3GPP TSG CN Plenary Meeting #19 **Tdoc NP-030073**
12th - 14th March 2003. Birmingham, U.K.

3GPP TSG-CN WG3 Meeting #27 **Tdoc N3-030185**

Version 1.1.0 of TR 29.962 is provided within this TDOC. Both a clean version that shall be presented to the CN plenary and a version with highlighted changes against version 1.0.0 is provided.
The cover sheet for the presentation to the CN plenary is provided in what follows.

Presentation of Specification to TSG or WG

Presentation to:	**TSG CN Meeting #19**
Document for presentation:	**TR 29.962, Version 1.1.0**
Presented for:	**Information**

Abstract of document:

TR 29.962 investigates the SIP signalling interworking between IMS network entities behaving as specified in the 3GPP profile of SIP in TS 24.229, and SIP network entities external to the 3GPP network, which may not adhere to the 3GPP profile of SIP.

The document focuses on scenarios where the non-3GPP UA does not support the SIP “Preconditions” extension (“Integration of Resource Management and SIP”, RFC 3312) and related SIP extensions, which are mandated in the IMS and required for a proper interaction between service based local policy, charging on IMS level and SIP signalling. Those SIP extensions are defined within the IETF, and are optional in external networks.

Two approaches to enable an interworking for such scenarios are developed and compared:

- Insert a “back to back user agent” as signalling interworking function.
- Allow modified call flows within the IMS and adopt the rules for service based local policy and charging on IMS level.

Changes since last presentation to TSG CN Meeting #18:

Changes detailing both approaches to resolve the interworking were agreed, filling the remaining empty sections of the document.

Outstanding Issues:

As agreed in a joint meeting between CN3#26 and CN1#27 in Bangkok, and endorsed by CN#18, the TR will be reviewed by CN1.
The review will take place during CN1#28 in Sophia Antipolis. The proposed changes will be endorsed by CN3#27 in San Diego. If required, a joined CN1 / CN3 meeting may also be organised in San Diego to complete the review.

Contentious Issues:

None

3GPP draft TR 29.962 V1.~~0~~1.0 (~~2002~~2003-

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)

Select keywords from list provided in specs database.

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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 IMS network entities behaving as specified in the 3GPP profile of SIP in TS 24.229 [1], with related ~~callflow~~call flow examples in TS 24.228 [2] and stage 2 work in TS 23.228 [3], and SIP network entities external to the 3GPP network, which may not adhere to the 3GPP profile of SIP.

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 3GPP network may feature different SIP capabilities, such as the support of arbitrary SIP packages

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]

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 3GPP network. Any SIP network entity within the 3GPP network 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, 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 3GPP network, which do not or only partly satisfy the 3GPP profile of SIP, in particular not fully 3GPP conformant SIP terminals, are out of scope of the present document.

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

It is ~~forseen~~foreseen that future non-3GPP SIP clients will support the above required SIP extensions, and it is envisaged that it is unlikely that interworking solutions will then be required.

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]

3.2 Abbreviations

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

4. Interworking Scenarios

Each topic is contained in an own subsection with the structure defined in Annex A. Further structure may be introduced to the present section by grouping related topics.

4.1 ~~calling~~ Calling 3GPP UA to ~~called~~ Called non-3GPP UA

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

- 3GPP UA to non-3GPP UA supporting the SIP precondition extension, but not supporting the SIP 100rel extension.
- 3GPP UA to non-3GPP UA supporting the SIP preconditions extension, but not supporting the SIP update extension.

4.1.1 3GPP UA to non-3GPP UA supporting the SIP 100rel extension and the SIP update extension, but not supporting the SIP preconditions extension

4.1.1.1 Description of interworking issue

The call fails, as detailed in Section 4.1.2.2

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.

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 ~~callflow~~call flow.

4.1.1.2 Proposed Resolutions to interworking issue

4.1.1.2.1 B2B UA

A B2B UA is used.

4.1.1.2.1.1 Insertion of B2B UA

How the B2B UA is inserted is discussed within Section 4.1.2.4.1.1.

4.1.1.2.1.2 Functionality of B2B UA

4.1.1.2.1.2.1 Description

The functionality of the B2B UA is as discussed in Section 4.1.2.4.1.2.1.

The B2B UA shall pass additional UPDATE messages, which are not related to the precondition extension, and related provisional acknowledge messages.

4.1.1.2.1.2.2 Advantages

General advantages of the B2B UA are discussed in Section 4.1.2.4.1.2.2.

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

4.1.1.2.1.2.3 Disadvantages

General disadvantages of the B2B UA are discussed in Section 4.1.2.4.1.2.3.

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

4.1.1.2.2 Modified end-to-end ~~callflow~~call flow

4.1.1.2.2.1 Description

The rules described in Section 4.1.3.2.2.1 are applied.

The resulting call flow is similar to Figure 4.1.2.4.2.1/1, possibly with additional update messages.

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

4.1.1.2.2.2 Advantages

See Section 4.1.3.2.2.2.

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

4.1.1.2.2.3 Disadvantages

See Section 4.1.3.2.2.3.~~Editor's Note: This section list possible disadvantages of this suggestion compared to competing suggestions.~~

4.1.2 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension

4.1.2.1 Description of interworking issue

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

4.1.2.2 Flow diagram

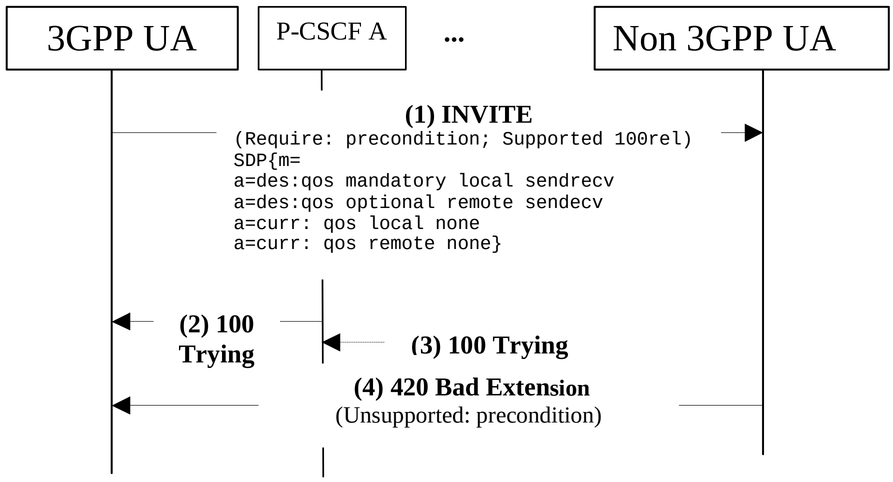


Figure 4.1.2.2/1: 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension.

4.1.2.3 Impact of Identified interworking issue

The call fails.

4.1.2.4 Proposed resolutions to interworking issue

A B2B UA is used.

4.1.2.4.1 B2B UA

4.1.2.4.1.1 Insertion of B2B UA

4.1.2.4.1.1.1 Static Insertion of B2B UA

4.1.2.4.1.1.1.1 Description

A B2B UA is permanently inserted at connections between the IMS and a given external network. This B2B UA handles all calls, including calls where the ~~callflow~~call flows may be passed without modification.

The B2B UA shall be inserted in the home IMS for all calls leaving the home IMS, which are not routed to another IMS via direct interconnection.

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

The B2B UA becomes active only when receiving a 420 Bad Extension (Unsupported precondition) response from the Non-3GPP UA, as depicted in Figure 4.1.2.4.1.1.1.1/1. Otherwise, the B2B UA passes all SIP messages received at one side to the other side.

The B2B UA shall store the SDP offer in initial invites for all calls until receiving a provisional or final response from the Non-3GPP UA.

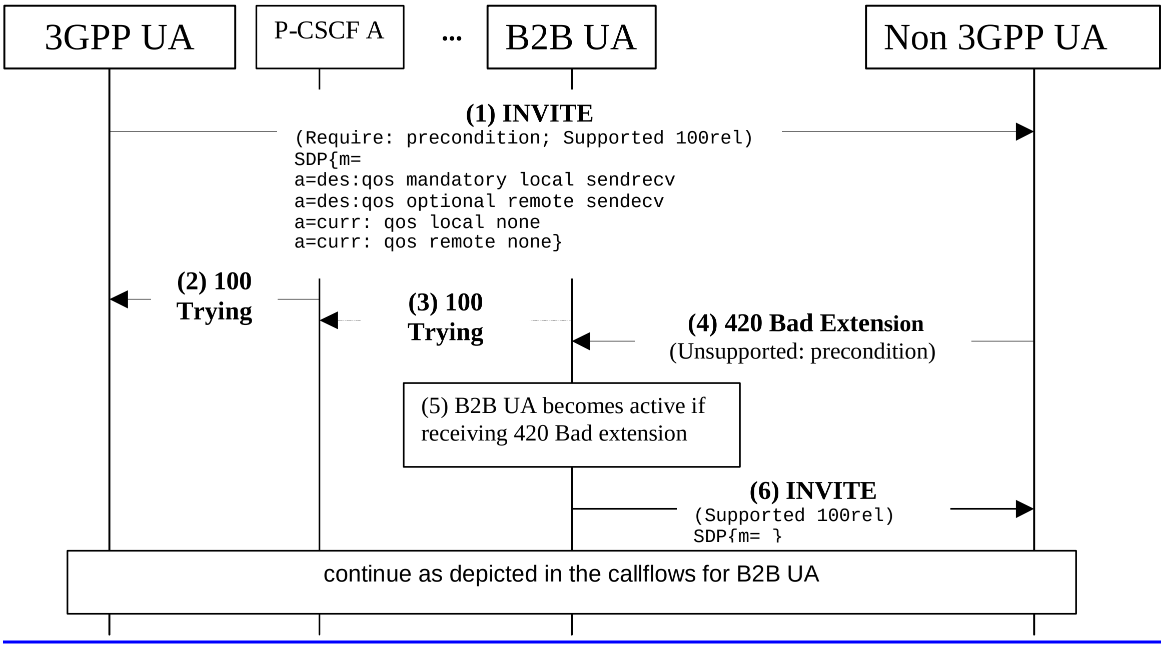


Figure 4.1.2.4.1.1.1.1/1: Activation of static B2B UA connecting 3GPP UA to non-3GPP UA not supporting the SIP preconditions extension

~~Editor´s Note: It has to be clarified how the network finds out that the call terminates outside a 3GPP IMS.~~

4.1.2.4.1.1.1.2 Advantages

4.1.2.4.1.1.1.3 Disadvantages

Additional processing load.

Additional delay.

Lack of signalling transparency may restrict the compatibility with future extensions for all calls.

4.1.2.4.1.2 Functionality of B2B UA

4.1.2.4.1.2.1 Description

Editor's Note: The following rules have been agreed only as basis for further contributions and have not yet been investigated in detail.

The B2B UA shall apply the following rules:

1. The B2B UA 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 B2B UA shall also comply to the SIP 100rel and update extensions.

3. On the IMS side, the B2B UA shall also comply to and apply the SIP 100rel, update and precondition extensions.

4. The B2B UA shall pass SIP requests arriving at one call leg to the call leg at the other side, with the exceptions below.

5. The B2B UA shall pass 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 B2B UA shall pass SIP final responses arriving at one call leg to the call leg at the other side, with the exceptions below.

7. The B2B UA shall not require the SIP preconditions extension on the non-IMS side.
8. The B2B UA shall insert preconditions in SDP sent from non-3GPP UA to 3GPP UA.
9. The B2B UA should remove preconditions in SDP sent from 3GPP UA to non-3GPP UA
10. The B2B UA shall not pass PRACK and 200 OK(PRACK) messages.
11. The B2B UA shall delay forwarding a 200 OK(INVITE) message from the non-IMS side to the IMS side until the mandatory preconditions are met on the IMS side.
12. The B2B UA shall handle subsequent SDP offers on the IMS side in an INVITE transaction locally, if only the preconditions are modified
13. If the B2B UA receives a subsequent SDP offers on the IMS side with modified media, it shall suspend the transaction on the IMS side and pass this SDP offer to a re-invite transaction on the non-IMS Side. The B2B UA shall pass the SDP answer received in the re-invite transaction 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 B2B UA shall pass an SDP answer within the 200 OK(INVITE) message of the original INVITE transaction from the non-IMS side to a provisional response on the IMS side.
15. For a re-Invite from the Non-IMS side to the IMS side, the B2B UA shall apply the rules in Section 4.2.2.4.1.2.1.

The B2B UA relies messages as indicated by the red dotted arrows in the figures below.

The called UA may also send no “Session progress” message and include the SDP answer in the “200 OK(INVITE)” instead. This case is discussed in Section 4.1.3.4.1.2.

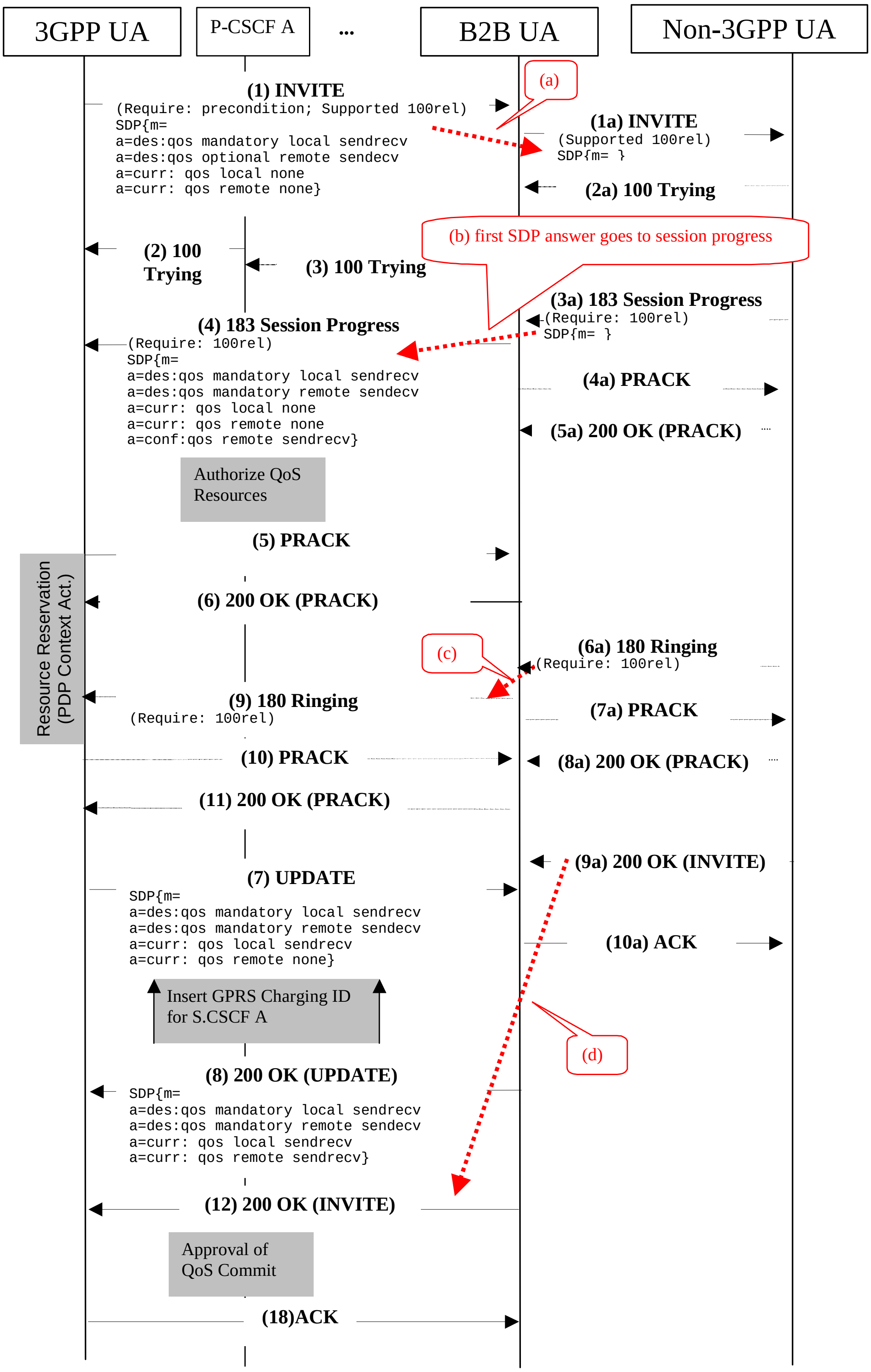


Figure 4.1.2.4.1.2.1/1: Functionality of B2B UA connecting 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension.
Calling UA includes SDP answer in 183 "Session Progress". Calling UA sends no second SDP offer.

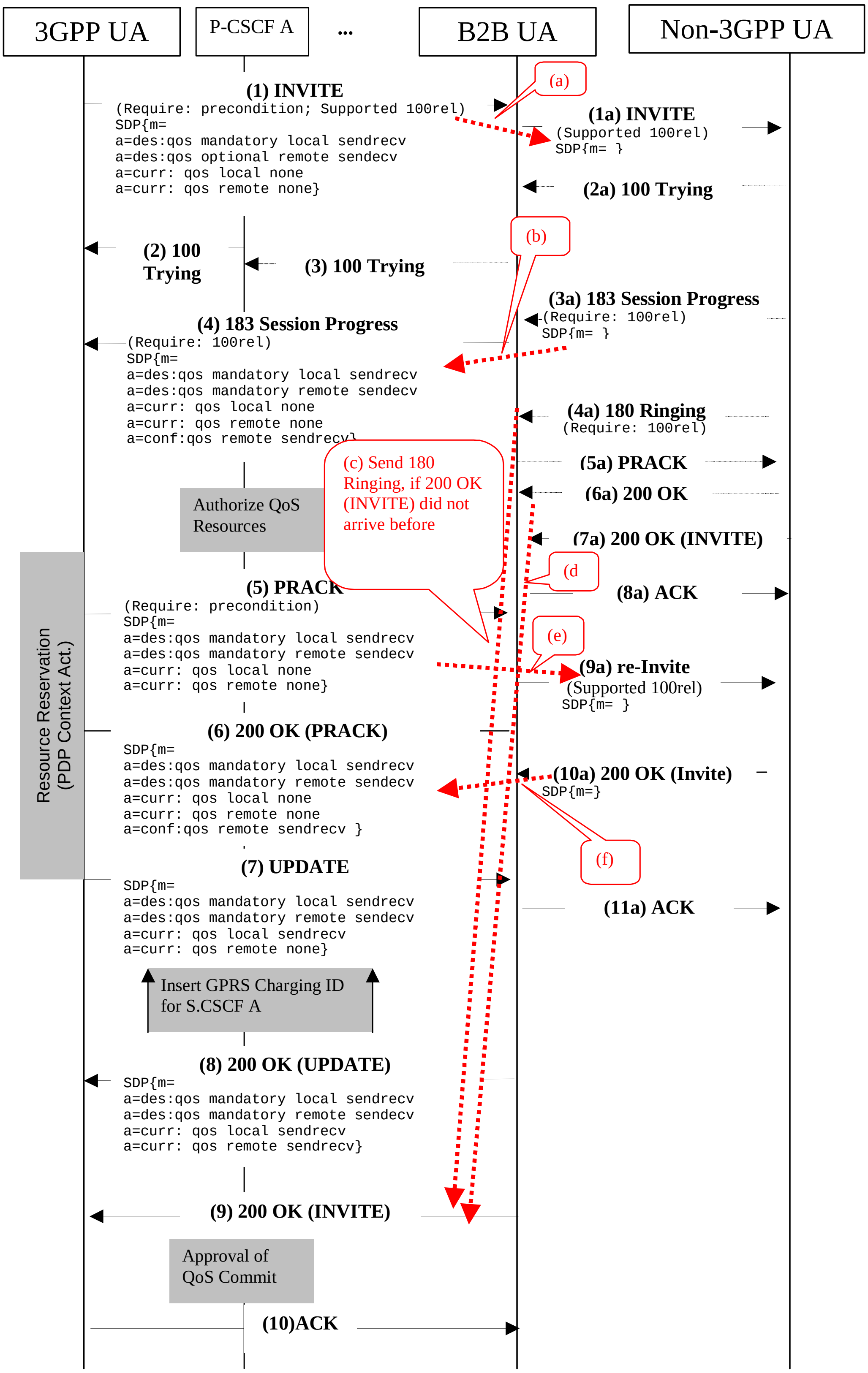


Figure 4.1.2.4.1.2.1/2: Functionality of B2B UA connecting 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension.
Called UA includes SDP answer in 183 "Session Progress". Calling UA sends second SDP offer.

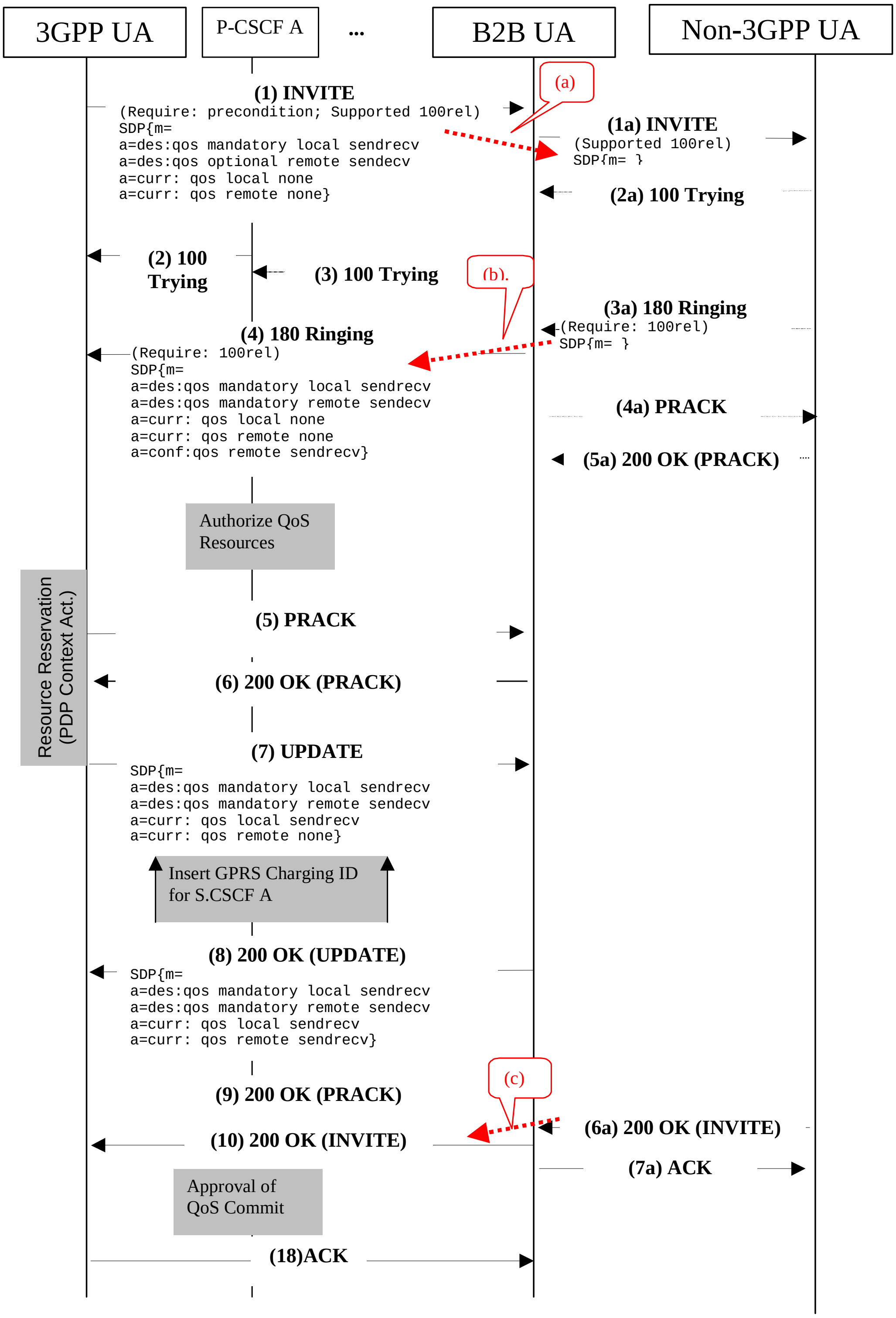


Figure 4.1.2.4.1.2.1/3: Functionality of B2B UA connecting 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension.
Called UA includes SDP answer in 180 "Ringing". Calling UA sends no second SDP offer.

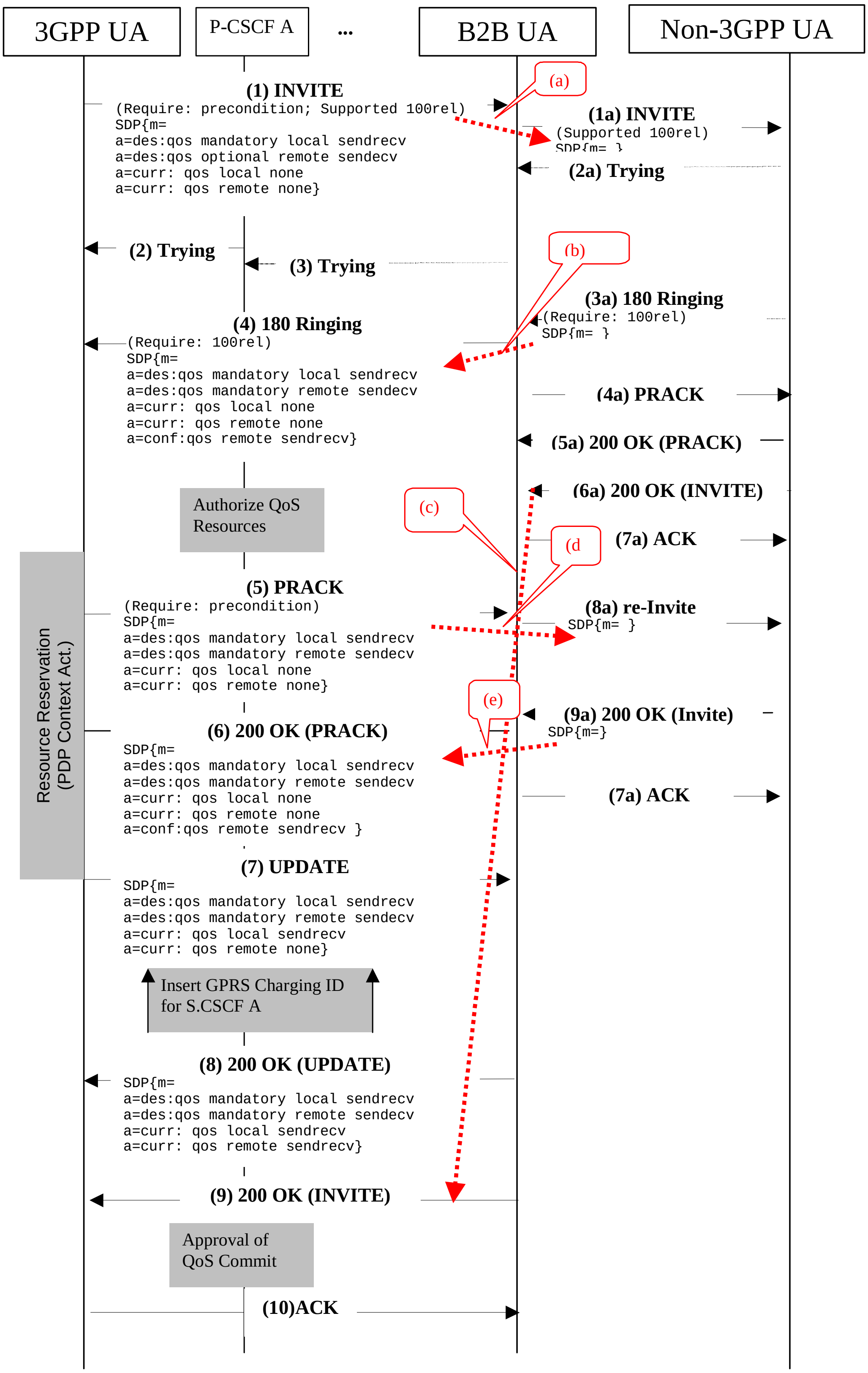


Figure 4.1.2.4.1.2.1/4: Functionality of B2B UA connecting calling 3GPP UA to called non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension.
Called UA includes SDP answer in 180 "Ringing". Calling UA sends second SDP offer.

4.1.2.4.1.2.2 Advantages

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

4.1.2.4.1.2.3 Disadvantages

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

The functionality and implementation of the B2B UA is complicated and will have to include the following features:

- Storage of SDP.
- Own timer supervision of dialogues on both call legs.

The compatibility with future SIP extensions may be limited by the need to update the B2B UA. This may limit the network's ability to deploy new IP multimedia applications.

4.1.2.4.2 Modified end-to-end ~~callflow~~call flow

4.1.2.4.2.1 Description

The rules described in Section 4.1.3.2.2.1 are applied.

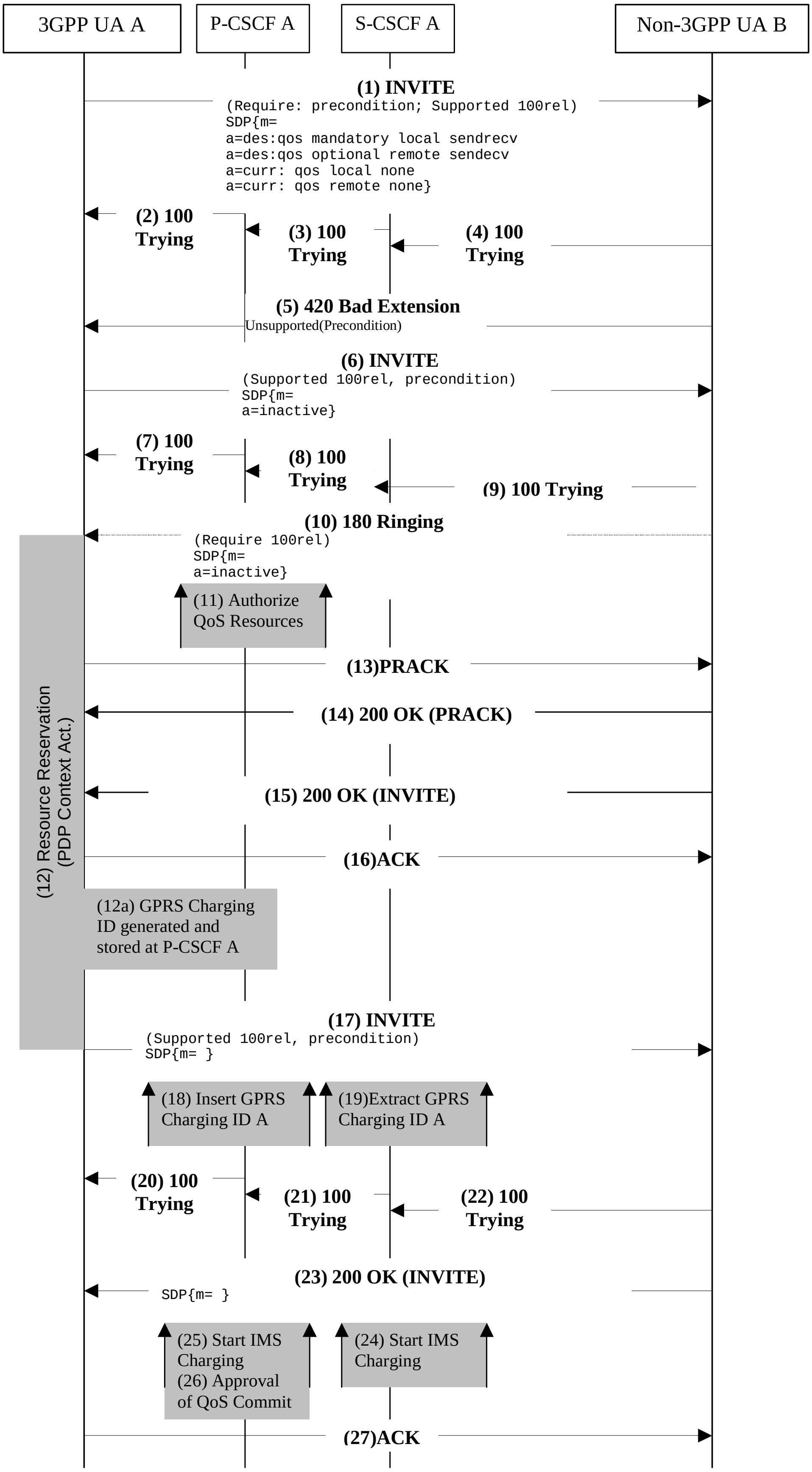


Figure 4.1.2.4.2.1/1: Using re-invite to connect calling 3GPP UA to called non-3GPP UA not supporting the SIP preconditions extension, but supporting the SIP 100rel extension.

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

4.1.2.4.2.2 Advantages

See Section 4.1.3.2.2.2.

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

4.1.2.4.2.3 Disadvantages

See Section 4.1.3.2.2.3.

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

4.1.3 3GPP UA to non-3GPP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

4.1.3.1 Description of interworking issue

The call fails, as detailed in Section 4.1.2.2.

4.1.3.2 Proposed Resolutions to interworking issue

4.1.3.2.1 B2B UA

A B2B UA is used.

4.1.3.2.1.1 Insertion of B2B UA

How the B2B UA is inserted is discussed within Section 4.1.2.4.1.1.

4.1.3.2.1.2 Functionality of B2B UA

4.1.3.2.1.2.1 Description

The B2B UA shall apply the rules given in section 4.1.2.4.1.2.1.

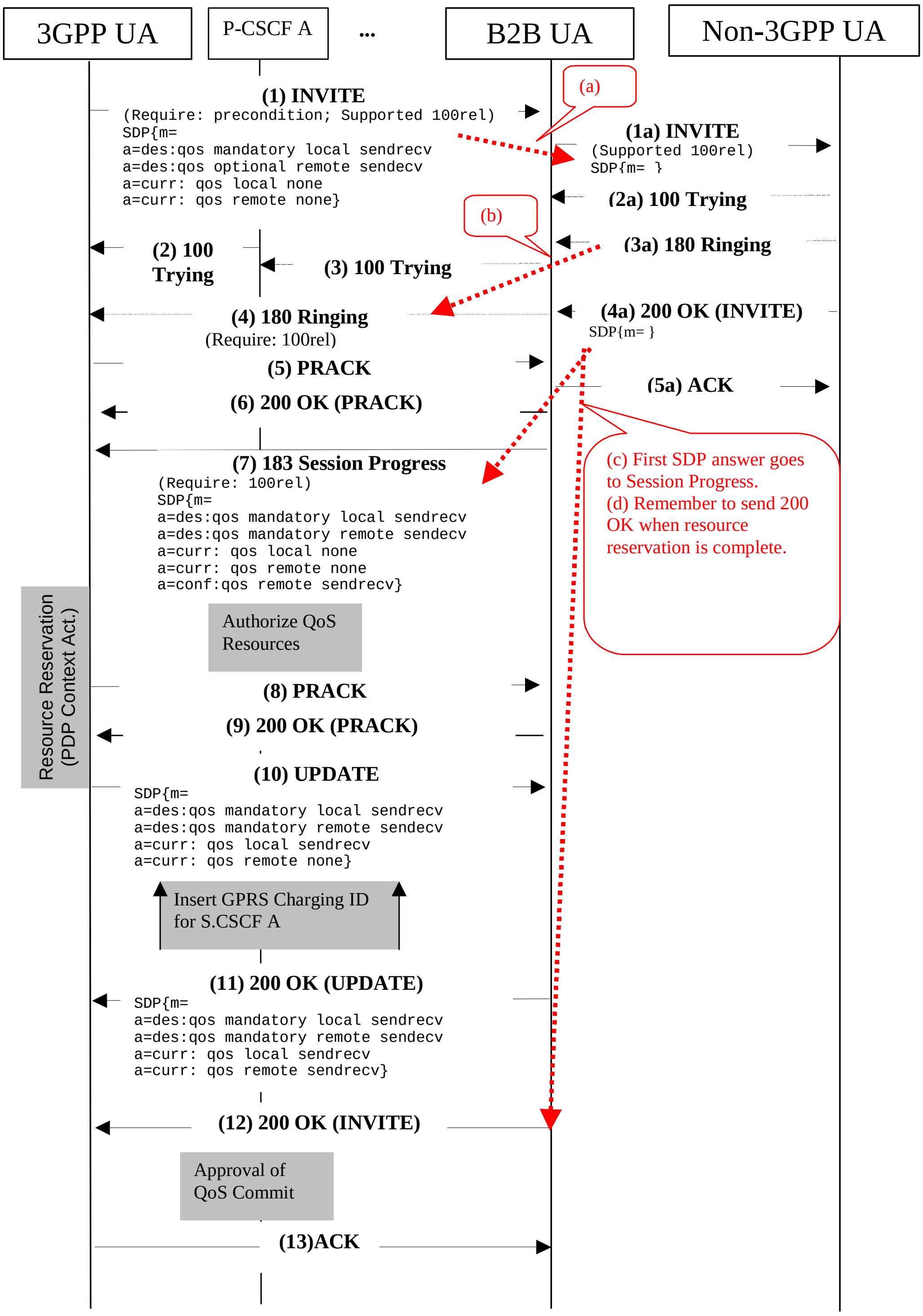


Figure 4.1.3.2.1.2/1: Functionality of B2B UA connecting calling 3GPP UA to called non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension. Calling UA sends no second SDP offer

There may be re-transmissions of the INVITE (1) by the 3GPP UA, which should be passed transparently by the B2B UA, as indicated in interaction (a).

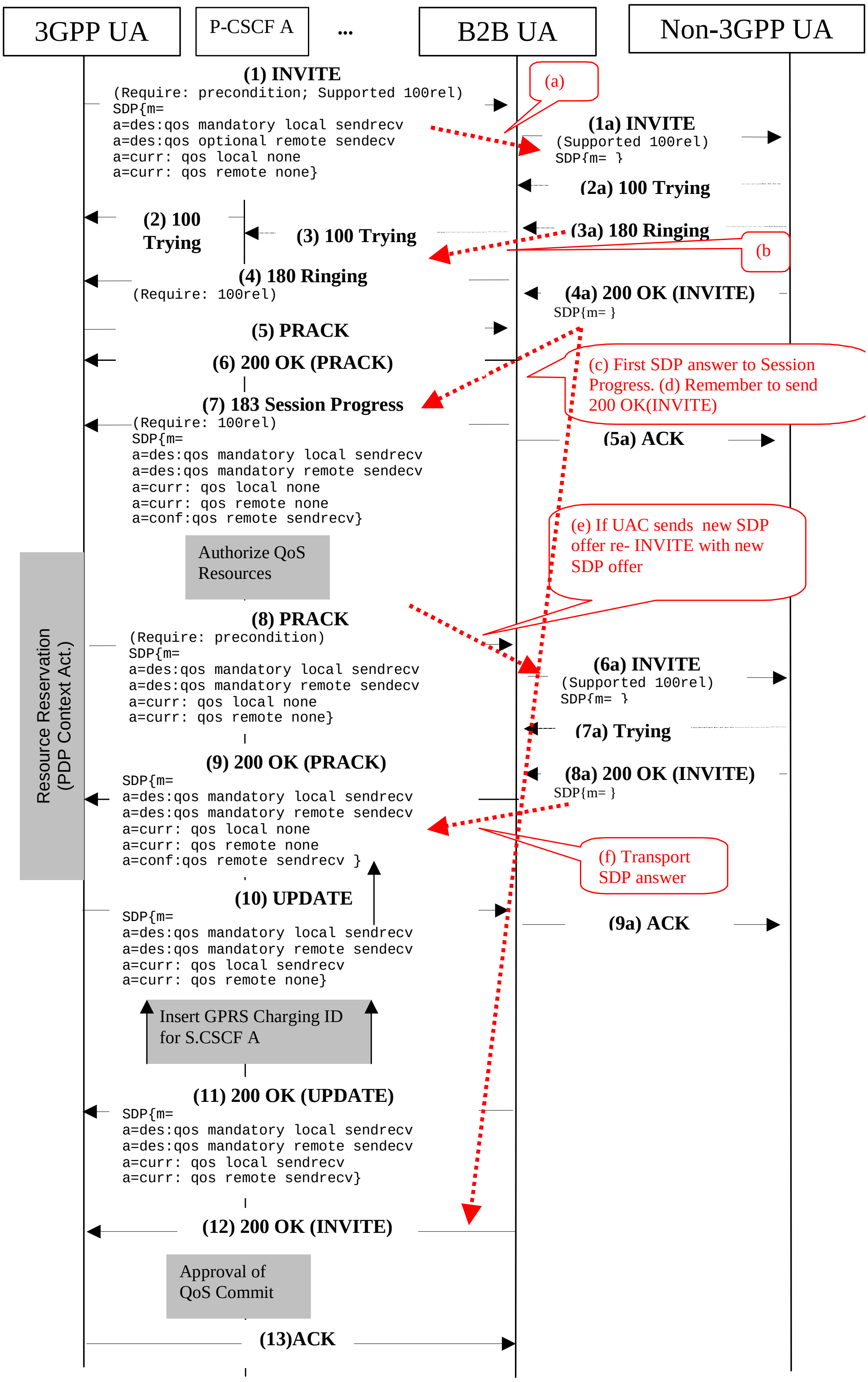


Figure 4.1.3.2.1.2/2: Functionality of B2B UA connecting calling 3GPP UA to called non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension. Calling UA sends second SDP offer

~~Editor´s Note: It has to be clarified how the B2B UA decides that the called non-3GPP UA is not supporting preconditions.~~

4.1.3.2.1.2.2 Advantages

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

General advantages of the B2B UA are discussed in Section 4.1.2.4.1.2.2.

4.1.3.2.1.2.3 Disadvantages

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

General disadvantages of the B2B UA are discussed in Section 4.1.2.4.1.2.3.

The 3GPP user perception suffers if the non-3GPP UA does not answer the call immediately, but does not send a ringing message.

The non-3GPP UA may suffer clipping.

4.1.3.2.2 Modified end-to-end ~~callflow~~call flow

4.1.3.2.2.1 Description

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

1. (e.g. in TS 24.229) The calling 3GPP UA should (not shall) require preconditions in an initial INVITE request. The calling 3GPP UA may (re-)INVITE an external UA without requiring preconditions, e.g. if receiving a 420 Bad Extension(precondition) error response. In this case, the 3GPP UA shall set the media to "inactive" when generating an SDP offer. The 3GPP UA shall 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 called 3GPP UA may accept invites not containing a "Require(precondition)" header.

3. (e.g. in TS 24.229) The called 3GPP UA may send provisional responses without requiring the 100rel extension, if the calling party did not indicate the support of the 100rel extension. In this case, the called 3GPP UA may also send a 200 OK(INVITE) before the resource reservation is complete, but shall set the media to inactive in the SDP offer or answer within this 200 OK(INVITE). The 3GPP UA shall send a re-INVITE activating the media by setting them to "send", "recv", or "sendrecv" in SDP once the local resource reservation is complete.

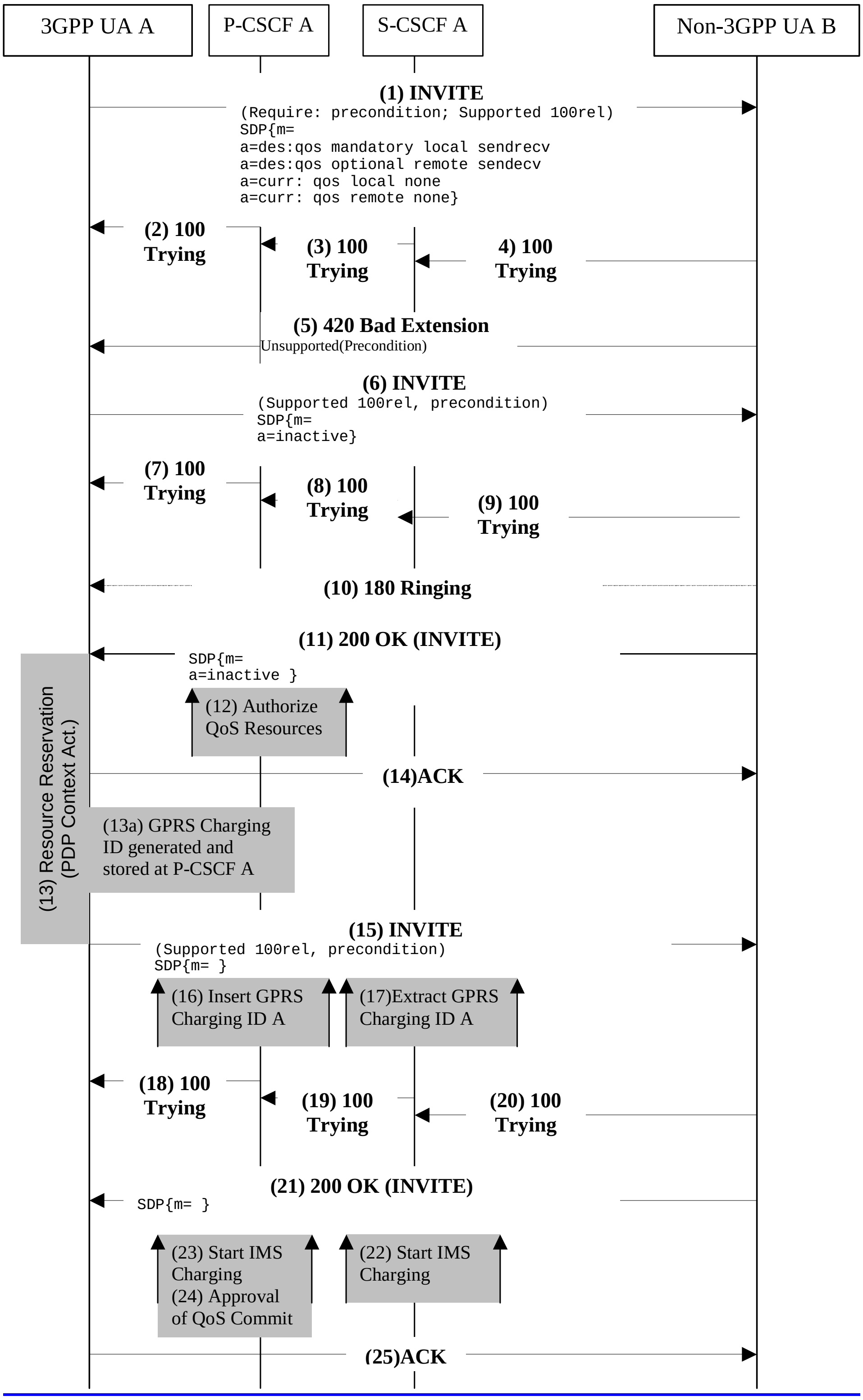
4. (e.g. in TS 29.207 and 29.208) The P-CSCF(PDF) shall approve the QoS Commit when receiving a 200 OK(INVITE) only, if media streams are active ("send", "recv", or "sendrecv" in SDP). To avoid fraud, P-CSCF(PDF) should not open gates for inactive media streams.

5. (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(INVITE) response and media streams are active ("send", "recv", or "sendrecv" in SDP).

6. (e.g. in TS 24.229): GPRS Charging ID may be 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

7. (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. As an alternative, both an SDP offer and an SDP answer with "inactive" media shall trigger the same actions with respect to the rules above.


3GPP UA A
P-CSCF A
S-CSCF A
Non-3GPP UA B
(1) INVITE
(Require: precondition; Supported 100rel)
SDP{m=
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendecv
a=curr: qos local none
a=curr: qos remote none}
(2) 100 Trying
(3) 100 Trying
4) 100 Trying
(5) 420 Bad Extension
Unsupported(Precondition)
(6) INVITE
(Supported 100rel, precondition)
SDP{m=
a=inactive}
(7) 100 Trying
(8) 100 Trying
(9) 100 Trying
(10) 180 Ringing
(11) 200 OK (INVITE)
SDP{m=
a=inactive }
(12) Authorize QoS Resources
(14)ACK
(13) Resource Reservation (PDP Context Act.)
(13a) GPRS Charging ID generated and stored at P-CSCF A
(15) INVITE
(Supported 100rel, precondition)
SDP{m= }
(16) Insert GPRS Charging ID A
(17)Extract GPRS Charging ID A
(18) 100 Trying
(19) 100 Trying
(20) 100 Trying
(21) 200 OK (INVITE)
SDP{m= }
(23) Start IMS Charging
(24) Approval of QoS Commit
(22) Start IMS Charging
(25)ACK

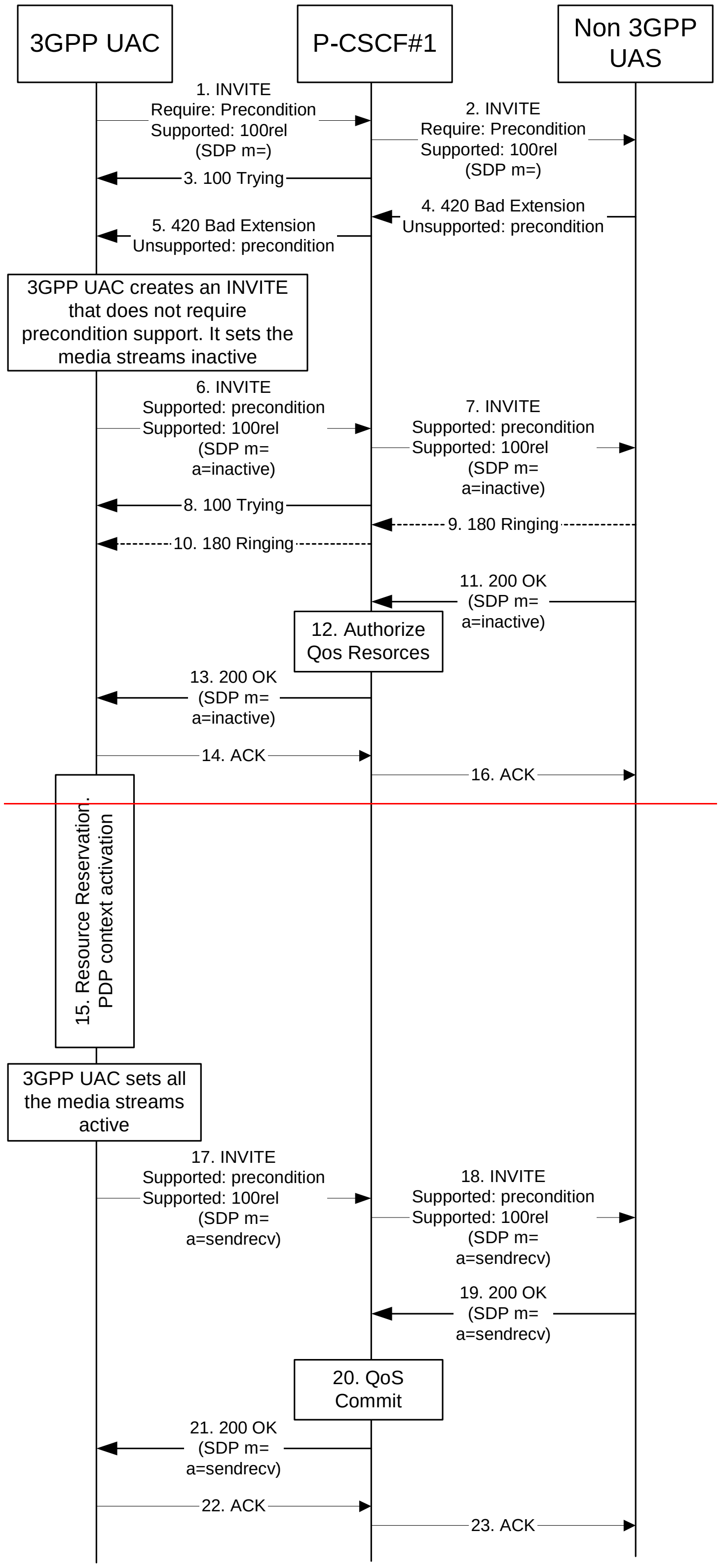


Figure 4.1.3.2.2.2/1: Using re-invite to connect calling 3GPP UA to called non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension.

~~**Editor's Note: This call flow is for futher investigation. The correctness has to be verified. Rules to decide when to insert GPRS Charging ID and when to commit QoS at the P-CSCF/PCF, at when to start IMS service charging at the S-CSCF have to be provided.**~~

4.1.3.2.2.2 Advantages

Only relatively minor changes are required.

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

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

4.1.3.2.2.3 Disadvantages

Changes have to be performed in various network entities.

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

4.2 Calling non-3GPP UA to Called 3GPP UA

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

- Non-3GPP UA supporting the SIP precondition extension, but not supporting the SIP 100rel extension, to 3GPP UA.
- Non-3GPP UA supporting the SIP preconditions extension, but not supporting the SIP update extension, to 3GPP UA.

4.2.1 Non-3GPP UA supporting the SIP 100rel extension and the SIP update extension, but not supporting the SIP preconditions extension, to 3GPP UA

4.2.1.1 Description of interworking issue

The call fails, as detailed in Section 4.2.2.2.

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.

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 ~~callflow~~call flow.

4.2.1.2 Proposed Resolutions to interworking issue

4.2.1.2.1 B2B UA

A B2B UA is used.

4.2.1.2.1.1 Insertion of B2B UA

How the B2B UA is inserted is discussed within Section 4.2.2.4.1.1.

4.2.1.2.1.2 Functionality of B2B UA

4.2.1.2.1.2.1 Description

The functionality of the B2B UA is as discussed in Section 4.2.2.4.1.2.1.

The B2B UA shall pass additional UPDATE messages, which are not related to the precondition extension, and related provisional acknowledge messages.

4.2.1.2.1.2.2 Advantages

General advantages of the B2B UA are discussed in Section 4.2.2.4.1.2.2.

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

4.2.1.2.1.2.3 Disadvantages

General disadvantages of the B2B UA are discussed in Section 4.2.2.4.1.2.3.

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

4.2.1.2.2 Modified end-to-end ~~callflow~~call flow

4.2.1.2.2.1 Description

The restriction to disallow a direct communication with a calling non-3GPP UA, which does not indicate the support or requirement of the SIP preconditions extension is removed from TS 24.229.

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

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

The resulting call flows are similar to the flows in Section 4.2.2.4.2.1, possibly with additional UPDATE messages inserted.

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

4.2.1.2.2.2 Advantages

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

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

4.2.1.2.2.3 Disadvantages

No disadvantages have been identified.

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

4.2.2 Non-3GPP SIP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA

4.2.2.1 Description of interworking issue

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

4.2.2.2 Flow diagram

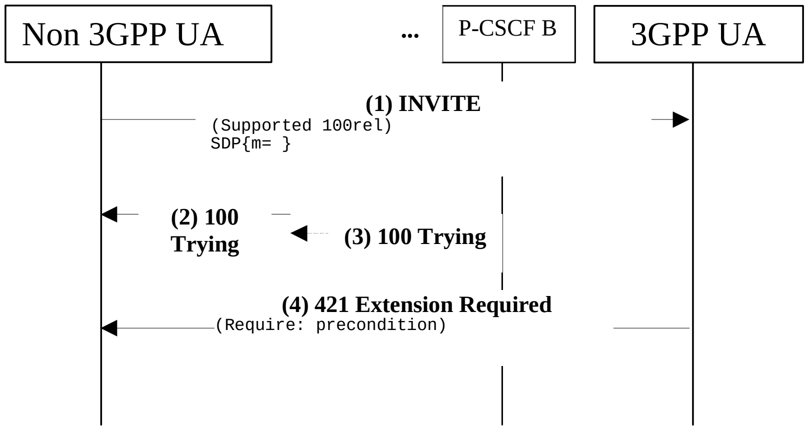


Figure 4.2.2.2/1: Non-3GPP SIP UA not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA.

4.2.2.3 Implications of Identified interworking issue

The call fails.

4.2.2.4 Proposed resolutions to interworking issue

4.2.2.4.1 B2B UA

A B2B UA is used.

4.2.2.4.1.1 Insertion of B2B UA

4.2.2.4.1.1.1 Static Insertion of B2B UA

4.2.2.4.1.1.1.1 Description

A B2B UA is permanently inserted at connection between IMS and a given external network. This B2B UA handles all calls, including calls where the ~~callflow~~call flows may be passed without modification.

The B2B UA shall be inserted in the border of the IMS for all calls entering the IMS from an external network (except for another IMS).

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 B2B UA becomes active only when receiving an INVITE message without an indication of the support or requirement of the preconditions extension from the Non-3GPP UA, as depicted in Figure 4.2.2.4.1.1.1.1/1. Otherwise, the B2B UA passes all SIP messages received at one side to the other side.

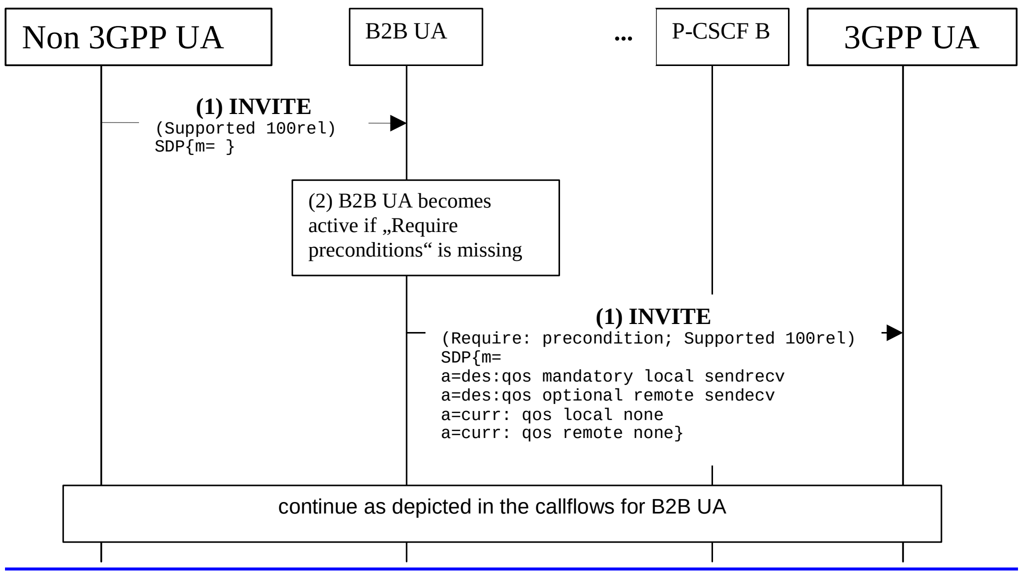


Figure 4.2.2.4.1.1.1.1/1: Activation of static B2B UA connecting Non-3GPP UA not indicating support of the SIP preconditions extension to 3GPP UA.

4.2.2.4.1.1.1.2 Advantages

4.2.2.4.1.1.1.3 Disadvantages

Additional processing load.

Additional delay.

Lack of signalling transparency may restrict the compatibility with future extensions for all calls.

The B2B UA may be activated unnecessarily, if the Non-3GPP UA supports the precondition extension, but fails to indicate this in the INVITE message.

4.2.2.4.1.2 Functionality of B2B UA

4.2.2.4.1.2.1 Description

Editor's Note: The following rules have been agreed only as basis for further contributions and have not yet been investigated in detail.

The B2B UA shall apply the following rules:

1. The B2B UA 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 B2B UA shall also comply with the SIP 100rel and update extensions.
3. On the IMS side, the B2B UA shall also comply to and apply the SIP 100rel, update and precondition extensions.
4. The B2B UA shall pass SIP requests arriving at one call leg to the call leg at the other side, with the exceptions below.
5. The B2B UA shall pass 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 B2B UA shall pass SIP final responses arriving at one call leg to the call leg at the other side, with the exceptions below.

7. The B2B UA shall not require the SIP preconditions extension on the non-IMS side.

8. The B2B UA shall insert preconditions in SDP sent from non-3GPP UA to 3GPP UA.

9. The B2B UA should remove preconditions in SDP sent from 3GPP UA to non-3GPP UA

10. The B2B UA shall not pass PRACK and 200 OK(PRACK) messages.

11. The B2B UA shall inspect an INVITE message 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 B2B UA receives an SDP offer in a provisional response from the IMS side, the B2B UA shall send the SDP offer in a 200 OK(invite) message at the non-IMS side. The B2B UA shall then forward the SDP answer received in the ACK message from the non-IMS side to the PRACK message for the provisional response on the IMS-side.

13. If the 100rel extension is not supported on the non-IMS side, and the B2B UA receives an SDP answer in a provisional response from the IMS side, the B2B UA shall send the SDP answer in a 200 OK(invite) message at the non-IMS side.

14. For a re-Invite from the IMS side to the Non-IMS side, the B2B UA shall apply the rules in Section 4.1.2.4.1.2.1.

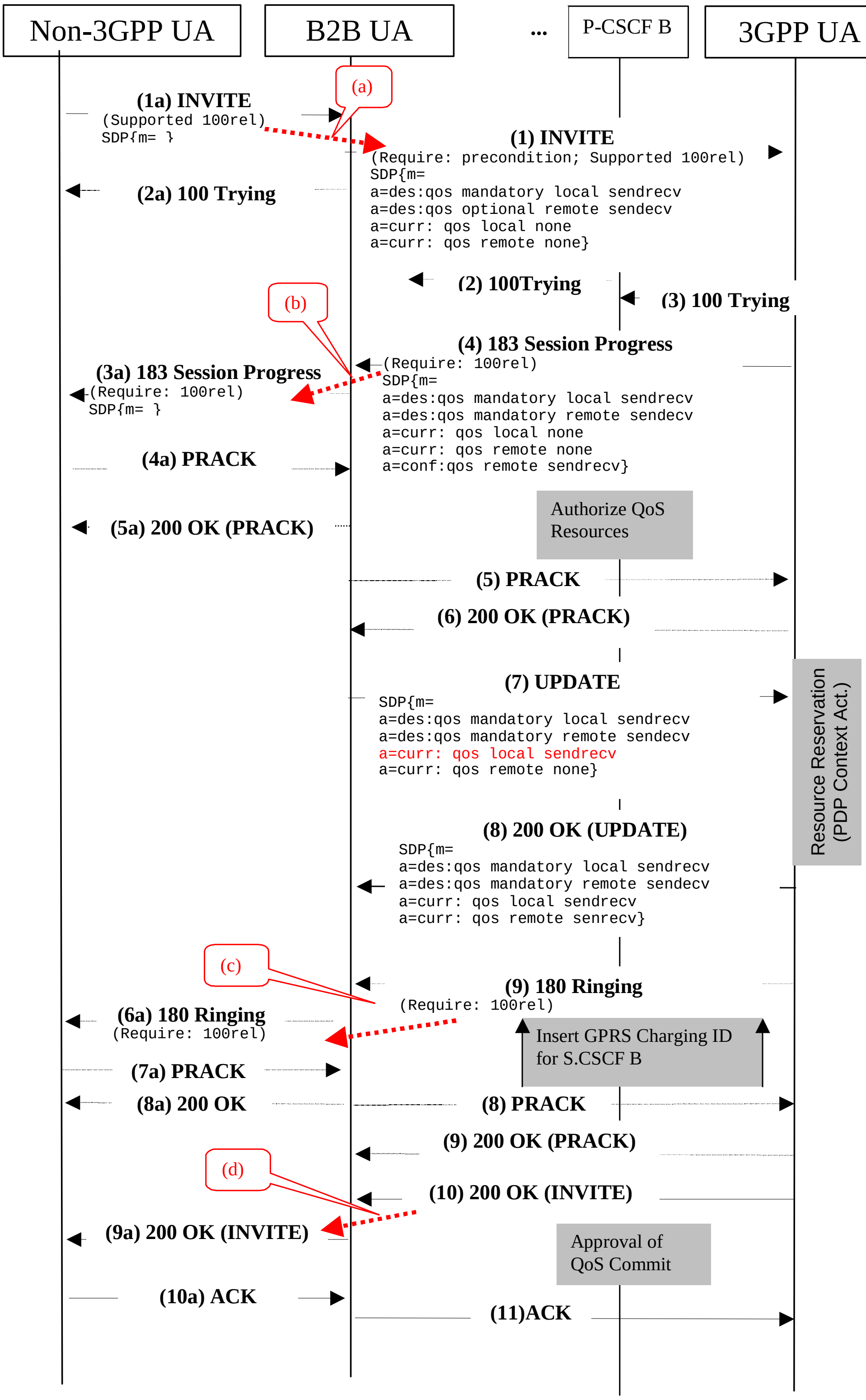


Figure 4.2.2.4.1.2.1/1: Functionality of B2B UA connecting calling non-3GPP SIP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension, to called 3GPP UA.

4.2.2.4.1.2.2 Advantages

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

4.2.2.4.1.2.3 Disadvantages

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

The functionality and implementation of the B2B UA is complicated and will have to include the following features:

- Storage of SDP.
- Own timer supervision of dialogues on both call legs.

The compatibility with future SIP extensions may be limited by the need to update the B2B UA. This may limit the network's ability to deploy new IP multimedia applications.

4.2.2.4.2 Modified end-to-end ~~callflow~~call flow

4.2.2.4.2.1 Description

The restriction to disallow a direct communication with a calling non-3GPP UA, which does not indicate the support or requirement of the SIP preconditions extension is removed from TS 24.229.

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

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

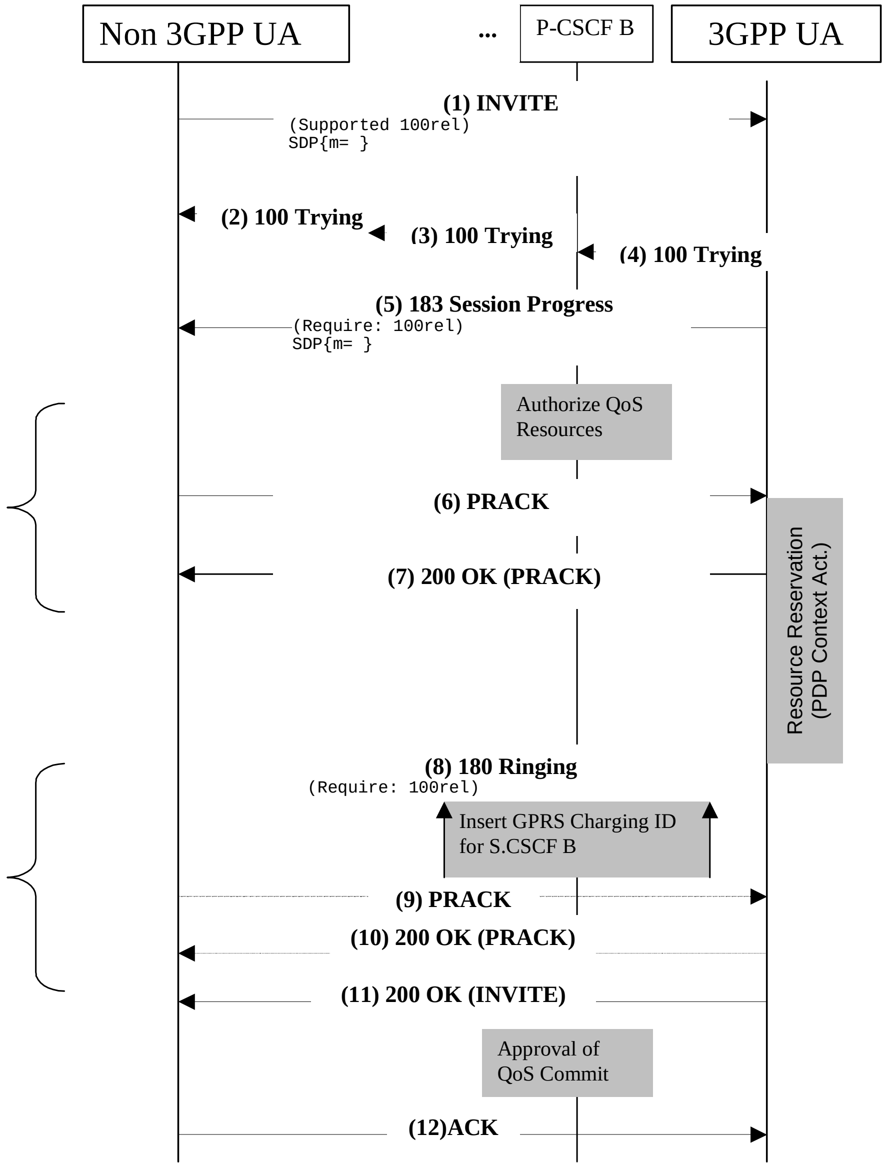


Figure 4.2.2.4.2.1/1: Modified end-to-end call flow for Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA. SDP offer in Invite.

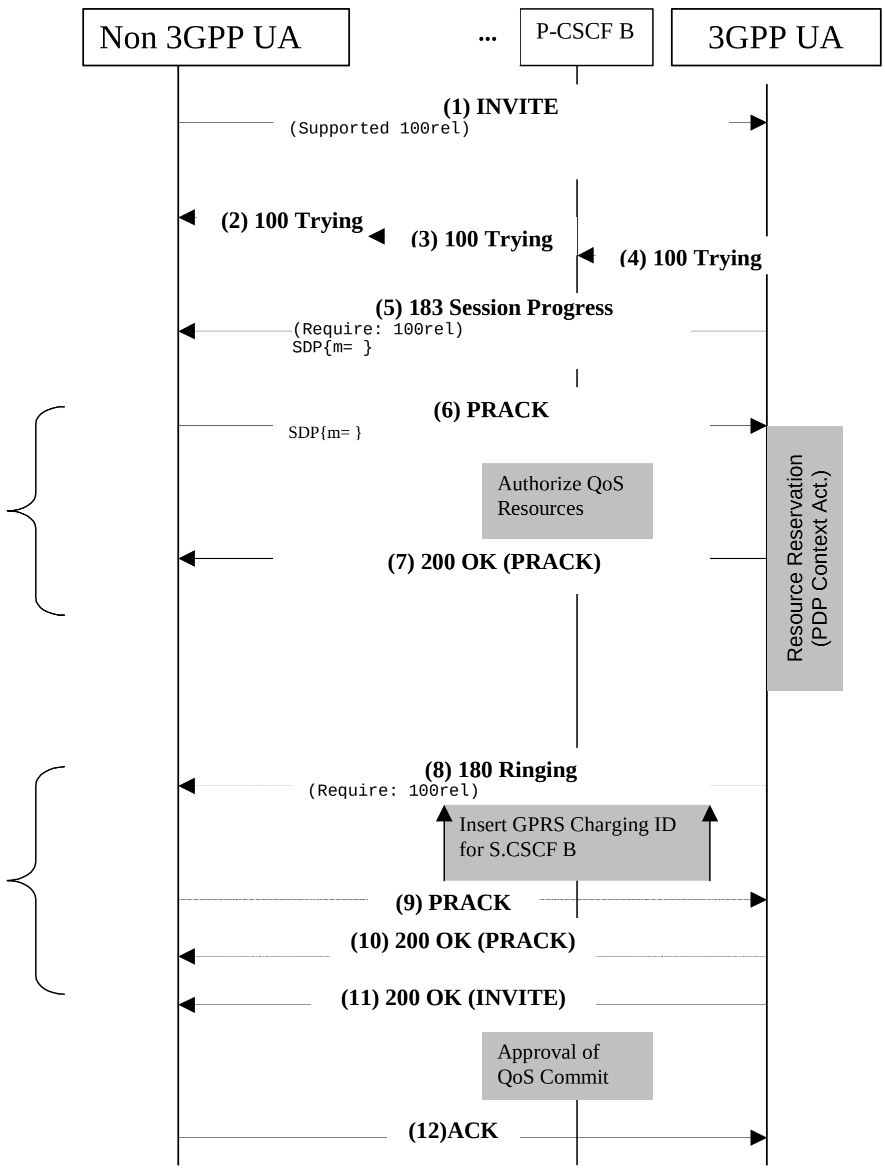


Figure 4.2.2.4.2.1/2: Modified end-to-end call flow for Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA. No SDP offer in Invite.

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

4.2.2.4.2.2 Advantages

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

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

4.2.2.4.2.3 Disadvantages

No disadvantages have been identified.

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

4.2.3 Non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to 3GPP UA

4.2.3.1 Description of interworking issue

The call fails, as detailed in Section 4.2.2.2.

4.2.3.2 Proposed Resolutions to interworking issue

4.2.3.2.1 B2B UA

A B2B UA is used.

4.2.3.2.1.1 Insertion of B2B UA

4.2.3.3.1.1.1 Static Insertion of B2B UA

4.2.3.3.1.1.1.1 Description

A B2B UA is permanently inserted at connection between IMS and a given external network. This B2B UA handles all calls, including calls where the call flows may be passed without modification.

The B2B UA shall be inserted in the border of the IMS for all calls entering the IMS the IMS from an external network (except for another IMS).

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 B2B UA becomes active only when receiving an INVITE message without an indication of the support or requirement of the 100rel extension from the Non-3GPP UA, as depicted in Figure 4.2.3.3.1.1.1.1/1. Otherwise, the B2B UA passes all SIP messages received at one side to the other side.

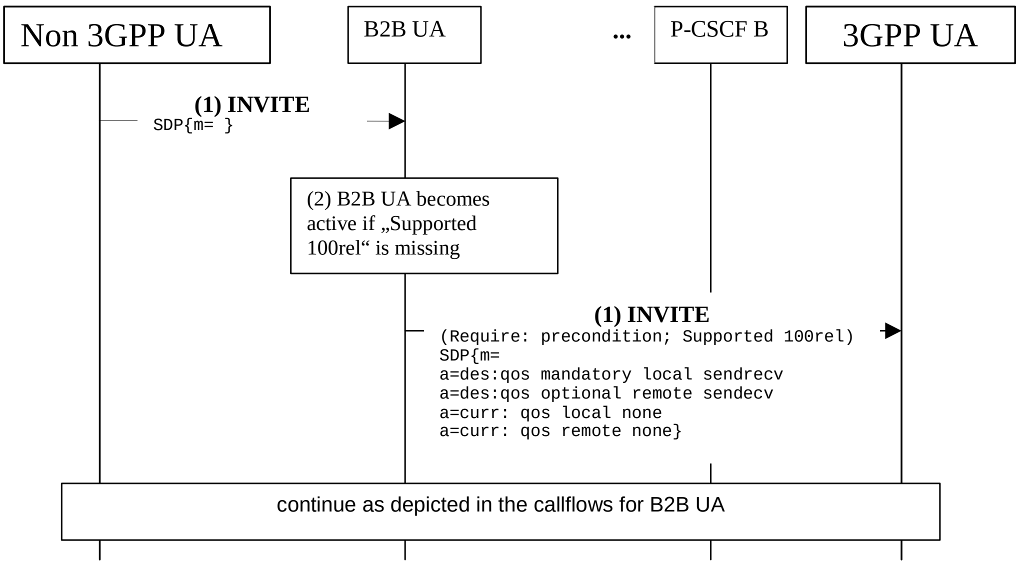


Figure 4.2.3.3.1.1.1.1/1: Activation of static B2B connecting Non-3GPP SIP UA not indicating support of the SIP preconditions extension to 3GPP UA.

4.2.3.3.1.1.1.2 Advantages

4.2.3.3.1.1.1.3 Disadvantages

Additional processing load.

Additional delay.

Lack of signalling transparency may restrict the compatibility with future extensions for all calls.

The B2B UA may be activated unnecessarily, if the Non-3GPP UA supports the 100 rel extension, but fails to indicate this in the INVITE message. RFC 3262 [6] recommends that a UAC supporting the 100rel extension indicates this capability in the INVITE message, but does not mandate the UAC to do so.

~~How the B2B UA is inserted is discussed within Section 4.2.2.4.1.1~~

4.2.3.2.1.2 Functionality of B2B UA

4.2.3.2.1.2.1 Description

The B2B UA shall apply the rules given in section 4.2.2.4.1.2.1.

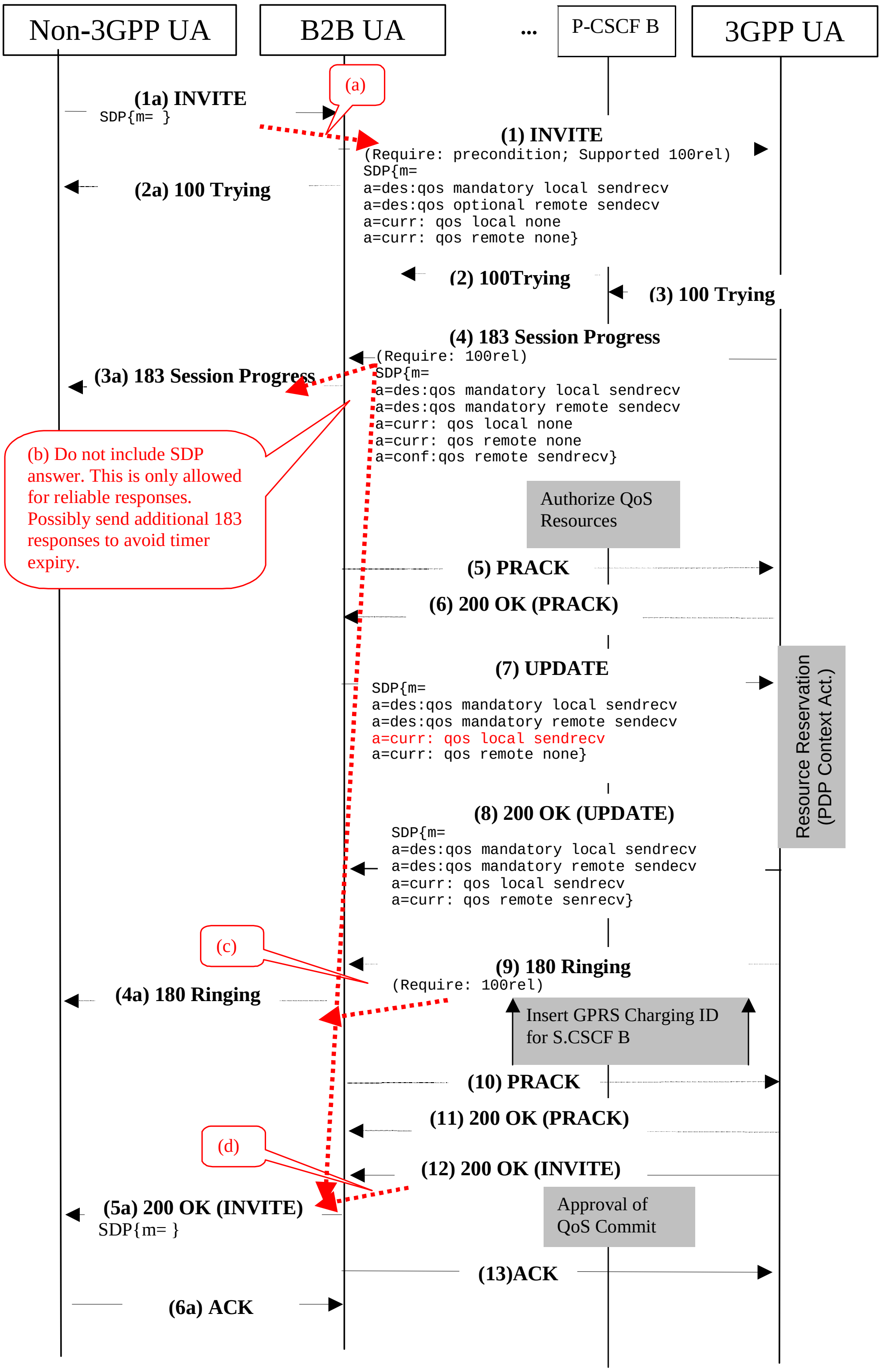


Figure 4.2.3.2.1.2.1/1: Functionality of B2B UA connecting calling non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to called 3GPP UA. SDP offer in INVITE request.

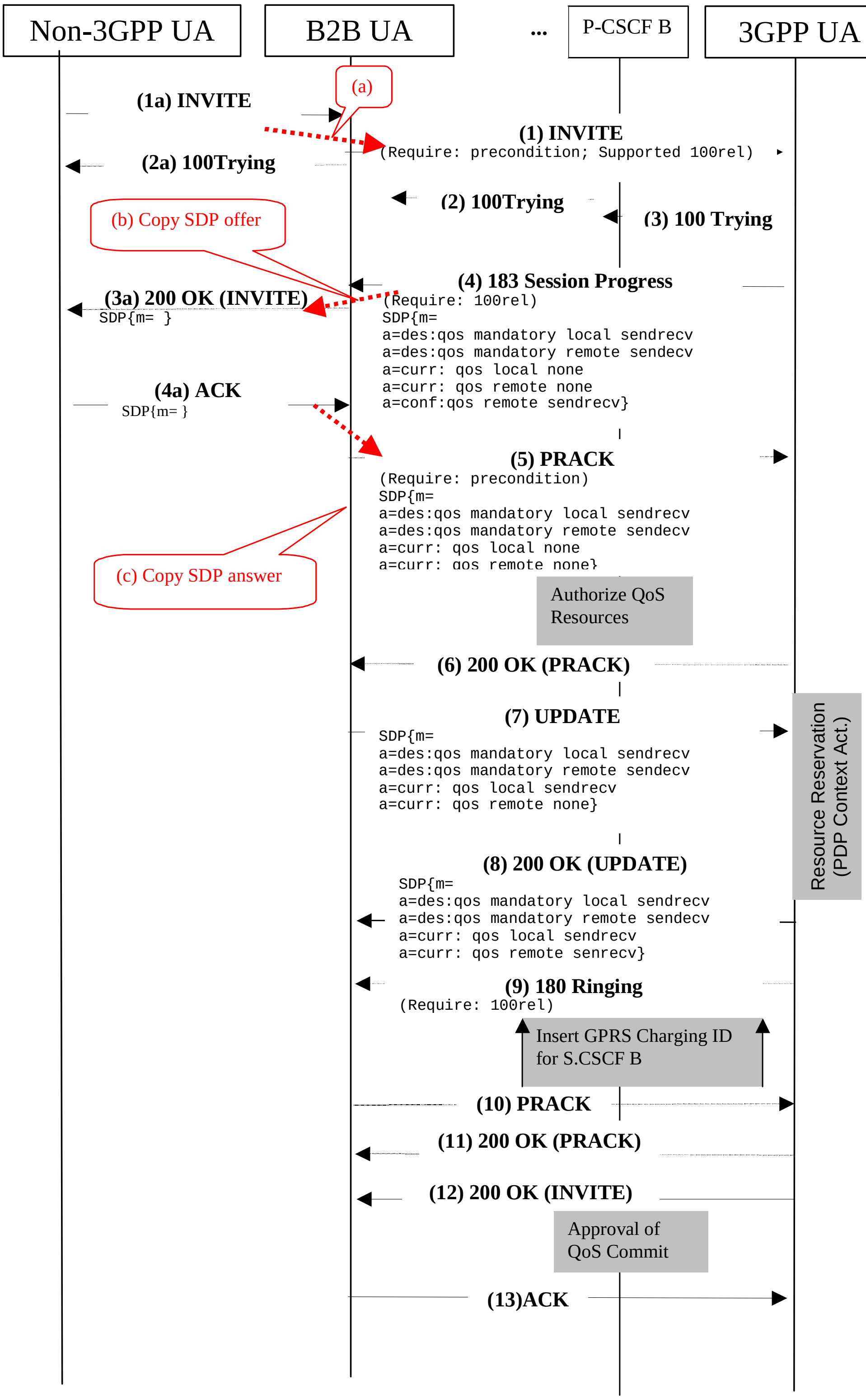


Figure 4.2.3.2.1.2.1/2: Functionality of B2B UA connecting calling non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to called 3GPP UA. SDP offer in OK response.

4.2.3.2.1.2.2 Advantages

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

General advantages of the B2B UA are discussed in Section 4.2.2.4.1.2.2.

4.2.3.2.1.2.3 Disadvantages

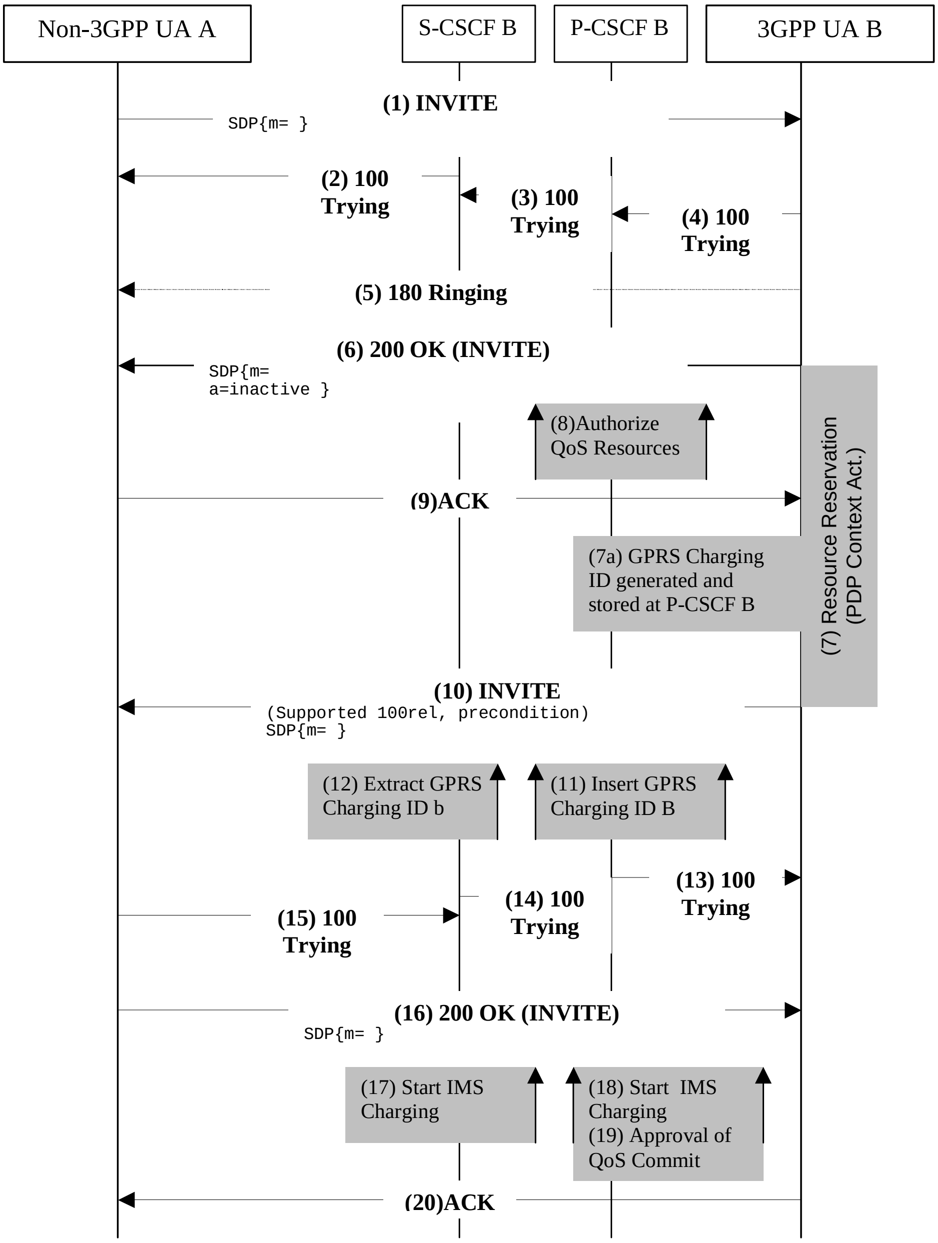
~~Editor's Note: This section list possible disadvantages of this suggestion compared to competing suggestions.~~

General disadvantages of the B2B UA are discussed in Section 4.2.3.4.1.2.3.

4.2.3.2.2 Modified end-to-end ~~callflow~~call flow

4.2.3.2.2.1 Description

The rules described in Section 4.1.3.2.2.1 are applied.



Non-3GPP UA A
S-CSCF B
P-CSCF B
3GPP UA B
(1) INVITE
SDP{m= }
(2) 100 Trying
(3) 100 Trying
(4) 100 Trying
(5) 180 Ringing
(6) 200 OK (INVITE)
SDP{m=
a=inactive }
(8)Authorize QoS Resources
(9)ACK
(7a) GPRS Charging ID generated and stored at P-CSCF B
(7) Resource Reservation (PDP Context Act.)
(10) INVITE
(Supported 100rel, precondition)
SDP{m= }
(12) Extract GPRS Charging ID b
(11) Insert GPRS Charging ID B
(13) 100 Trying
(14) 100 Trying
(15) 100 Trying
(16) 200 OK (INVITE)
SDP{m= }
(17) Start IMS Charging
(18) Start IMS Charging
(19) Approval of QoS Commit
(20)ACK

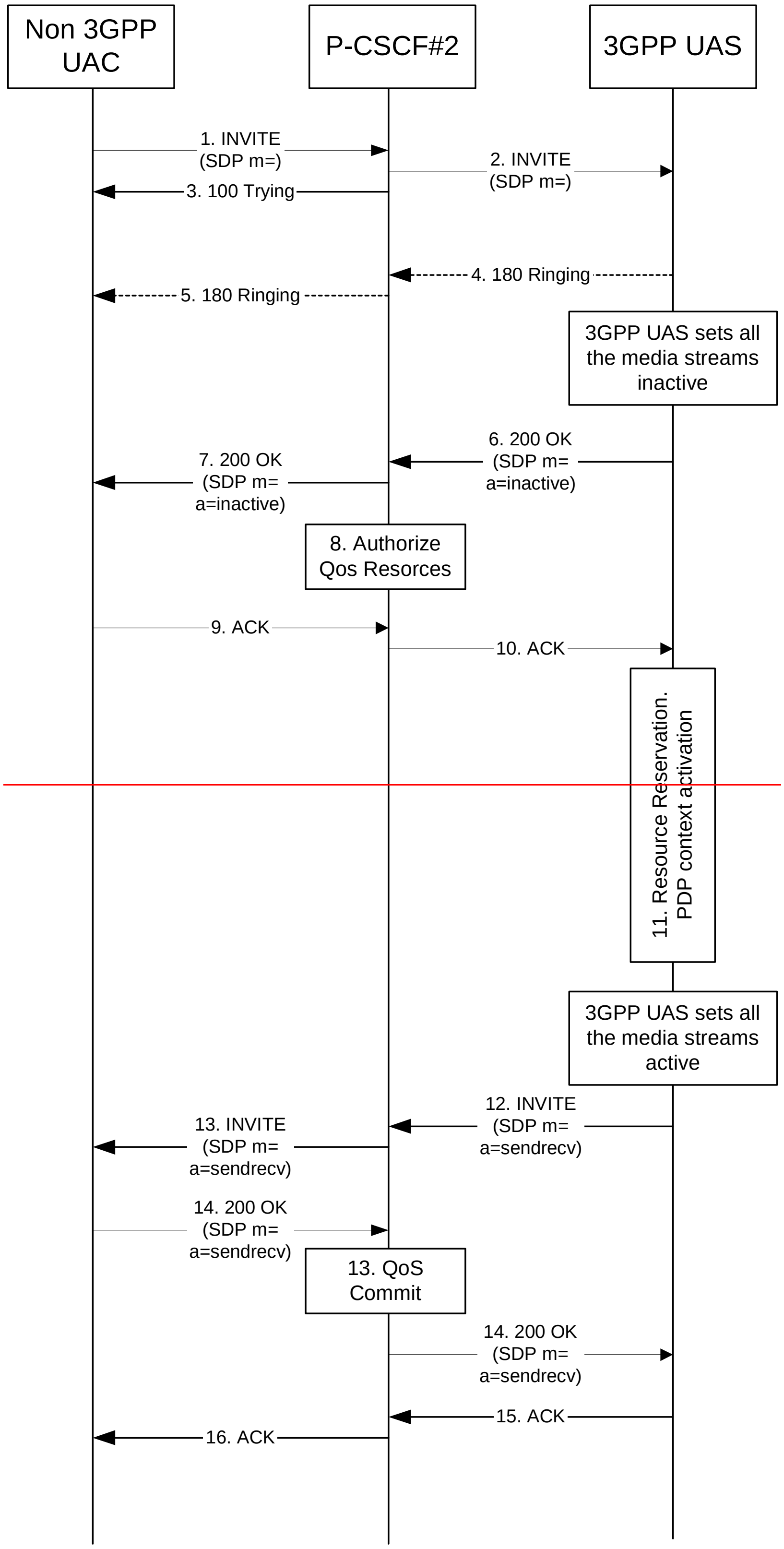
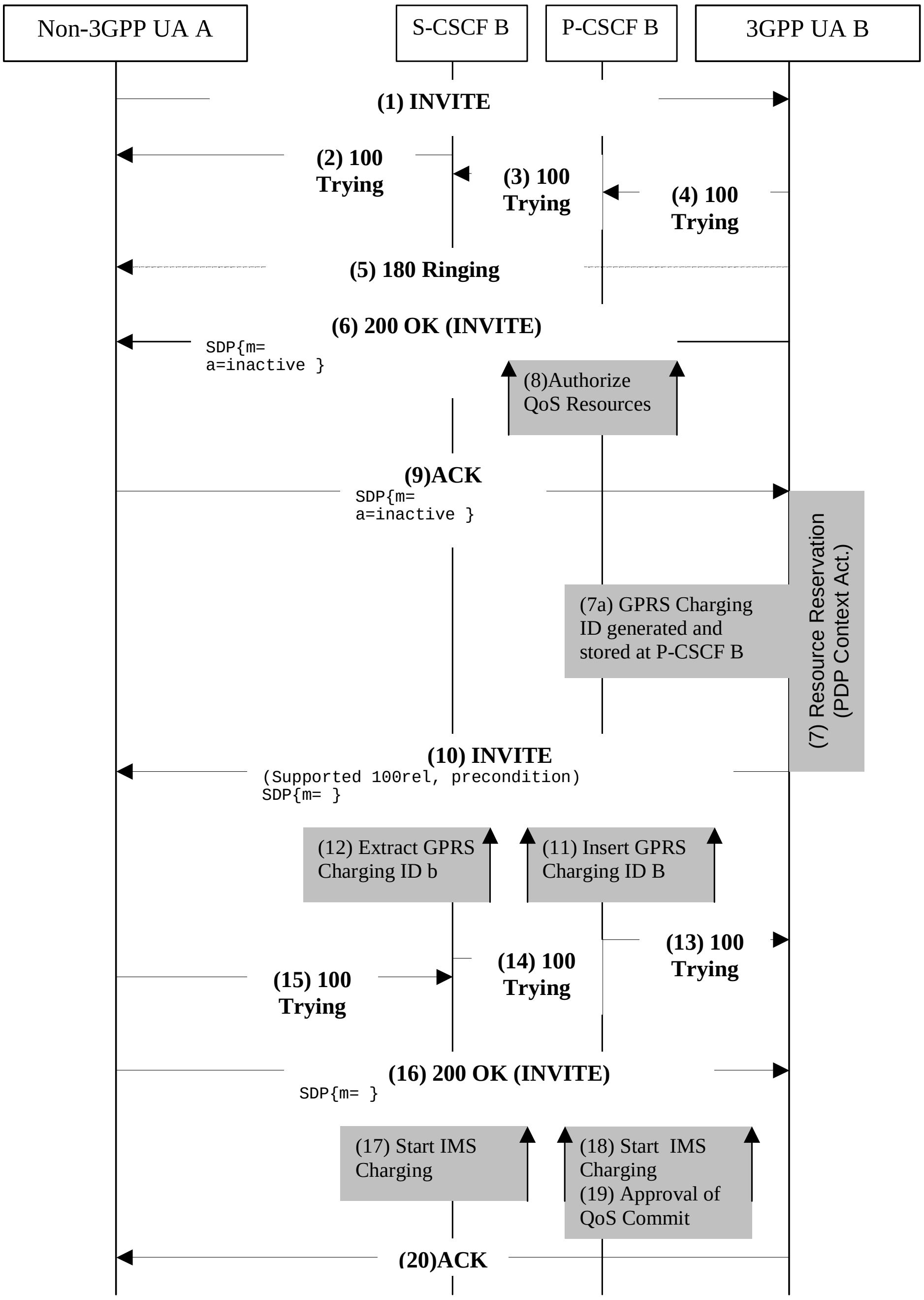


Figure 4.2.3.2.2.2/1: Using re-invite to connect calling non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension to called 3GPP UA. The INVITE contains SDP.



Non-3GPP UA A
S-CSCF B
P-CSCF B
3GPP UA B
(1) INVITE
(2) 100 Trying
(3) 100 Trying
(4) 100 Trying
(5) 180 Ringing
(6) 200 OK (INVITE)
SDP{m= a=inactive }
(8)Authorize QoS Resources
(9)ACK
SDP{m= a=inactive }
(7a) GPRS Charging ID generated and stored at P-CSCF B
(7) Resource Reservation (PDP Context Act.)
(10) INVITE
(Supported 100rel, precondition)
SDP{m= }
(12) Extract GPRS Charging ID b
(11) Insert GPRS Charging ID B
(13) 100 Trying
(14) 100 Trying
(15) 100 Trying
(16) 200 OK (INVITE)
SDP{m= }
(17) Start IMS Charging
(18) Start IMS Charging
(19) Approval of QoS Commit
(20)ACK

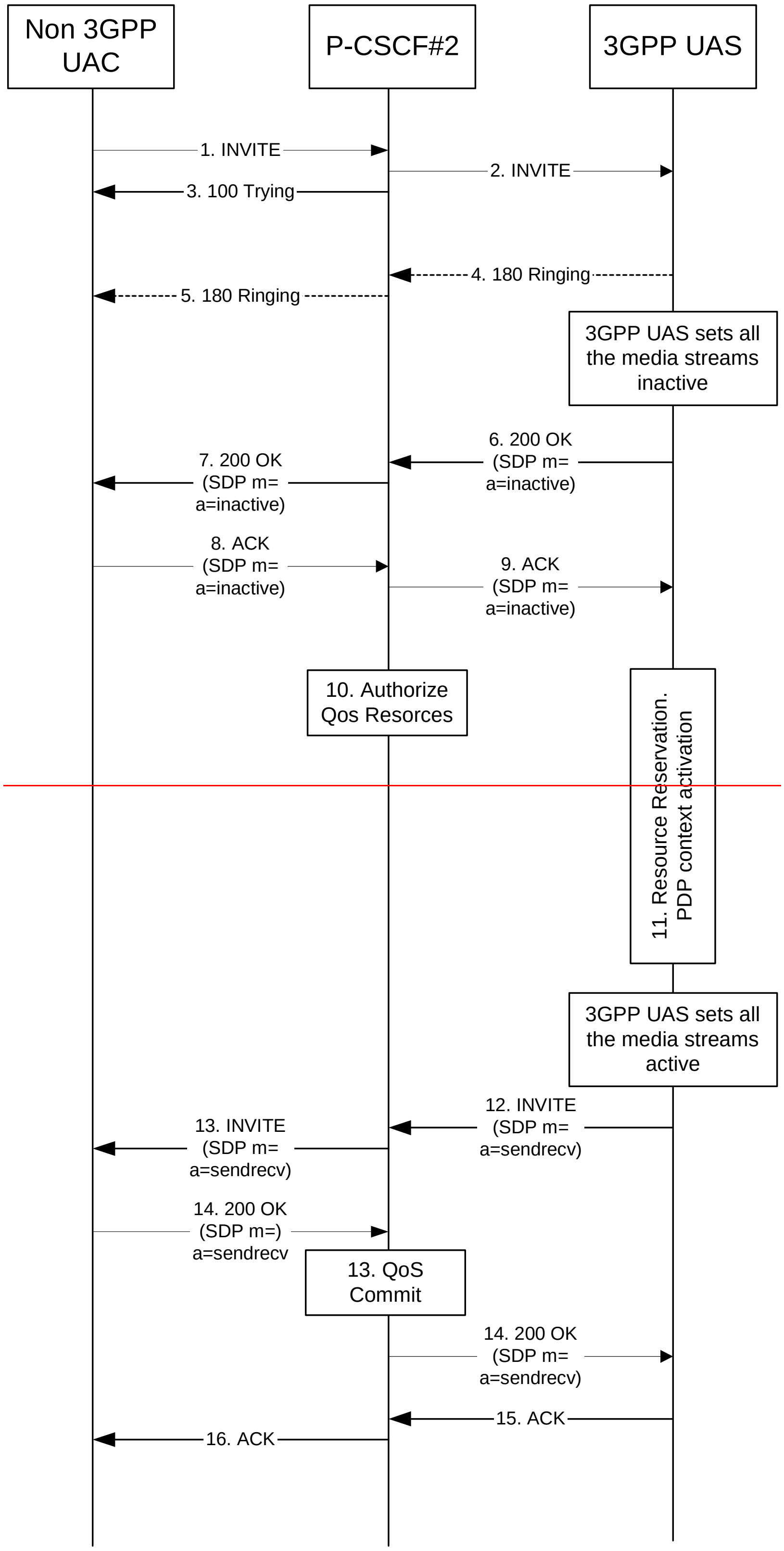


Figure 4.2.3.2.2.2/2: Using re-invite to connect calling non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension to called 3GPP UA. The INVITE contains no SDP.

~~**Editor's Note: These call flows are for futher investigation. The correctness has to be verified. Rules to decide when to insert GPRS Charging ID and when to commit QoS at the P-CSCF/PCF, at when to start IMS service charging at the S-CSCF have to be provided.**~~

4.2.3.2.2.2 Advantages

See Section 4.1.3.2.2.2.

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

4.2.3.2.2.3 Disadvantages

See Section 4.1.3.2.2.3.~~Editor's Note: This section list possible disadvantages of this suggestion compared to competing suggestions.~~

Annex A: Interworking topic template

4.x *Topic Name*

4.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.

4.x.2 Flow diagram

Editor's Note: This section contains a flow diagram illustrating the technical description of the possible interworking topic.

4.x.3 Impact of Identified interworking issue

Editor's Note: Identified interworking issues to be considered

- User interaction (call setup time, delay etc)
- Charging and Billing Implications (no charging etc)
- SIP Media ~~authorization~~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 ~~occurance~~occurrence

4.x.4 Proposed Resolutions to interworking issue

Editor's Note: This section contains one or more suggestions how an interworking may be performed.

4.x.4.y *Suggestion yy*

4.x.4.y.1 Description

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

4.x.4.y.1 Advantages

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

4.x.4.y.1 Disadvantages

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

4.x.5 Preferred Suggestion

Editor's Note: This section identifies the preferred of the above suggestions, if a consensus has been found.

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~~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: ~~Callflow~~Call flows between rogue 3GPP UA to non-3GPP UA, if 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 does not support or use these extensions.

This annex details the consequences, in case a rogue 3GPP UA does not behave according to TS 24.229 and does not apply some or all of the above SIP extensions.

The numbering of this Annex corresponds to the numbering of Section 4. For example, Sections C.2.1 and 4.2.1 consider the same ~~seecnario~~scenario.

C.1 ~~calling~~ Calling ~~rogue~~ Rogue 3GPP UA to ~~called~~ Called non-3GPP UA

C.1.1 ~~rogue~~ Rogue 3GPP UA to non-3GPP UA supporting the SIP 100rel extension and the SIP update extension, but not supporting 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 ~~callflow~~call flow:

- The calling or the called UA may send UPDATE messages at various places within the ~~callflow~~call flow. Those messages may include additional SDP offers. Due to the large number of possibilities, such ~~callflow~~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.1.2 ~~rogue~~ Rogue 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting 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 Section C.1.1, 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.1.3 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.

C.1.2.2 Flow diagram

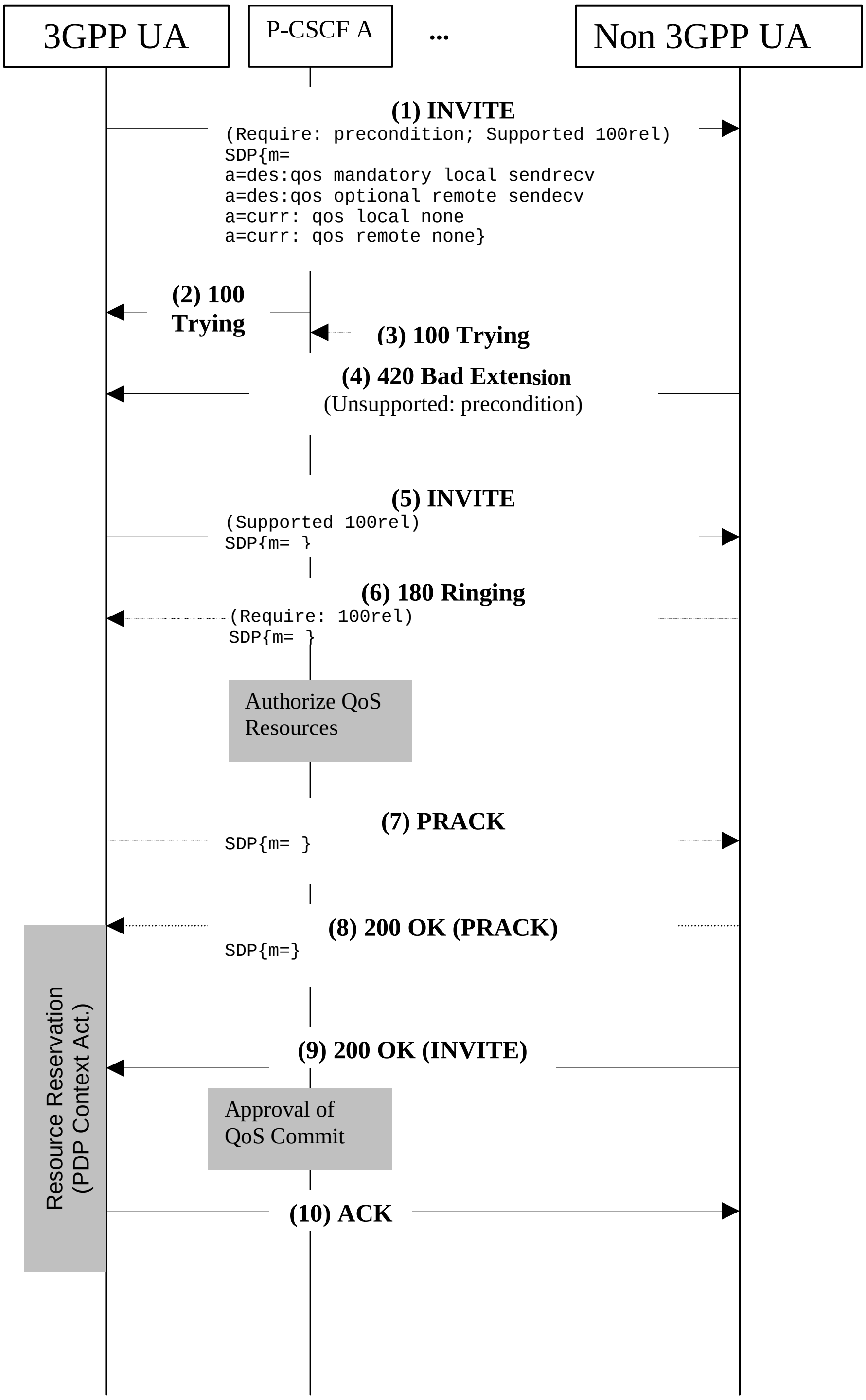


Figure C.1.2.2/1: rogue 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension

C.1.2.3 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 Rogue 3GPP UA to non-3GPP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

C.1.3.1 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.

C.1.3.2 Flow diagram

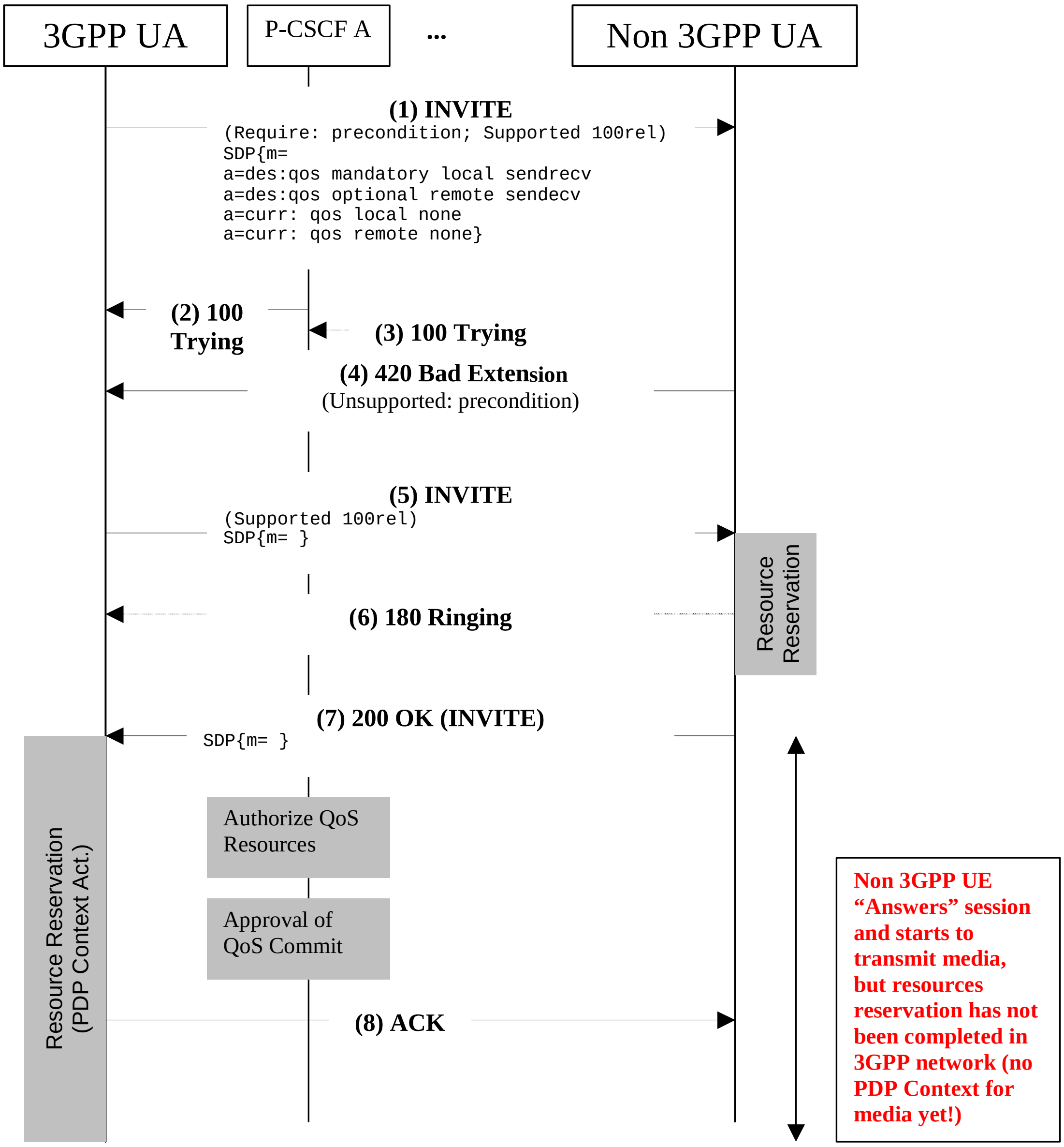


Figure C.1.3.2/1: 3GPP UA to non-3GPP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension.

~~1.~~(5) INVITE

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

~~2.~~(6) 180 Ringing

The ~~Non~~ non-3GPP ~~UE~~ UA **may optionally** send the "180 Ringing" message to the 3GPP UE. As the non-3GPP ~~UE~~ 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.

~~3.~~(7) 200 OK (Answer)

The ~~Non~~ non-3GPP ~~UE~~ UA sends the "200 OK" message to the 3GPP UE to indicate that the called party has answered. As the ~~Non~~ non-3GPP ~~UE~~ 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.

~~4.~~(8) ACK

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

C.1.3.3 Impacts of Identified interworking issue

C.1.3.3.1 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.

C.1.3.3.2 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.

C.1.3.3.3 SIP Media ~~authorization~~authorisation

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

C.1.3.3.4 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 ~~PCF~~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(~~PCF~~PDF).

C.1.3.3.5 Fraudulent and security risks

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

C.2 Calling non-3GPP UA to ~~called~~ Called ~~rogue~~ Rogue 3GPP UA

C.2.1 Non-3GPP UA supporting the SIP 100rel extension and the SIP update extension, but not supporting the SIP preconditions extension, to rogue 3GPP UA

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 ~~callflow~~call flow. The UA, may send UPDATE messages at various places within the ~~callflow~~call flow. Those messages may include additional

SDP offers. Due to the large number of possibilities, such ~~callflow~~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.

C.2.2 Non-3GPP SIP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA

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.

C.2.2.2 Flow diagram

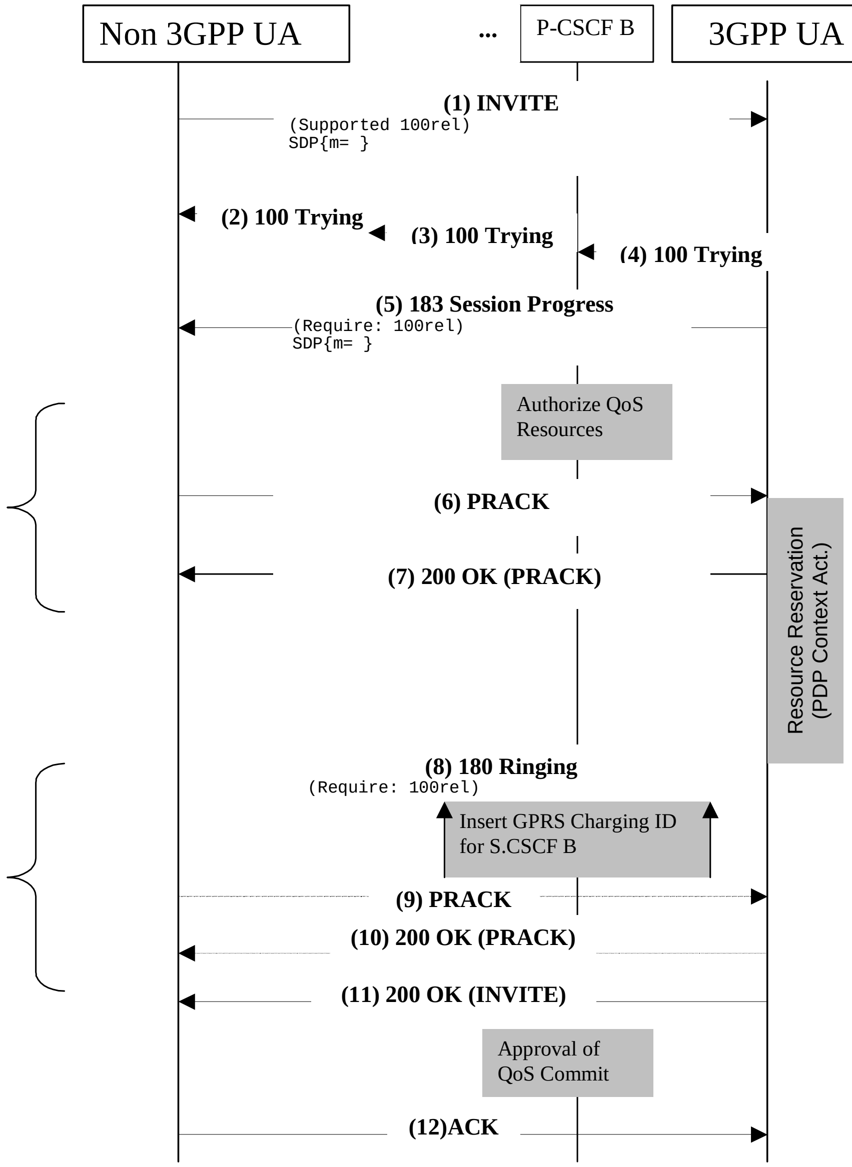


Figure C.2.2.2/1: Non-3GPP UA supporting the 100rel SIP extension, but not supporting the ~~the~~ SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in Invite.

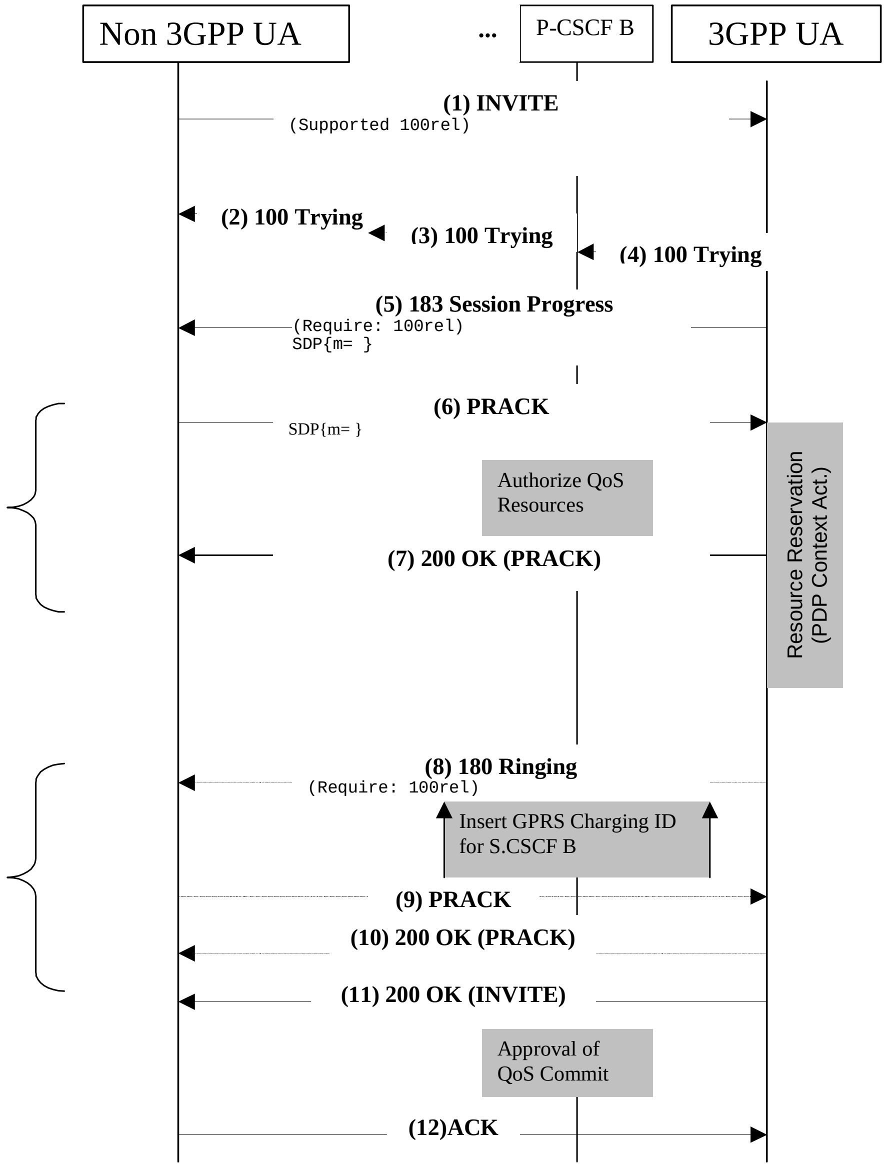


Figure C.2.2.2/2: Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. No SDP offer in Invite.

C.2.2.3 Impacts of Identified interworking issue

No negative impacts have been identified.

C.2.3 Non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA

C.2.3.1 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.

C.2.3.2 Flow diagram

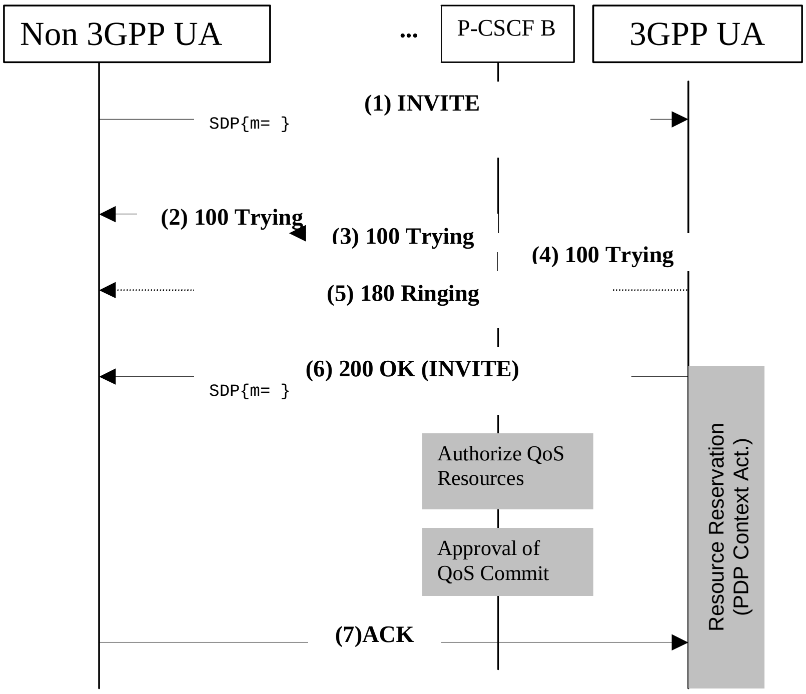


Figure C.2.3.2/1: Non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in INVITE request.

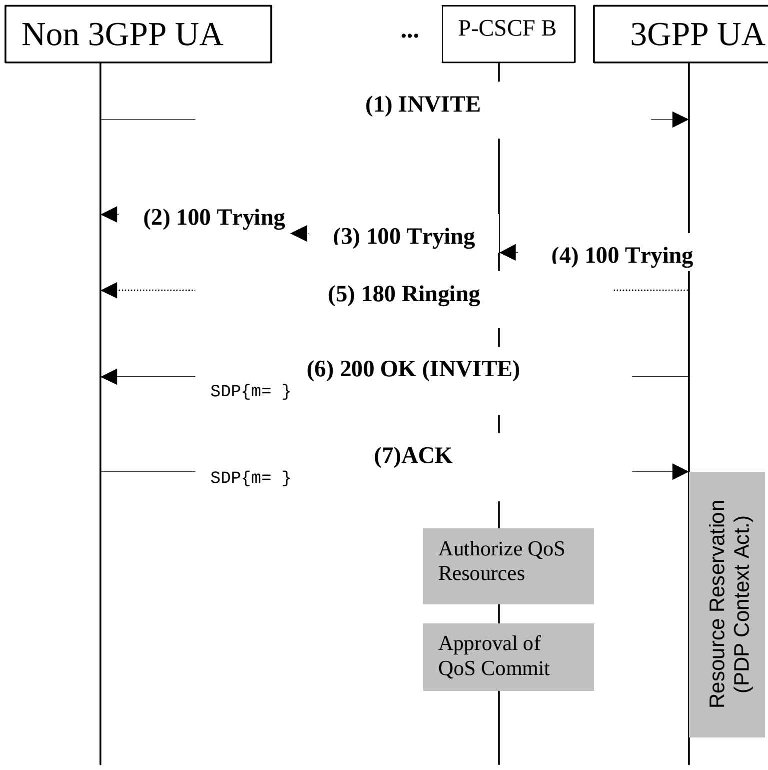


Figure C.2.3.2/2: Non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in OK response.

C.2.3.3 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~~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.

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

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

~~Notes:~~

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

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

~~3.~~NOTE 3: The 100 TRYING response (2), (3), (4) to the INVITE message (1) is send hop-by-hop, as indicated in this flow diagram. ~~It will be omitted in all subsequent flow diagrams for simplicity.~~ All other messages are generated by the 3GPP UEs.

~~4.~~NOTE 4: Most parts of the SIP messages are omitted for simplicity. Only the “require”, “supported” and “allowed” header fields are depicted.

~~5.~~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

~~6.~~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.

~~7.~~NOTE 7: The use of the “183 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 called UA is not capable of meeting unilaterally is included in the initial INVITE (1), a 101-199 provisional response, such as the “183 Session Progress”, is required to transport the SDP answer including the mandated “confirmation status” SDP attribute (Ref. [6], Section 6). Moreover, the “180 Ringing” message 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 (11) and OK(PRACK) (12) messages. A calling 3GPP UA will refrain from generating a new SDP offer within PRACK (11), if it does not wish to further restrict the set of codecs selected within the first offer/answer pair.

~~9.~~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.

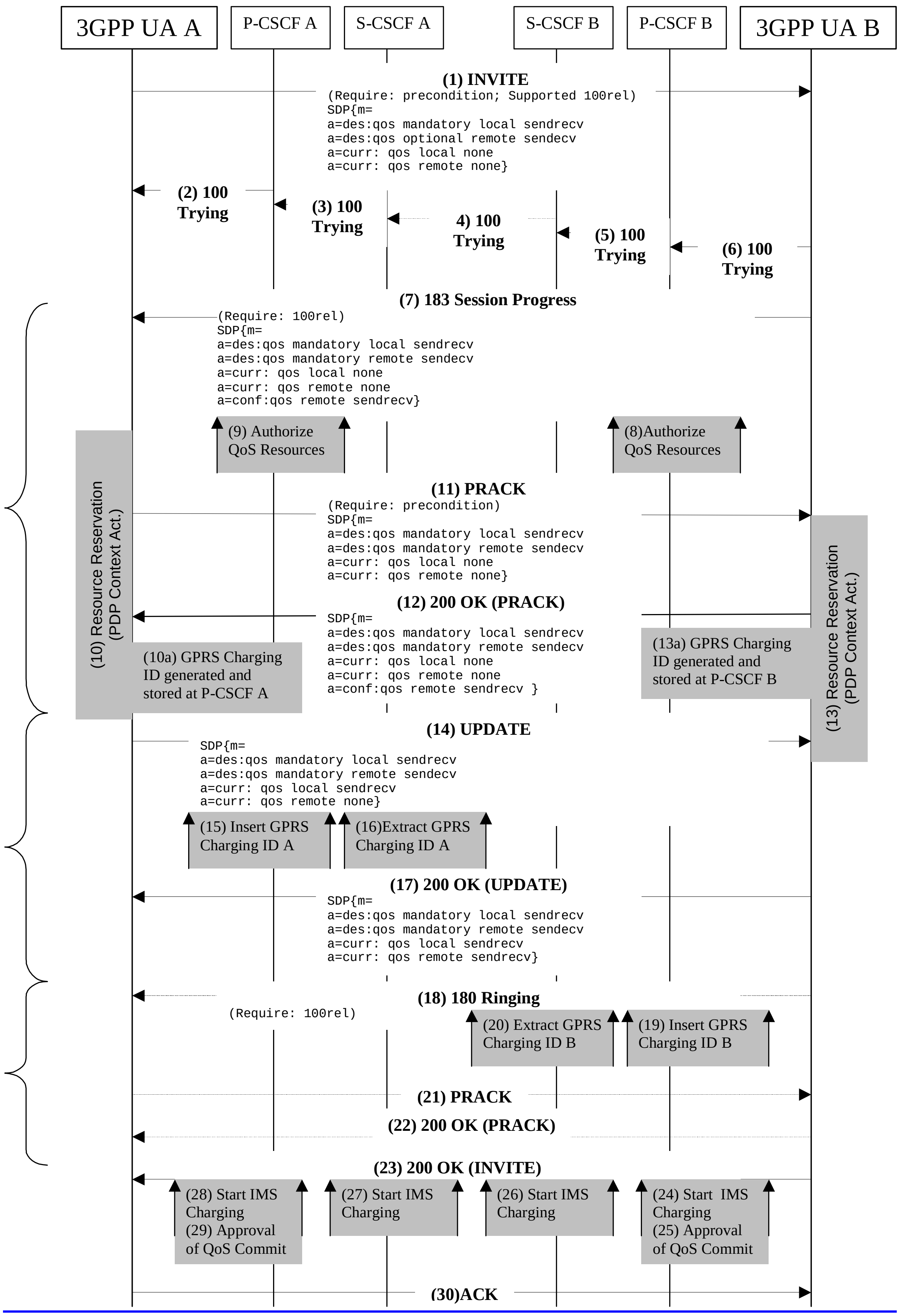
~~10.~~NOTE 10: The use of the “Update” Request (14) is optional according to IETF specifications [5], [7], unless certain conditions make it mandatory. In particular, if a threshold specified in a previous SDP “confirm-status” attribute (e.g. in message (7)) is reached or surpassed, the UA must send 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.

~~11.~~NOTE 11: If the “UPDATE” (14) request is not used, the subsequent” OK(UPDATE)” (17) response is also not present.

~~12.~~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.

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

~~14.~~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.



3GPP UA A
P-CSCF A
S-CSCF A
S-CSCF B
P-CSCF B
3GPP UA B
(1) INVITE
(Require: precondition; Supported 100rel)
SDP{m=
a=des:qos mandatory local sendrecv
a=des:qos optional remote sendecv
a=curr: qos local none
a=curr: qos remote none}
(2) 100 Trying
(3) 100 Trying
4) 100 Trying
(5) 100 Trying
(6) 100 Trying
(7) 183 Session Progress
(Require: 100rel)
SDP{m=
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendecv
a=curr: qos local none
a=curr: qos remote none
a=conf:qos remote sendrecv}
(9) Authorize QoS Resources
(8)Authorize QoS Resources
(10) Resource Reservation (PDP Context Act.)
(11) PRACK
(Require: precondition)
SDP{m=
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendecv
a=curr: qos local none
a=curr: qos remote none}
(12) 200 OK (PRACK)
SDP{m=
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendecv
a=curr: qos local none
a=curr: qos remote none
a=conf:qos remote sendrecv }
(10a) GPRS Charging ID generated and stored at P-CSCF A
(13a) GPRS Charging ID generated and stored at P-CSCF B
(13) Resource Reservation (PDP Context Act.)
(14) UPDATE
SDP{m=
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendecv
a=curr: qos local sendrecv
a=curr: qos remote none}
(15) Insert GPRS Charging ID A
(16)Extract GPRS Charging ID A
(17) 200 OK (UPDATE)
SDP{m=
a=des:qos mandatory local sendrecv
a=des:qos mandatory remote sendecv
a=curr: qos local sendrecv
a=curr: qos remote sendrecv}
(18) 180 Ringing
(Require: 100rel)
(20) Extract GPRS Charging ID B
(19) Insert GPRS Charging ID B
(21) PRACK
(22) 200 OK (PRACK)
(23) 200 OK (INVITE)
(28) Start IMS Charging
(29) Approval of QoS Commit
(27) Start IMS Charging
(26) Start IMS Charging
(24) Start IMS Charging
(25) Approval of QoS Commit
(30)ACK

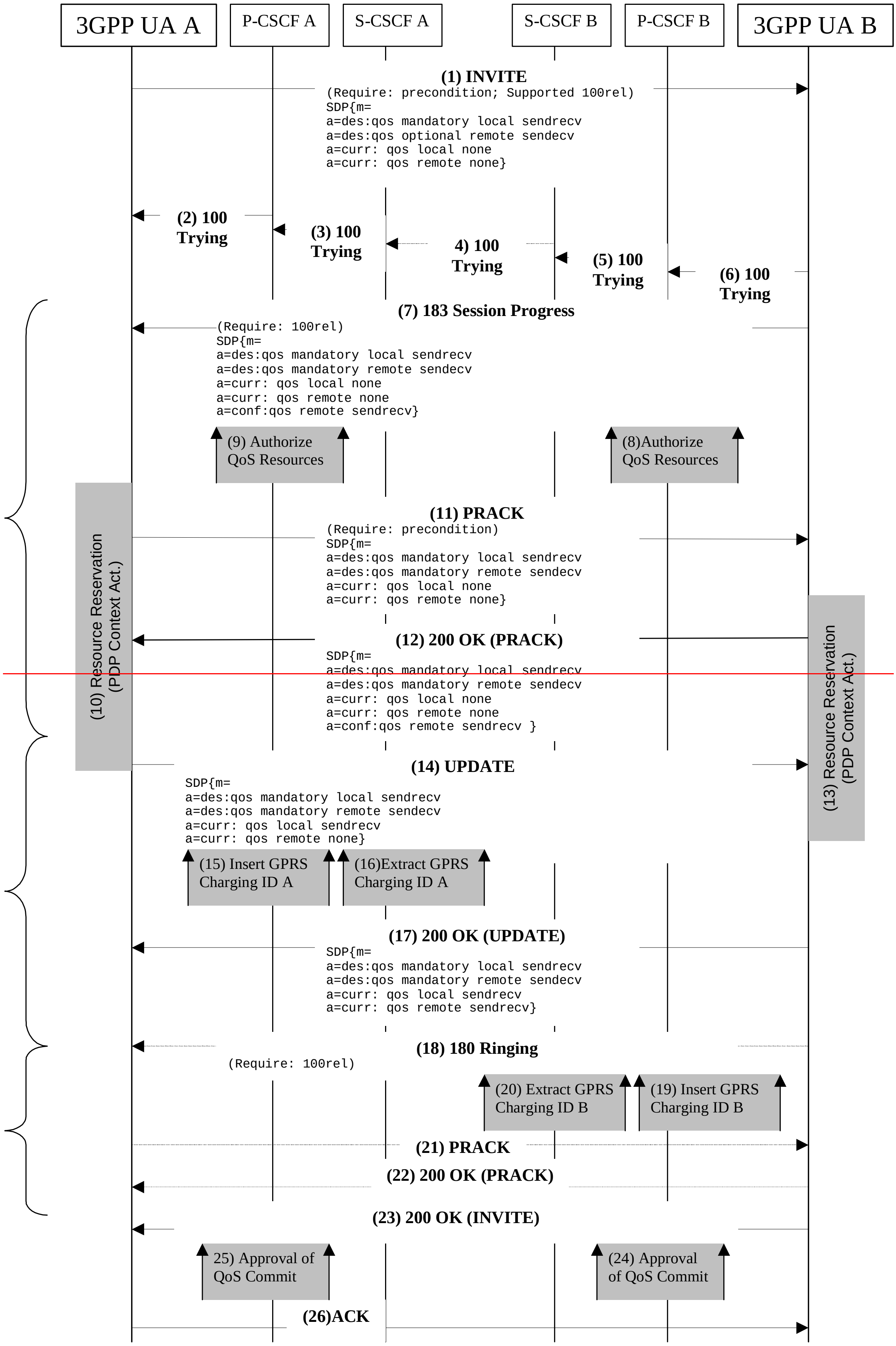


Figure D/1: 3GPP UA to 3GPP UA Call flow

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 ~~callflow~~call flows that deviate to some extent from the reference ~~callflow~~call flow in Annex D. These ~~scenaorios~~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.

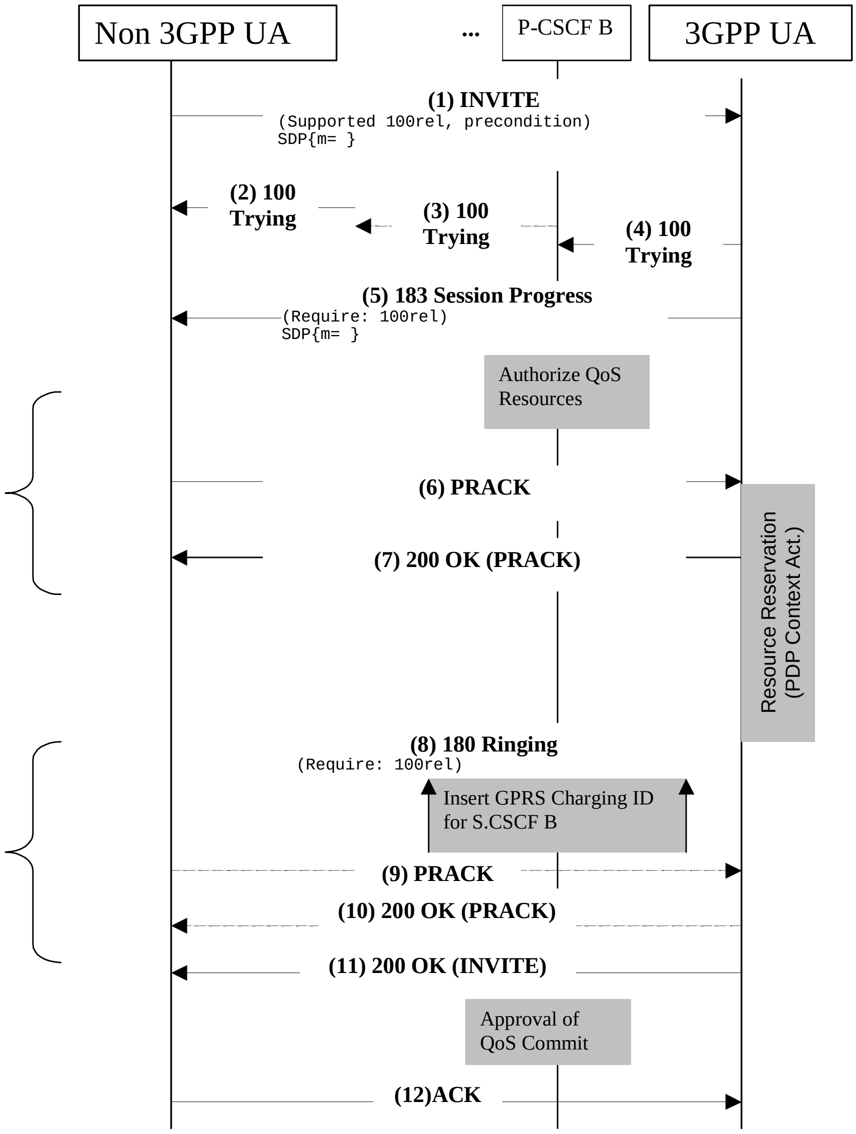


Figure E.1/1: Non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, and not performing QoS reservation, to 3GPP UA

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, 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(INVITE)) 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".

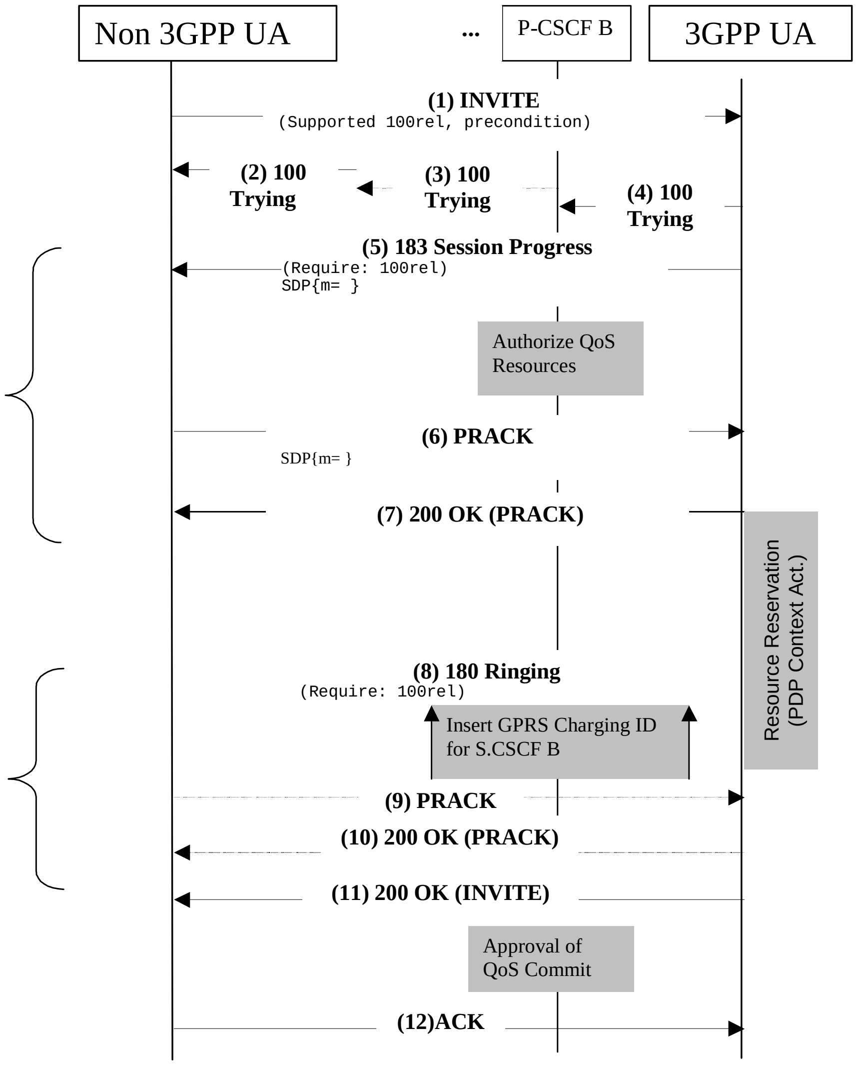


Figure E.2/1: Non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not including the SDP offer in the initial invite, to 3GPP UA

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 ~~B2BUA~~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 M3-030157			Agreed changes are included.	1.0.0	1.1.0
2003-03	NP-19				Presented to Plenary NP#19 for information	1.1.0	

3GPP draft TR 29.962 V1.1.0 (2003-02)

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)

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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 IMS network entities behaving as specified in the 3GPP profile of SIP in TS 24.229 [1], with related call flow examples in TS 24.228 [2] and stage 2 work in TS 23.228 [3], and SIP network entities external to the 3GPP network, which may not adhere to the 3GPP profile of SIP.

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 3GPP network may feature different SIP capabilities, such as the support of arbitrary SIP packages

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]

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 3GPP network. Any SIP network entity within the 3GPP network 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, 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 3GPP network, which do not or only partly satisfy the 3GPP profile of SIP, in particular not fully 3GPP conformant SIP terminals, are out of scope of the present document.

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

It is foreseen that future non-3GPP SIP clients will support the above required SIP extensions, and it is envisaged that it is unlikely that interworking solutions will then be required.

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]

3.2 Abbreviations

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

4. Interworking Scenarios

Each topic is contained in an own subsection with the structure defined in Annex A. Further structure may be introduced to the present section by grouping related topics.

4.1 Calling 3GPP UA to Called non-3GPP UA

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

- 3GPP UA to non-3GPP UA supporting the SIP precondition extension, but not supporting the SIP 100rel extension.
- 3GPP UA to non-3GPP UA supporting the SIP preconditions extension, but not supporting the SIP update extension.

4.1.1 3GPP UA to non-3GPP UA supporting the SIP 100rel extension and the SIP update extension, but not supporting the SIP preconditions extension

4.1.1.1 Description of interworking issue

The call fails, as detailed in Section 4.1.2.2

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.

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.

4.1.1.2 Proposed Resolutions to interworking issue

4.1.1.2.1 B2B UA

A B2B UA is used.

4.1.1.2.1.1 Insertion of B2B UA

How the B2B UA is inserted is discussed within Section 4.1.2.4.1.1.

4.1.1.2.1.2 Functionality of B2B UA

4.1.1.2.1.2.1 Description

The functionality of the B2B UA is as discussed in Section 4.1.2.4.1.2.1.

The B2B UA shall pass additional UPDATE messages, which are not related to the precondition extension, and related provisional acknowledge messages.

4.1.1.2.1.2.2 Advantages

General advantages of the B2B UA are discussed in Section 4.1.2.4.1.2.2.

Both the calling and 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 can not be depicted. The dialog state is not altered by UPDATE requests, and thus these messages probably do not have harmful side effects.

4.1.1.2.1.2.3 Disadvantages

General disadvantages of the B2B UA are discussed in Section 4.1.2.4.1.2.3.

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

4.1.1.2.2 Modified end-to-end call flow

4.1.1.2.2.1 Description

The rules described in Section 4.1.3.2.2.1 are applied.

The resulting call flow is similar to Figure 4.1.2.4.2.1/1, possibly with additional update messages.

4.1.1.2.2.2 Advantages

See Section 4.1.3.2.2.2.

4.1.1.2.2.3 Disadvantages

See Section 4.1.3.2.2.3.

4.1.2 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension

4.1.2.1 Description of interworking issue

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

4.1.2.2 Flow diagram

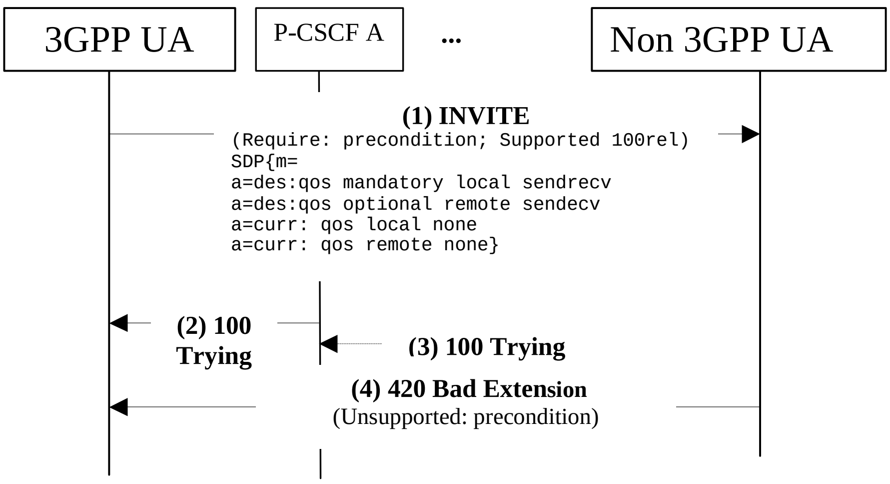

Figure 4.1.2.2/1: 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension.

4.1.2.3 Impact of Identified interworking issue

The call fails.

4.1.2.4 Proposed resolutions to interworking issue

A B2B UA is used.

4.1.2.4.1 B2B UA

4.1.2.4.1.1 Insertion of B2B UA

4.1.2.4.1.1.1 Static Insertion of B2B UA

4.1.2.4.1.1.1.1 Description

A B2B UA is permanently inserted at connections between the IMS and a given external network. This B2B UA handles all calls, including calls where the call flows may be passed without modification.

The B2B UA shall be inserted in the home IMS for all calls leaving the home IMS, which are not routed to another IMS via direct interconnection.

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

The B2B UA becomes active only when receiving a 420 Bad Extension (Unsupported precondition) response from the Non-3GPP UA, as depicted in Figure 4.1.2.4.1.1.1.1/1. Otherwise, the B2B UA passes all SIP messages received at one side to the other side.

The B2B UA shall store the SDP offer in initial invites for all calls until receiving a provisional or final response from the Non-3GPP UA.

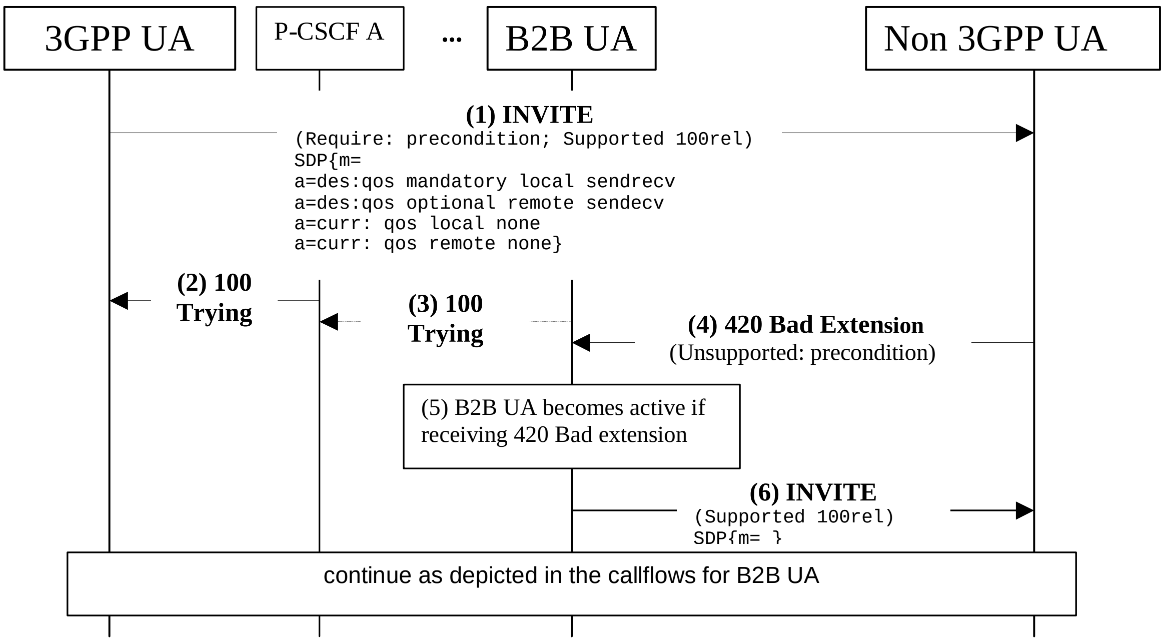

Figure 4.1.2.4.1.1.1.1/1: Activation of static B2B UA connecting 3GPP UA to non-3GPP UA not supporting the SIP preconditions extension

4.1.2.4.1.1.1.2 Advantages

4.1.2.4.1.1.1.3 Disadvantages

Additional processing load.

Additional delay.

Lack of signalling transparency may restrict the compatibility with future extensions for all calls.

4.1.2.4.1.2 Functionality of B2B UA

4.1.2.4.1.2.1 Description

Editor's Note: The following rules have been agreed only as basis for further contributions and have not yet been investigated in detail.

The B2B UA shall apply the following rules:

1. The B2B UA 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 B2B UA shall also comply to the SIP 100rel and update extensions.
3. On the IMS side, the B2B UA shall also comply to and apply the SIP 100rel, update and precondition extensions.
4. The B2B UA shall pass SIP requests arriving at one call leg to the call leg at the other side, with the exceptions below.
5. The B2B UA shall pass 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 B2B UA shall pass SIP final responses arriving at one call leg to the call leg at the other side, with the exceptions below.
7. The B2B UA shall not require the SIP preconditions extension on the non-IMS side.
8. The B2B UA shall insert preconditions in SDP sent from non-3GPP UA to 3GPP UA.
9. The B2B UA should remove preconditions in SDP sent from 3GPP UA to non-3GPP UA
10. The B2B UA shall not pass PRACK and 200 OK(PRACK) messages.
11. The B2B UA shall delay forwarding a 200 OK(INVITE) message from the non-IMS side to the IMS side until the mandatory preconditions are met on the IMS side.
12. The B2B UA shall handle subsequent SDP offers on the IMS side in an INVITE transaction locally, if only the preconditions are modified
13. If the B2B UA receives a subsequent SDP offers on the IMS side with modified media, it shall suspend the transaction on the IMS side and pass this SDP offer to a re-invite transaction on the non-IMS Side. The B2B UA shall pass the SDP answer received in the re-invite transaction 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 B2B UA shall pass an SDP answer within the 200 OK(INVITE) message of the original INVITE transaction from the non-IMS side to a provisional response on the IMS side.
15. For a re-Invite from the Non-IMS side to the IMS side, the B2B UA shall apply the rules in Section 4.2.2.4.1.2.1.

The B2B UA relies messages as indicated by the red dotted arrows in the figures below.

The called UA may also send no "Session progress" message and include the SDP answer in the "200 OK(INVITE)" instead. This case is discussed in Section 4.1.3.4.1.2.

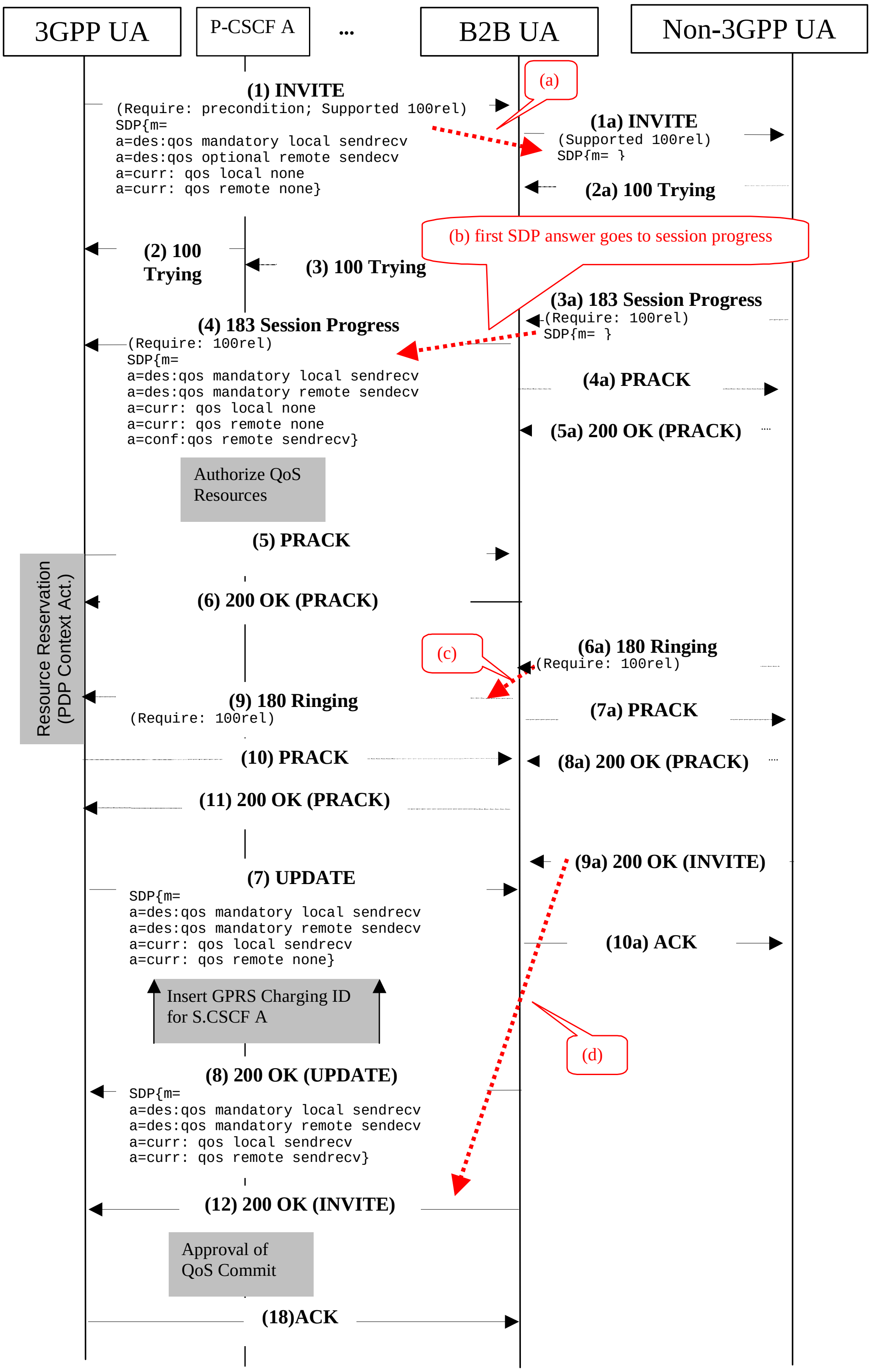

Figure 4.1.2.4.1.2.1/1: Functionality of B2B UA connecting 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. Calling UA includes SDP answer in 183 "Session Progress". Calling UA sends no second SDP offer.

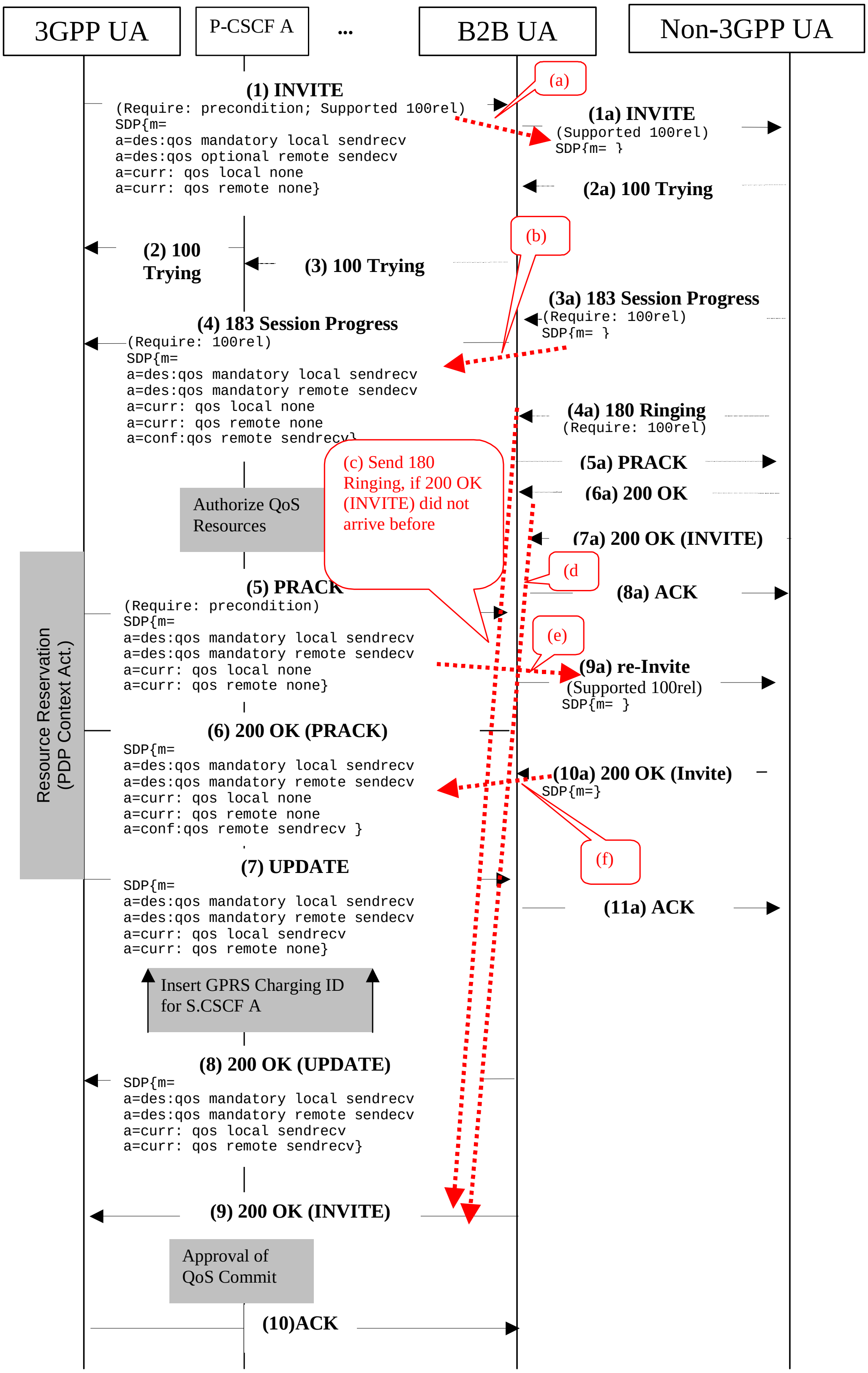

Figure 4.1.2.4.1.2.1/2: Functionality of B2B UA connecting 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. Called UA includes SDP answer in 183 "Session Progress". Calling UA sends second SDP offer.

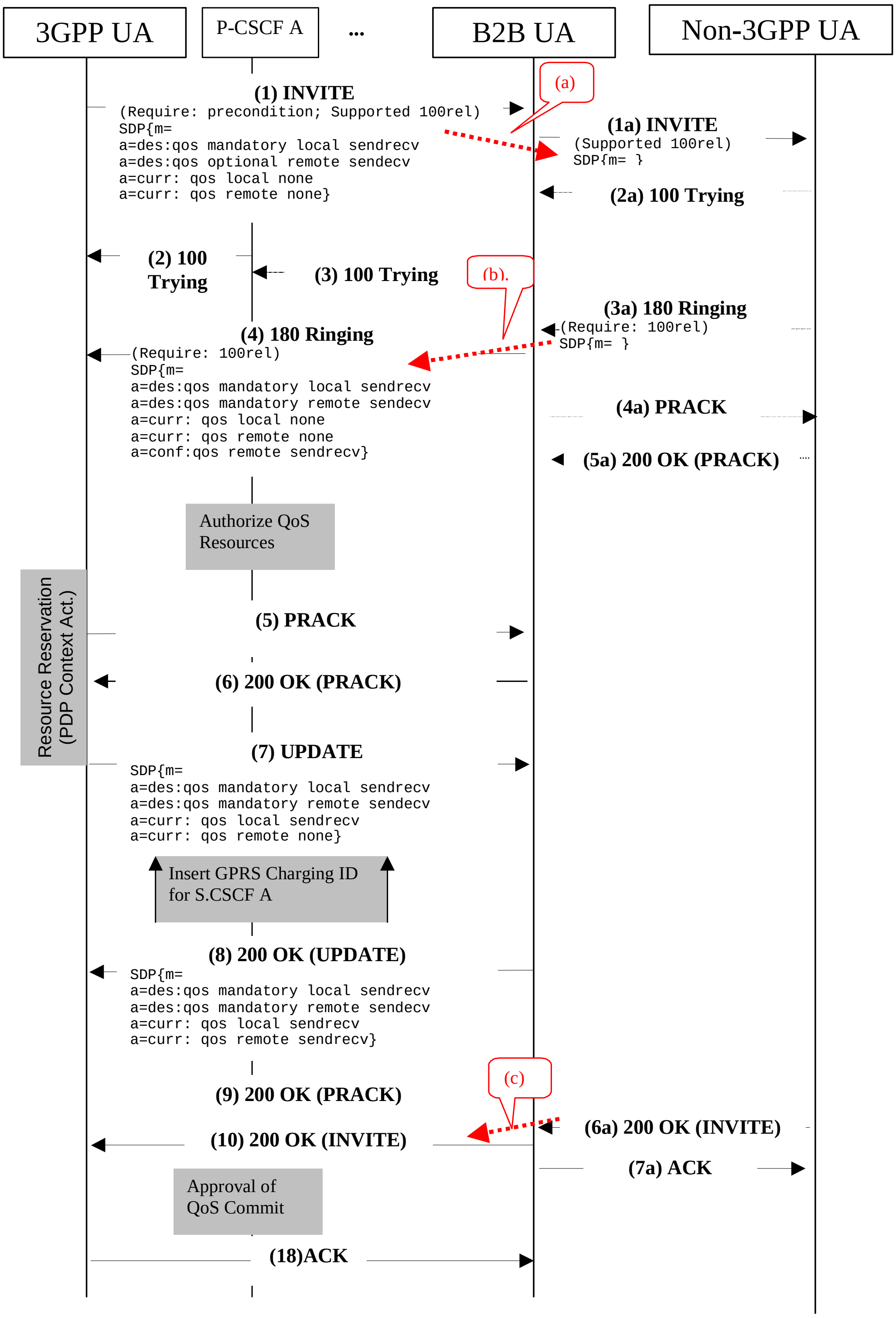

Figure 4.1.2.4.1.2.1/3: Functionality of B2B UA connecting 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. Called UA includes SDP answer in 180 "Ringing". Calling UA sends no second SDP offer.

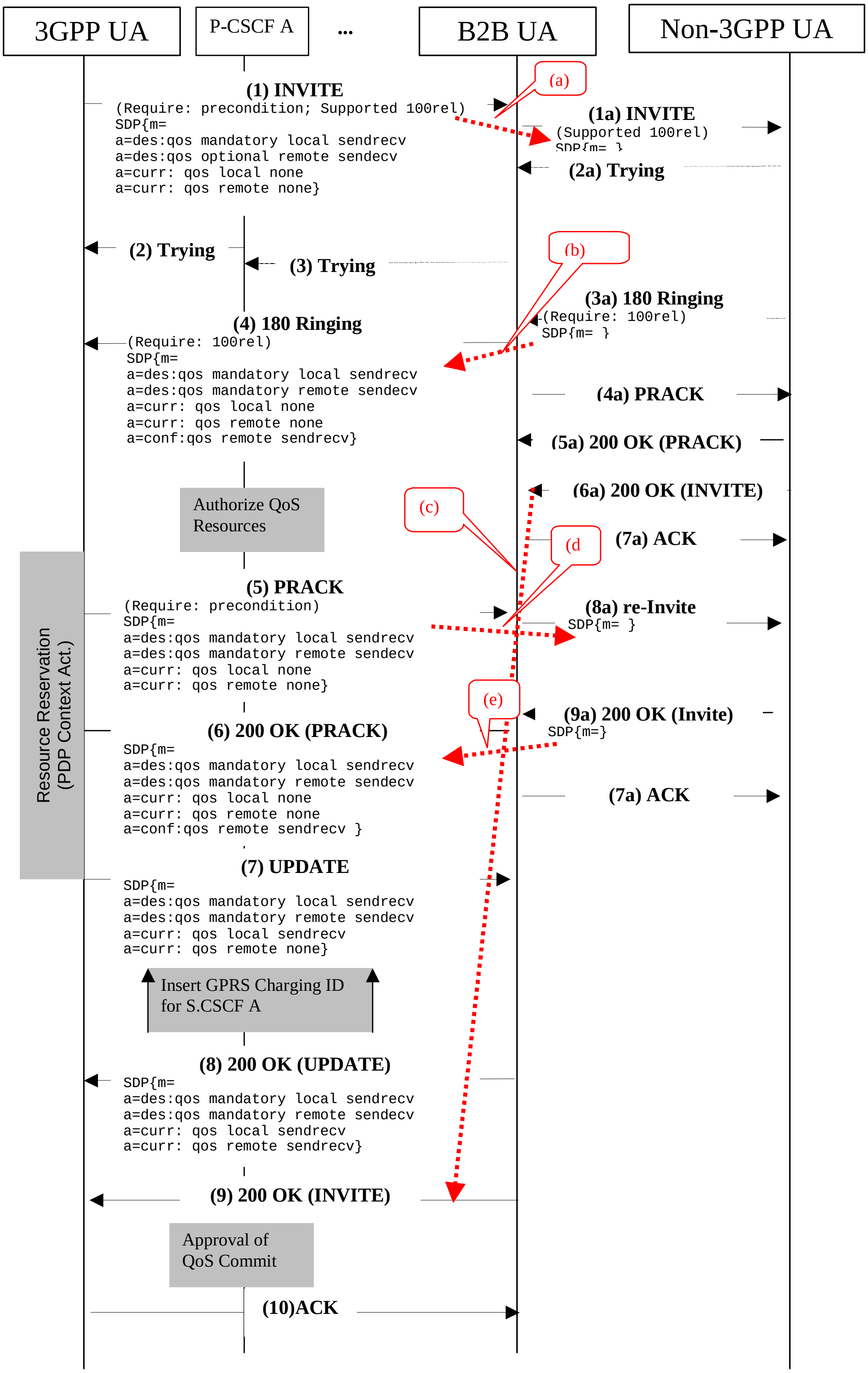

Figure 4.1.2.4.1.2.1/4: Functionality of B2B UA connecting calling 3GPP UA to called non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. Called UA includes SDP answer in 180 "Ringing". Calling UA sends second SDP offer.

4.1.2.4.1.2.2 Advantages

4.1.2.4.1.2.3 Disadvantages

The functionality and implementation of the B2B UA is complicated and will have to include the following features:

- Storage of SDP.
- Own timer supervision of dialogues on both call legs.

The compatibility with future SIP extensions may be limited by the need to update the B2B UA. This may limit the network's ability to deploy new IP multimedia applications.

4.1.2.4.2 Modified end-to-end call flow

4.1.2.4.2.1 Description

The rules described in Section 4.1.3.2.2.1 are applied.

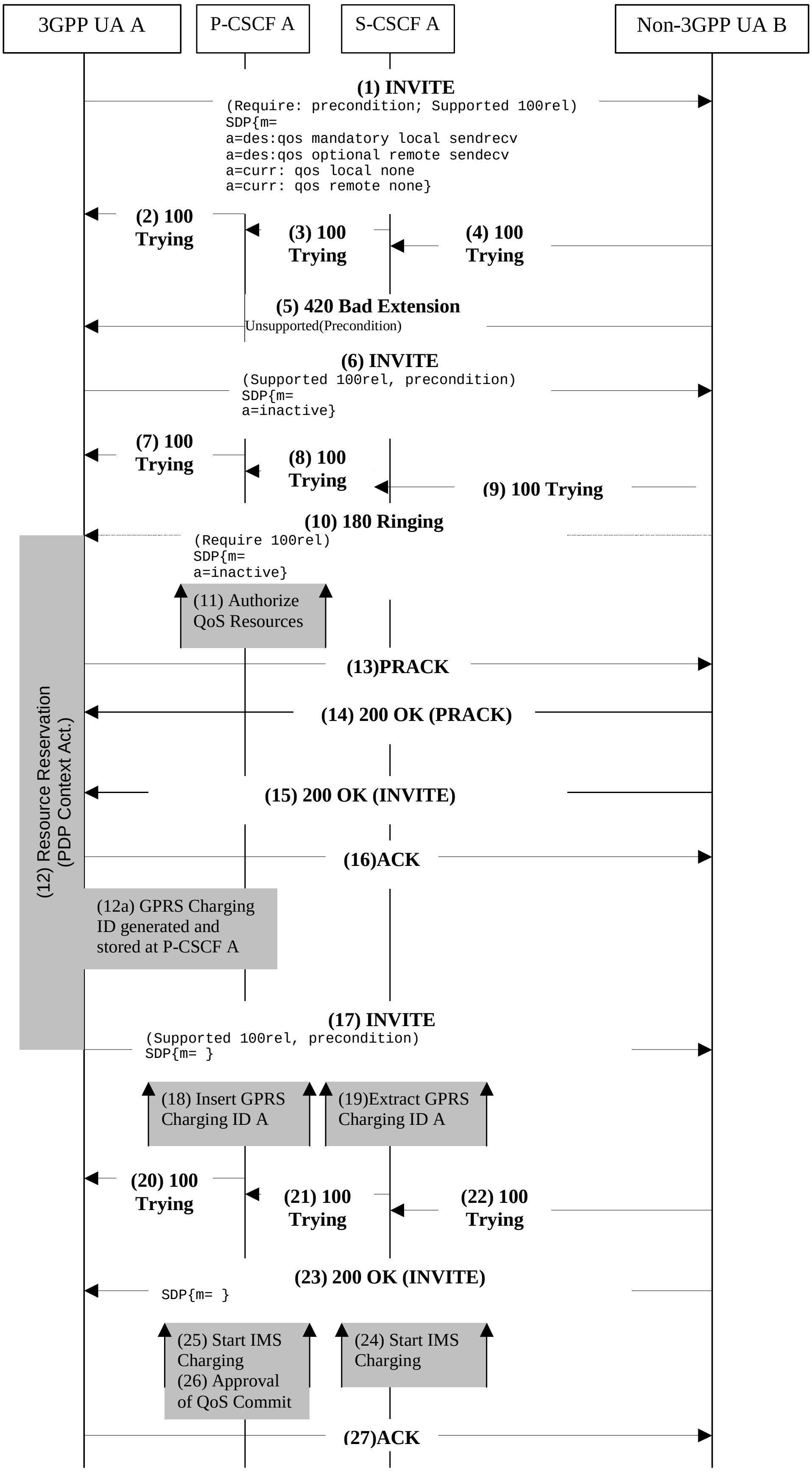

Figure 4.1.2.4.2.1/1: Using re-invite to connect calling 3GPP UA to called non-3GPP UA not supporting the SIP preconditions extension, but supporting the SIP 100rel extension.

4.1.2.4.2.2 Advantages

See Section 4.1.3.2.2.2.

4.1.2.4.2.3 Disadvantages

See Section 4.1.3.2.2.3.

4.1.3 3GPP UA to non-3GPP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

4.1.3.1 Description of interworking issue

The call fails, as detailed in Section 4.1.2.2.

4.1.3.2 Proposed Resolutions to interworking issue

4.1.3.2.1 B2B UA

A B2B UA is used.

4.1.3.2.1.1 Insertion of B2B UA

How the B2B UA is inserted is discussed within Section 4.1.2.4.1.1.

4.1.3.2.1.2 Functionality of B2B UA

4.1.3.2.1.2.1 Description

The B2B UA shall apply the rules given in section 4.1.2.4.1.2.1.

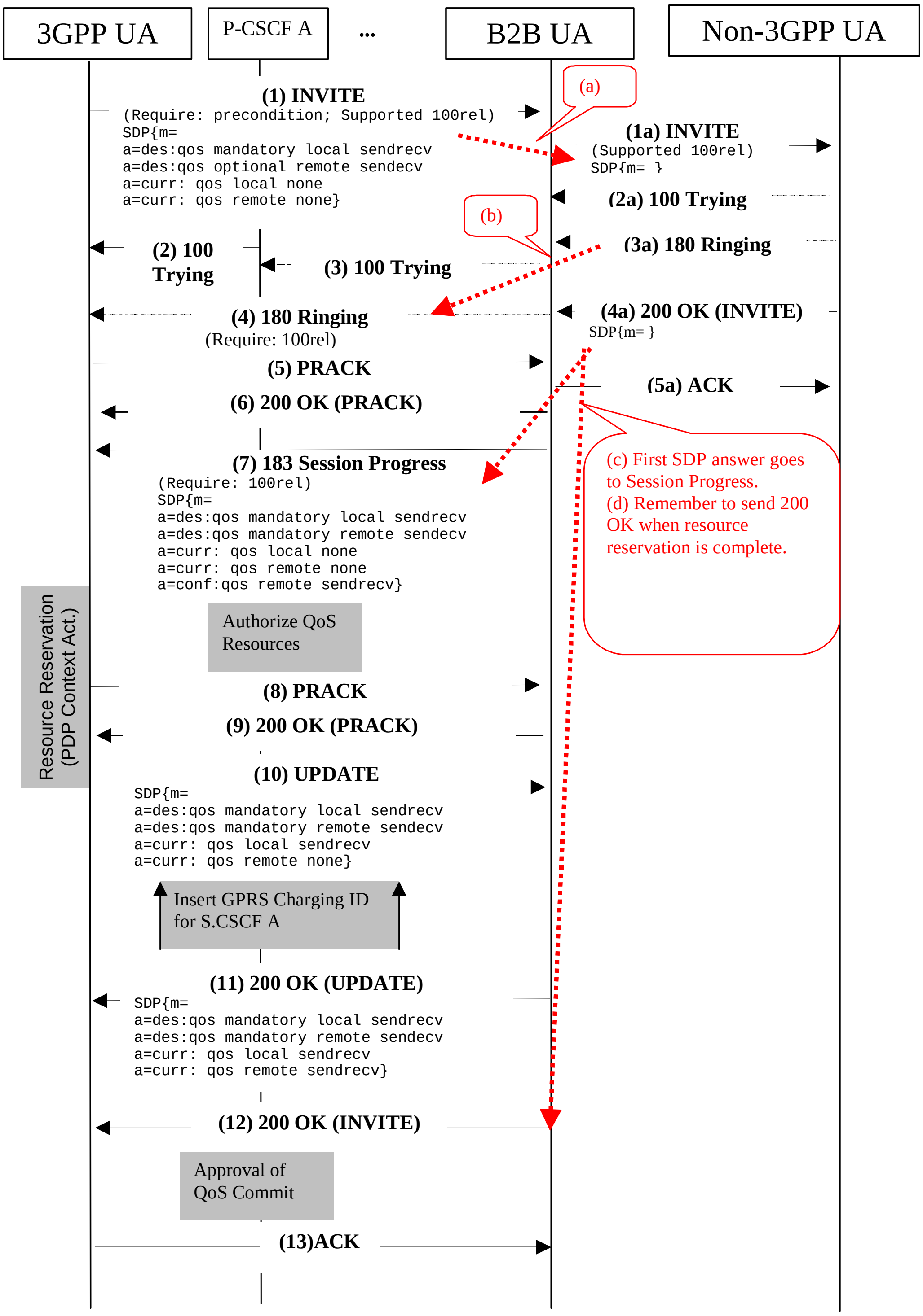

Figure 4.1.3.2.1.2/1: Functionality of B2B UA connecting calling 3GPP UA to called non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension. Calling UA sends no second SDP offer

There may be re-transmissions of the INVITE (1) by the 3GPP UA, which should be passed transparently by the B2B UA, as indicated in interaction (a).

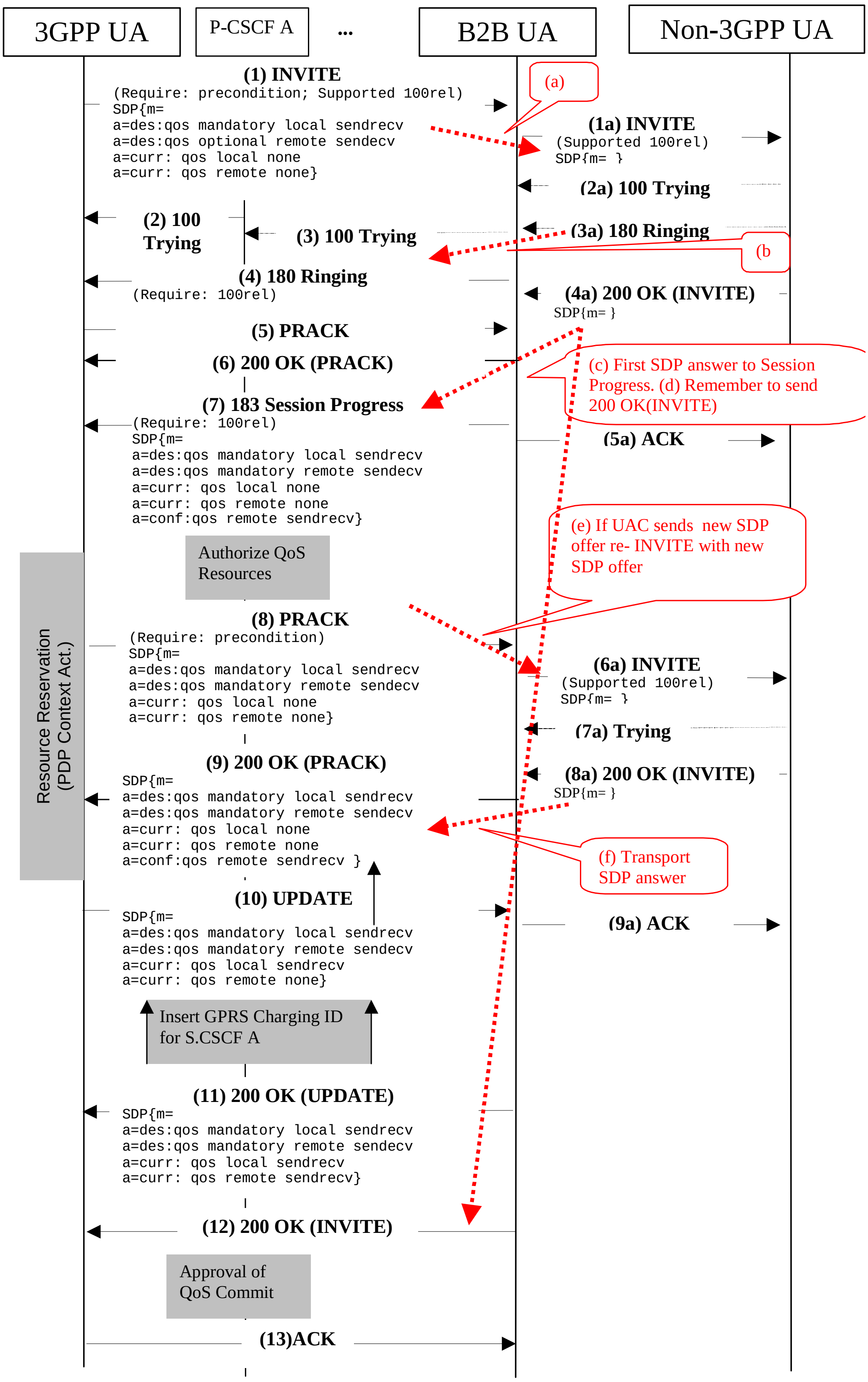

Figure 4.1.3.2.1.2/2: Functionality of B2B UA connecting calling 3GPP UA to called non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension. Calling UA sends second SDP offer

4.1.3.2.1.2.2 Advantages

General advantages of the B2B UA are discussed in Section 4.1.2.4.1.2.2.

4.1.3.2.1.2.3 Disadvantages

General disadvantages of the B2B UA are discussed in Section 4.1.2.4.1.2.3.

The 3GPP user perception suffers if the non-3GPP UA does not answer the call immediately, but does not send a ringing message.

The non-3GPP UA may suffer clipping.

4.1.3.2.2 Modified end-to-end call flow

4.1.3.2.2.1 Description

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

1. (e.g. in TS 24.229) The calling 3GPP UA should (not shall) require preconditions in an initial INVITE request. The calling 3GPP UA may (re-)INVITE an external UA without requiring preconditions, e.g. if receiving a 420 Bad Extension(precondition) error response. In this case, the 3GPP UA shall set the media to “inactive” when generating an SDP offer. The 3GPP UA shall 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 called 3GPP UA may accept invites not containing a “Require(precondition)” header.

3. (e.g. in TS 24.229) The called 3GPP UA may send provisional responses without requiring the 100rel extension, if the calling party did not indicate the support of the 100rel extension. In this case, the called 3GPP UA may also send a 200 OK(INVITE) before the resource reservation is complete, but shall set the media to inactive in the SDP offer or answer within this 200 OK(INVITE). The 3GPP UA shall send a re-INVITE activating the media by setting them to “send”, ”recv”, or “sendrecv” in SDP once the local resource reservation is complete.

4. (e.g. in TS 29.207 and 29.208) The P-CSCF(PDF) shall approve the QoS Commit when receiving a 200 OK(INVITE) 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.

5. (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(INVITE) response and media streams are active (“send”, ”recv”, or “sendrecv” in SDP).

6. (e.g. in TS 24.229): GPRS Charging ID may be 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

7. (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. As an alternative, both an SDP offer and an SDP answer with “inactive” media shall trigger the same actions with respect to the rules above.

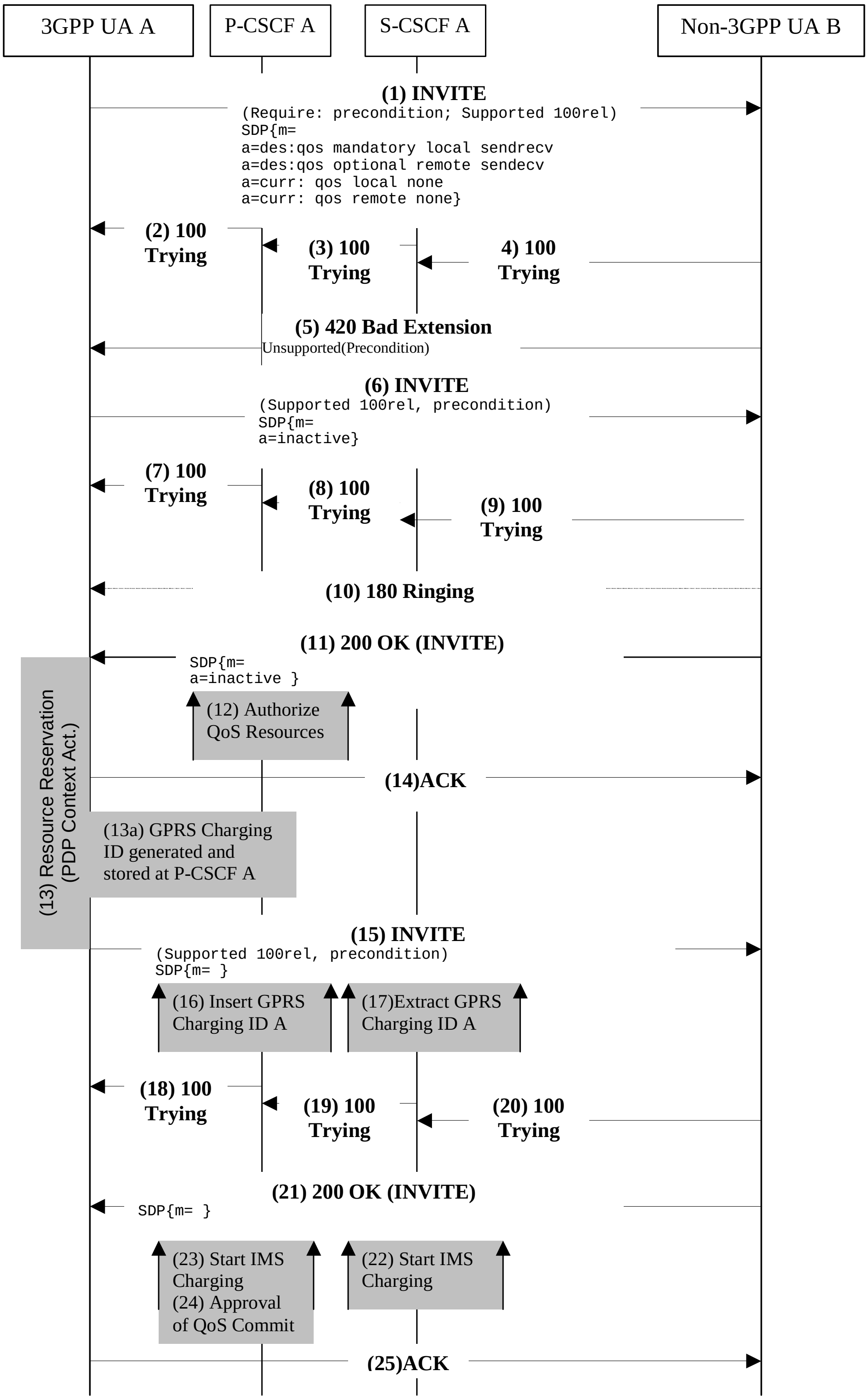

Figure 4.1.3.2.2.2/1: Using re-invite to connect calling 3GPP UA to called non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension.

4.1.3.2.2.2 Advantages

Only relatively minor changes are required.

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

4.1.3.2.2.3 Disadvantages

Changes have to be performed in various network entities.

4.2 Calling non-3GPP UA to Called 3GPP UA

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

- Non-3GPP UA supporting the SIP precondition extension, but not supporting the SIP 100rel extension, to 3GPP UA.
- Non-3GPP UA supporting the SIP preconditions extension, but not supporting the SIP update extension, to 3GPP UA.

4.2.1 Non-3GPP UA supporting the SIP 100rel extension and the SIP update extension, but not supporting the SIP preconditions extension, to 3GPP UA

4.2.1.1 Description of interworking issue

The call fails, as detailed in Section 4.2.2.2.

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.

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.

4.2.1.2 Proposed Resolutions to interworking issue

4.2.1.2.1 B2B UA

A B2B UA is used.

4.2.1.2.1.1 Insertion of B2B UA

How the B2B UA is inserted is discussed within Section 4.2.2.4.1.1.

4.2.1.2.1.2 Functionality of B2B UA

4.2.1.2.1.2.1 Description

The functionality of the B2B UA is as discussed in Section 4.2.2.4.1.2.1.

The B2B UA shall pass additional UPDATE messages, which are not related to the precondition extension, and related provisional acknowledge messages.

4.2.1.2.1.2.2 Advantages

General advantages of the B2B UA are discussed in Section 4.2.2.4.1.2.2.

The calling and 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 can not be depicted. The dialog state is not altered by UPDATE requests, and thus these messages probably do not have harmful side effects.

4.2.1.2.1.2.3 Disadvantages

General disadvantages of the B2B UA are discussed in Section 4.2.2.4.1.2.3.

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

4.2.1.2.2 Modified end-to-end call flow

4.2.1.2.2.1 Description

The restriction to disallow a direct communication with a calling non-3GPP UA, which does not indicate the support or requirement of the SIP preconditions extension is removed from TS 24.229.

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

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

The resulting call flows are similar to the flows in Section 4.2.2.4.2.1, possibly with additional UPDATE messages inserted.

4.2.1.2.2.2 Advantages

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

4.2.1.2.2.3 Disadvantages

No disadvantages have been identified.

4.2.2 Non-3GPP SIP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA

4.2.2.1 Description of interworking issue

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

4.2.2.2 Flow diagram

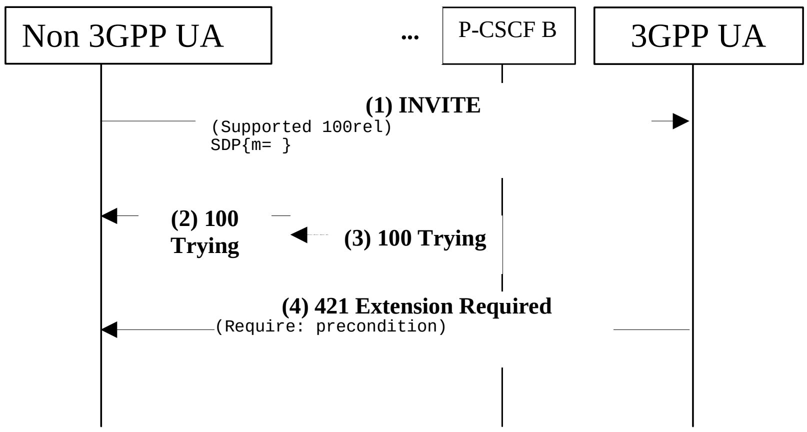

Figure 4.2.2.2/1: Non-3GPP SIP UA not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA.

4.2.2.3 Implications of Identified interworking issue

The call fails.

4.2.2.4 Proposed resolutions to interworking issue

4.2.2.4.1 B2B UA

A B2B UA is used.

4.2.2.4.1.1 Insertion of B2B UA

4.2.2.4.1.1.1 Static Insertion of B2B UA

4.2.2.4.1.1.1.1 Description

A B2B UA is permanently inserted at connection between IMS and a given external network. This B2B UA handles all calls, including calls where the call flows may be passed without modification.

The B2B UA shall be inserted in the border of the IMS for all calls entering the IMS from an external network (except for another IMS).

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 B2B UA becomes active only when receiving an INVITE message without an indication of the support or requirement of the preconditions extension from the Non-3GPP UA, as depicted in Figure 4.2.2.4.1.1.1.1/1. Otherwise, the B2B UA passes all SIP messages received at one side to the other side.

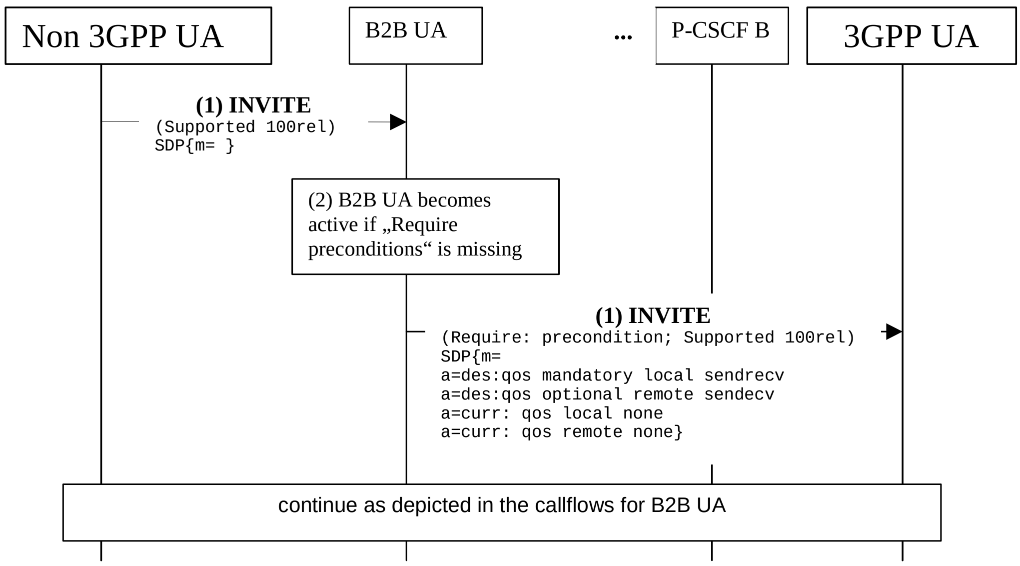

Figure 4.2.2.4.1.1.1.1/1: Activation of static B2B UA connecting Non-3GPP UA not indicating support of the SIP preconditions extension to 3GPP UA.

4.2.2.4.1.1.1.2 Advantages

4.2.2.4.1.1.1.3 Disadvantages

Additional processing load.

Additional delay.

Lack of signalling transparency may restrict the compatibility with future extensions for all calls.

The B2B UA may be activated unnecessarily, if the Non-3GPP UA supports the precondition extension, but fails to indicate this in the INVITE message.

4.2.2.4.1.2 Functionality of B2B UA

4.2.2.4.1.2.1 Description

Editor's Note: The following rules have been agreed only as basis for further contributions and have not yet been investigated in detail.

The B2B UA shall apply the following rules:

1. The B2B UA 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 B2B UA shall also comply with the SIP 100rel and update extensions.
3. On the IMS side, the B2B UA shall also comply to and apply the SIP 100rel, update and precondition extensions.
4. The B2B UA shall pass SIP requests arriving at one call leg to the call leg at the other side, with the exceptions below.
5. The B2B UA shall pass 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 B2B UA shall pass SIP final responses arriving at one call leg to the call leg at the other side, with the exceptions below.
7. The B2B UA shall not require the SIP preconditions extension on the non-IMS side.
8. The B2B UA shall insert preconditions in SDP sent from non-3GPP UA to 3GPP UA.
9. The B2B UA should remove preconditions in SDP sent from 3GPP UA to non-3GPP UA
10. The B2B UA shall not pass PRACK and 200 OK(PRACK) messages.
11. The B2B UA shall inspect an INVITE message 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 B2B UA receives an SDP offer in a provisional response from the IMS side, the B2B UA shall send the SDP offer in a 200 OK(invite) message at the non-IMS side. The B2B UA shall then forward the SDP answer received in the ACK message from the non-IMS side to the PRACK message for the provisional response on the IMS-side.
13. If the 100rel extension is not supported on the non-IMS side, and the B2B UA receives an SDP answer in a provisional response from the IMS side, the B2B UA shall send the SDP answer in a 200 OK(invite) message at the non-IMS side.
14. For a re-Invite from the IMS side to the Non-IMS side, the B2B UA shall apply the rules in Section 4.1.2.4.1.2.1.

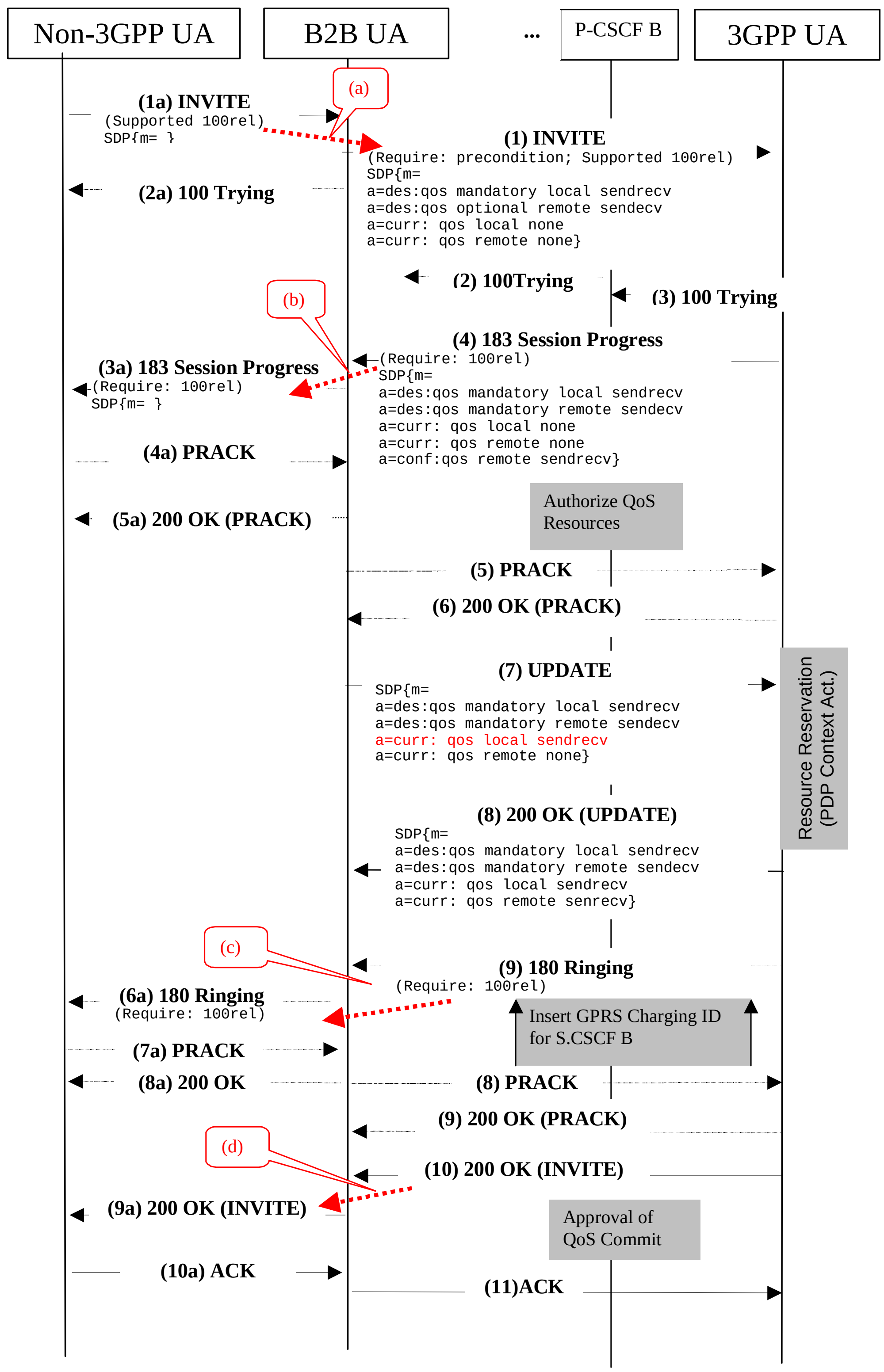

Figure 4.2.2.4.1.2.1/1: Functionality of B2B UA connecting calling non-3GPP SIP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension, to called 3GPP UA.

4.2.2.4.1.2.2 Advantages

4.2.2.4.1.2.3 Disadvantages

The functionality and implementation of the B2B UA is complicated and will have to include the following features:

- Storage of SDP.
- Own timer supervision of dialogues on both call legs.

The compatibility with future SIP extensions may be limited by the need to update the B2B UA. This may limit the network's ability to deploy new IP multimedia applications.

4.2.2.4.2 Modified end-to-end call flow

4.2.2.4.2.1 Description

The restriction to disallow a direct communication with a calling non-3GPP UA, which does not indicate the support or requirement of the SIP preconditions extension is removed from TS 24.229.

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

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

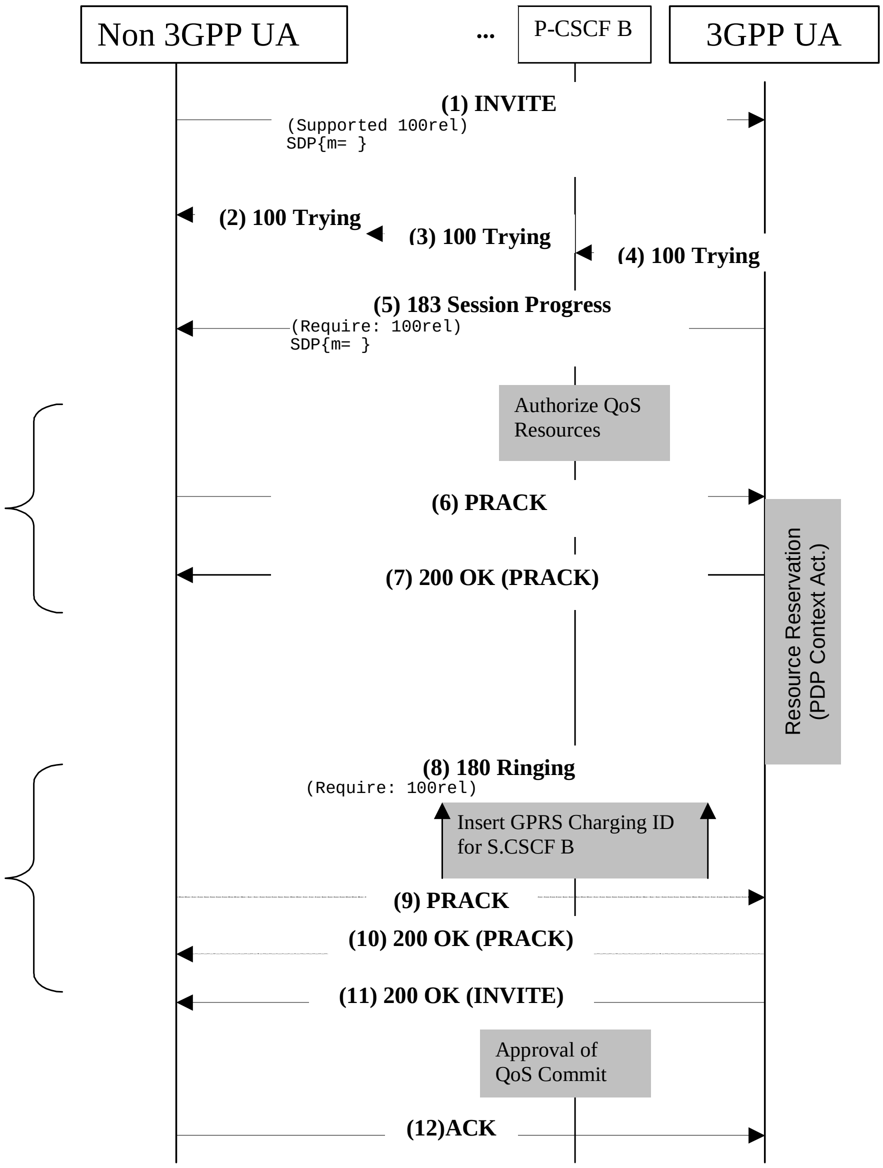

Figure 4.2.2.4.2.1/1: Modified end-to-end call flow for Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA. SDP offer in Invite.

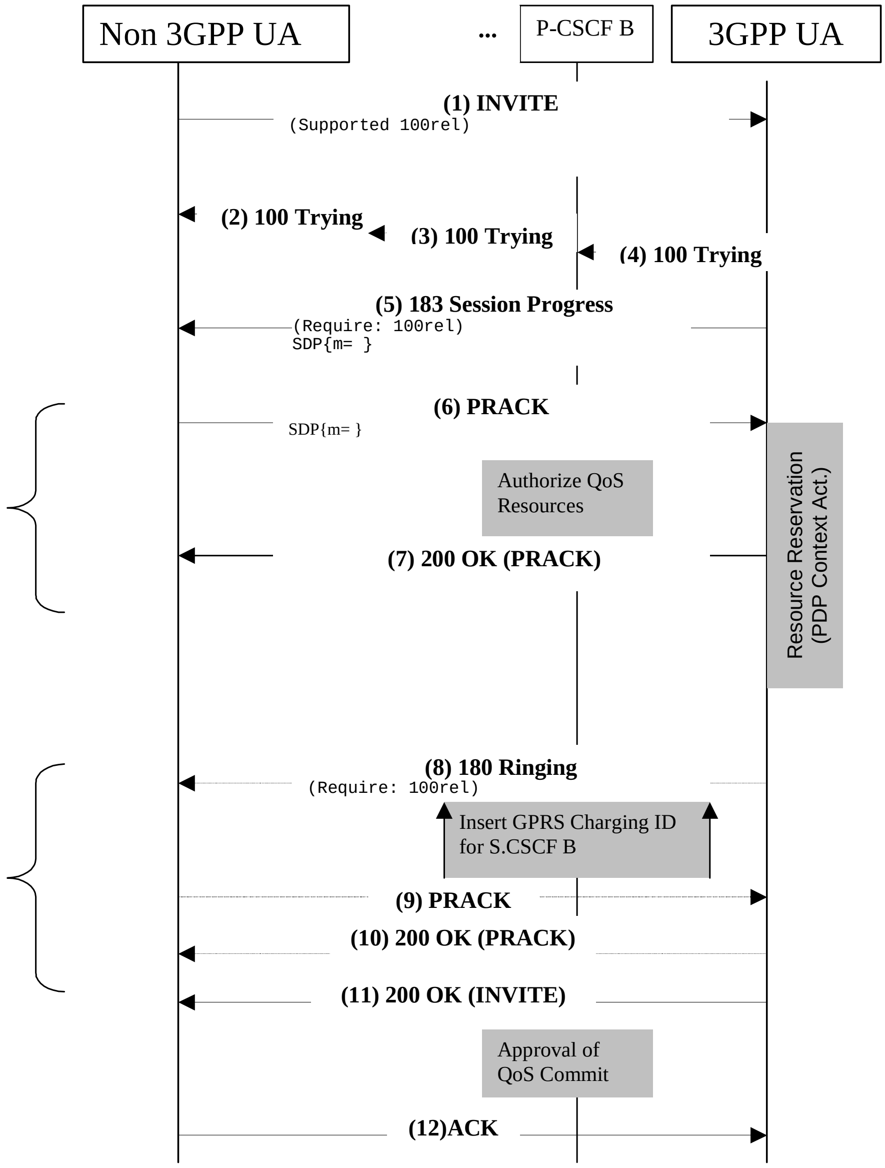

Figure 4.2.2.4.2.1/2: Modified end-to-end call flow for Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA. No SDP offer in Invite.

4.2.2.4.2.2 Advantages

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

4.2.2.4.2.3 Disadvantages

No disadvantages have been identified.

4.2.3 Non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to 3GPP UA

4.2.3.1 Description of interworking issue

The call fails, as detailed in Section 4.2.2.2.

4.2.3.2 Proposed Resolutions to interworking issue

4.2.3.2.1 B2B UA

A B2B UA is used.

4.2.3.2.1.1 Insertion of B2B UA

4.2.3.3.1.1.1 Static Insertion of B2B UA

4.2.3.3.1.1.1.1 Description

A B2B UA is permanently inserted at connection between IMS and a given external network. This B2B UA handles all calls, including calls where the call flows may be passed without modification.

The B2B UA shall be inserted in the border of the IMS for all calls entering the IMS the IMS from an external network (except for another IMS).

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 B2B UA becomes active only when receiving an INVITE message without an indication of the support or requirement of the 100rel extension from the Non-3GPP UA, as depicted in Figure 4.2.3.3.1.1.1.1/1. Otherwise, the B2B UA passes all SIP messages received at one side to the other side.

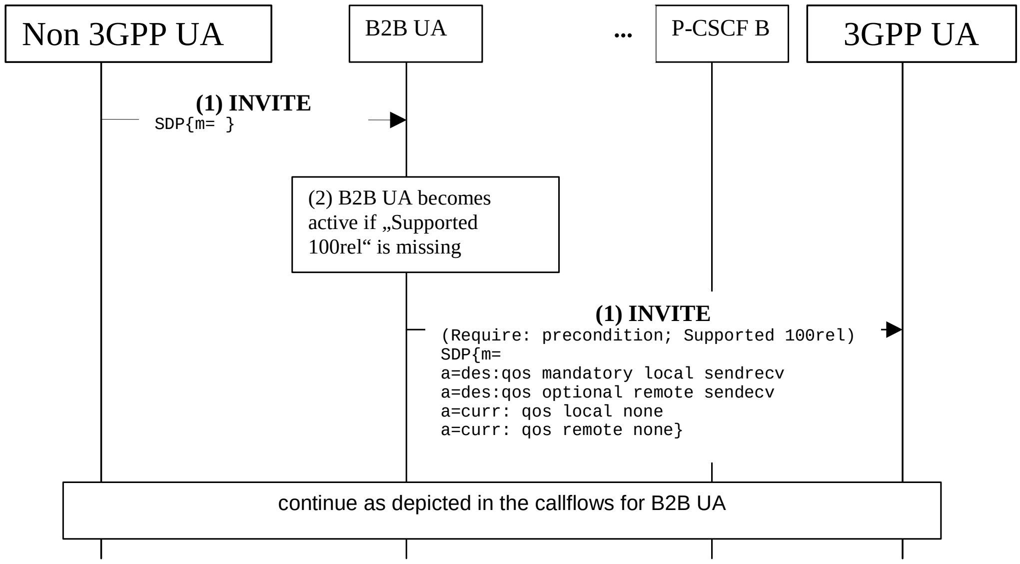

Figure 4.2.3.3.1.1.1.1/1: Activation of static B2B connecting Non-3GPP SIP UA not indicating support of the SIP preconditions extension to 3GPP UA.

4.2.3.3.1.1.1.2 Advantages

4.2.3.3.1.1.1.3 Disadvantages

Additional processing load.

Additional delay.

Lack of signalling transparency may restrict the compatibility with future extensions for all calls.

The B2B UA may be activated unnecessarily, if the Non-3GPP UA supports the 100 rel extension, but fails to indicate this in the INVITE message. RFC 3262 [6] recommends that a UAC supporting the 100rel extension indicates this capability in the INVITE message, but does not mandate the UAC to do so.

4.2.3.2.1.2 Functionality of B2B UA

4.2.3.2.1.2.1 Description

The B2B UA shall apply the rules given in section 4.2.2.4.1.2.1.

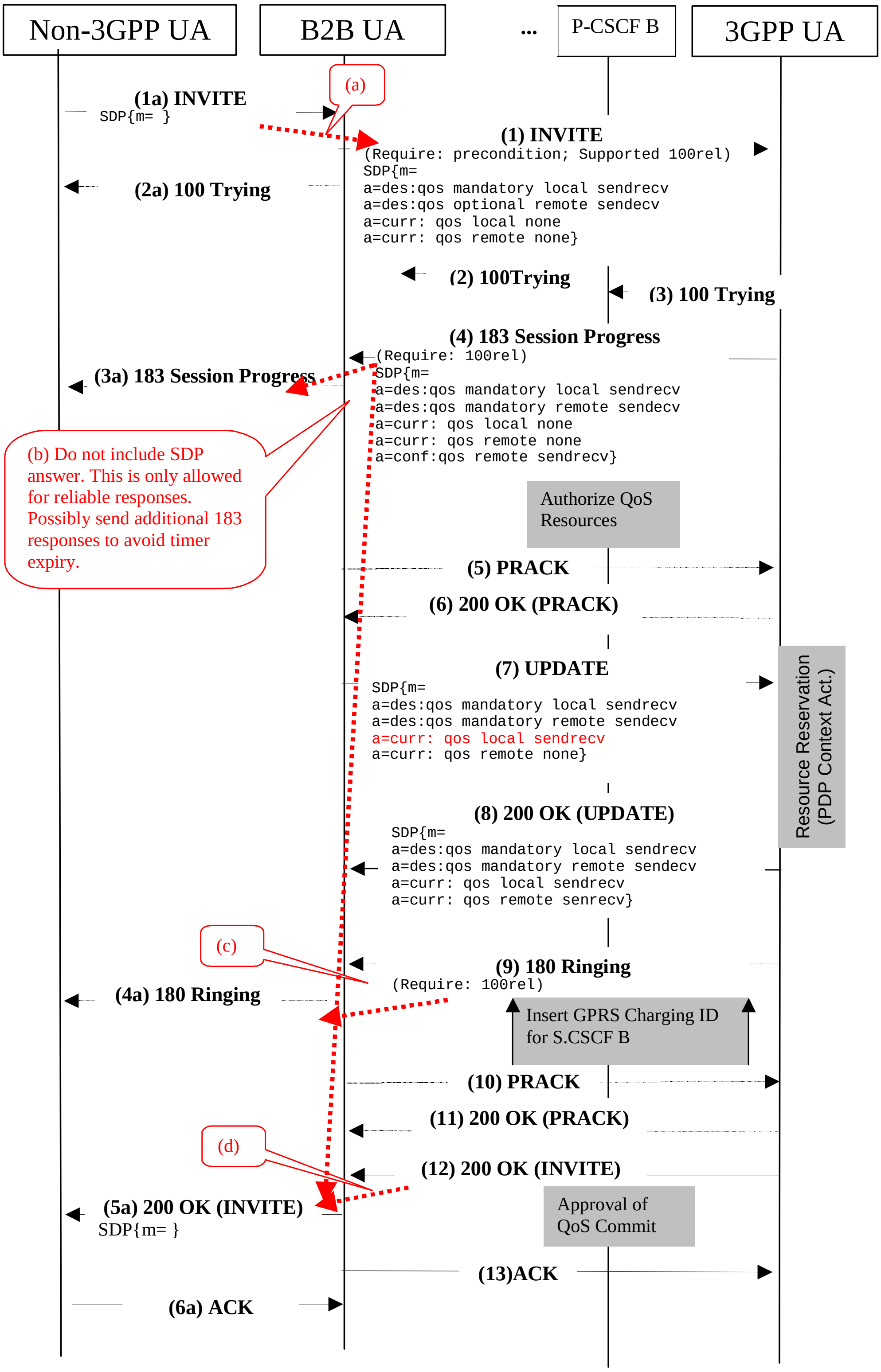

Figure 4.2.3.2.1.2.1/1: Functionality of B2B UA connecting calling non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to called 3GPP UA. SDP offer in INVITE request.

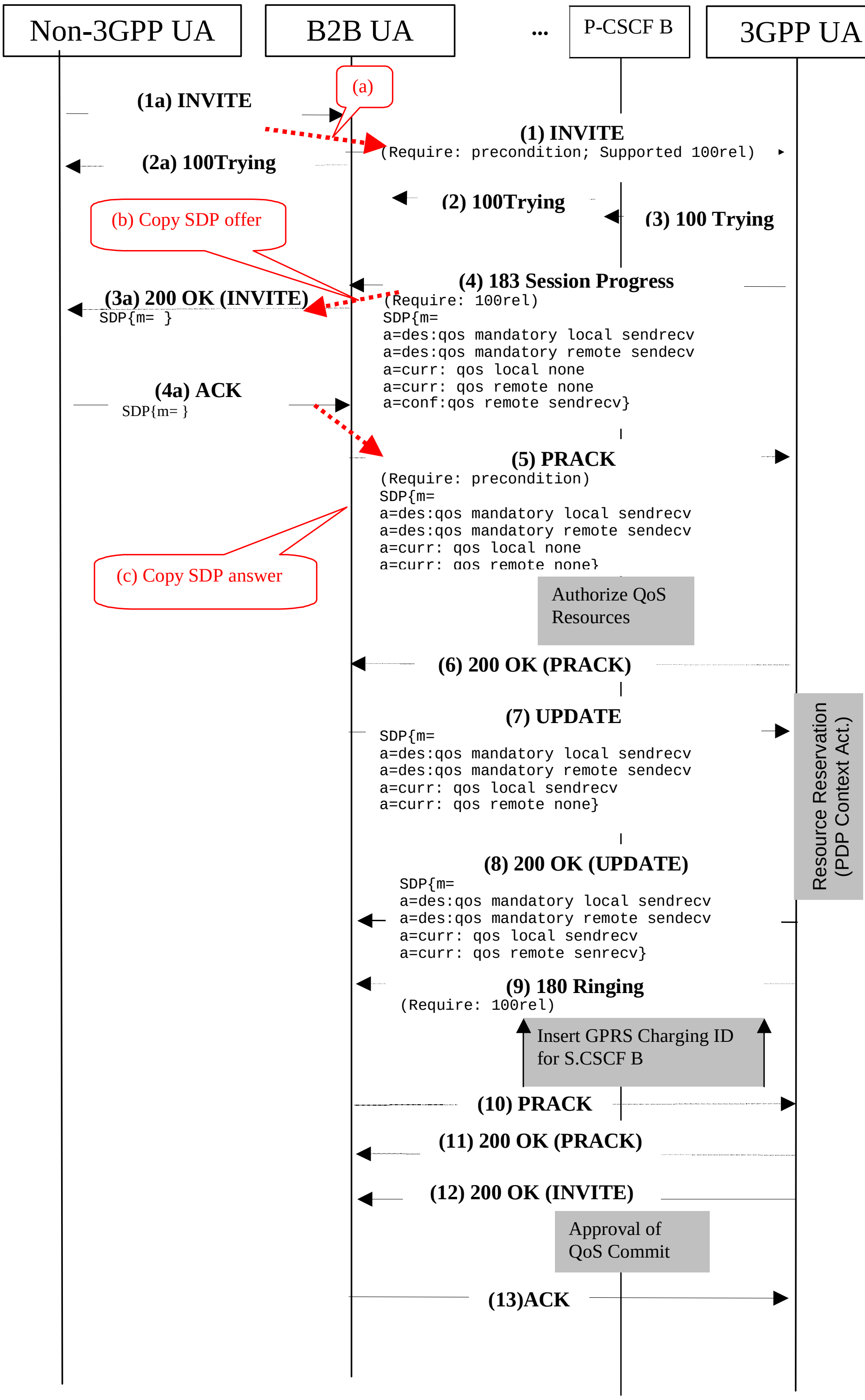

Figure 4.2.3.2.1.2.1/2: Functionality of B2B UA connecting calling non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to called 3GPP UA. SDP offer in OK response.

4.2.3.2.1.2.2 Advantages

General advantages of the B2B UA are discussed in Section 4.2.2.4.1.2.2.

4.2.3.2.1.2.3 Disadvantages

General disadvantages of the B2B UA are discussed in Section 4.2.3.4.1.2.3.

4.2.3.2.2 Modified end-to-end call flow

4.2.3.2.2.1 Description

The rules described in Section 4.1.3.2.2.1 are applied.

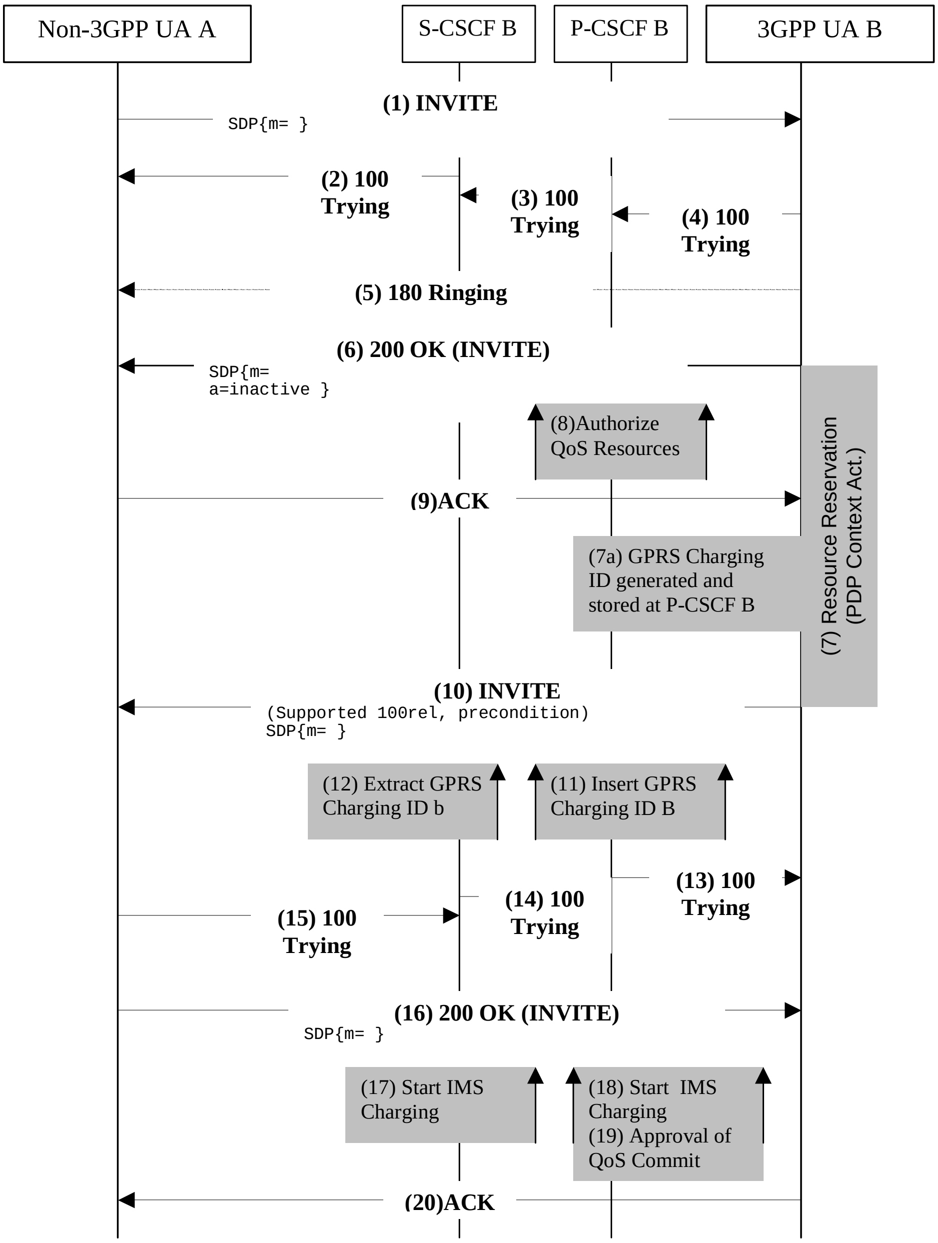

Figure 4.2.3.2.2.2/1: Using re-invite to connect calling non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension to called 3GPP UA. The INVITE contains SDP.

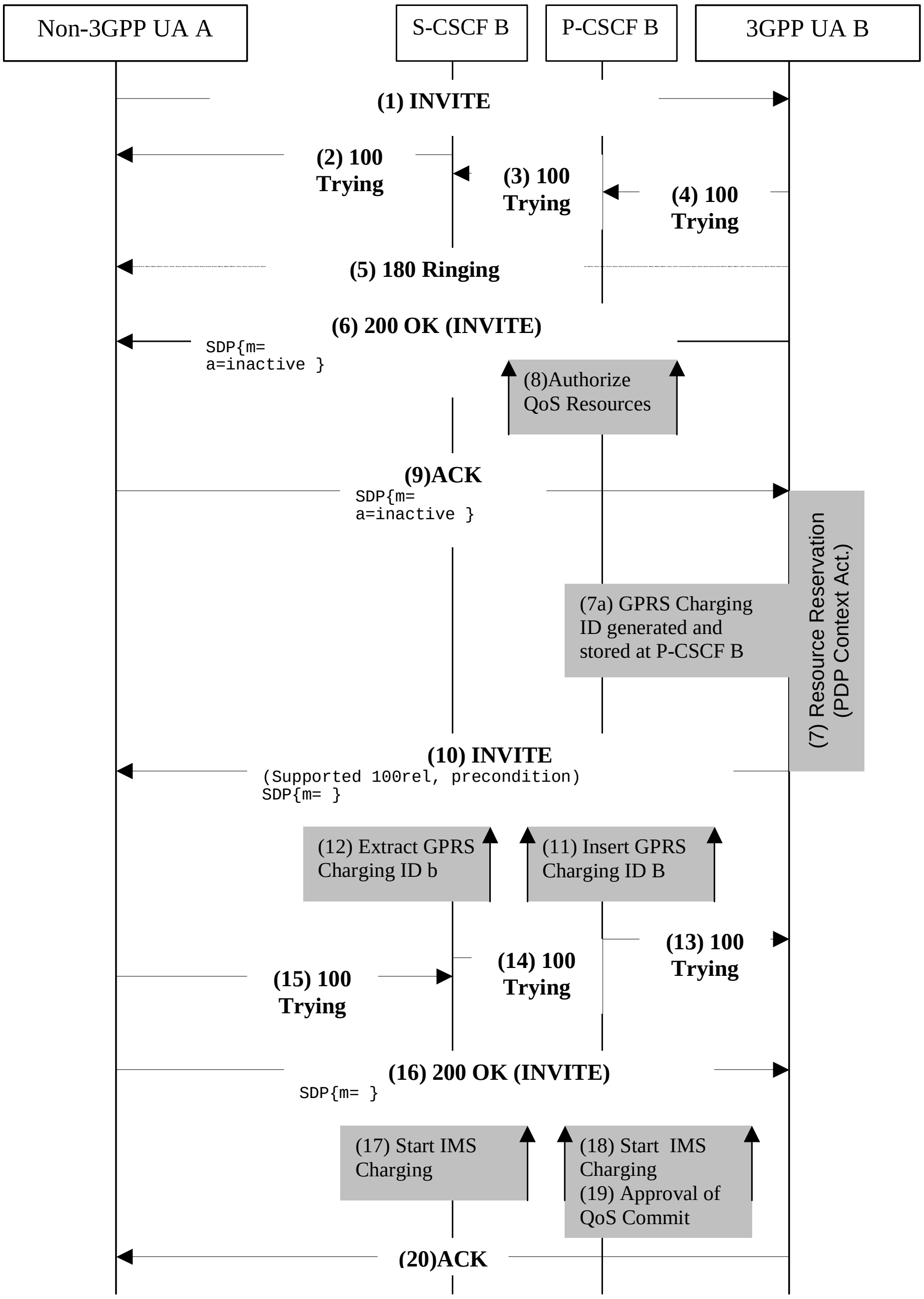

Figure 4.2.3.2.2.2/2: Using re-invite to connect calling non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension to called 3GPP UA. The INVITE contains no SDP.

4.2.3.2.2.2 Advantages

See Section 4.1.3.2.2.2.

4.2.3.2.2.3 Disadvantages

See Section 4.1.3.2.2.3.

Annex A: Interworking topic template

4.x *Topic Name*

4.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.

4.x.2 Flow diagram

Editor's Note: This section contains a flow diagram illustrating the technical description of the possible interworking topic.

4.x.3 Impact of Identified interworking issue

Editor's Note: Identified interworking issues to be considered

- 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

4.x.4 Proposed Resolutions to interworking issue

Editor's Note: This section contains one or more suggestions how an interworking may be performed.

4.x.4.y *Suggestion yy*

4.x.4.y.1 Description

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

4.x.4.y.1 Advantages

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

4.x.4.y.1 Disadvantages

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

4.x.5 Preferred Suggestion

Editor's Note: This section identifies the preferred of the above suggestions, if a consensus has been found.

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: Call flows between rogue 3GPP UA to non-3GPP UA, if 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 does not support or use these extensions.

This annex details the consequences, in case a rogue 3GPP UA does not behave according to TS 24.229 and does not apply some or all of the above SIP extensions.

The numbering of this Annex corresponds to the numbering of Section 4. For example, Sections C.2.1 and 4.2.1 consider the same scenario.

C.1 Calling Rogue 3GPP UA to Called non-3GPP UA

C.1.1 Rogue 3GPP UA to non-3GPP UA supporting the SIP 100rel extension and the SIP update extension, but not supporting 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.1.2 Rogue 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting 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 Section C.1.1, 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.1.3 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.

C.1.2.2 Flow diagram

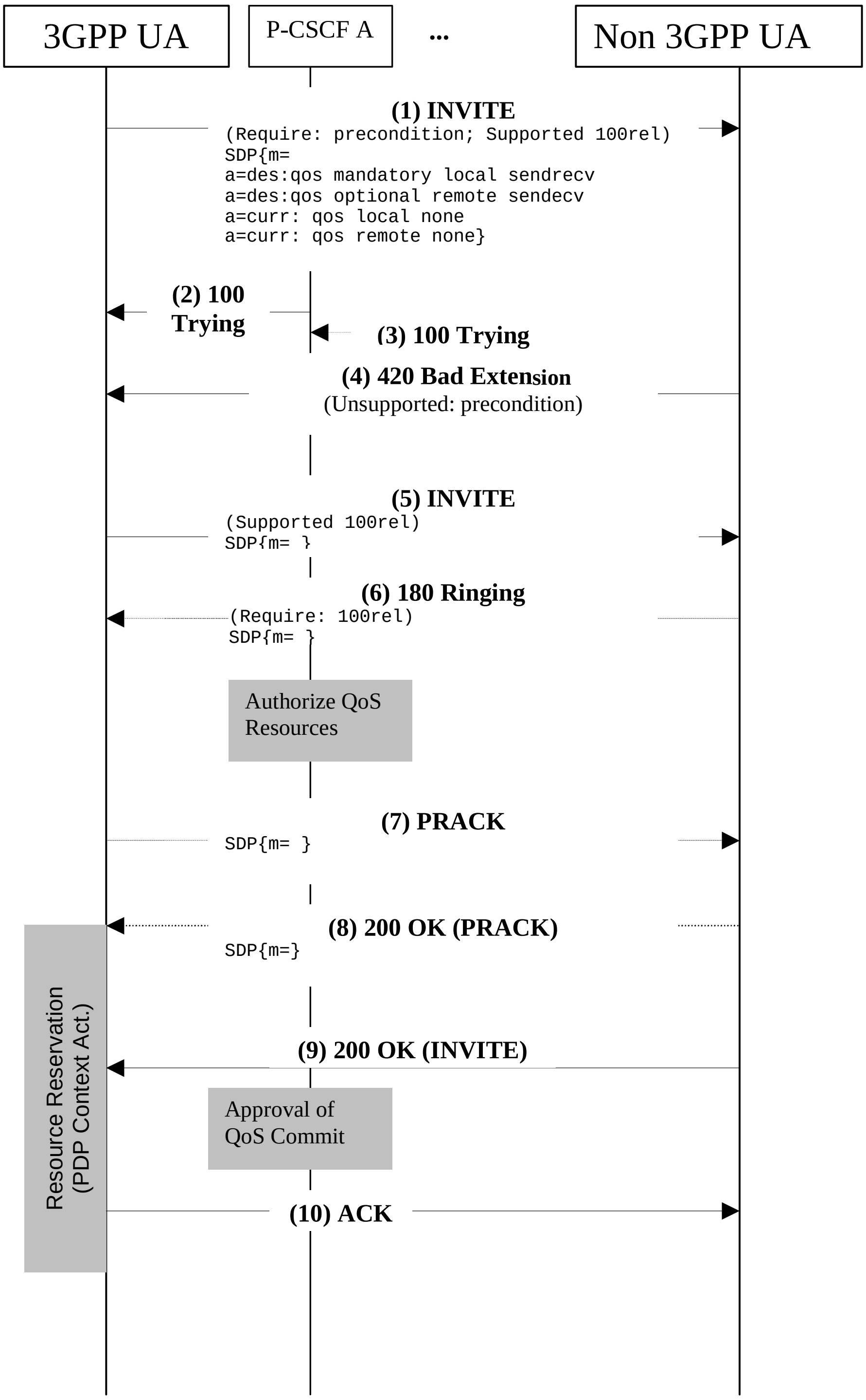

Figure C.1.2.2/1: rogue 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension

C.1.2.3 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 Rogue 3GPP UA to non-3GPP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

C.1.3.1 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.

C.1.3.2 Flow diagram

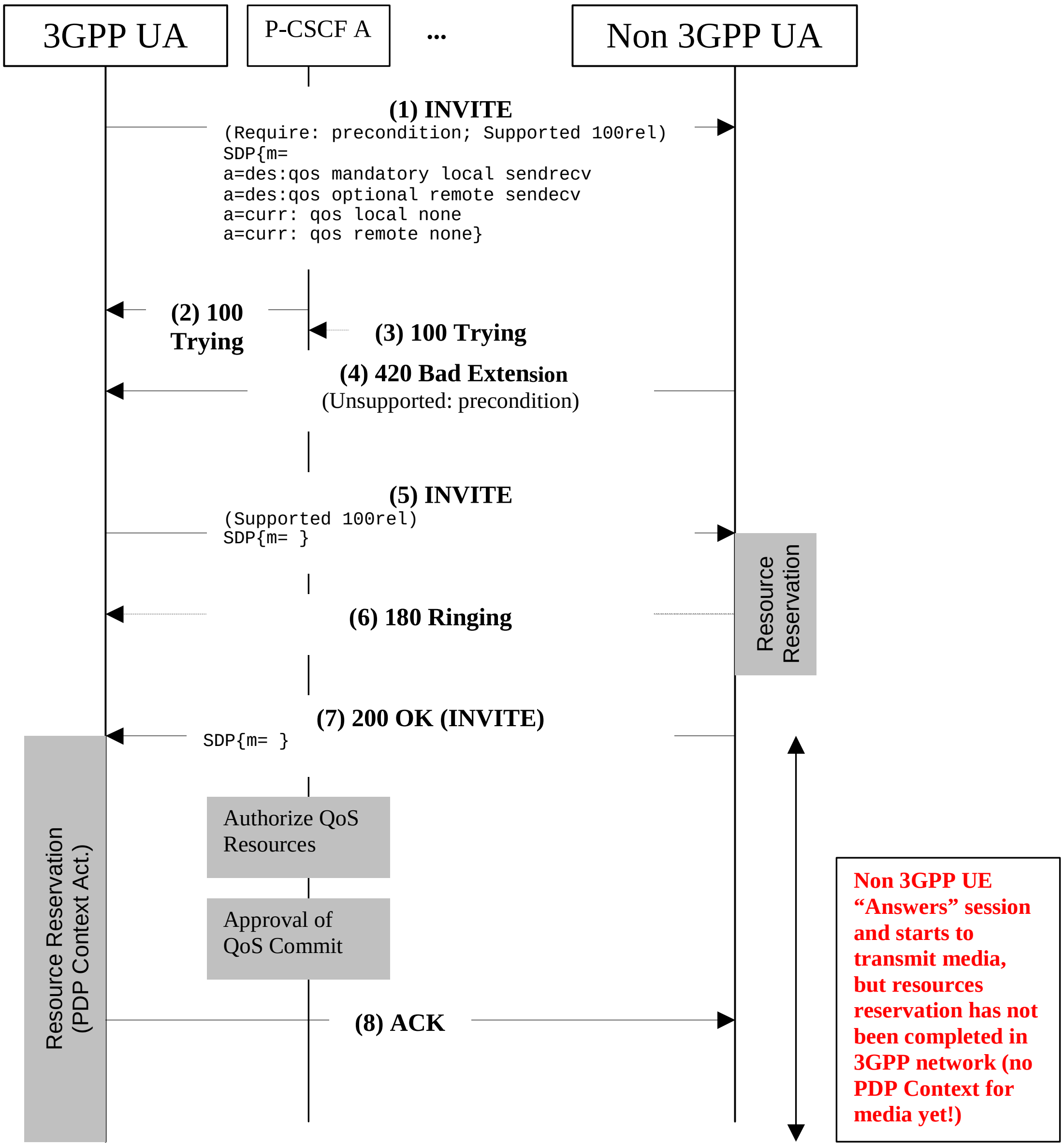

Figure C.1.3.2/1: 3GPP UA to non-3GPP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension.

(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.1.3.3 Impacts of Identified interworking issue

C.1.3.3.1 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.

C.1.3.3.2 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.

C.1.3.3.3 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

C.1.3.3.4 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).

C.1.3.3.5 Fraudulent and security risks

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

C.2 Calling non-3GPP UA to Called Rogue 3GPP UA

C.2.1 Non-3GPP UA supporting the SIP 100rel extension and the SIP update extension, but not supporting the SIP preconditions extension, to rogue 3GPP UA

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.

C.2.2 Non-3GPP SIP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA

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.

C.2.2.2 Flow diagram

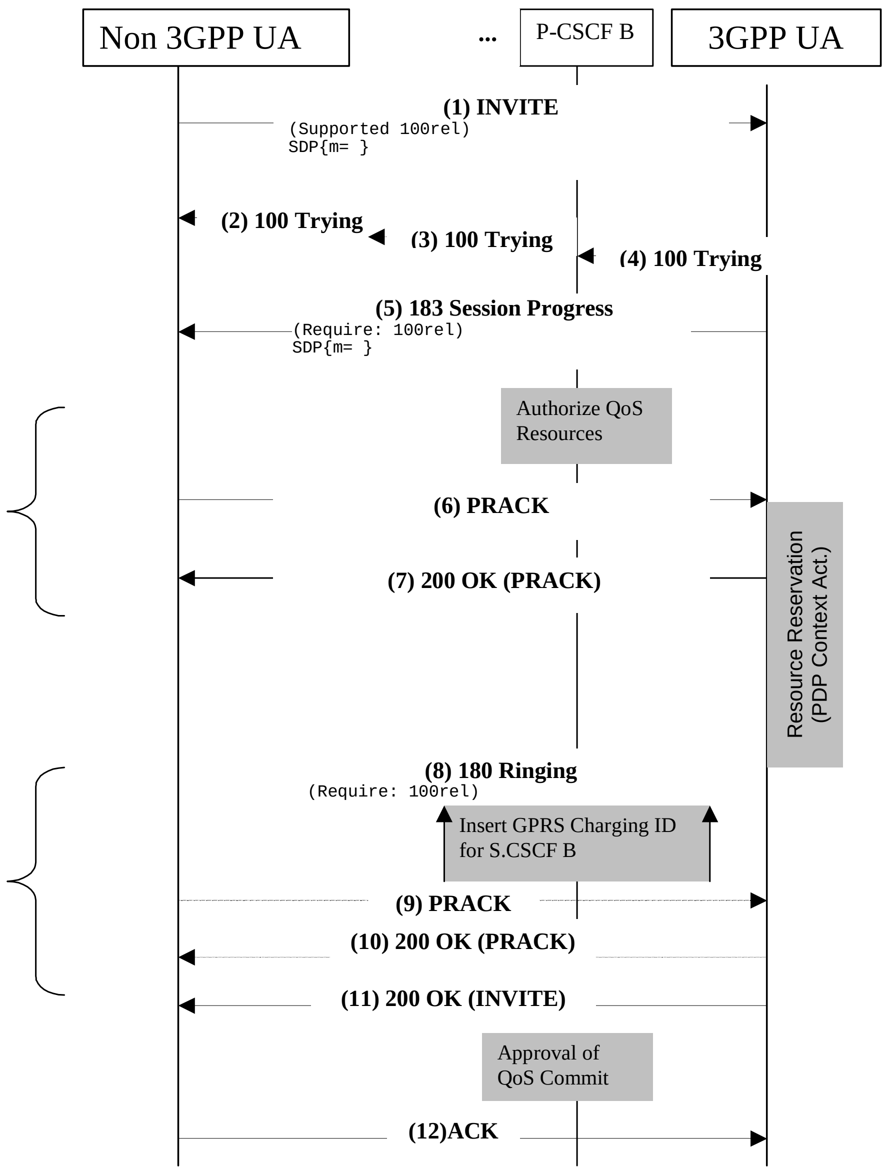

Figure C.2.2.2/1: Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in Invite.

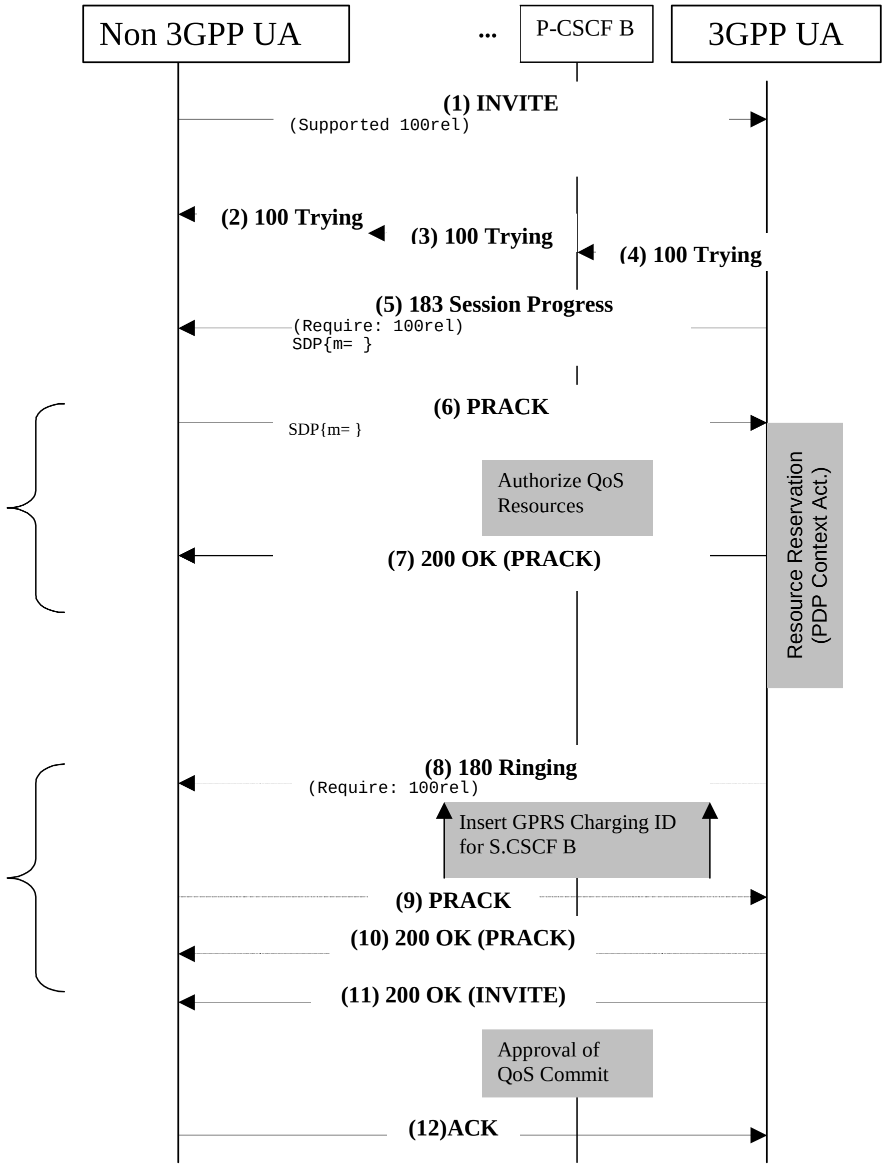

Figure C.2.2.2/2: Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. No SDP offer in Invite.

C.2.2.3 Impacts of Identified interworking issue

No negative impacts have been identified.

C.2.3 Non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA

C.2.3.1 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.

C.2.3.2 Flow diagram

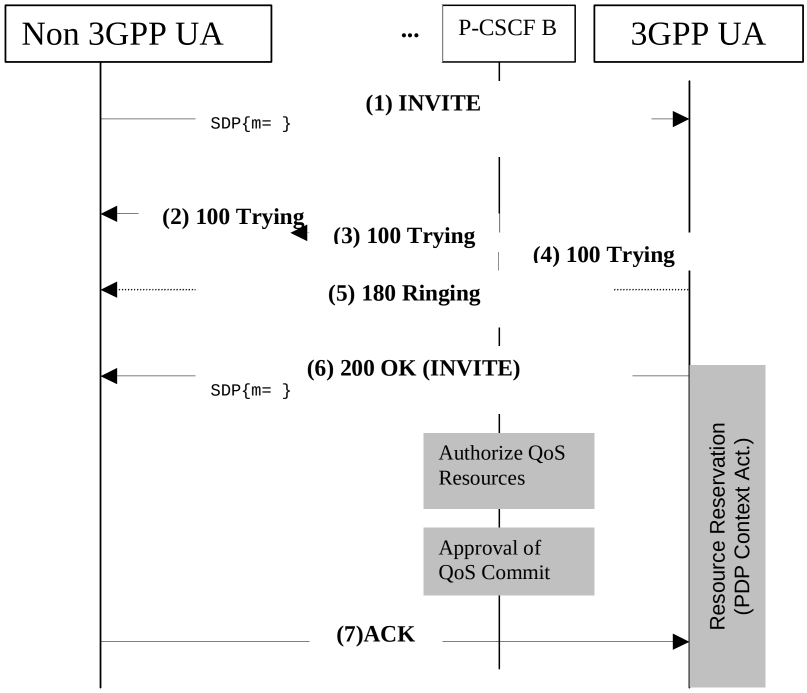

Figure C.2.3.2/1: Non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in INVITE request.

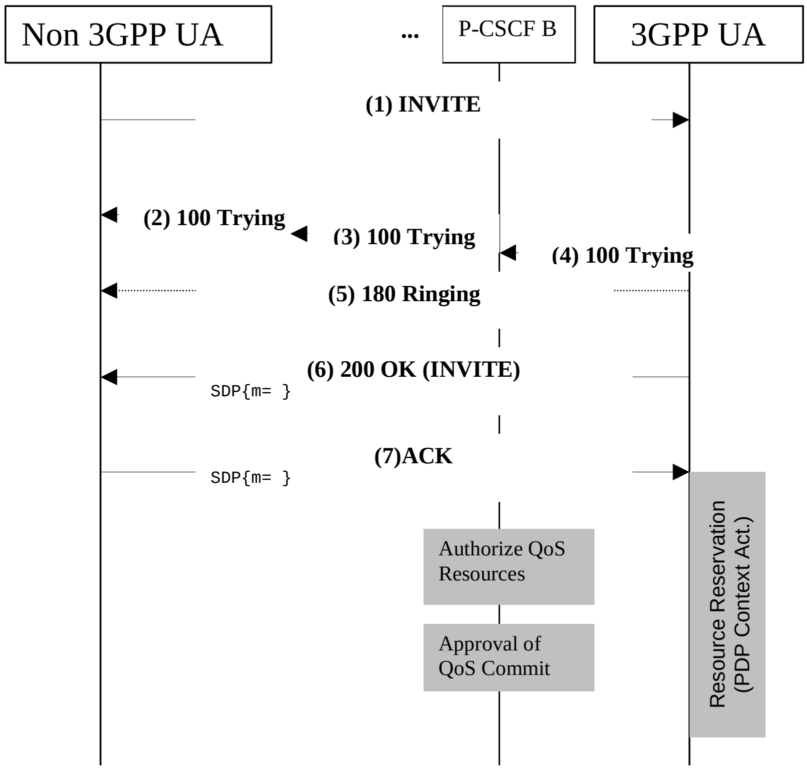

Figure C.2.3.2/2: Non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in OK response.

C.2.3.3 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.

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

The interworking between calling 3GPP UA and called 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 message (1) is send 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 the “183 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 called UA is not capable of meeting unilaterally is included in the initial INVITE (1), a 101-199 provisional response, such as the “183 Session Progress”, is required to transport the SDP answer including the mandated “confirmation status” SDP attribute (Ref. [6], Section 6). Moreover, the “180 Ringing” message 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 (11) and OK(PRACK) (12) messages. A calling 3GPP UA will refrain from generating a new SDP offer within PRACK (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” (14) request is not used, the subsequent” OK(UPDATE)” (17) response 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” (14) request 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.

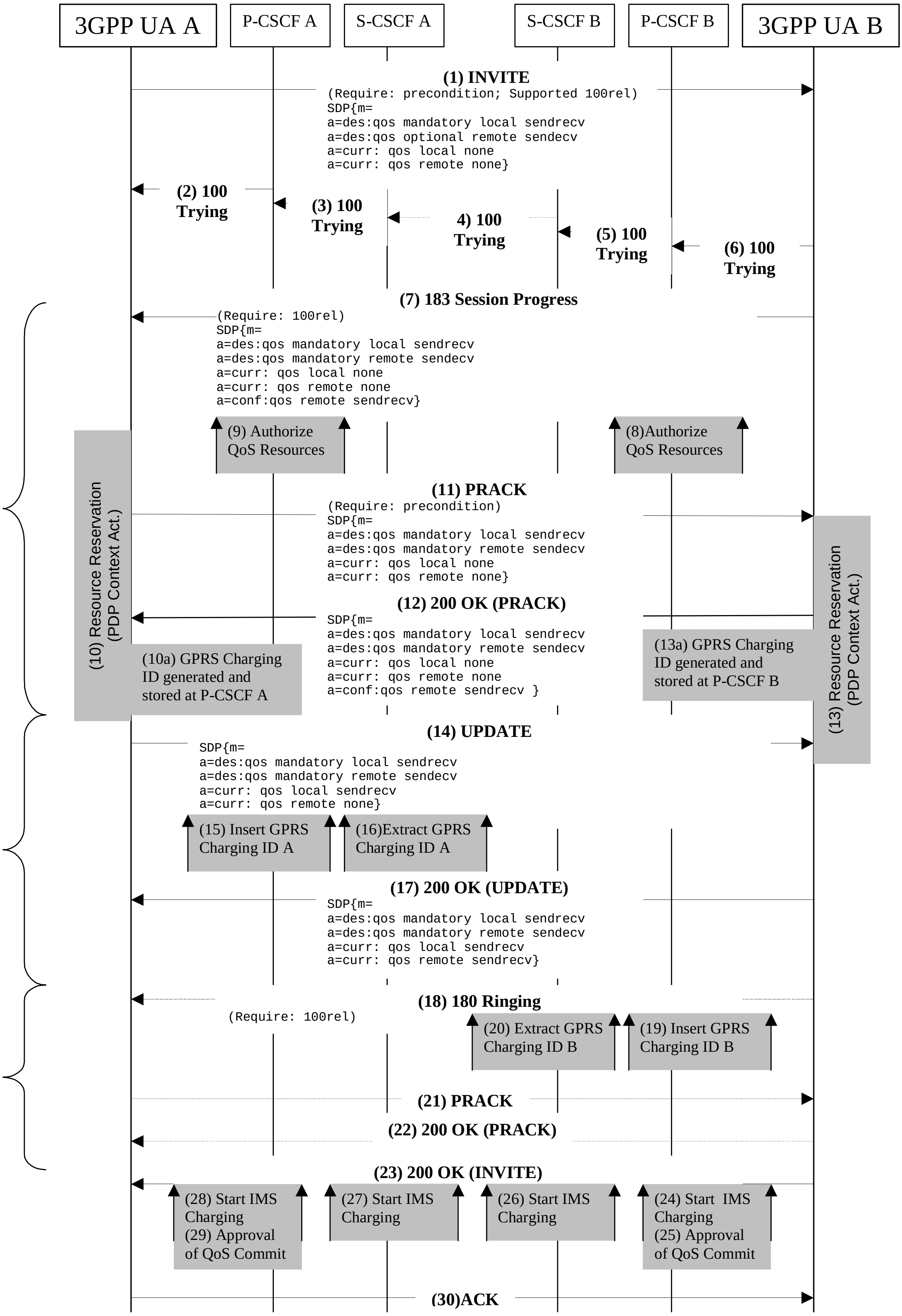

Figure D/1: 3GPP UA to 3GPP UA Call flow

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.

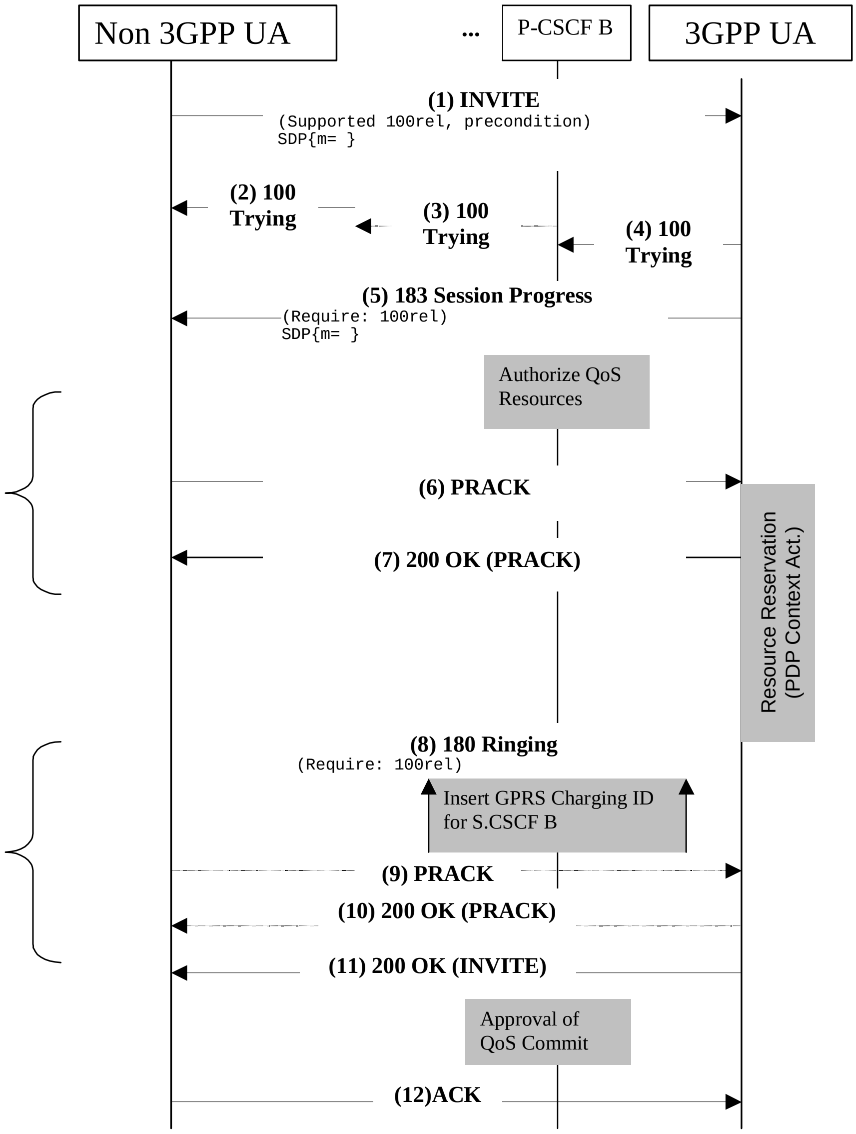

Figure E.1/1: Non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, and not performing QoS reservation, to 3GPP UA

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, 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(INVITE)) 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”.

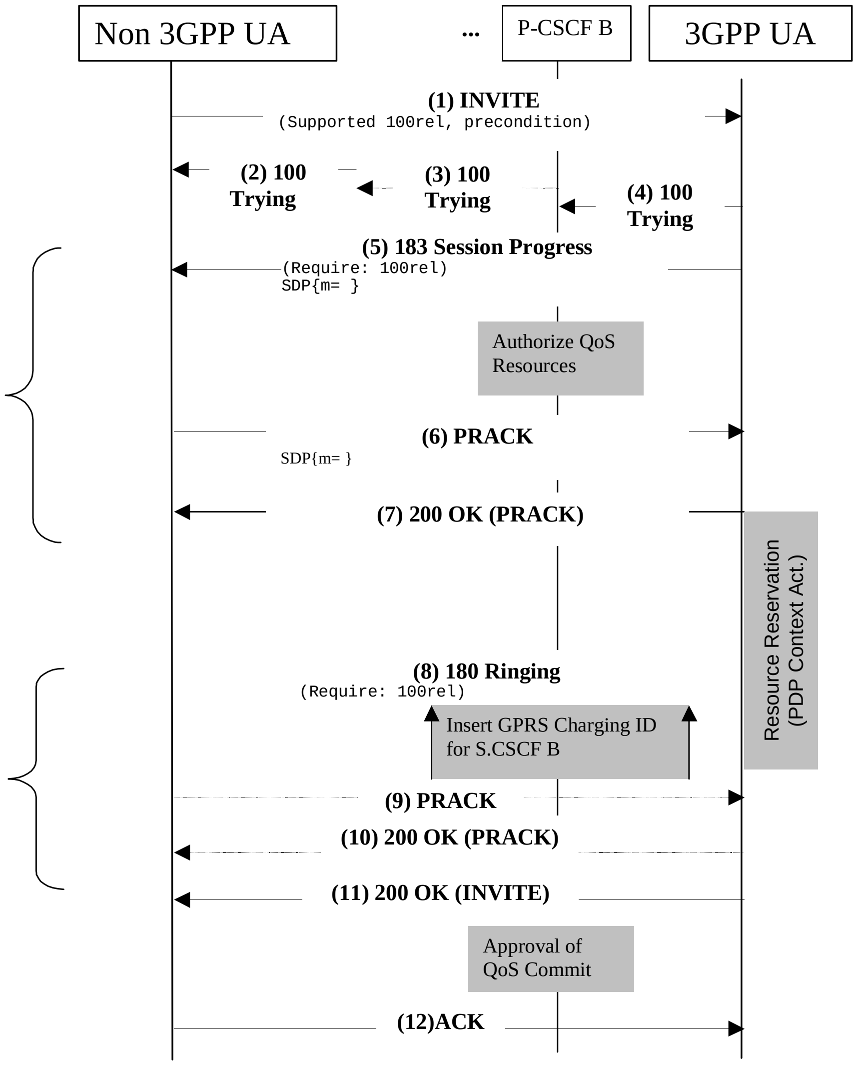

Figure E.2/1: Non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not including the SDP offer in the initial invite, to 3GPP UA

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 M3-030157			Agreed changes are included.	1.0.0	1.1.0
2003-03	NP-19				Presented to Plenary NP#19 for information	1.1.0	