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Technical Report

**Universal Mobile Telecommunications System (UMTS);
Signalling interworking between the 3GPP profile of the
Session Initiation Protocol (SIP) and non-3GPP SIP usage
(3GPP TR 29.962 version 6.1.0 Release 6)**

Reference

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Foreword

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Contents

Foreword

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

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

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1 Scope

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

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

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

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

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

Preconditions: "Integration of Resource Management and SIP" RFC 3312 [5];

Update: "The Session Initiation Protocol UPDATE Method" RFC 3311 [7];

100rel: "Reliability of Provisional Responses in SIP" RFC 3262 [6].

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

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

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

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

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

The present document is dedicated exclusively to issues inherent in the SIP and SDP signalling. Related topics in a wider sense, such as IPv6 to IPv4 address translation or user plane transcoding are out of scope.

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

2 References

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

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

[1] 3GPP TS 24.229: "IP Multimedia Call Control Protocol based on SIP and SDP; Stage 3".

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

[3] 3GPP TS 23.228: "IP Multimedia Subsystem (IMS); Stage 2".

[4] IETF RFC 3261: "SIP: Session Initiation Protocol".

[5] IETF RFC 3312: "Integration of Resource Management and Session Initiation Protocol (SIP)".

[6] IETF RFC 3262: "Reliability of Provisional Responses in Session Initiation Protocol (SIP)".

[7] IETF RFC 3311: "The Session Initiation Protocol (SIP) UPDATE Method".

[8] IETF RFC 3264: "An Offer/Answer Model with Session Description Protocol (SDP)".

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

[10] 3GPP TS 32.225: "Telecommunication management; Charging management; Charging data description for the IP Multimedia Subsystem (IMS)".

[11] 3GPP TS 29.207: "Policy control over Go interface".

3 Definitions and Abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in 3GPP TS 24.229 [1], RFC 3261 [4] and the following apply:

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

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

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

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

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

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

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

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in 3GPP TS 24.229 [1] and RFC 3261 [4] apply.

4 Session setup from calling 3GPP UA towards called non-3GPP UA

Each topic is contained in its own subclause with the structure defined in annex A.

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

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

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

A UA that supports the SIP preconditions extension shall also support the SIP 100rel extension and the SIP update extension. Therefore it includes the "precondition" tag in the Require or in the Supported header, the "100rel" tag in the Supported header and the "Update" tag in the Allow header.

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

4.1.1 Description of interworking issue

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

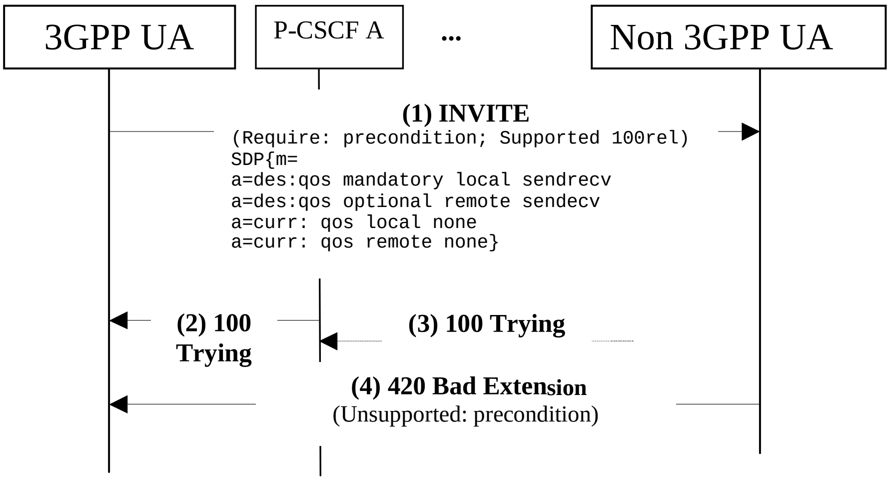


Figure 4.1.1/1: Session Setup towards a non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension.

4.1.2 Proposed Resolution B2BUA

A B2BUA is used.

<u>Insertion of B2BUA</u>

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

In the ideal case, the originating S-CSCF should insert the B2BUA for the entire SIP signalling attempts when the destination network is outside 3GPP. However, the originating S-CSCF does not have any means, according to 3GPP TS 24.229 [1], to decide when the call is destined for a 3GPP network or not. As a consequence, the only solution is for the originating S-CSCF to statically insert the B2BUA for all the signalling that it is leaving the home network.

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

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

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

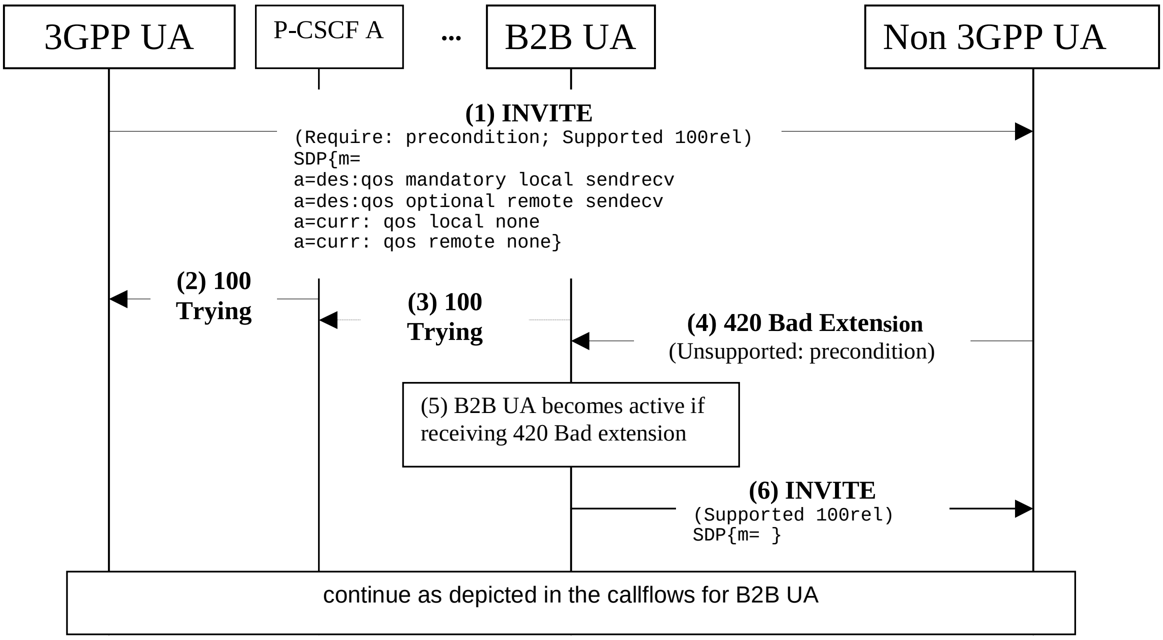


Figure 4.1.2/1: Activation of static B2BUA connecting 3GPP UA to non-3GPP UA not making use of the SIP preconditions extension

Functionality of B2BUA

The B2BUA shall apply the following rules:

1. The B2BUA shall behave as a SIP UA according to RFC 3261 and the SIP 100rel and update extensions on both call legs.
2. On the non-IMS side, the B2BUA shall also comply to the SIP 100rel and update extensions.
3. On the IMS side, the B2BUA shall also comply to and apply the SIP 100rel, update and precondition extensions.
4. The B2BUA shall forward SIP requests arriving at one call leg to the call leg at the other side, with the exceptions below.
5. The B2BUA shall forward SIP provisional responses arriving at one call leg to the call leg at the other side, provided that a transaction at the other call leg is open and with the exceptions below.
6. The B2BUA shall forward SIP final responses arriving at one call leg to the call leg at the other side, with the exceptions below.
7. The B2BUA shall not require the SIP preconditions extension on the non-IMS side.
8. The B2BUA shall insert preconditions in SDP sent from non-3GPP UA to 3GPP UA.
9. The B2BUA should remove preconditions in SDP sent from 3GPP UA to non-3GPP UA.
10. The B2BUA shall not forward PRACK requests and 200 (OK) responses for the PRACK request.
11. The B2BUA shall delay forwarding a 200 (OK) response for an INVITE request from the non-IMS side to the IMS side until the mandatory preconditions are met on the IMS side.
12. The B2BUA shall handle subsequent SDP offers on the IMS side in an INVITE transaction locally, if only the preconditions are modified
13. If the B2BUA receives a subsequent SDP offers on the IMS side with modified media, it shall suspend the transaction on the IMS side and forward this SDP offer to a re-INVITE request on the non-IMS Side. The

B2BUA shall forward the SDP answer received in the re-INVITE request on the non-IMS side to the appropriate message according to the rules for the transport of SDP offer answer pairs in RFC 3261 and continue with the transaction on the IMS side.

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

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

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

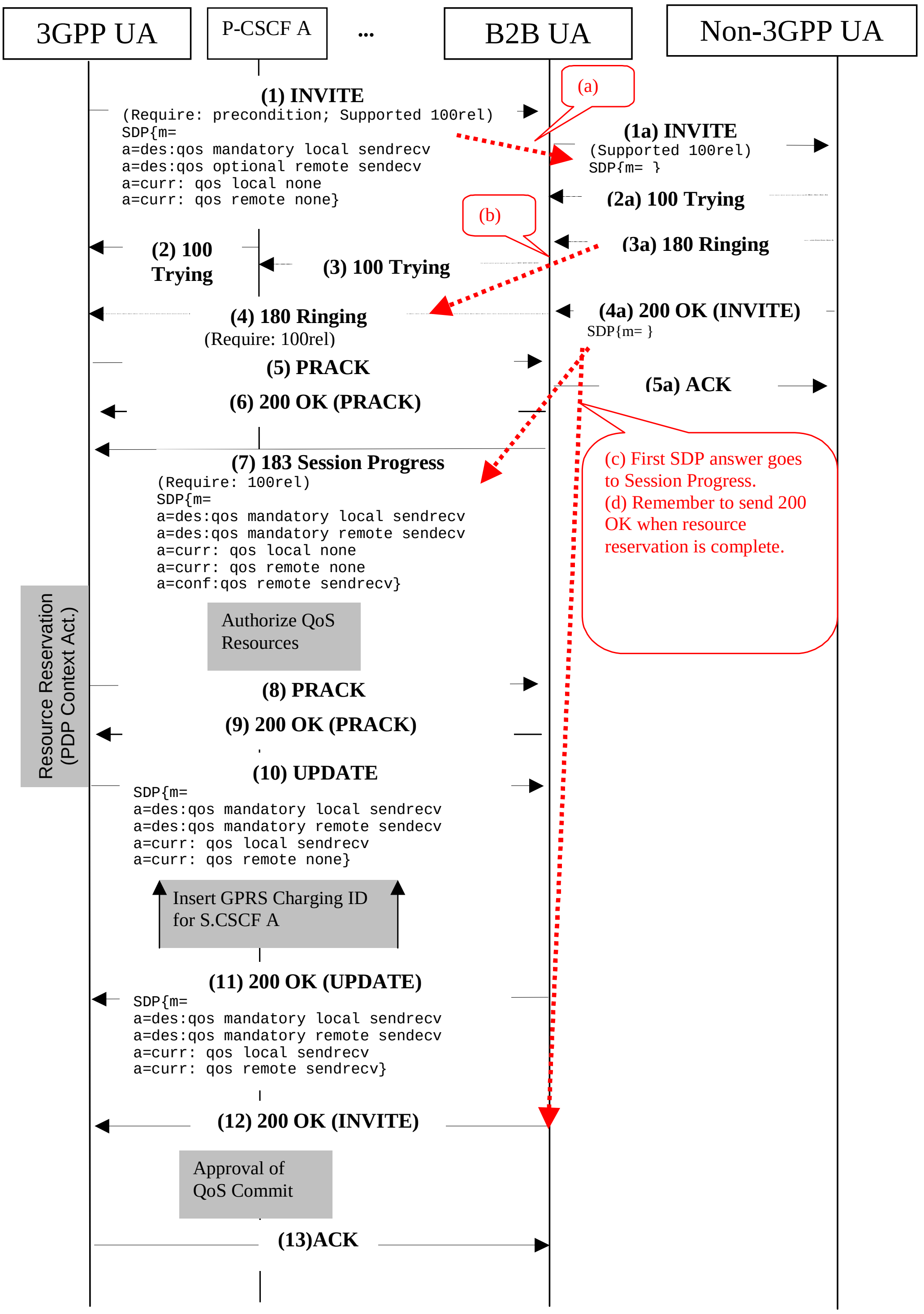


Figure 4.1.2/2: Functionality of B2BUA connecting an originating 3GPP UA to a terminating non-3GPP UA not making use of the SIP preconditions extension, the SIP update extension and the SIP 100rel extension. The originating UA sends no second SDP offer.

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

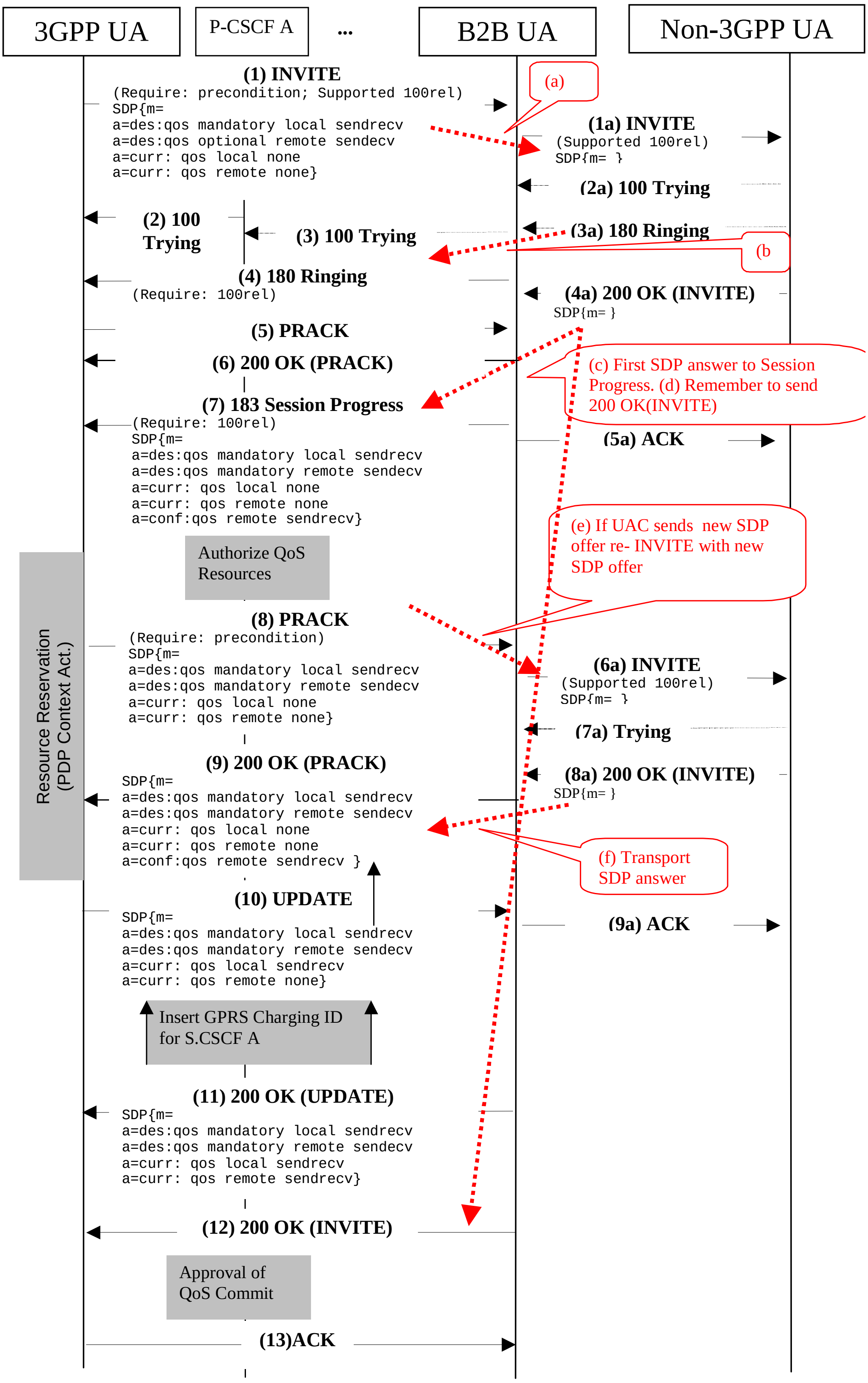


Figure 4.1.2/3: Functionality of B2BUA connecting an originating 3GPP UA to a terminating non-3GPP UA not making use of the SIP preconditions extension, the SIP update extension and the SIP 100rel extension. The originating UA sends second SDP offer.

Implications of the above solution are discussed in clause 6.1.

4.1.3 Proposed Resolution Modified end-to-end call flow

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

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

2. (e.g. in 3GPP TS 24.229 [1]) The terminating non-3GPP UA may send provisional responses without requiring the 100rel extension. The terminating non-3GPP UA may also send a 200 (OK) response for an INVITE request before the 3GPP UA has complete the resource reservation, but will not send media, because it was requested in the SDP offer (media was inactive) by the 3GPP UA. The 3GPP UA may send a re-INVITE activating the media by setting them to "send", "recv", or "sendrecv" in SDP once the local resource reservation is complete.

3. (e.g. in 3GPP TS 29.207 [11] and 3GPP TS 29.208) The P-CSCF(PDF) shall approve the QoS Commit when receiving a 200 (OK) response for an INVITE request only, if media streams are active ("send", "recv", or "sendrecv" in SDP). To avoid fraud, P-CSCF(PDF) should not open gates for inactive media streams.

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

5. (e.g. in 3GPP TS 24.229): GPRS Charging ID is transported in INVITE request.

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

6. (e.g. 3GPP TS 29.207 [11] and 3GPP TS 29.208): P-CSCF and S-CSCF shall treat media in a SDP answer as "inactive" with respect to the rules above, ignoring any other setting, if the media were set to "inactive" in the SDP offer.

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

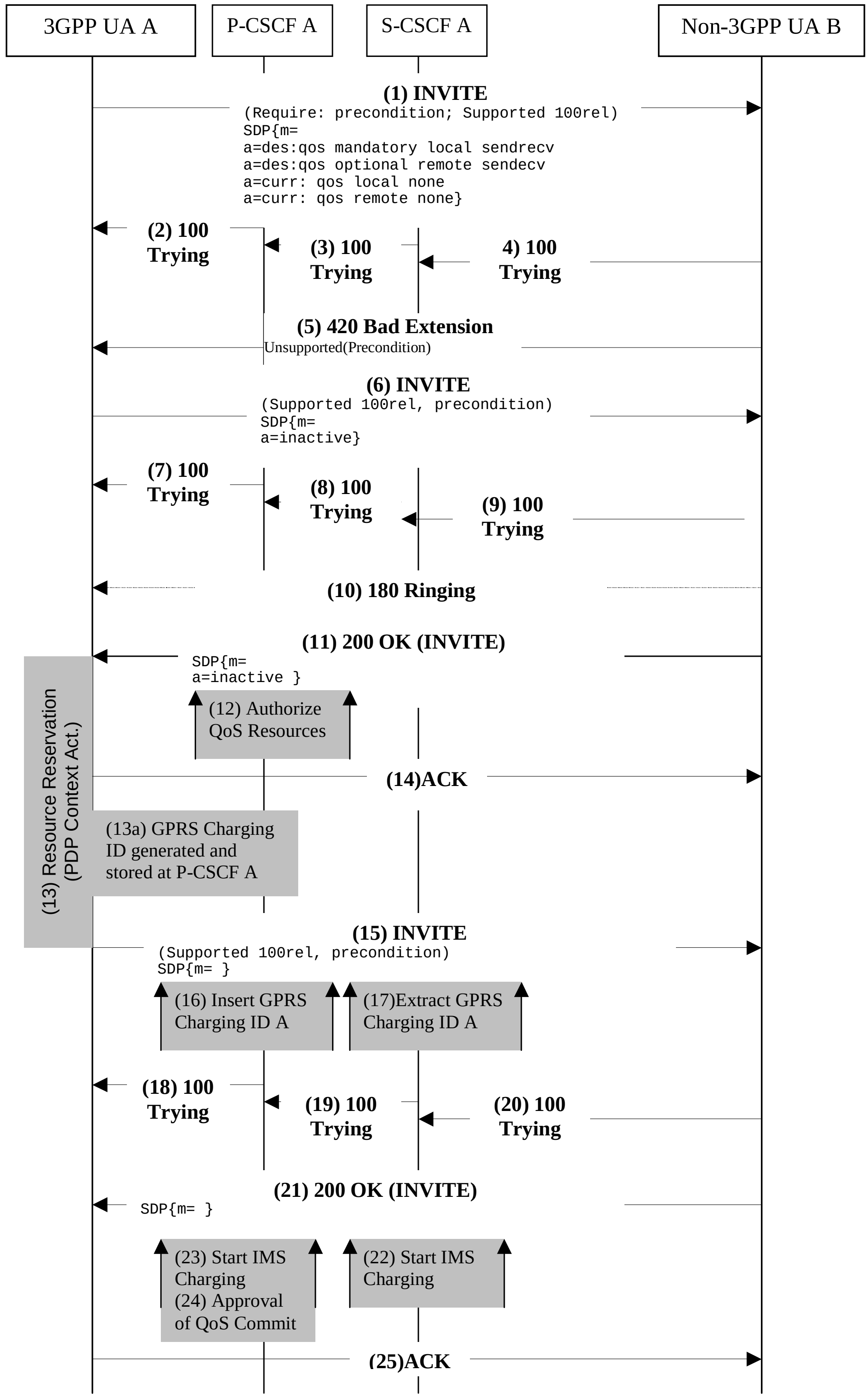


Figure 4.1.3/1: Using re-INVITE to connect an originating 3GPP UA to a terminating non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension.

Implications of the above solution are discussed in clause 6.2.

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

4.2.1 Description of interworking issue

The call fails, as detailed in clause 4.1.1.

4.2.2 Proposed resolution B2BUA

Insertion of B2BUA

The Insertion of the B2BUA is detailed in clause 4.1.2.

Functionality of B2BUA

The B2BUA applies the rules detailed in clause 4.1.2.

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

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

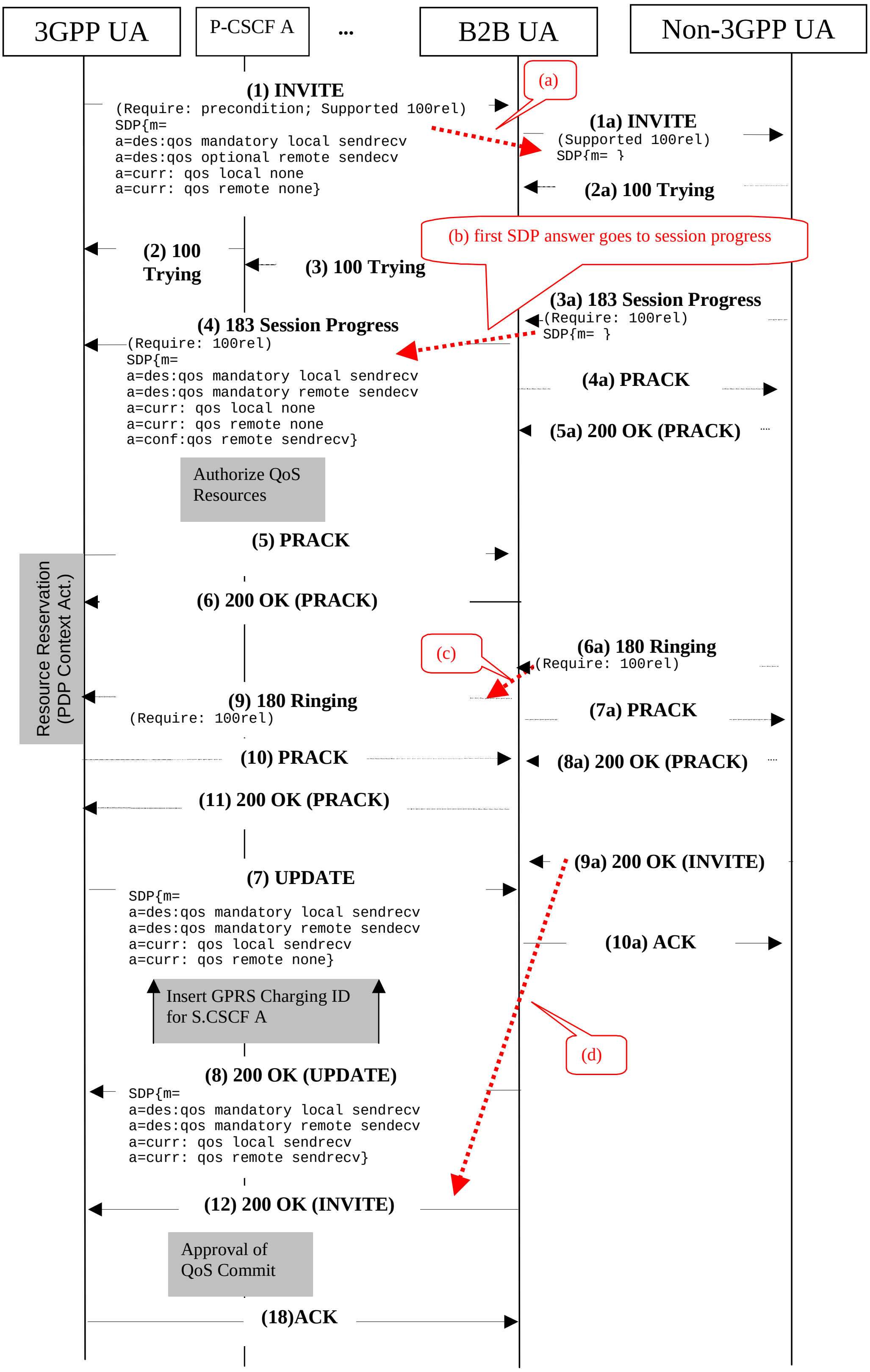


Figure 4.2.2/1: Functionality of B2BUA connecting 3GPP UA to non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension. The originating UA includes SDP answer in 183 "Session Progress". The terminating UA sends no second SDP offer.

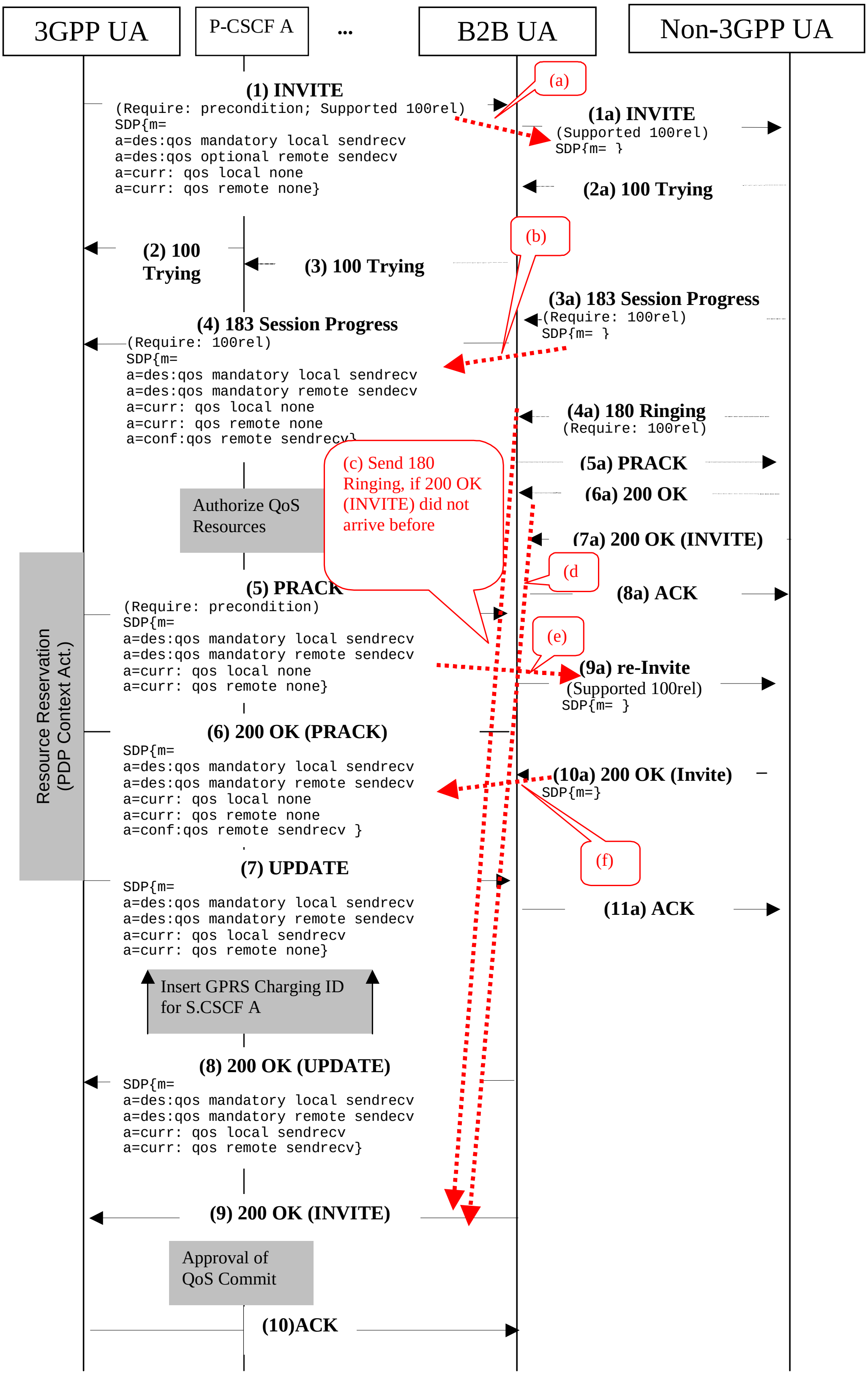


Figure 4.2.2/2: Functionality of B2BUA connecting 3GPP UA to non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension. The terminating UA includes SDP answer in 183 "Session Progress". The originating UA sends second SDP offer.

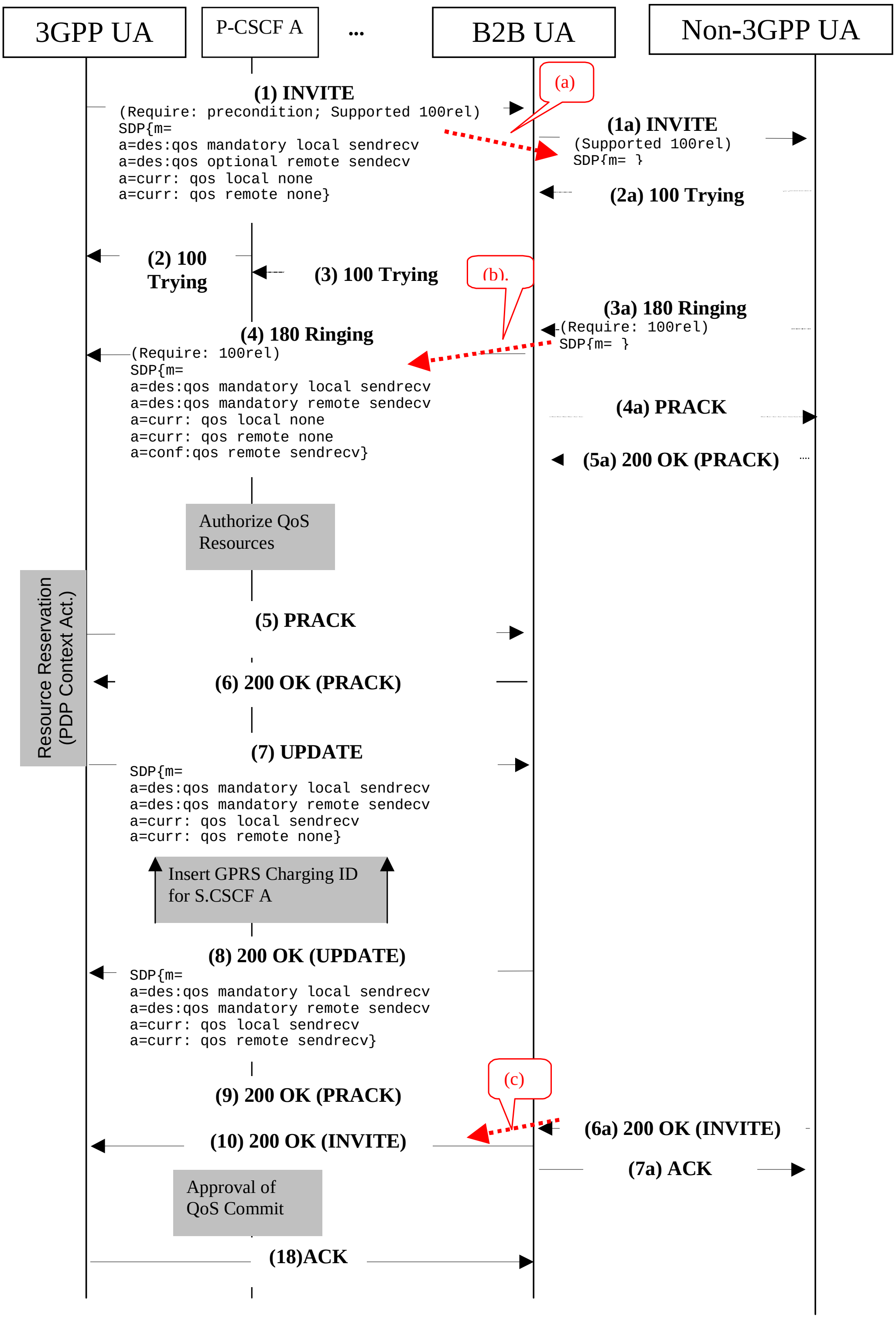


Figure 4.2.2/3: Functionality of B2BUA connecting 3GPP UA to non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension. The terminating UA includes SDP answer in 180 "Ringing". The originating UA sends no second SDP offer.

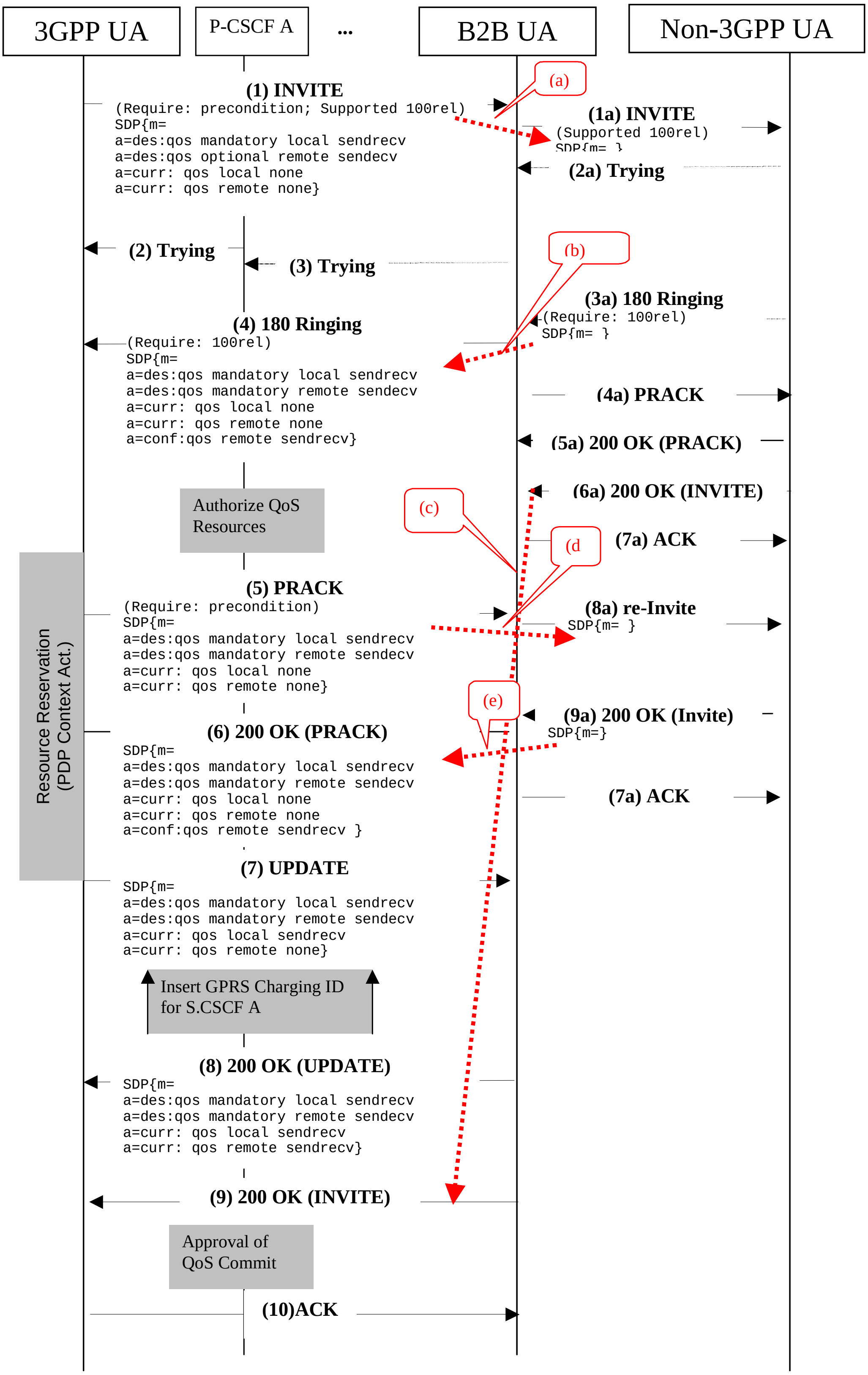


Figure 4.2.2/4: Functionality of B2BUA connecting an originating 3GPP UA to terminating non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension. The terminating UA includes SDP answer in 180 "Ringing". The terminating UA sends second SDP offer.

Implications of the above solution are detailed in Clause 6.1.

4.2.3 Proposed resolution modified end-to-end call flow

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

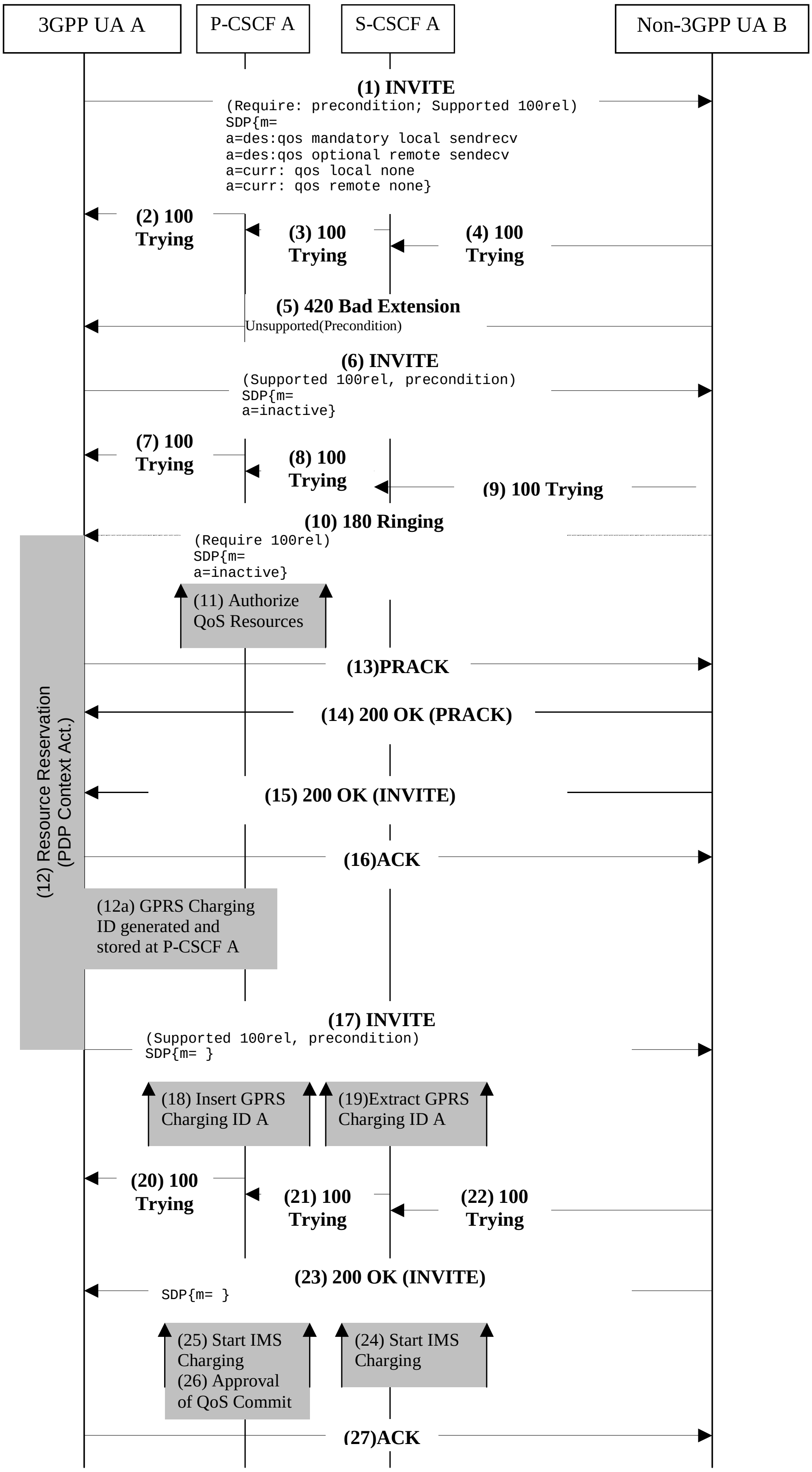


Figure 4.2.3/1: Using re-INVITE to connect an originating 3GPP UA to a terminating non-3GPP UA not making use of the SIP preconditions extension.

Implications of the above solution are detailed in Clause 6.2.

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

4.3.1 Description of interworking issue

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

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

4.3.2 Proposed Resolution B2BUA

A B2BUA is used.

Insertion of B2BUA

How the B2BUA is inserted is discussed within Clause 4.1.2.

Functionality of B2BUA

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

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

Implications of the above solution

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

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

4.3.3 Proposed Resolution Modified end-to-end call flow

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

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

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

Implications of the above solution are detailed in Clause 6.2.

5 Session setup from calling non-3GPP UA towards called 3GPP UA

Each topic is contained in its own subclause with the structure defined in Annex A.

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

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

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

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

5.1.1 Description of interworking issue

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

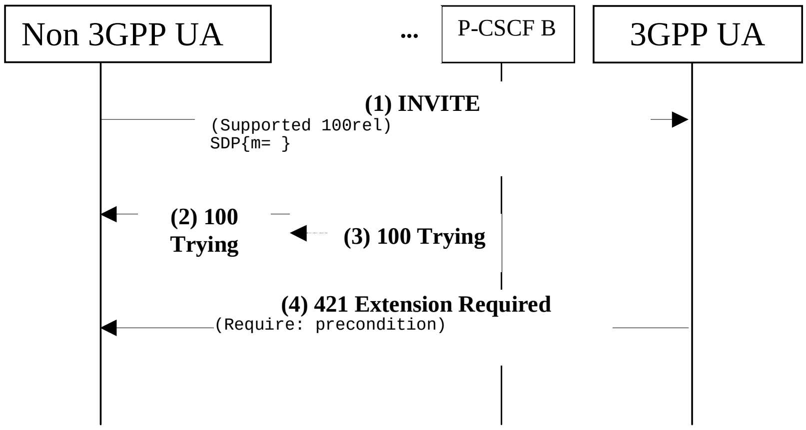


Figure 5.1.1/1: Session Setup from non-3GPP SIP UA not making use of the SIP preconditions extension and the SIP update extension, to 3GPP UA.

5.1.2 Proposed Resolution B2BUA

A B2BUA is used.

<u>Insertion of B2BUA</u>

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

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

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

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

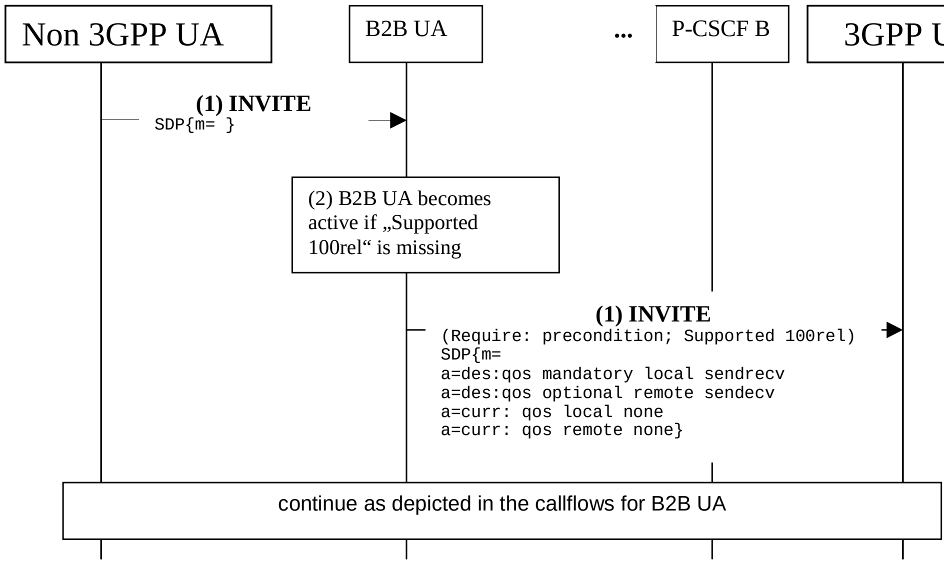


Figure 5.1.2/1: Activation of static B2B connecting Non-3GPP SIP UA not indicating support of the SIP preconditions extension to 3GPP UA.

Functionality of B2BUA

The B2BUA shall apply the following rules:

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

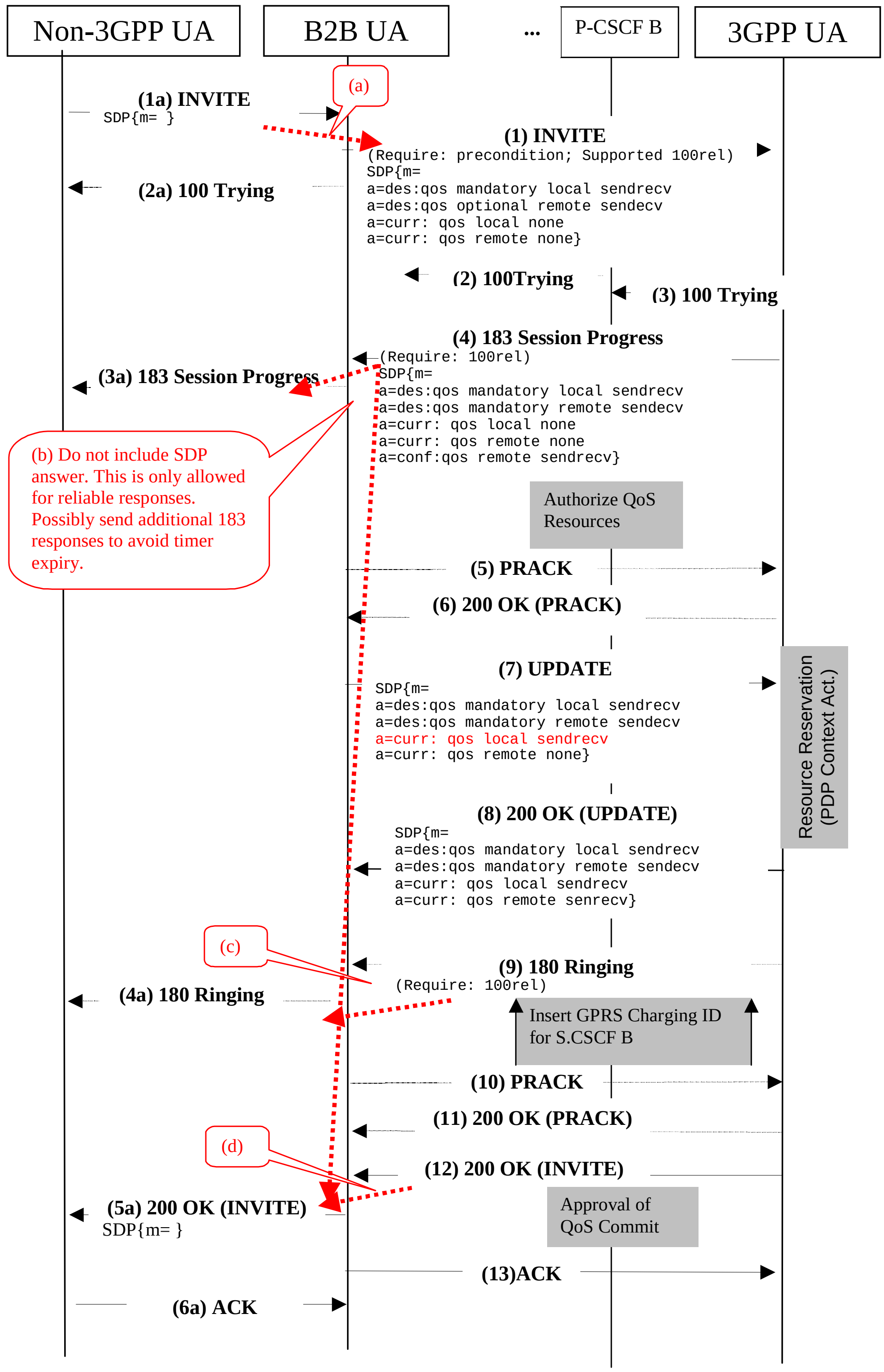


Figure 5.1.2/2: Functionality of B2BUA connecting an originating non-3GPP SIP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to a terminating 3GPP UA. SDP offer in INVITE request.

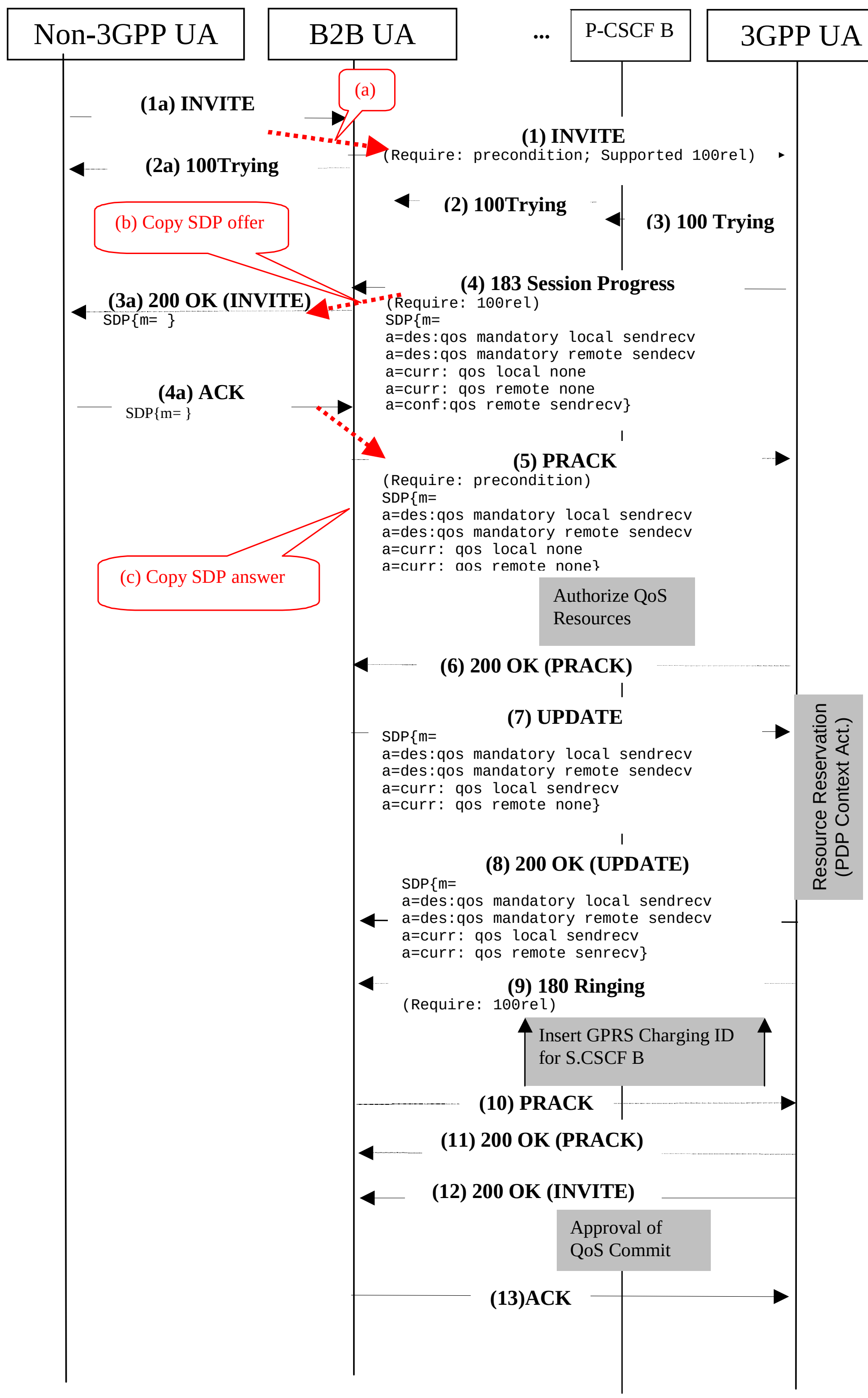


Figure 5.1.2/3: Functionality of B2BUA connecting an originating non-3GPP SIP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to a terminating 3GPP UA. SDP offer in OK response.

Implications of the above Solution are discussed in Clause 6.1.

5.1.3 Proposed Resolution Modified end-to-end call flow

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

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

2. (e.g. in 3GPP TS 24.229 [1]) If a 3GPP UA receives an INVITE request without the support for preconditions and without any SDP payload, the 3GPP UA shall put the media on inactive in the SDP offer when answering the INVITE request. When an ACK is received with an SDP answer, and the media in it requires the 3GPP UA to reserve resources, the 3GPP UA shall start reserving resources and send a re-INVITE to resume the inactive media whenever it is ready with the resource reservation.

3. (e.g. in 3GPP TS 29.207 [11] and 3GPP TS 29.208 [9]) The P-CSCF(PDF) shall approve the QoS Commit when receiving a 200 (OK) response for an INVITE request only, if media streams are active ("send", "recv", or "sendrecv" in SDP). To avoid fraud, P-CSCF(PDF) should not open gates for inactive media streams.

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

5. (e.g. in 3GPP TS 24.229 [1]): GPRS Charging ID is transported in the re-INVITE request .

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

6. (e.g. TS 29.207 [11] and 29.208 [9]): P-CSCF and S-CSCF shall treat media in a SDP answer as "inactive" with respect to the rules above, ignoring any other setting, if the media were set to "inactive" in the SDP offer.

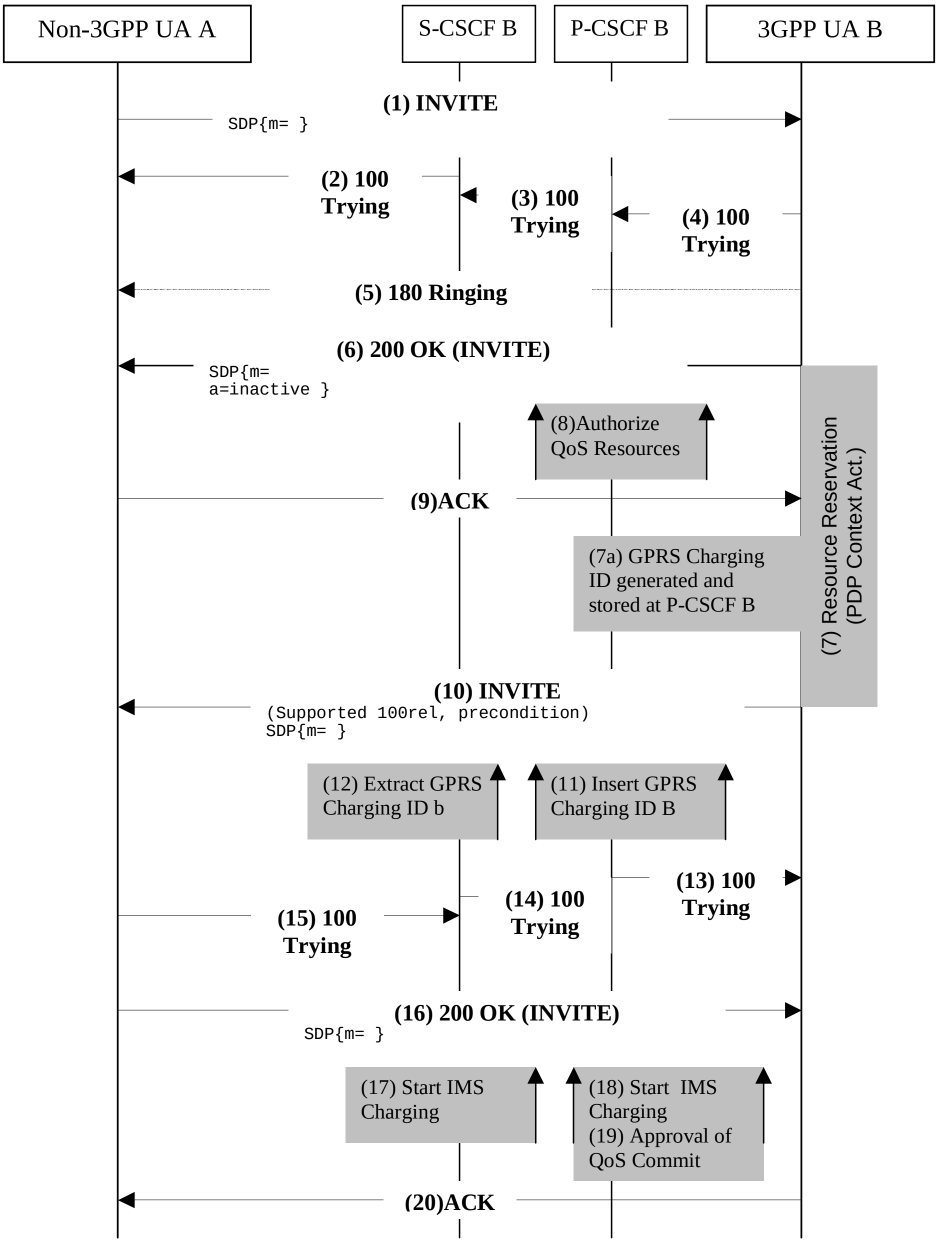


Figure 5.1.3/1: Using re-INVITE to connect an originating non-3GPP UA not making use of the SIP preconditions extension, the SIP update extension and the SIP 100rel extension to a terminating 3GPP UA. The INVITE request contains SDP.

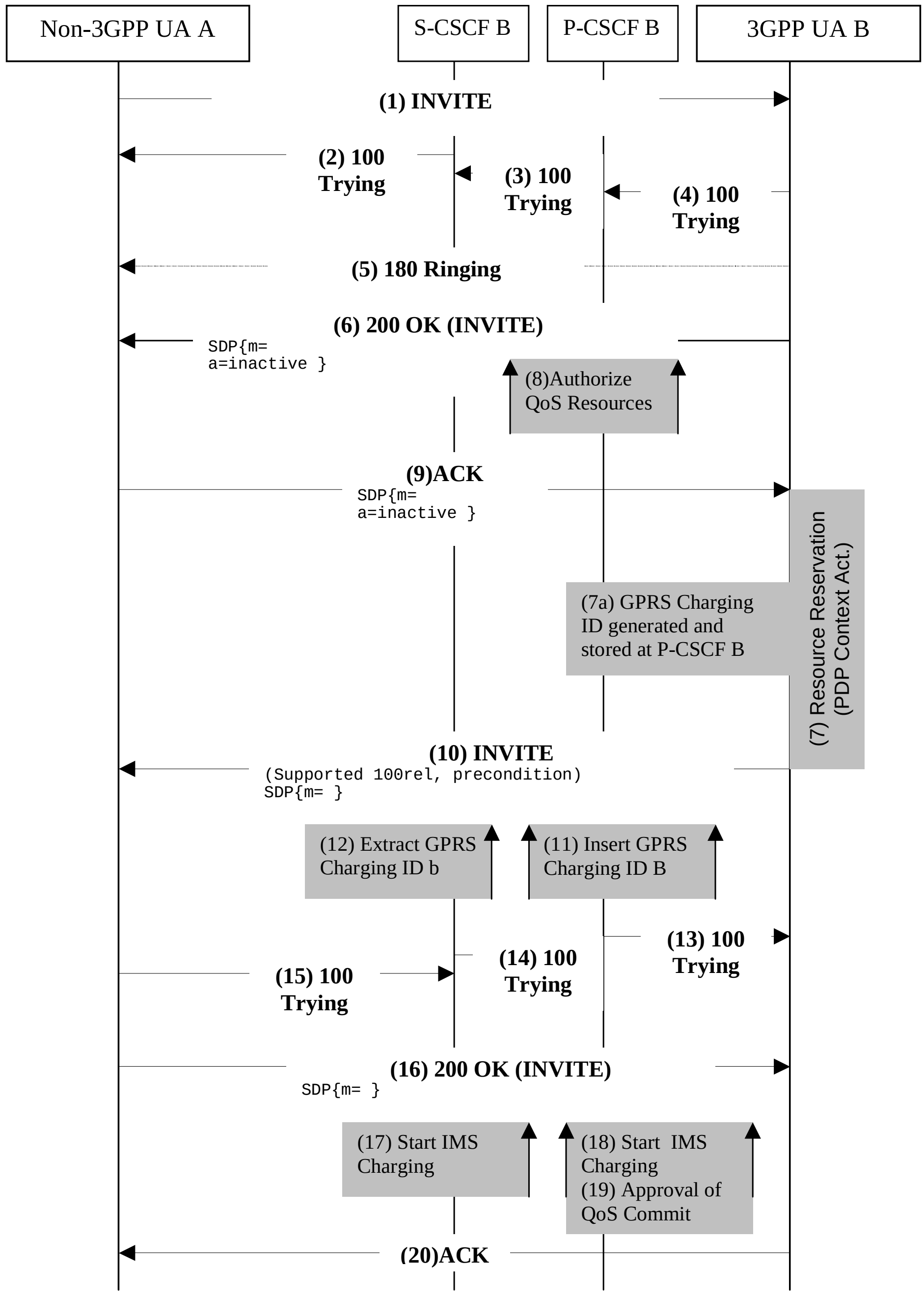


Figure 5.1.3/2: Using re-INVITE to connect an originating non-3GPP UA not making use of the SIP preconditions extension, the SIP update extension and the SIP 100rel extension to a terminating 3GPP UA. The INVITE request contains no SDP.

Implications of the above Solution are discussed in Clause 6.2.

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

5.2.1 Description of interworking issue

The call fails, as detailed in Clause 5.1.1.

5.2.2 Proposed resolution B2BUA

A B2BUA is used.

Insertion of B2BUA

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

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

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

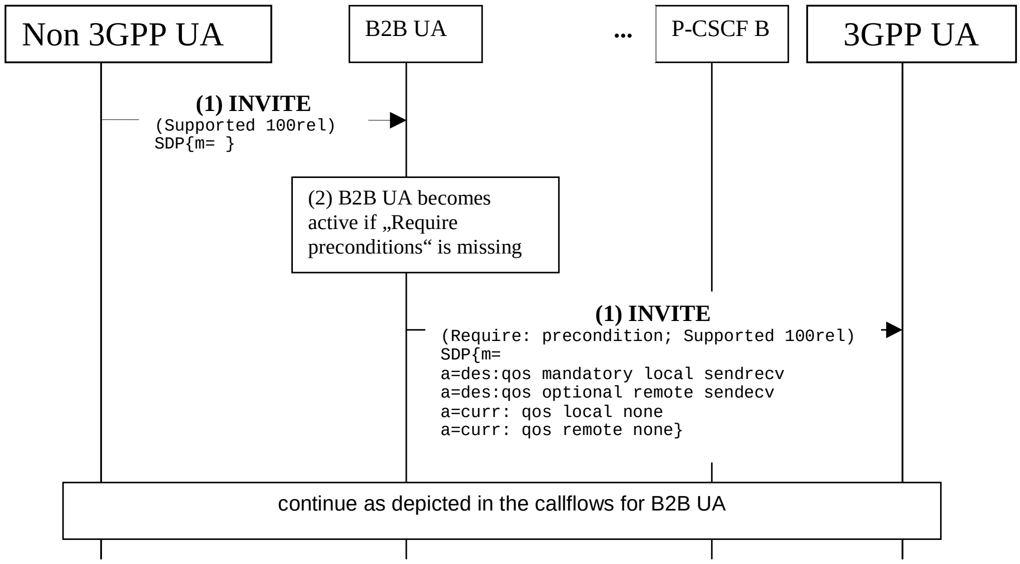


Figure 5.2.2/1: Activation of static B2BUA connecting Non-3GPP UA not indicating support of the SIP preconditions extension to 3GPP UA.

Functionality of B2BUA

The B2BUA shall apply the rules detailed in Clause 5.1.2.

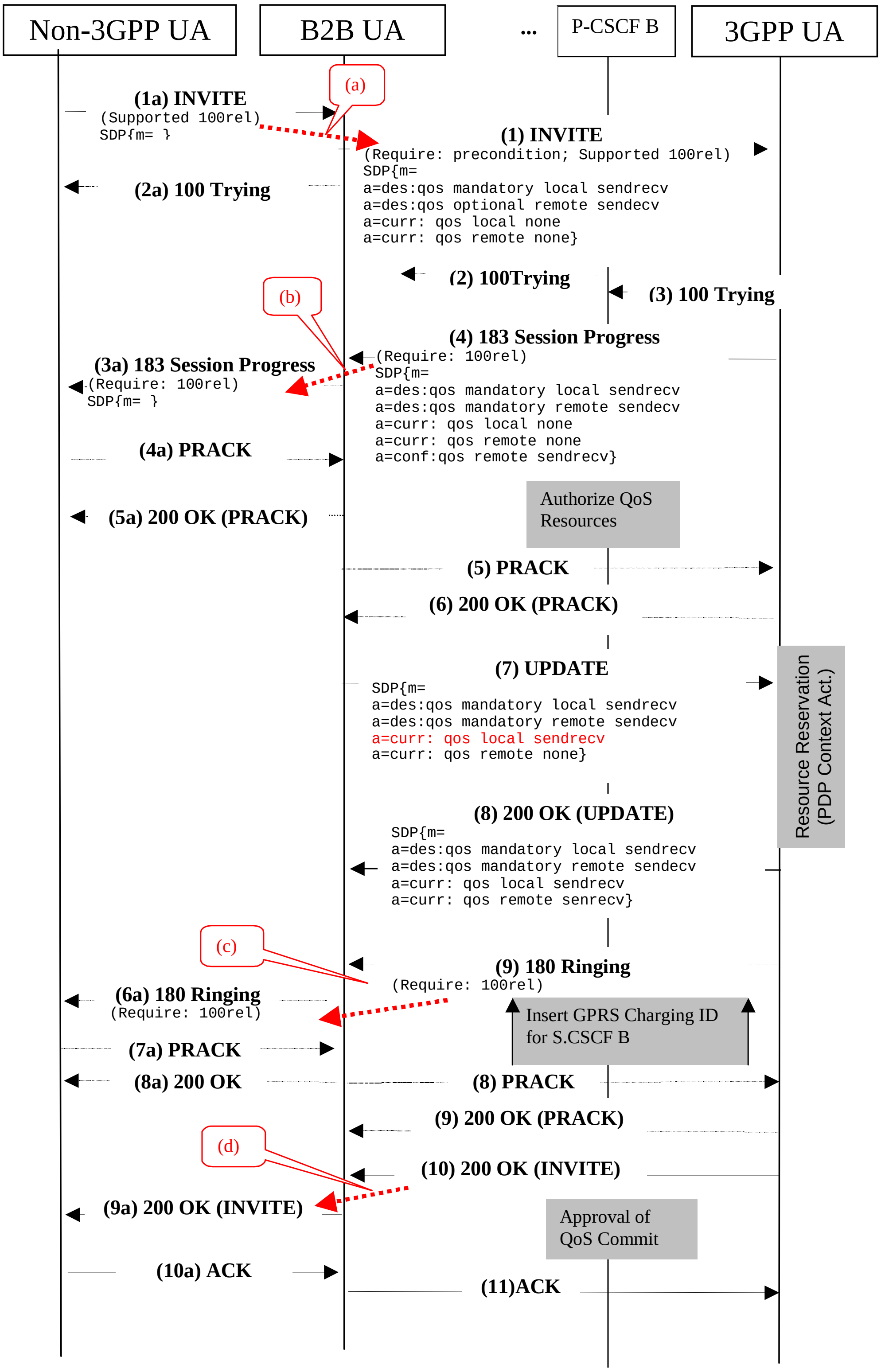


Figure 5.2.2/2: Functionality of B2BUA connecting an originating non-3GPP SIP UA not making use of the SIP preconditions extension and the SIP update extension, to a terminating 3GPP UA.

Implications of the above solution are discussed in Clause 6.1.

5.2.3 Proposed Resolution Modified end-to-end call flow

Currently, 3GPP TS 24.229 [1] mandates the usage of the SIP preconditions extension for incoming session attempts to a 3GPP UA. The usage of preconditions is intended to provide extra capabilities to the originating side of the session, but it is not really required at the terminating side. The restriction in 3GPP TS 24.229 [1] to support preconditions at the terminating side does not have any effect on terminating session attempts. Therefore, it is proposed to remove such restriction.

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

The restriction to mandate the usage of the SIP preconditions extension for terminating session attempts is removed from 3GPP TS 24.229 [1].

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

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

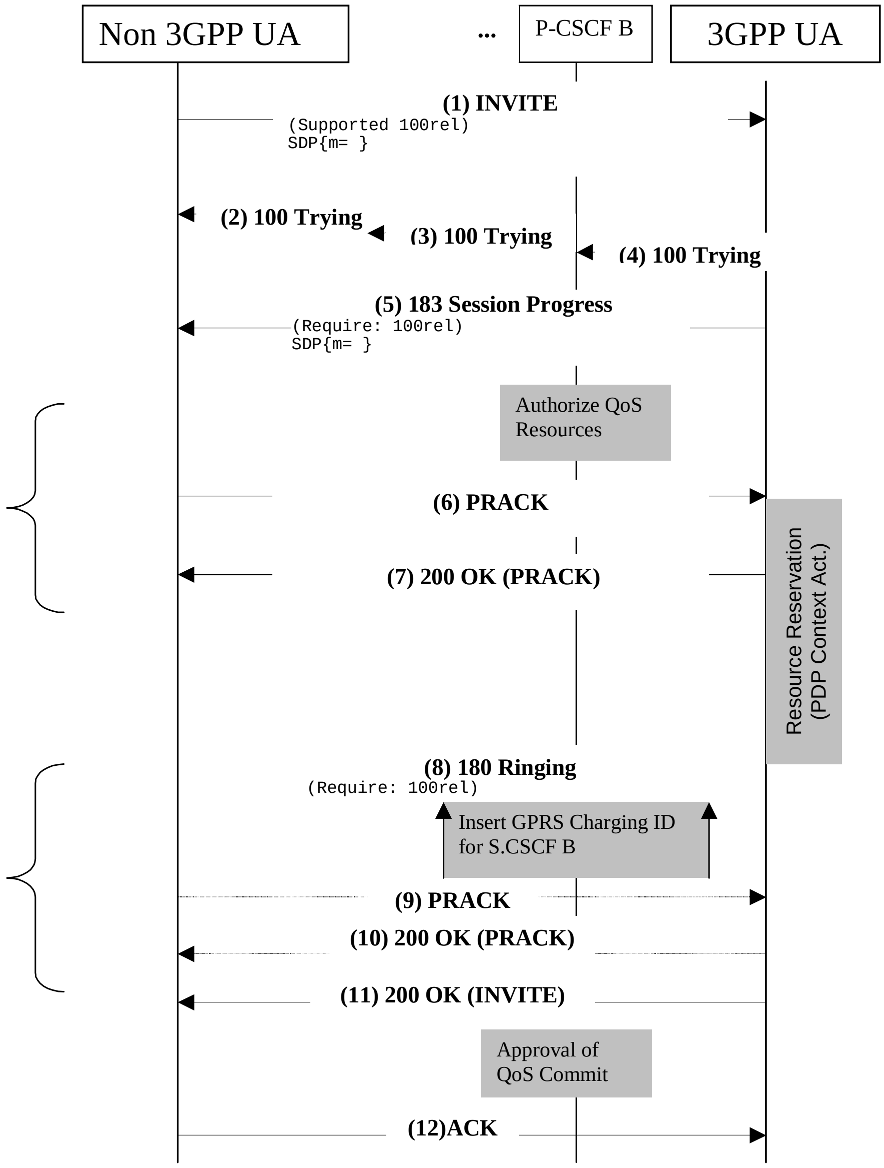


Figure 5.2.3/1: Modified end-to-end call flow for non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension, to 3GPP UA. SDP offer in INVITE request.

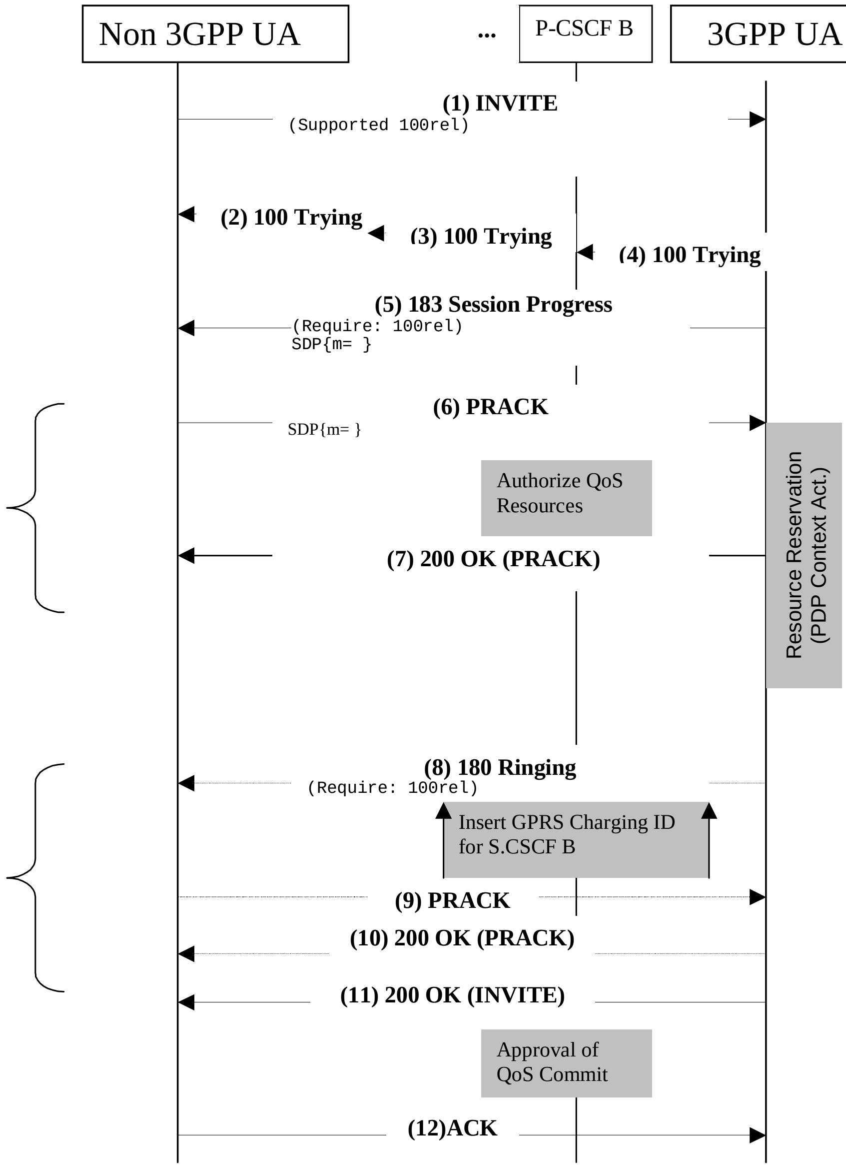


Figure 5.2.3/2: Modified end-to-end call flow for non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension, to 3GPP UA. No SDP offer in INVITE request.

<u>Impacts of the above solution</u>

No modifications or extra functionality compared to Release 5 required.

No disadvantages have been identified.

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

5.3.1 Description of interworking issue

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

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

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

5.3.2 Proposed Resolution B2BUA

A B2BUA is used.

How the B2BUA is inserted is discussed within Clause 5.2.2.

Functionality of B2BUA

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

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

Impacts of the above solution.

General impacts of the B2BUA are discussed in Clause 6.1.

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

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

5.3.3 Proposed Resolution Modified end-to-end call flow

The restriction in 3GPP TS 24.229 [1] to mandate the usage of preconditions at the terminating side does not have any effect on terminating session attempts. Therefore, it is proposed to remove this restriction, as detailed in Clause 5.2.3.

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

Impacts of the above solution

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

No disadvantages have been identified.

6 Implications of the Proposed Solutions

6.1 B2BUA

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

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

Additional processing load and additional delay may result.

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

The B2BUA is automatically in the signalling path for all communications. For a session setup from a calling non-3GPP UA towards called 3GPP UA, the B2BUA may be activated unnecessarily, if the non-3GPP UA supports the 100rel extension, but fails to indicate this in the INVITE request. RFC 3262 [6] recommends that a UAC supporting the 100rel extension indicates this capability in the INVITE request, but does not mandate the UAC to do so. The B2BUA may also be activated unnecessarily, if the non-3GPP UA supports the precondition extension, but fails to indicate this in the INVITE request.

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

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

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

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

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

6.2 Modified end-to-end call flow

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

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

Changes have to be performed in various network entities.

Annex A: Interworking topic template

A.1 Description of interworking issue

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

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

A.2 Proposed Resolution *yy*

Description

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

Implications of the above solution

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

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

Annex B: Mechanisms allowing optional Additions within SIP

Excerpts from RFC 3261

8.1 UAC Behavior

…

8.1.1.9 Supported and Require. If the UAC supports extensions to SIP that can be applied by the server to the response, the UAC SHOULD include a Supported header field in the request listing the option tags (Clause 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.

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 (Clause 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 Clause 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 Clause 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 Clause 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 Clause 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 Clause 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 Clause 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 (Clause 20.29) or Require (Clause 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 Clause 8.1.3.5.

21.4.16 421 Extension Required

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

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

According to 3GPP TS 24.229 [1], a 3GPP UA shall use the SIP 100rel, update, and precondition extensions. The 3GPP UA shall abort the call set-up in manners detailed in the main part of this TR, if the non-3GPP UA is not making use of these extensions.

This annex aims to explain why 3GPP TS 24.229 [1] introduces these restrictions. It details the consequences, if a 3GPP UA would not behave according to 3GPP TS 24.229 [1] and would not apply some or all of the above SIP extensions.

C.1 Impacts of session setup call flows from calling 3GPP UA

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

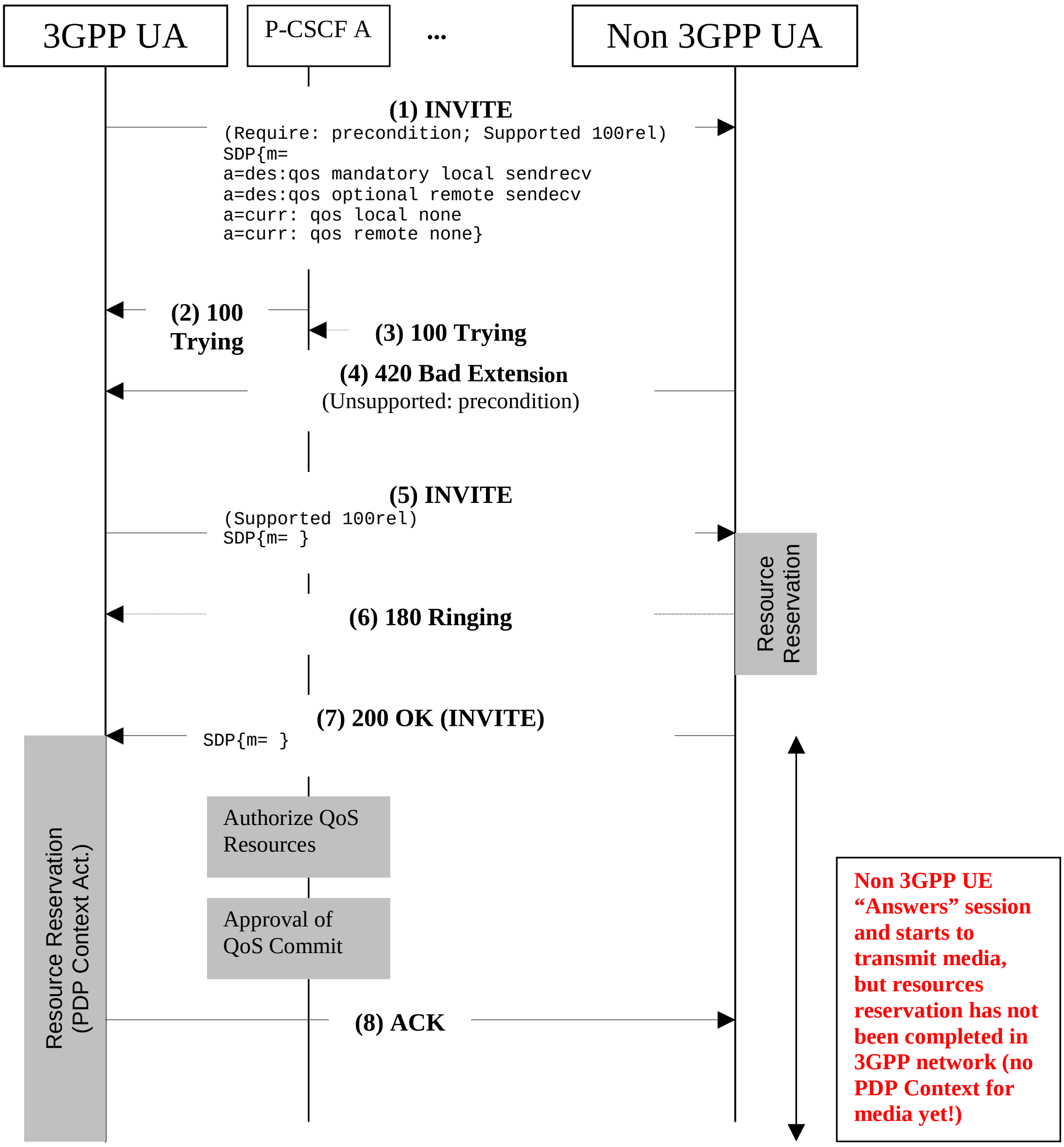
C.1.1.1 Description of interworking issue

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

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

According to RFC3261 [4], Clause 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.



(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.

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

C.1.1.2 Impacts of Identified interworking issue

User interaction

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

Charging and Billing Implications

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

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

SIP Media authorisation

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

SIP Media allocation

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

Fraudulent and security risks

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

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

C.1.2.1 Description of interworking issue

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

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

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

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

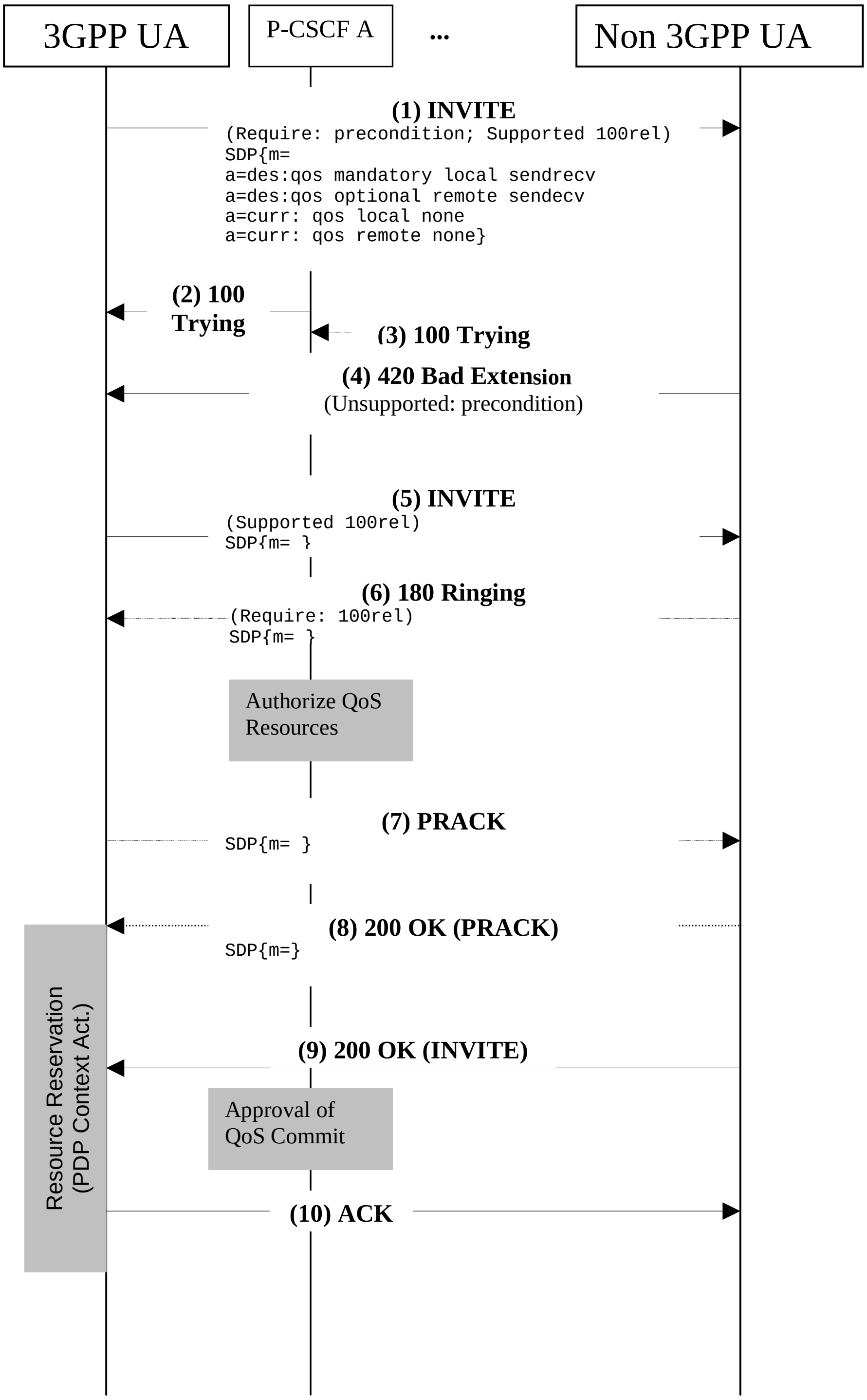


Figure C.1.2.1/1: Session Setup towards non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension

C.1.2.2 Impacts of identified interworking issue

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

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

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

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

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

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

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

C.2 Impacts of session setup towards called 3GPP UA

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

C.2.1.1 Description of interworking issue

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

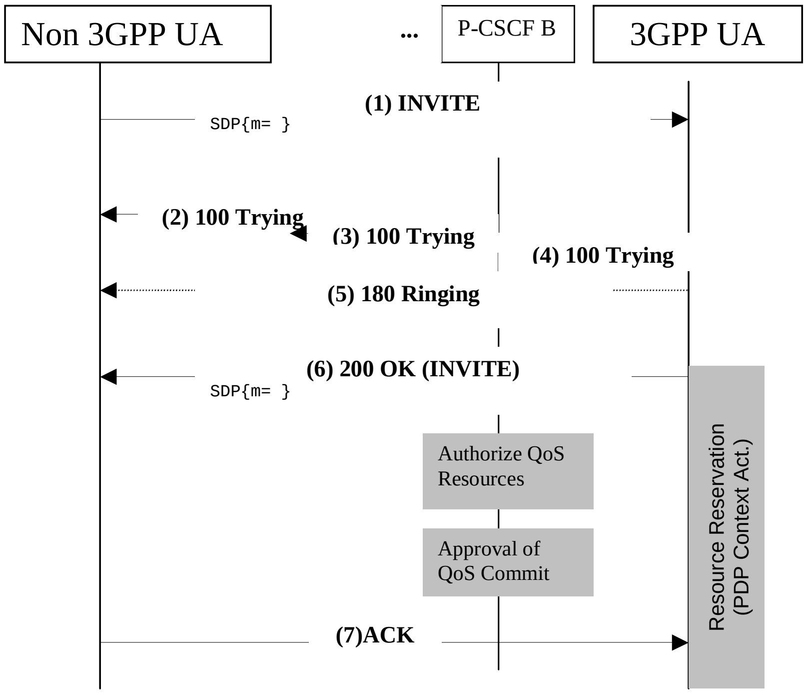


Figure C.2.1.1/1: Non-3GPP SIP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension. SDP offer in INVITE request.

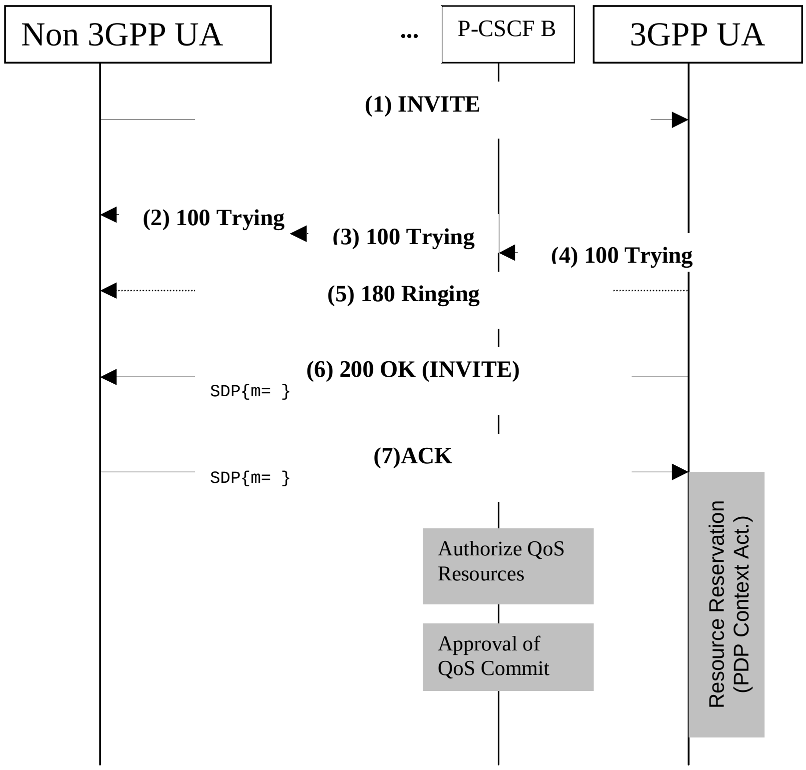


Figure C.2.1.1/2: Non-3GPP SIP UA not making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension. SDP offer in OK response.

C.2.1.2 Impacts of identified interworking issue

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

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

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

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

C.2.2.1 Description of interworking issue

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

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

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

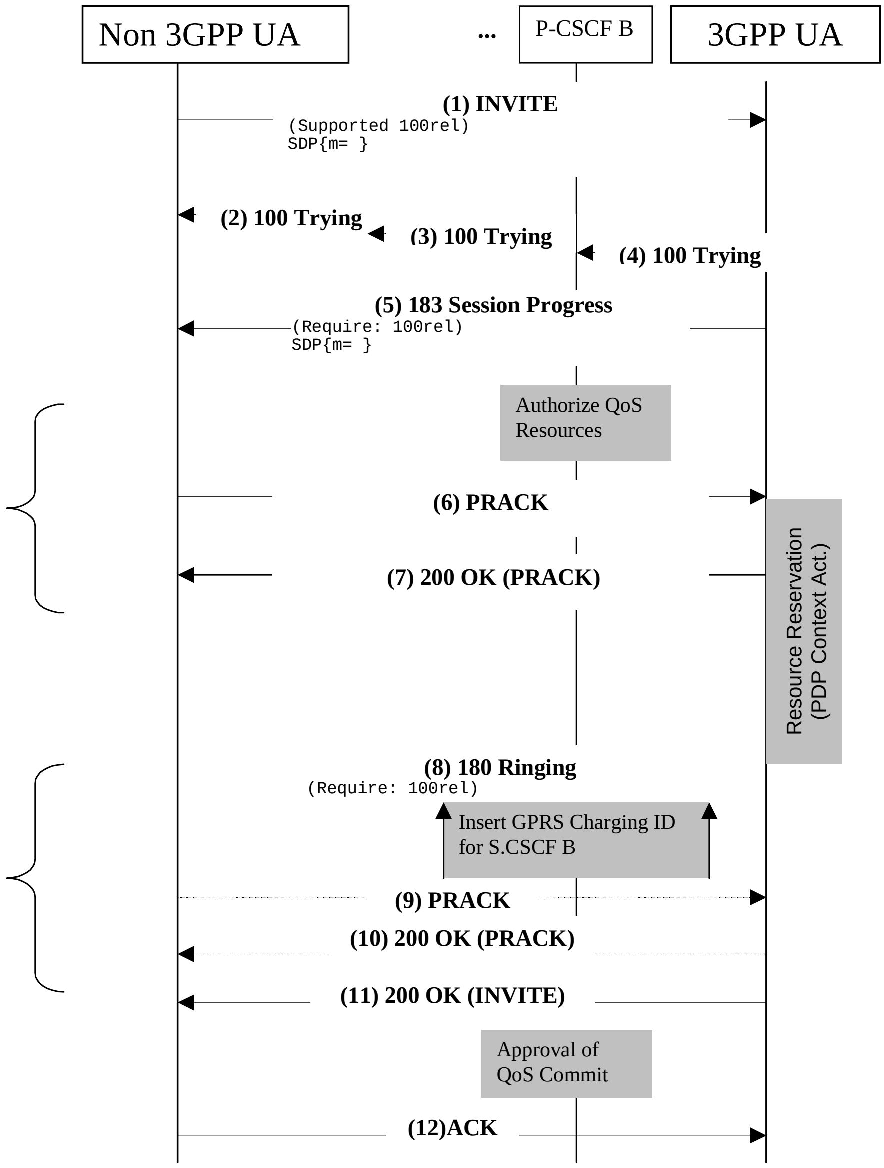


Figure C.2.2.1/1: Non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in Invite.

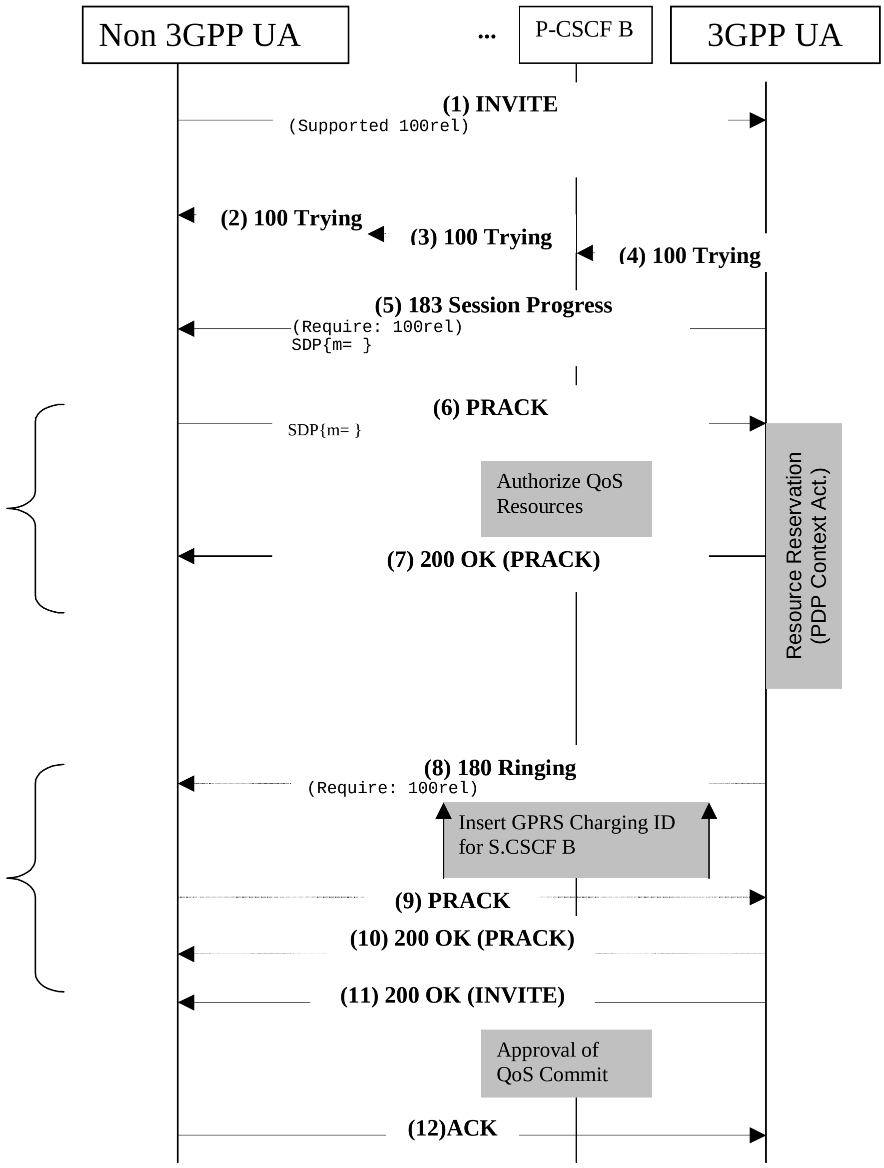


Figure C.2.2.1/2: Non-3GPP UA not making use of the SIP preconditions extension and the SIP update extension. No SDP offer in Invite.

C.2.2.2 Impacts of identified interworking issue

No negative impacts have been identified.

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

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

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

Annex D: Reference call flow from 3GPP UA to 3GPP UA

The interworking between an originating 3GPP UA and a terminating 3GPP UA is as defined in 3GPP TS 24.229 [1]. No interworking issues exist, but the flow diagram is depicted here for comparison.

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

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

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

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

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

NOTE 6: The P-CSCF inspects each SDP, in order to identify offer/answer pairs (RFC 3264 [8]). The P-CSCF may modify the QoS authorisation (8,9) when processing each SDP answer.

NOTE 7: The use of the183 (Session Progress) (7) provisional response is optional according to IETF specifications. However, if the SIP precondition extension [6] is used and SDP with mandatory preconditions that the terminating UA is not capable of meeting unilaterally is included in the initial INVITE request (1), a 101-199 provisional response, such as the 183 (Session Progress) response, is required to transport the SDP answer including the mandated "confirmation status" SDP attribute RFC 3262[6], Clause 6). Moreover, the 180 (Ringing) response is not suitable because the user should not be alerted until the preconditions are met.

NOTE 8: It is optional to convey a new SDP offer/answer within the PRACK request (11) and 200 (OK) for a PRACK request) (12). An originating 3GPP UA will refrain from generating a new SDP offer within PRACK request (11), if it does not wish to further restrict the set of codecs selected within the first offer/answer pair.

NOTE 9: According to RFC 3262[6], Clause 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 RFC 3312[5], RFC 3311[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 (RFC 3312[5], Clause 7). The only suitable way to convey the new SDP offer/answer may be via an UPDATE request.

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

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

NOTE 13: The UPDATE request (14)is used to convey the GPRS charging ID from P-CSCF A to S-CSCF-A. 3GPP TS 24.229 [1]

NOTE 14: According to 3GPP 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.

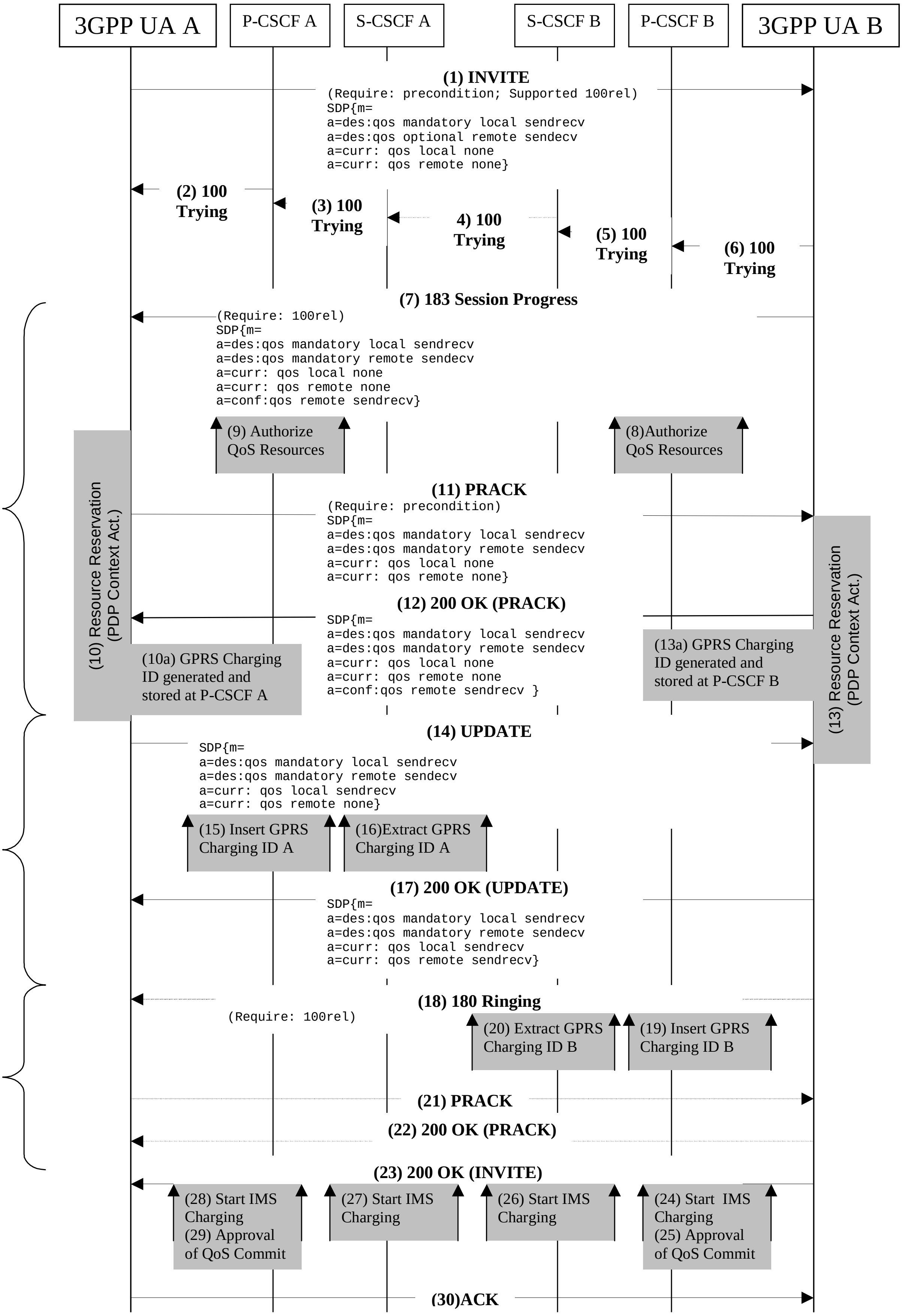


Figure D/1: 3GPP UA to 3GPP UA Call flow

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

Annex E: Scenarios without identified interworking issues

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

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

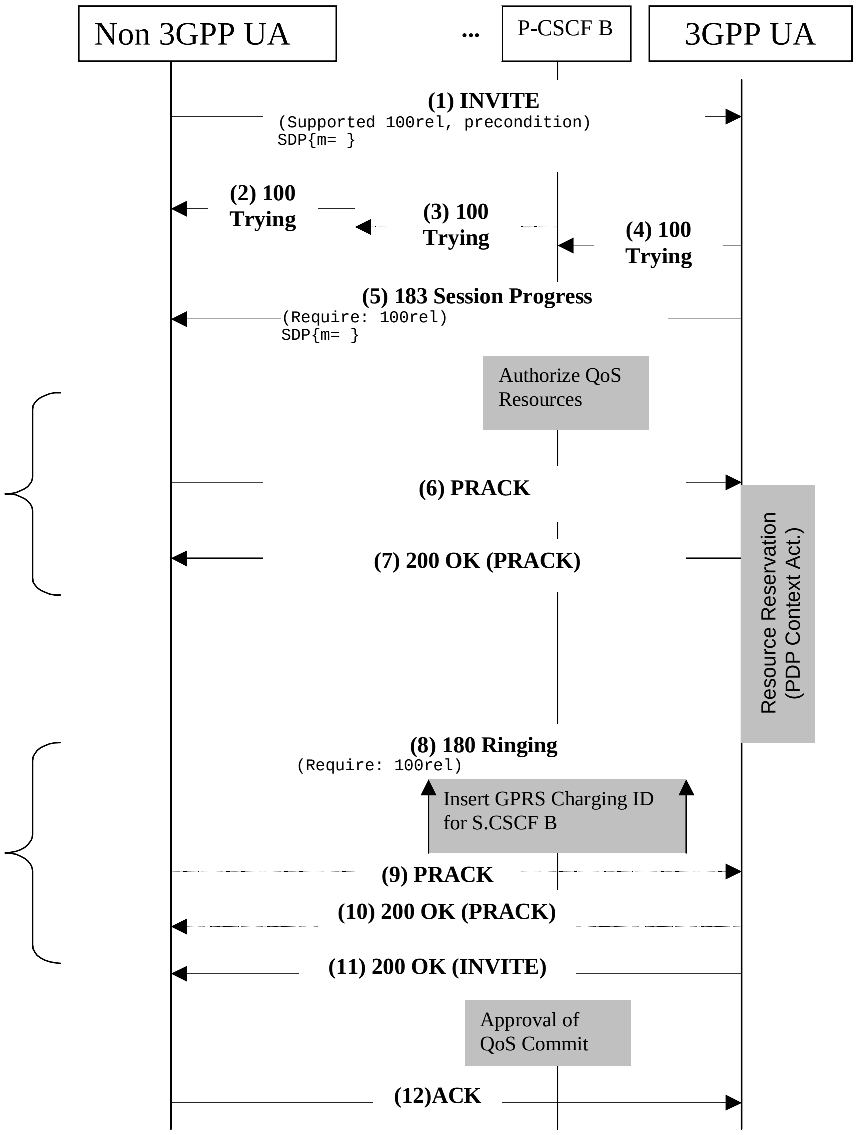


Figure E.1/1: Non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, and not performing QoS reservation, to 3GPP UA

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

According to 3GPP TS 24.229 [1], clause 5.1.4.1, the called 3GPP UA must send a provisional response (otherwise it can not complete the resource reservation before sending the 200 (OK) response for an INVITE request) and require the 100rel extension within this message. According to RFC 3261, clause 13.2.1, "the initial offer MUST be in either an INVITE or, if not there, in the first reliable non-failure message from the UAS back to the UAC".

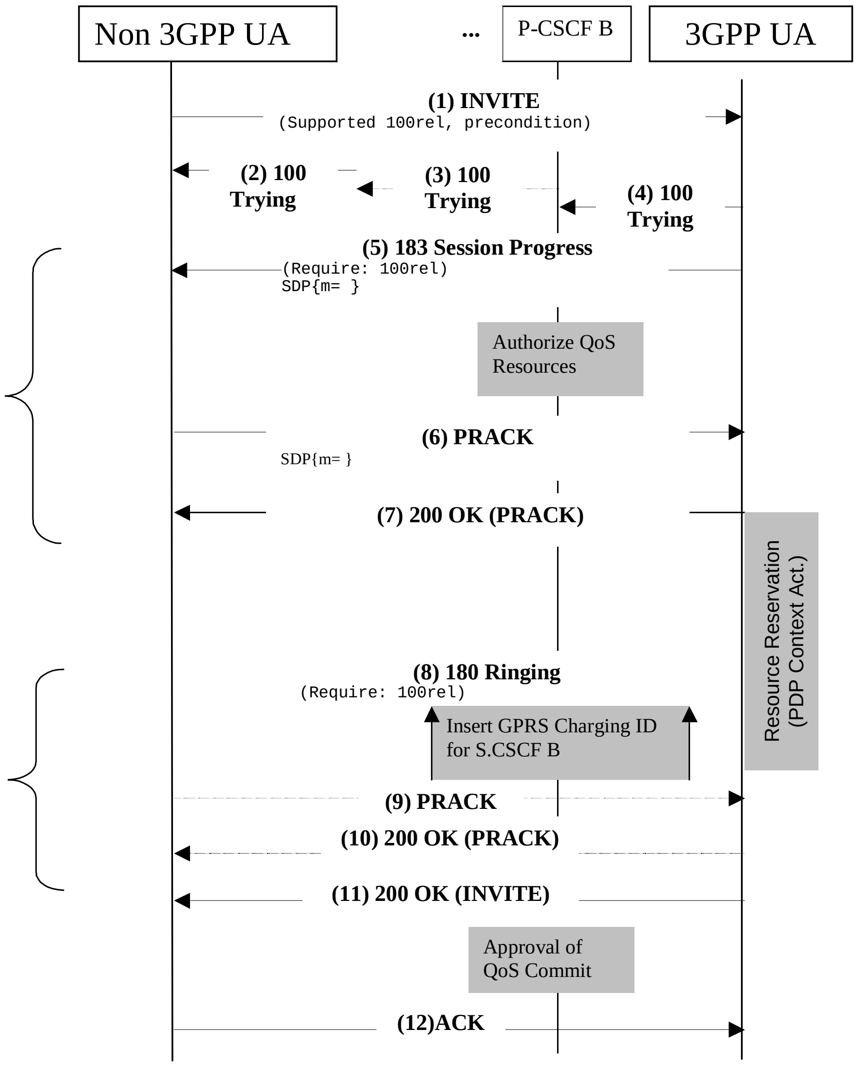


Figure E.2/1: Non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not including the SDP offer in the initial INVITE request, to 3GPP UA

Annex F:
Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2002-01	CN3#21				Creation of document	-	0.1.0
2002-07	CN3#24				Include suggestions for B2B UA	0.1.0	0.2.0
2002-11	CN3#26				Output of drafting group included, presented to CN#18 for information	0.2.0	1.0.0
2002-12	NP-18	NP-020610			Presented to Plenary NP#18 for information	1.0.0	
2002-02	CN3#27	N3-030152 N3-030153 N3-030154 N3-030156 N3-030157			Agreed changes are included.	1.0.0	1.1.0
2003-03	NP-19				Presented to Plenary NP#19 for information	1.1.0	
2003-05	CN1#29	N1-030487 N1-030535			Review by CN1. Proposed Changes and new version of TR require endorsement by CN3	1.1.0	1.2.0
2003-05	CN3#29	N3-030454 N3-030456 N3-030458 N3-030463			CN3 endorsed version 1.2.0. Agreed changes based on this version are included. CN3 agreed to send TR to CN plenary for approval	1.2.0	2.0.0
2003-06	NP#20	NP-030293			Approved at NP#20 and placed under change control	2.0.0	6.0.0
2003-09	NP#21	NP-030343	001		Editorial Corrections	6.0.0	6.1.0

History

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V6.1.0	September 2003	Publication