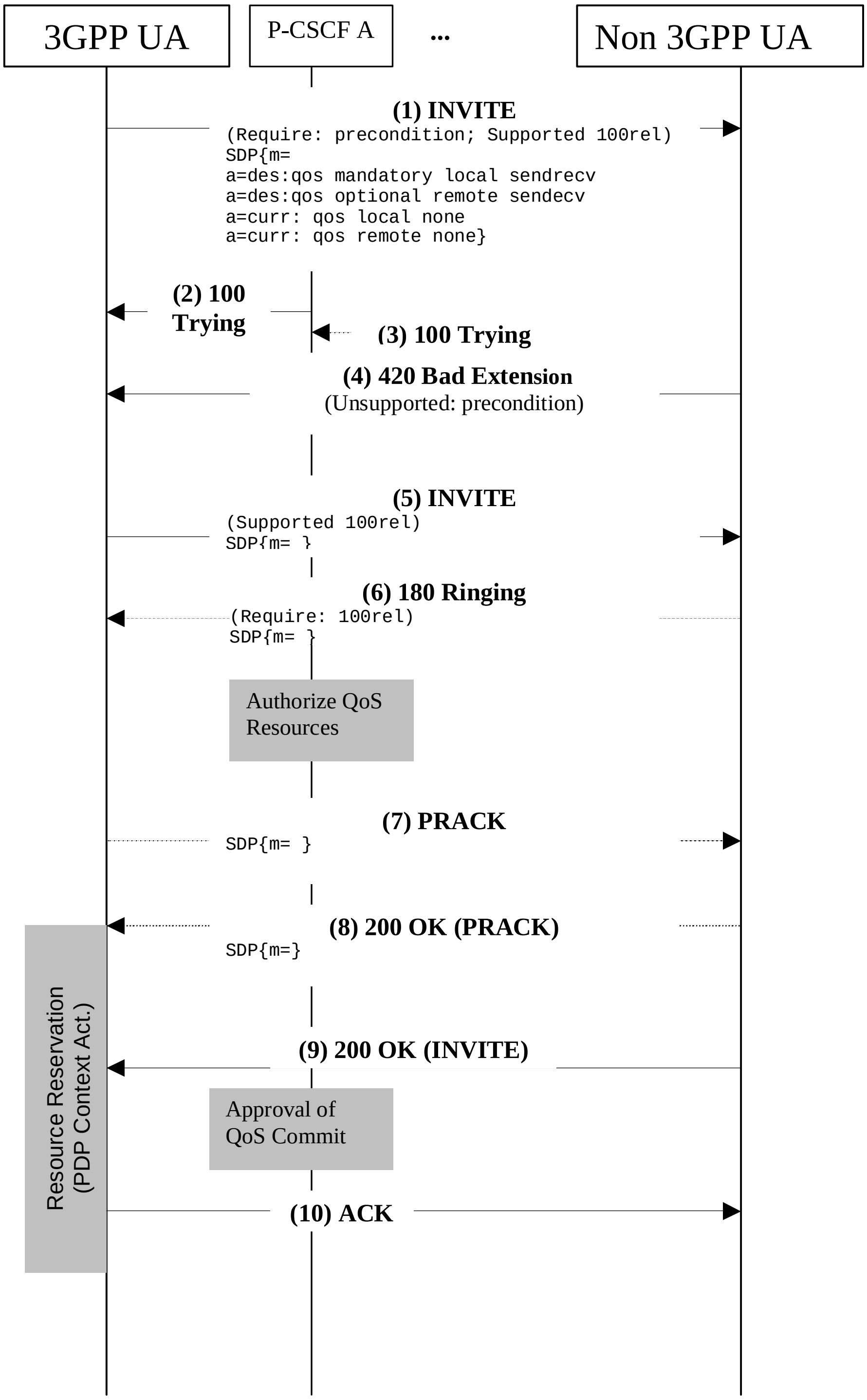

Figure C.1.2.1/1: Session Setup towards non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension

C.1.2.2 Impacts of Identified interworking issue

User at the called non-3GPP UA is alerted before resource reservation at the calling rogue 3GPP UA is complete. The call may still fail at this stage.

IMS Charging fails because the Update message is not available to transport GPRS-Charging ID from P-CSCF A to S-CSCF A. There is no other message to replace the UPDATE request for this purpose, which is certain to be sent after the interaction at the Go interface, which delivers the GPRS-Charging-ID to the P-CSCF.

Moreover, if this problem is solved, the Charging (triggered by the 200 OK(INVITE) response) may commence before the resources are available.

A user might invoke this scenario with the purpose to avoid charging.

C.1.3 Session Setup towards non-3GPP UA not making use of the SIP preconditions extension

Within 3GPP, the SIP update extension is only required to convey SDP offer/answer with SDP attributes defined in the SIP precondition extension with the help of the SIP UPDATE method.

If the non-3GPP SIP UA supports the SIP update extension, but does not use them, the situation is similar to Section C.1.2 and the discussion in this Section is applicable for the present scenario.

A fixed UE supporting the SIP update extension may use features of this extension for purposes not related to the SIP precondition extension.

As a result, various extra messages may be inserted into the call flow:

- The calling or the called UA may send UPDATE messages at various places within the call flow. Those messages may include additional SDP offers. Due to the large number of possibilities, such call flows are not depicted. The dialog state is not altered by UPDATE requests, and thus they probably do not have harmful side effects. Again, the discussion in Section C.1.2 applies.

C.2 Imacts of Session Setup towards Called 3GPP UA

C.2.1 Non-3GPP SIP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

C.2.1.1 Description of interworking issue

According to the SIP 100rel extension, Section 3, “the UAS may send any non-100 provisional response to INVITE reliably, so long as the initial INVITE request contained a Supported header field with option tag 100rel.” Thus, the 3GPP UAS must not send any provisional responses reliably.

Two cases may occur, and are discussed in what follows:

- According to RFC3261 [5], Section 13.2.1, “If the initial (SDP) offer is an INVITE request, the answer MUST be in a reliable non-failure message from UAS back to UAC which is correlated to that INVITE.” UAS must include the SDP answer in the 200 OK message.

- According to RFC3261 [5], Section 13.2.1, the initial (SDP) offer must be, if not in an INVITE, in the first reliable non-failure message send from UAS back to UAC. If the SIP 100rel extension is not supported, this is the final 2xx response. The SDP answer must be in the ACK message.

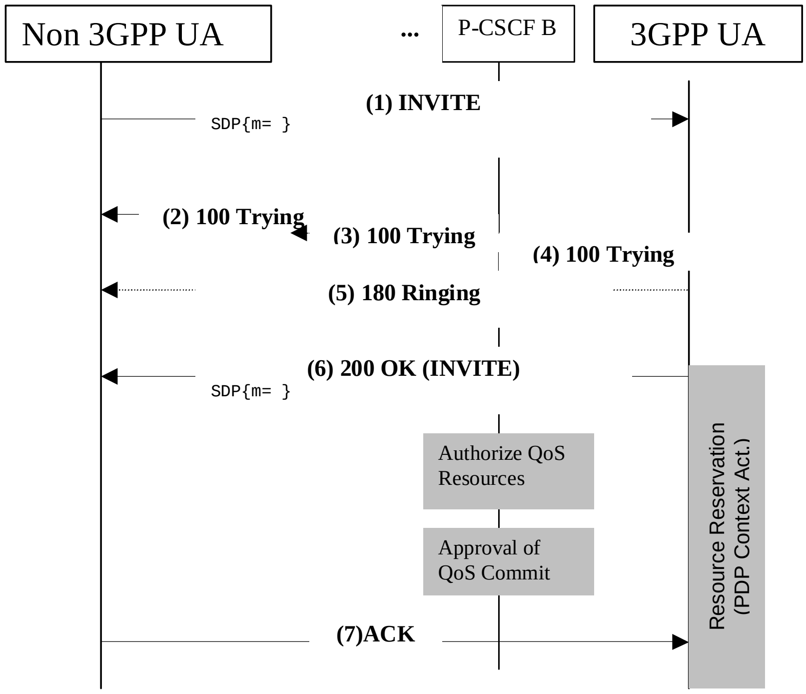

Figure C.2.1.1/1: Non-3GPP SIP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension. SDP offer in INVITE request.

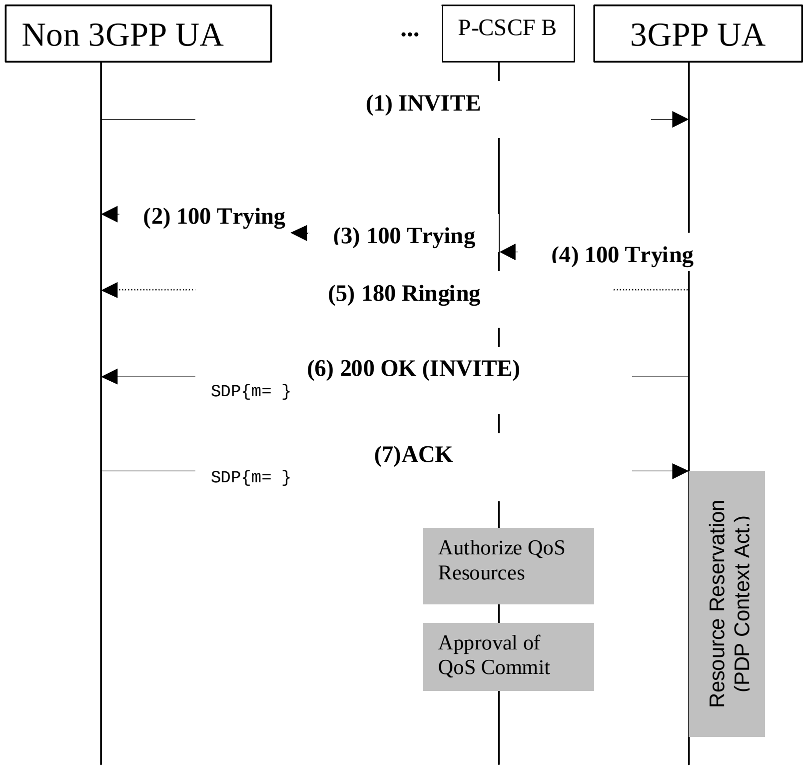

Figure C.2.1.1/2: Non-3GPP SIP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension. SDP offer in OK response.

C.2.1.2 Impacts of Identified interworking issue

3GPP user may be alerted before resources are available. Calls may fail after this point. Moreover, if media offer is transported within 200 OK (Invite) Response Message, user may be alerted before the success of the media negotiation.

IMS Charging is likely to fail, because there are no means to transport the GPRS-Charging-ID from P-CSCF B to S-CSCF B.

A user might invoke this scenario on purpose to avoid charging.

C.2.2 Non-3GPP SIP UA not making use of the SIP preconditions extension and the SIP update extension

C.2.2.1 Description of interworking issue

The called rogue 3GPP UA accepts the INVITE, although no support of preconditions is indicated.

The called rogue 3GPP UA does not need to send UPDATE requests requiring preconditions, because this would not alter the behaviour of the calling UA. Note that, according to the SIP precondition extension, only the called UA is required to suspend the session set-up until mandatory preconditions are met.

According to TS 24.229 [3], the called 3GPP UA shall send the 200 (OK) response to the initial INVITE request only after the local resource reservation has been completed.

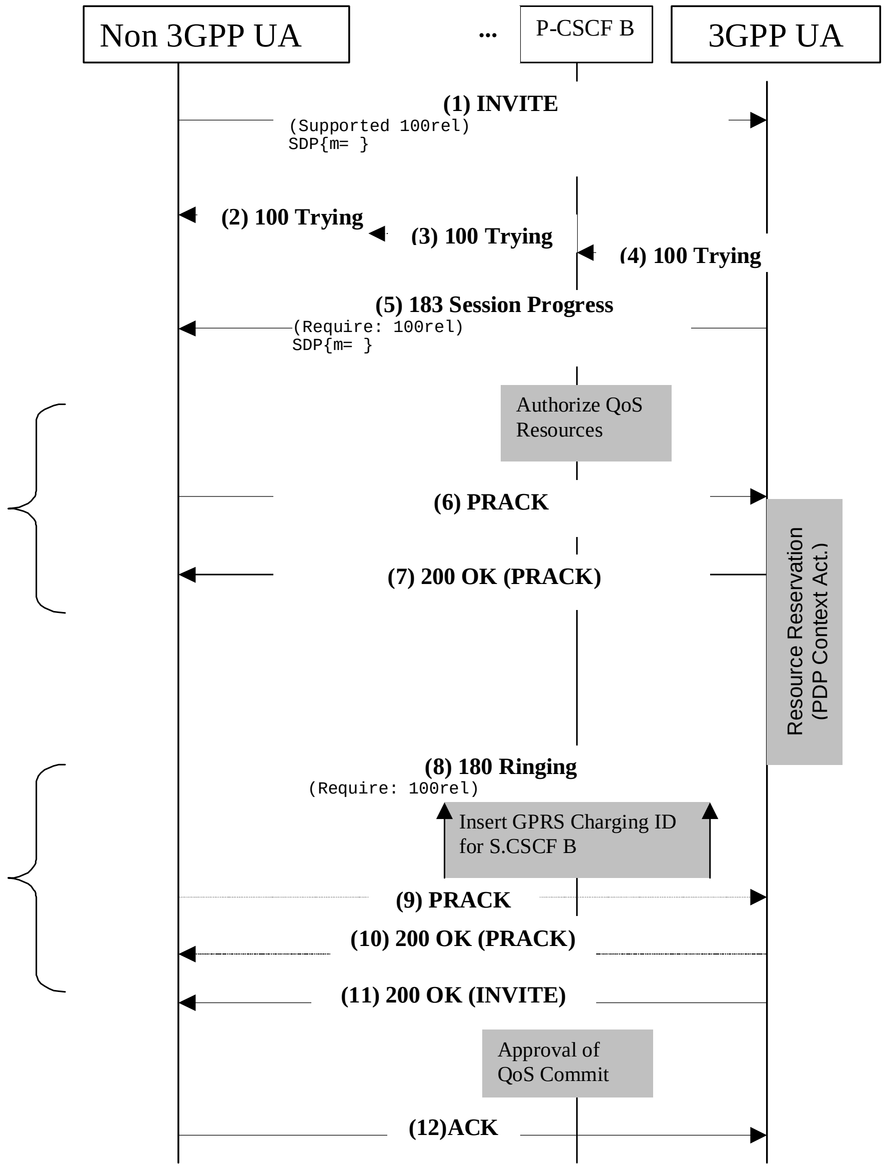

Figure C.2.2.1/1: Non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in Invite.

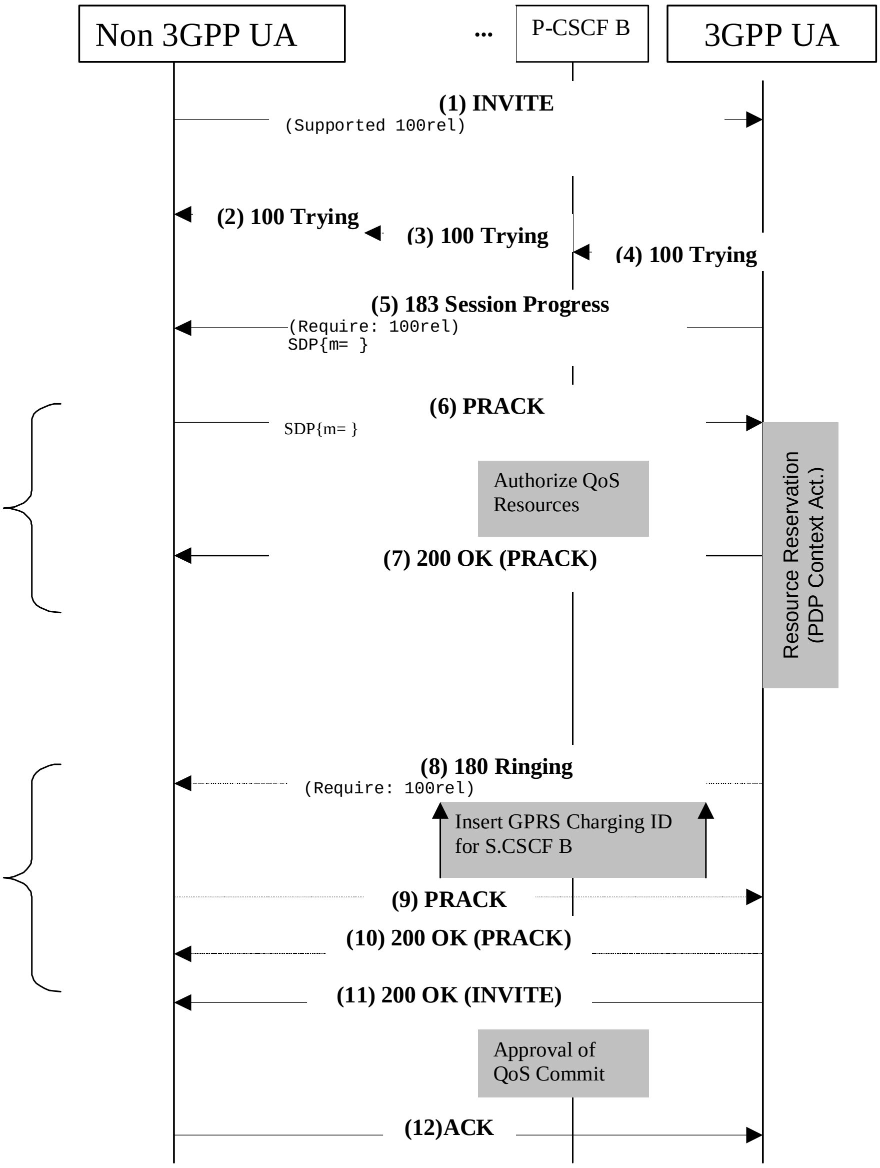

Figure C.2.2.1/2: Non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension. No SDP offer in Invite.

C.2.2.2 Impacts of Identified interworking issue

No negative impacts have been identified.

C.2.3 Non-3GPP UA not making use of the SIP preconditions extension

Within 3GPP, the SIP update extension is only required to convey SDP offer/answer with SDP attributes defined in the SIP precondition extension with the help of the SIP UPDATE method.

If the non-3GPP SIP UA supports the SIP update extension, but does not use them, the situation is similar to Section C.2.2 and the discussion in this Section is applicable for the present scenario.

A non-3GPP UA supporting the SIP update extension may use features of this extension for purposes not related to the SIP precondition extension. As a result, various extra messages may be inserted into the call flow. The UA, may send UPDATE messages at various places within the call flow. Those messages may include additional SDP offers. Due to the large number of possibilities, such call flows are not depicted. The dialog state is not altered by UPDATE requests, and thus they probably do not have harmful side effects. Again, the discussion in Section C.2.2 applies.

Annex D: Reference Call Flow from 3GPP UA to 3GPP UA

The interworking between an originating 3GPP UA and a terminating 3GPP UA is as defined in 3GPP TS 24.229. No interworking issues exist, but the flow diagram is depicted here for comparison.

NOTE 1: The message flow between the 3GPP UEs is depicted.

NOTE 2: SIP proxies are omitted with the exception of the P-CSCFs and the S-CSCFs, which are depicted in this call flow but will be omitted in most other call flows.

NOTE 3: The 100 (Trying) response (2), (3), (4) to the INVITE request (1) is sent hop-by-hop, as indicated in this flow diagram. All other messages are generated by the 3GPP UEs.

NOTE 4: Most parts of the SIP messages are omitted for simplicity. Only the "Require", "Supported" and "Allowed" header fields are depicted.

NOTE 5: Most parts of the SDP are omitted for simplicity. Only the SDP preconditions for one medium are depicted. There may be an arbitrary number of number of media, each with own preconditions.

NOTE 6: The P-CSCF inspects each SDP, in order to identify offer/answer pairs [8]. The P-CSCF may modify the QoS authorisation (8,9) when processing each SDP answer.

NOTE 7: The use of the183 (Session Progress) (7) provisional response is optional according to IETF specifications. However, if the SIP precondition extension [6] is used and SDP with mandatory preconditions that the terminating UA is not capable of meeting unilaterally is included in the initial INVITE request (1), a 101-199 provisional response, such as the 183 (Session Progress) response, is required to transport the SDP answer including the mandated “confirmation status” SDP attribute (Ref. [6], Section 6). Moreover, the 180 (Ringing) response is not suitable because the user should not be alerted until the preconditions are met.
NOTE 8: It is optional to convey a new SDP offer/answer within the PRACK request (11) and 200 (OK) for a PRACK request) (12). An originating 3GPP UA will refrain from generating a new SDP offer within PRACK request (11), if it does not wish to further restrict the set of codecs selected within the first offer/answer pair.

NOTE 9: According to IETF Ref. 6, Section 5, the called UA should start the resource reservation (13) immediately after having send the SDP answer within of the 183 (Session Progress) (7) provisional response. However, a called 3GPP UA may expect a second SDP offer, and thus wait for the next message until starting resource reservation. The called 3GPP UA can be certain to receive an new message soon, since it demands the PRACK message with the “Require 100rel” SIP header within the 183 (Session Progress) (7) provisional response.

NOTE 10: The use of the UPDATE request (14) is optional according to IETF specifications [5], [7], unless certain conditions make it mandatory. In particular, if a threshold specified in a previous SDP “confirm-status” attribute (e.g. in message (7)) is reached or surpassed, the UA must send an SDP offer reflecting the new current status (Ref. [5], Section 7). The only suitable way to convey the new SDP offer/answer may be via an UPDATE request.

NOTE 11: If the UPDATE request (14) is not used, the subsequent 200 (OK) response for an UPDATE request(17) is also not present.

NOTE 12: The use of the 180 (Ringing) provisional response (18) is optional according to IETF and 3GPP specifications. The 180 (Ringing) provisional response is used to convey the GPRS charging ID from P-CSCF B to S-CSCF-B. If the 180 (Ringing) provisional response is omitted, the GPRS Charging ID is transported within the “200 OK(INVITE)” (23) response.

NOTE 13: The UPDATE request (14)is used to convey the GPRS charging ID from P-CSCF A to S-CSCF-A. [1]

NOTE 14: According to TS 24.229 [3], the called 3GPP UA shall send the 200 (OK) response to the initial INVITE request only after the local resource reservation has been completed.

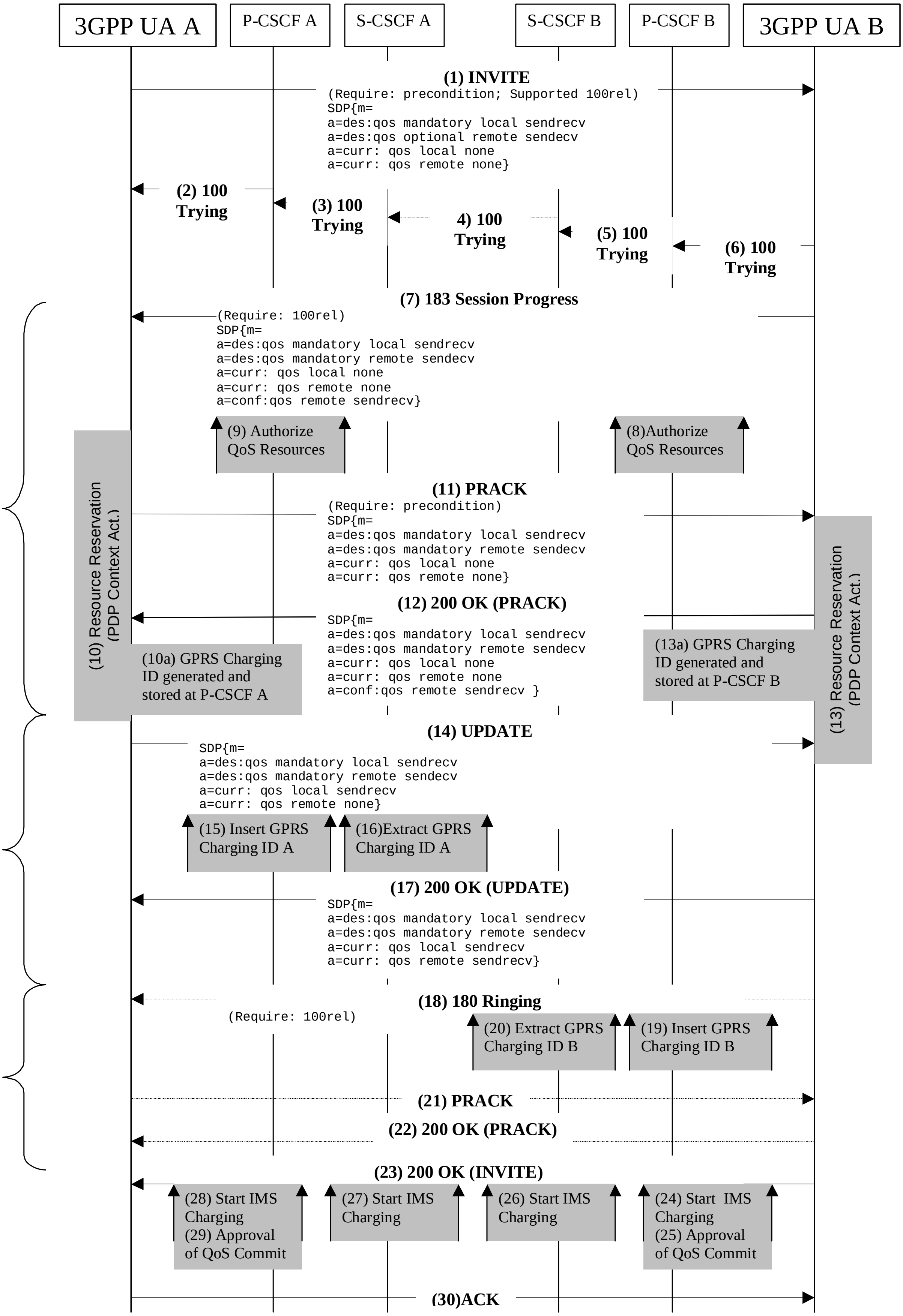

Figure C/1: 3GPP UA to 3GPP UA Call flow

The following dependencies between SIP signalling and mechanisms related to service based local policy and charging on IMS level have been identified. The listed steps have to be performed in the indicated order both for mobile originated and mobile terminated calls.

1. The P-CSCF stores information about authorised media learned from SDP offer-answer exchange (8, 9)
2. A UE set up a PDP context after SDP offer-answer exchange (10, 13). User Plane data may only be transported after PDP context is set up.
3. While a PDP context is set up, the GGSN asks the P-CSCF(PDF) for a decision to authorise the media. The GGSN also sends the GPRS Charging ID to the PDF in this request. (10a, 13a)
4. The P-CSCF(PDF) sends the GPRS Charging ID to the P-CSCF(S-CSCF) in a suitable SIP message (14,15,16 and 18,19,20)
5. The S-CSCF(PDF) sends the GPRS Charging ID to the charging system, which uses it to correlate IMS and GPRS charging.(16,20)
6. The 200 OK(INVITE) SIP message triggers S-SCSF and P-CSCF to inform the charging subsystem that the SIP session is established. The charging subsystem may use this as trigger to start service based charging. (23,24,26,27,28)
7. The 200 OK(INVITE) SIP message triggers P-CSCF(PDF) to open gates at GGSN. (23,25,29). User Plane data may only be transported after gates are open.

Annex E: Scenarios without identified interworking issues

This Annex contains scenarios, which result in call flows that deviate to some extent from the reference call flow in Annex D. These scenarios have been investigated, but no interworking problems have been identified.

E.1 Calling non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not performing QoS reservation, to called 3GPP UA.

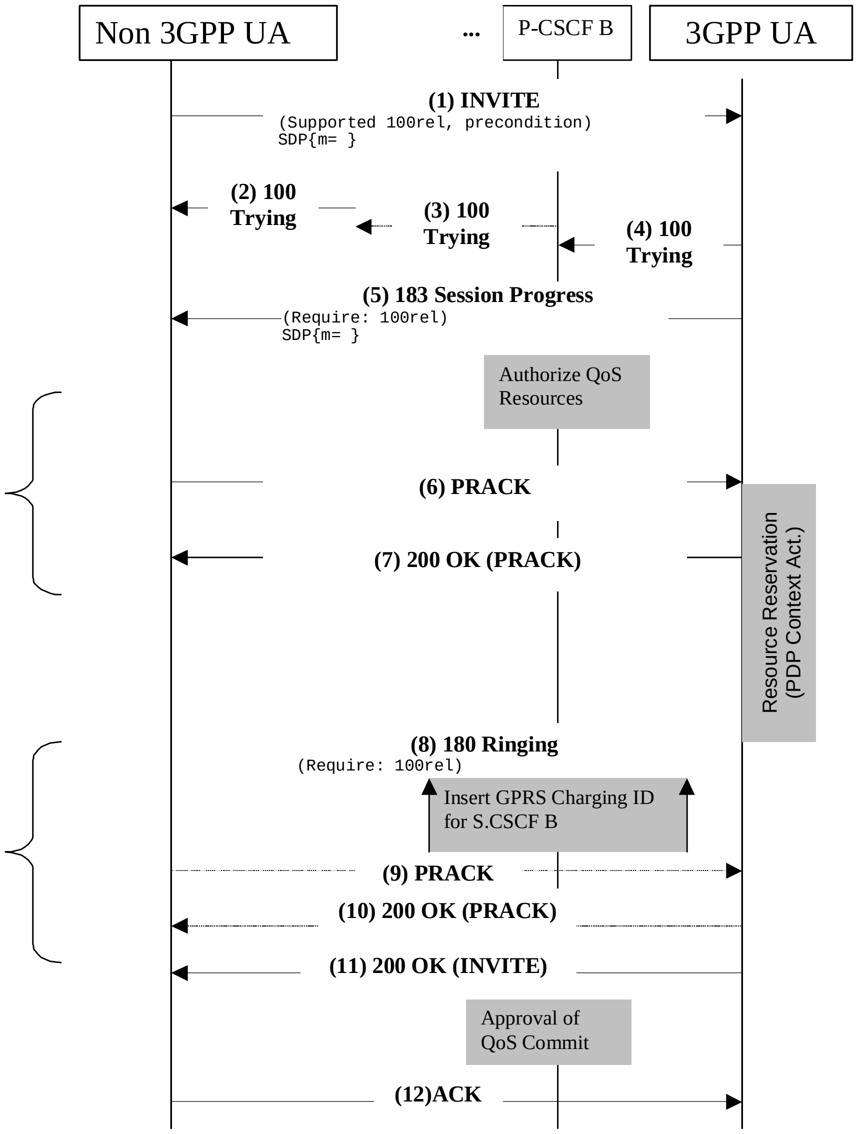

Figure E.1/1: Non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, and not performing QoS reservation, to 3GPP UA

E.2 Calling non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not including the SDP offer in the initial INVITE request, to called 3GPP UA.

According to TS 24.229, Section 5.1.4.1, the called 3GPP UA must send a provisional response (otherwise it can not complete the resource reservation before sending the 200 (OK) response for an INVITE request) and require the 100rel extension within this message. According to RFC 3261, Section 13.2.1, “the initial offer MUST be in either an INVITE or, if not there, in the first reliable non-failure message from the UAS back to the UAC”.

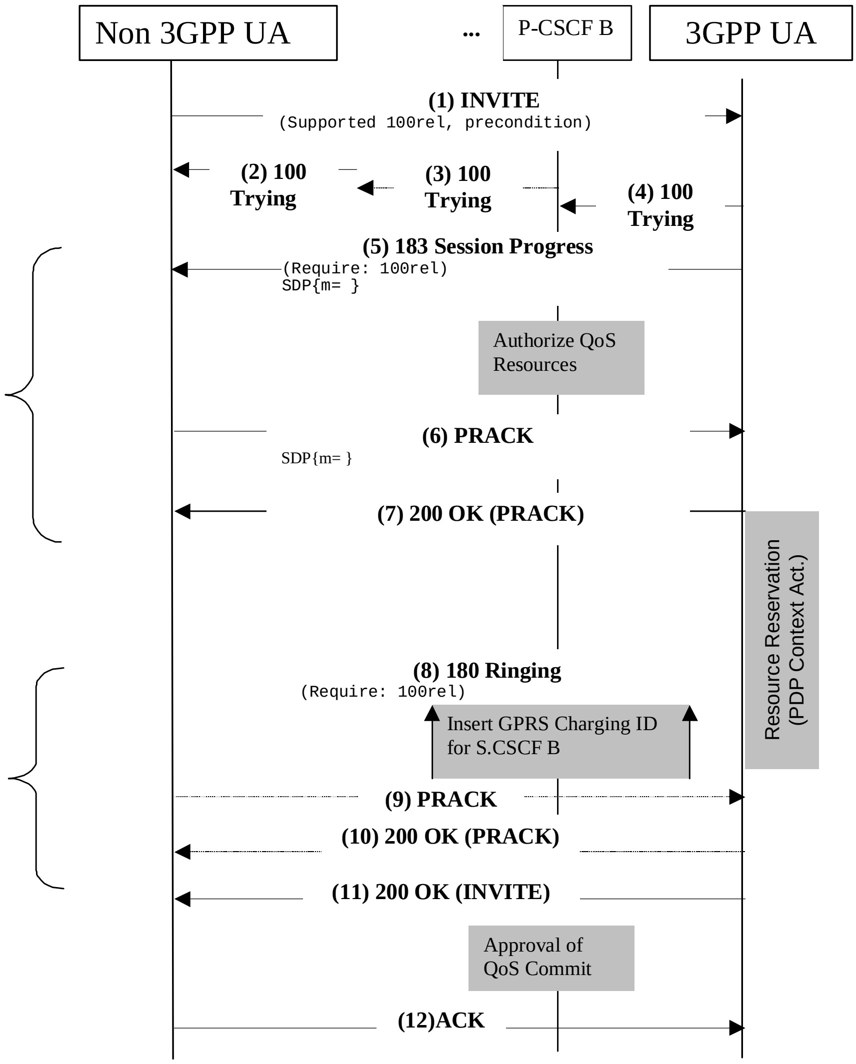

Figure E.2/1: Non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not including the SDP offer in the initial INVITE request, to 3GPP UA

Annex F: Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2002-01	CN3#21				Creation of document	-	0.1.0
2002-07	CN3#24				Include suggestions for B2B UA	0.1.0	0.2.0
2002-11	CN3#26				Output of drafting group included, presented to CN#18 for information	0.2.0	1.0.0
2002-12	NP-18	NP-020610			Presented to Plenary NP#18 for information	1.0.0	
2002-02	CN3#27	N3-030152 N3-030153 N3-030154 N3-030156 N3-030157			Agreed changes are included.	1.0.0	1.1.0
2003-03	NP-19				Presented to Plenary NP#19 for information	1.1.0	
2003-05	CN1#29	N1-030487 N1-030535			Review by CN1. Proposed Changes and new version of TR require endorsement by CN3	1.1.0	1.2.0
2003-05	CN3#29	N3-030454 N3-030456 N3-030458 N3-030463			CN3 endorsed version 1.2.0. Agreed changes based on this version are included. CN3 agreed to send TR to CN plenary for approval	1.2.0	2.0.0

3GPP TSG-CN WG3 Meeting #28 **_N3-030461_**
San Diego, USA, 19th - 23rd May 2003.

Title:	**LS on SIP signalling interworking between IM CN subsystem entities and SIP network entities external to the IN CN subsystem**
Release:	**Rel-6**
Source:	**CN3**
To:	**SA2**
Cc:	**CN1**

Contact Person:

Name: Thomas Belling
Tel. Number: +49 89 636 75207
E-mail Address: **Thomas.Belling@siemens.com**

Attachments: N3-030460 TR 29.962v200

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1. Overall Description:

CN3 is responsible for the stage 3 work on interworking between the IM CN subsystems and external IP networks.

In TR 29.962, CN3 investigated the SIP signalling interworking between IM CN subsystem entities behaving as specified in the 3GPP profile of SIP in TS 24.229, and SIP network entities external to the IM CN subsystem, which may not adhere to the 3GPP profile of SIP. The document focuses on scenarios where the non-3GPP UA does not support one or more of the following SIP extensions:

- Preconditions: “Integration of Resource Management and SIP” RFC 3312
- Update: “The Session Initiation Protocol UPDATE Method”, RFC 3311
- 100rel: “Reliability of Provisional Responses in SIP”, RFC 3262

An interworking with such host is currently disallowed, since essential functionality within the IM CN subsystem, such as service based charging, might otherwise not be possible. TR 29.962 investigates possibilities to interwork with such hosts while maintaining this essential functionality. A proposal using a B2BUA and another proposal making use of modified end-to-end callflows are detailed.

CN3 would like to ask SA2 to study the architectural impacts of the proposed solutions. SA2 is asked to decide if any of the above proposals shall be further pursued and implemented in normative specifications.

CN3 considers the work on TR 29.962 as finalized. TR 29.962 was also recently reviewed by CN1, which is responsible for the stage 3 of the SIP signalling within the IM CN subsystem.

2. Actions:

To SA2 group.

ACTION: SA2 is asked to decide if any of the above proposals shall be further pursued and implemented in normative specifications.

3. Date of Next CN3 Meetings:

CN3 #29	25th - 29th August 2003	Sophia Antipolis, France.
CN3 #30	27th – 31st October 2003	t.b.a