3GPP TSG CN Plenary Meeting #20 **NP-030265**
4th - 6th June 2003. HÄMEENLINNA, Finland.

Source: **MCC**

Title: **All LSs sent from CN1 since TSG CN#19 meeting,- pack 1**

Agenda item: **6.1.1**

Document for: **INFORMATION**

Introduction:

This document contains **9 agreed** LSs sent from **TSG CN WG1#29**, and are forwarded to TSG CN Plenary meeting #20 for information only.

TDoc #	Status	Source	Tdoc Title	Comments
N1-030509	AGREED	Peter/Siemens	LS on Handling of DTMF	Linked to 317 and 394. To: CN3, Cc:,
N1-030528	AGREED	Peter/Siemens	LS CN1 review of SIP interworking TR29.962	Linked to 487 and 527. To: CN3, Cc:,
N1-030543	AGREED	Robert/Siemens	Reply LS on Early UE handling	Linked to 475. Revised from 484. To: SA2, Cc: RAN3
N1-030547	AGREED	Miguel/Ericsson	Reply LS on Radio Access Bearer for PS conversational testing	Linked to 336. Revised from 478. To : SA4, Cc: RAN2, GERAN2
N1-030548	AGREED	Atle/Ericsson	LS on IPv6 DNS server discovery in release 99 and release 4	Related to 426. To: SA2, CN3, Cc: CN. Revised from 488.
N1-030549	AGREED	Atle/Ericsson	LS on change of IP address due to privacy	Related to 427. To: SA2, Cc: . Revised from 504.
N1-030560	AGREED	Keith/Lucent	LS on duration of ICID at IMS registration	Related to 397. To: SA5, Cc: SA2. Revised from 498
N1-030566	AGREED	Georg/Nokia	LS on Protocols over the Mt interface	Linked to 327. To: SA2, Cc: CN, SA1. Revised from 477.
N1-030567	AGREED	Inma/Nokia	LS on DRX parameters update	Linked to 341. To: SA2, Cc: GERAN1, GERAN2, RAN1, RAN2, RAN3, T2. Revised from 479

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030509***
Sophia-Antipolis, France, 31 March – 04 April 2003

Title: **LS on Handling of DTMF**
Response to: **N3-030184 = N1-030317on Handling of DTMF in IMS from WG CN3**
Release: **Rel-6**
Work Item:

Source: **CN1**
To: **CN3**
Cc: -

Contact Person:
Name: Peter Leis
Tel. Number: +49 89 636 75208
E-mail Address: peter.leis@siemens.com

Attachments: None

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1. Overall Description:

CN1 thanks CN3 for their liaison statement on Handling of DTMF in the IMS. CN1 has discussed the issue and provides the following answer:

24.229 will be updated so that for DTMF the MIME type will be limited to the subtype "telephone-event" according to RFC2833.

For DTMF orginating in the IMS CN3 suggested that it is desirable, that this is indicated with an appropriate SDP offer for telephone-event immediately before the DTMF is being sent (using UPDATE or Re-INVITE). CN1 sees no need for this change as detecting DTMF encoded in the RTP does not need any hardware resource. In addition it is seen that such an indication would be a misuse of SIP for media control.

CN1 agrees that support of DTMF terminating in the IMS is not required at the IM-MGW.

2. Actions:

To CN3 group.

ACTION: CN1 kindly asks CN3 to take the above given information into account in their specifications.

3. Date of Next TSG-CN1 Meetings:

CN1_30	19th – 23rd May 2003	San Diego, USA
CN1_31	25th – 29th August 2003	Sophia-Antipolis, France

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030528***
Sophia-Antipolis, France, 31 March – 04 April 2003

Title: LS CN1 review of SIP interworking TR29.962
Response to:
Release: Release-6
Work Item:

Source: CN1
To: CN3
Cc:

Contact Person:
Name: Peter Leis
Tel. Number: +49 89 636 75208
E-mail Address: peter.leis@siemens.com

Attachments: N1-030535.zip, N1-030487.zip

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1. Overall Description:

CN1 would like to inform that during CN1#29 the SIP interworking TR 29.962 was reviewed. After discussing the item CN1 came to the following conclusions:

- CN1 provides agreed proposals N1-030535 and N1-030487 against the TR.
- CRs against the TR will still be allowed in the next CN WG meetings in May 2003. The delegates are requested to submit all such CRs, if any, to CN3 meeting.
- It will be up to CN3 to decide whether to send the TR to TSG-CN #20 for approval.
- The rapporteur of 29.962 was requested to distribute a new version of the TR, with all CN1 proposed changes included, both for CN1 and CN3 to review (via email exploder).

2. Actions:

To CN3

ACTION:

CN1 kindly asks CN3 to approve the attached proposals on the TR 29.962 since CN3 has the maintenance responsibility of TR 29.962.

3. Date of Next TSG-CN1 Meetings:

CN1_30	19th – 23rd May 2003	San Diego, USA
CN1_31	25th – 29th August 2003	Sophia-Antipolis, France

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030487***
Sophia-Antipolis, France, 31 March – 04 April 2003

Source:	Ericsson, Nokia
Title:	CR 29.962: General revision
Agenda item:	7.9
Document for:	Approval

Introduction

This document reviews the TR 29.962 and proposes changes, clarifications, and additions and deletions.

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)

Select keywords from list provided in specs database.

Keywords

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Contents

Foreword

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

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

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

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

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

1 Scope

The present document investigates the SIP signalling interworking between 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~~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]

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

As the session attempt requires the support of the preconditions extension, and as the non-3GPP UA does not support such extension, ~~T~~the ~~call~~ session attempt 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~~ indicate the called party that the calling party has setup the bearers, and therefore, the terminating party can alert the user.

~~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~~B2BUA

A ~~B2B UA~~B2BUA is used.

4.1.1.2.1.1 Insertion of ~~B2B UA~~B2BUA

How the ~~B2B UA~~B2BUA is inserted is discussed within Section 4.1.2.4.1.1.

4.1.1.2.1.2 Functionality of ~~B2B UA~~B2BUA

4.1.1.2.1.2.1 Description

The functionality of the ~~B2B UA~~B2BUA is as discussed in Section 4.1.2.4.1.2.1.

The ~~B2B UA~~B2BUA shall ~~pass~~ forward additional UPDATE ~~messages~~requests, which are not related to the precondition extension, and related provisional acknowledgement (PRACK) ~~messages~~requests.

4.1.1.2.1.2.2 Advantages

General advantages of the ~~B2B UA~~B2BUA 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.~~

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.

4.1.1.2.1.2.3 Disadvantages

General disadvantages of the ~~B2B UA~~B2BUA 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

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 also sets the media streams in "inactive" to avoid receive media at this time. The session progresses, and eventually is setup. When the 3GPP UA gets all the bearers setup, it re-INVITEs the calling party removing the "inactive" media. This indicates the non-3GPP UA that the 3GPP UA is ready to receive media.

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~~ SIP UPDATE ~~messages~~requests.

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

4.1.2.2 Flow diagram

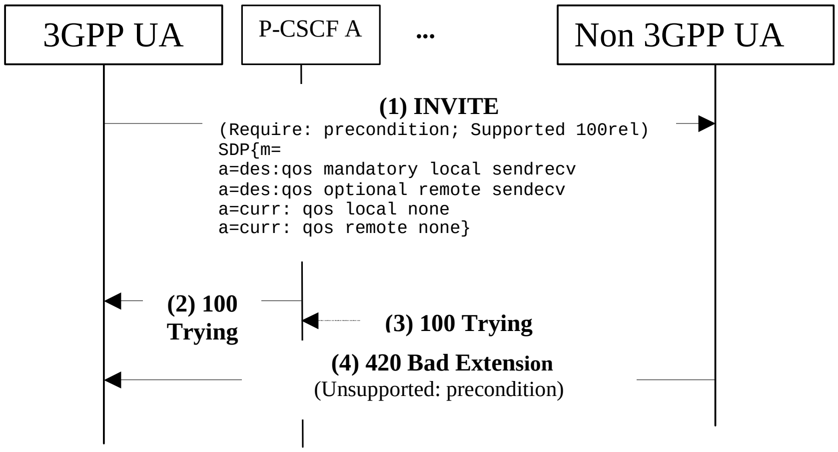

Figure 4.1.2.2/1: 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension.

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~~B2BUA

4.1.2.4.1.1 Insertion of ~~B2B UA~~B2BUA

4.1.2.4.1.1.1 Static Insertion of ~~B2B UA~~B2BUA

4.1.2.4.1.1.1.1 Description

A ~~B2B UA~~B2BUA is permanently inserted at connections between the IMS and a given external network. This ~~B2B UA~~B2BUA handles all ~~calls~~SIP signalling, including session attemps, subscriptions, instant messaging, etc, including ~~calls~~ signalling where the ~~call~~ flows may ~~be pass~~forward~~ed~~ without ~~modification~~B2BUA intervention.

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

The ~~B2B UA~~B2BUA becomes active only when receiving a 420 (Bad Extension) ~~(Unsupported precondition)~~ response with the "Unsupported" header field including the "preconditions" tag from the ~~Non~~non-3GPP UA, as depicted in Figure 4.1.2.4.1.1.1.1/1. Note however that this behaviour is a hack to the protocol SIP, because an entity should decide its behaviour (proxy or UA) prior to forwarding any request or generating any response. Among other things, population and interpretation of certain headers (such a Contact, Proxy-Require, Require, etc.) will depend on the behaviour of the entity. Therefore, it is not possible for a entity to start behaving as a SIP proxy, and upon the reception of a response, change its behaviour to a B2BUA.~~Otherwise, the B2B UA passes all SIP messages received at one side to the other side.~~

The ~~B2B UA~~B2BUA shall store the SDP offer in initial ~~invites~~ INVITE requests for all calls until receiving a provisional or final response from the Non-3GPP UA.

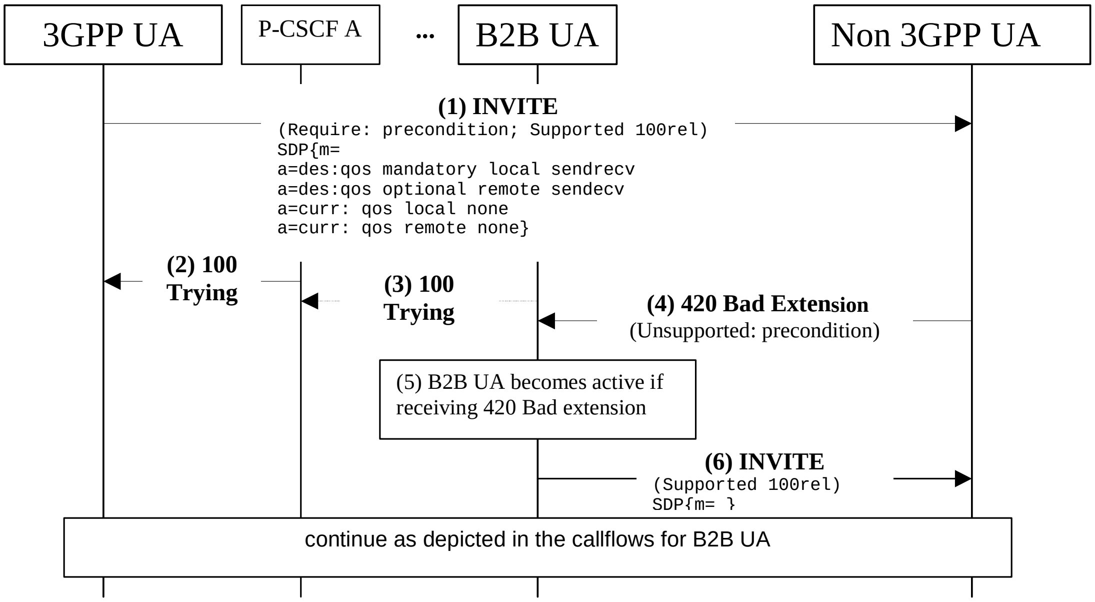

Figure 4.1.2.4.1.1.1.1/1: Activation of static ~~B2B UA~~B2BUA connecting 3GPP UA to non-3GPP UA not supporting the SIP preconditions extension

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~~session attempts, subscriptions, and instant messaging.

Trying to specify the behaviour of a B2BUA in a deterministic way seems to be complicated.

4.1.2.4.1.2 Functionality of ~~B2B UA~~B2BUA

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~~B2BUA shall apply the following rules:

1. The ~~B2B UA~~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 ~~B2B UA~~B2BUA shall also comply to the SIP 100rel and update extensions.

3. On the IMS side, the ~~B2B UA~~B2BUA shall also comply to and apply the SIP 100rel, update and precondition extensions.

4. The ~~B2B UA~~B2BUA shall ~~pass~~ forward SIP requests arriving at one call leg to the call leg at the other side, with the exceptions below.

5. The ~~B2B UA~~B2BUA shall ~~pass~~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 ~~B2B UA~~B2BUA shall ~~pass~~forward SIP final responses arriving at one call leg to the call leg at the other side, with the exceptions below.

7. The ~~B2B UA~~B2BUA shall not require the SIP preconditions extension on the non-IMS side.

8. The ~~B2B UA~~B2BUA shall insert preconditions in SDP sent from non-3GPP UA to 3GPP UA.

9. The ~~B2B UA~~B2BUA should remove preconditions in SDP sent from 3GPP UA to non-3GPP UA

10. The ~~B2B UA~~B2BUA shall not ~~pass~~forward PRACK requests and 200 (OK) responses for the PRACK request~~(PRACK) messages~~.

11. The ~~B2B UA~~B2BUA shall delay forwarding a 200 (OK) response for an INVITE request ~~(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~~B2BUA 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~~B2BUA receives a subsequent SDP offers on the IMS side with modified media, it shall suspend the transaction on the IMS side and ~~pass~~forward this SDP offer to a re-~~invite~~ INVITE ~~transaction~~ request on the non-IMS Side. The ~~B2B UA~~B2BUA shall ~~pass~~forward the SDP answer received in the re-~~invite~~ INVITE ~~transaction~~ 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 ~~B2B UA~~B2BUA shall ~~pass~~ forward an SDP answer within the 200 (OK) response for the INVITE request ~~(INVITE) message~~ of the original INVITE ~~transaction~~ request from the non-IMS side to a provisional response on the IMS side.

15. For a re-~~Invite~~ INVITE request from the Non-IMS side to the IMS side, the ~~B2B UA~~B2BUA shall apply the rules in Section 4.2.2.4.1.2.1.

The ~~B2B UA~~B2BUA relies ~~messages~~ requests and responses as indicated by the red dotted arrows in the figures below.

The ~~called~~ terminating UA may also send no 183 ~~“~~(Session progress~~”~~) ~~message~~ response and include the SDP answer in the ~~“~~200 (OK) response for the INVITE request ~~(INVITE)”~~ instead. This case is discussed in Section 4.1.3.4.1.2.

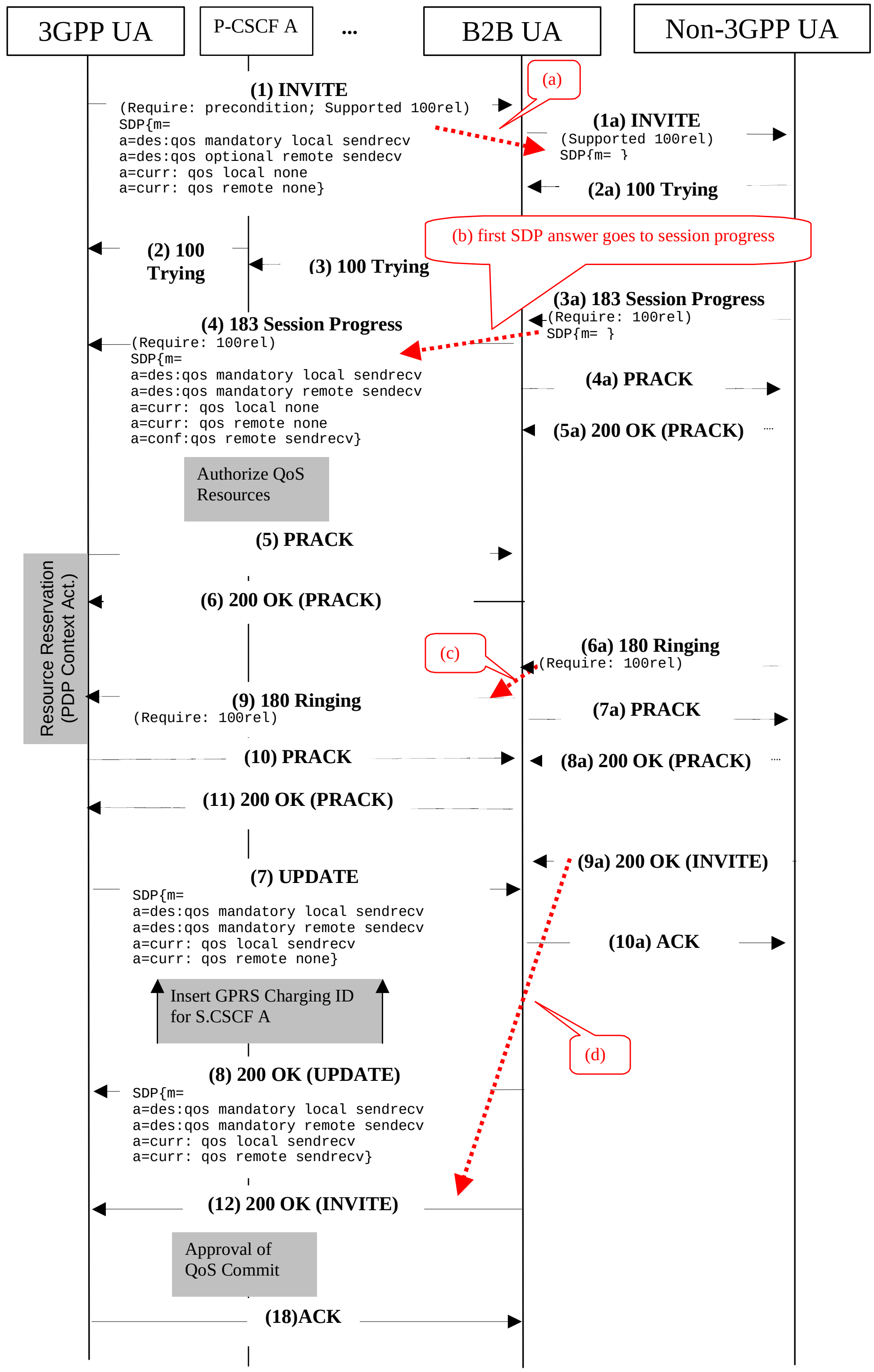

Figure 4.1.2.4.1.2.1/1: Functionality of ~~B2B UA~~B2BUA connecting 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. ~~Calling~~ The originating UA includes SDP answer in 183 “Session Progress”. ~~Calling~~ The terminating UA sends no second SDP offer.

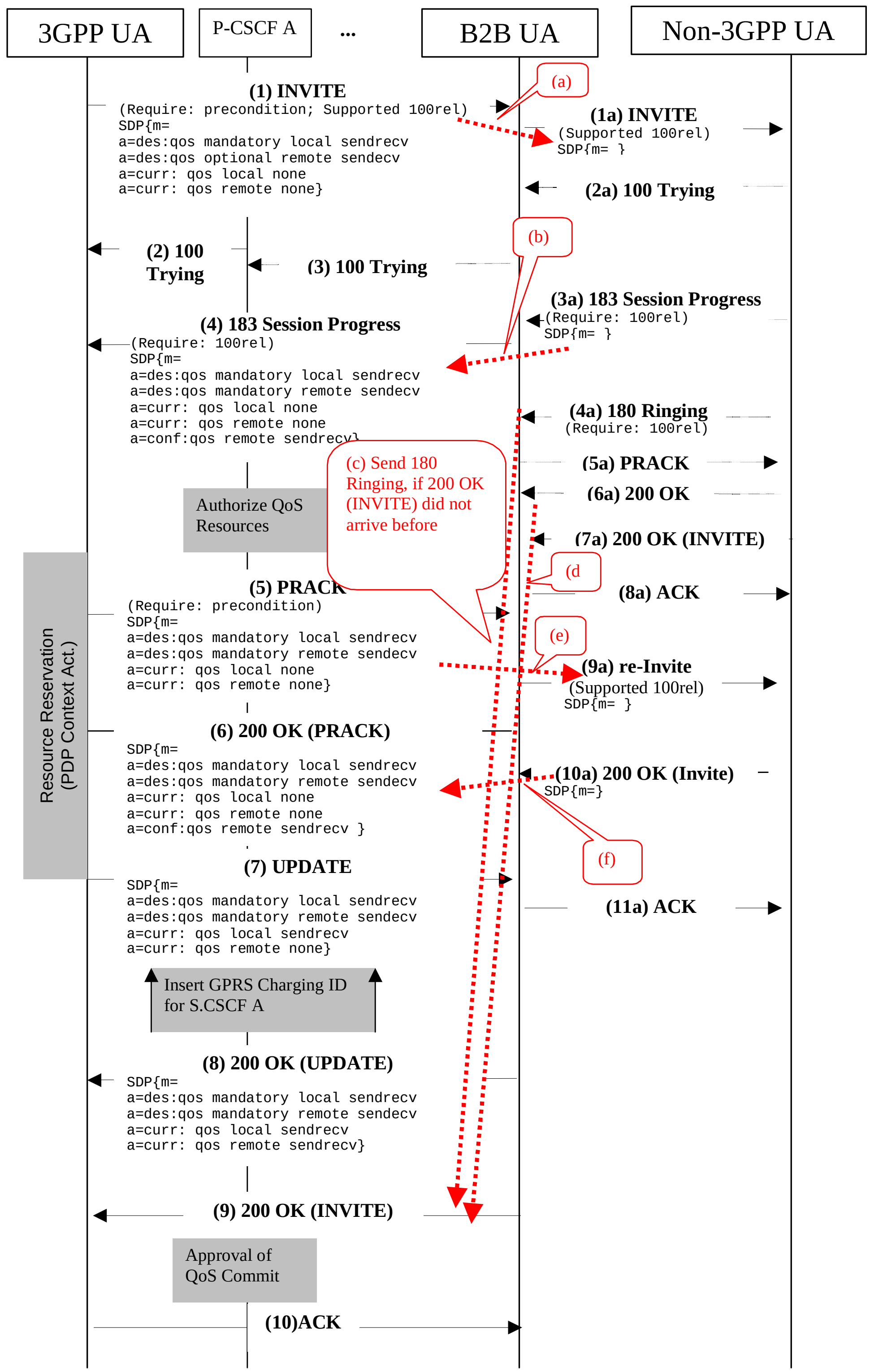

Figure 4.1.2.4.1.2.1/2: Functionality of ~~B2B UA~~B2BUA connecting 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. ~~Called~~ The terminating UA includes SDP answer in 183 "Session Progress". ~~Calling~~ The originating UA sends second SDP offer.

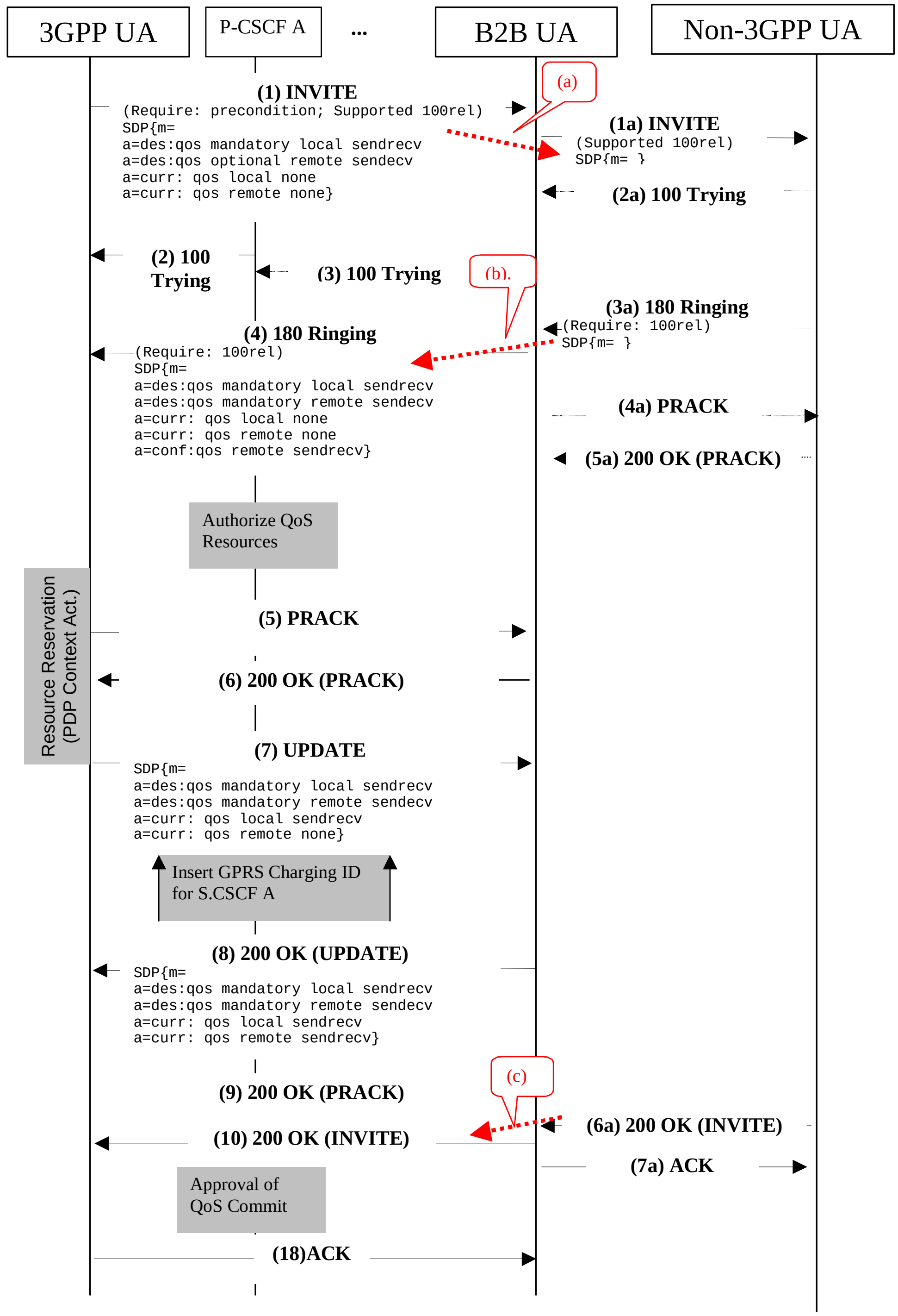

Figure 4.1.2.4.1.2.1/3: Functionality of ~~B2B UA~~B2BUA connecting 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. ~~Called~~ The terminating UA includes SDP answer in 180 “Ringing”. ~~Calling~~ The originating UA sends no second SDP offer.

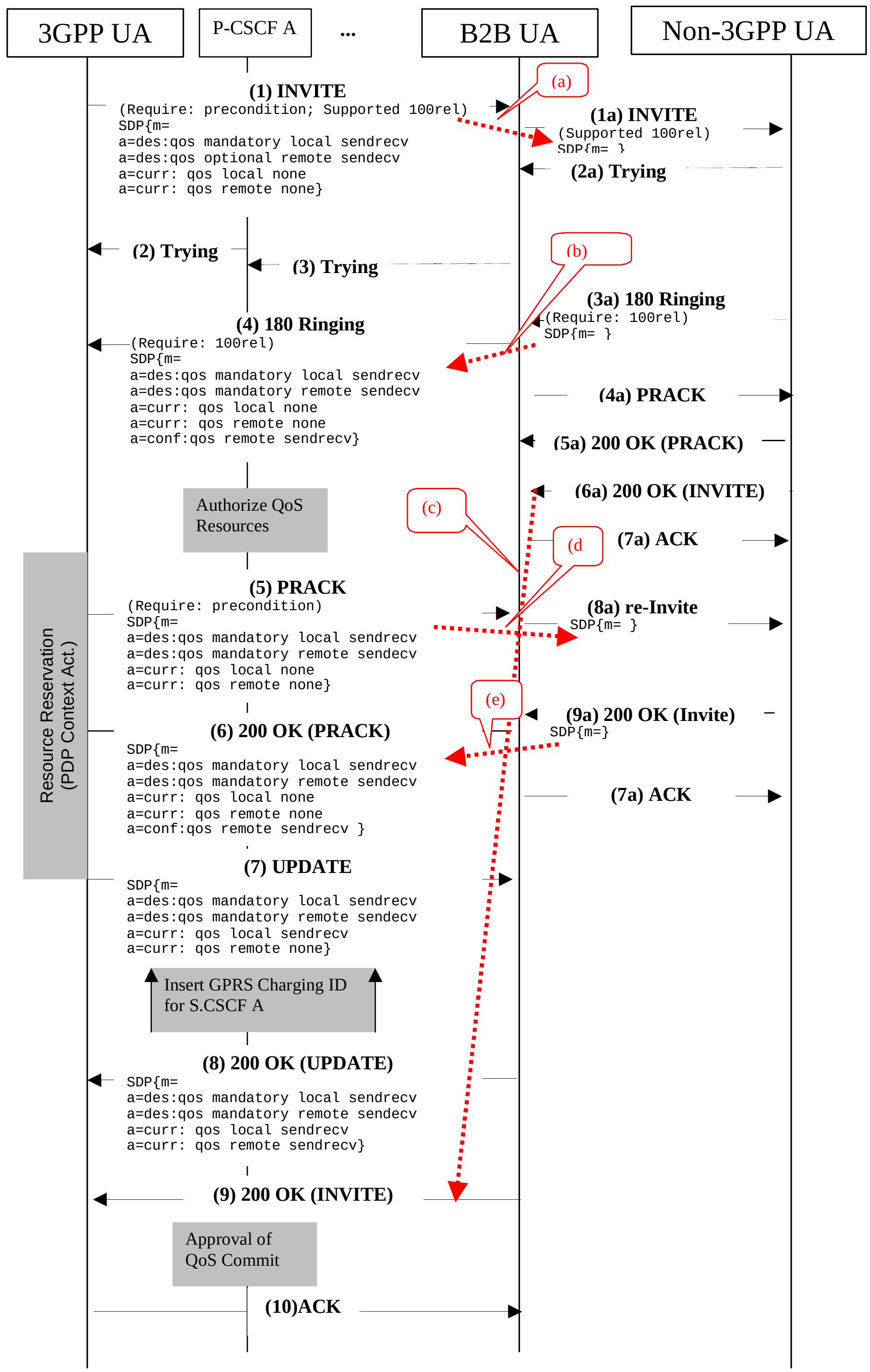

Figure 4.1.2.4.1.2.1/4: Functionality of ~~B2B UA~~B2BUA connecting ~~calling~~ an originating 3GPP UA to ~~called~~ terminating non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. ~~Called~~ The terminating UA includes SDP answer in 180 "Ringing". ~~Calling~~ The terminating 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~~B2BUA 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~~B2BUA. 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.

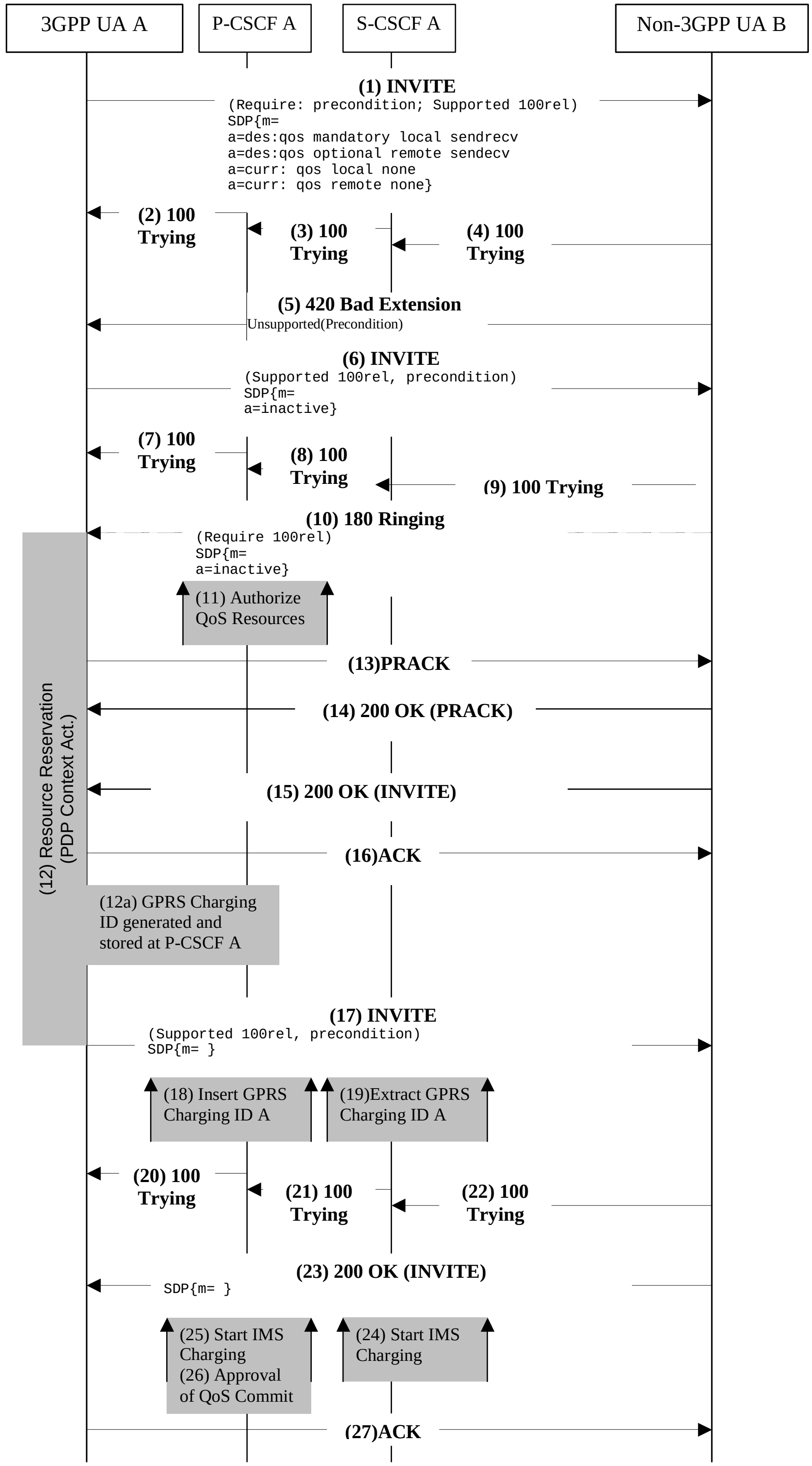

Figure 4.1.2.4.2.1/1: Using re-~~invite~~ INVITE to connect an ~~calling~~ originating 3GPP UA to a ~~called~~ terminating non-3GPP UA not supporting the SIP preconditions extension, but supporting the SIP 100rel extension.

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~~B2BUA

A ~~B2B UA~~B2BUA is used.

4.1.3.2.1.1 Insertion of ~~B2B UA~~B2BUA

How the ~~B2B UA~~B2BUA is inserted is discussed within Section 4.1.2.4.1.1.

4.1.3.2.1.2 Functionality of ~~B2B UA~~B2BUA

4.1.3.2.1.2.1 Description

The ~~B2B UA~~B2BUA shall apply the rules given in section 4.1.2.4.1.2.1.

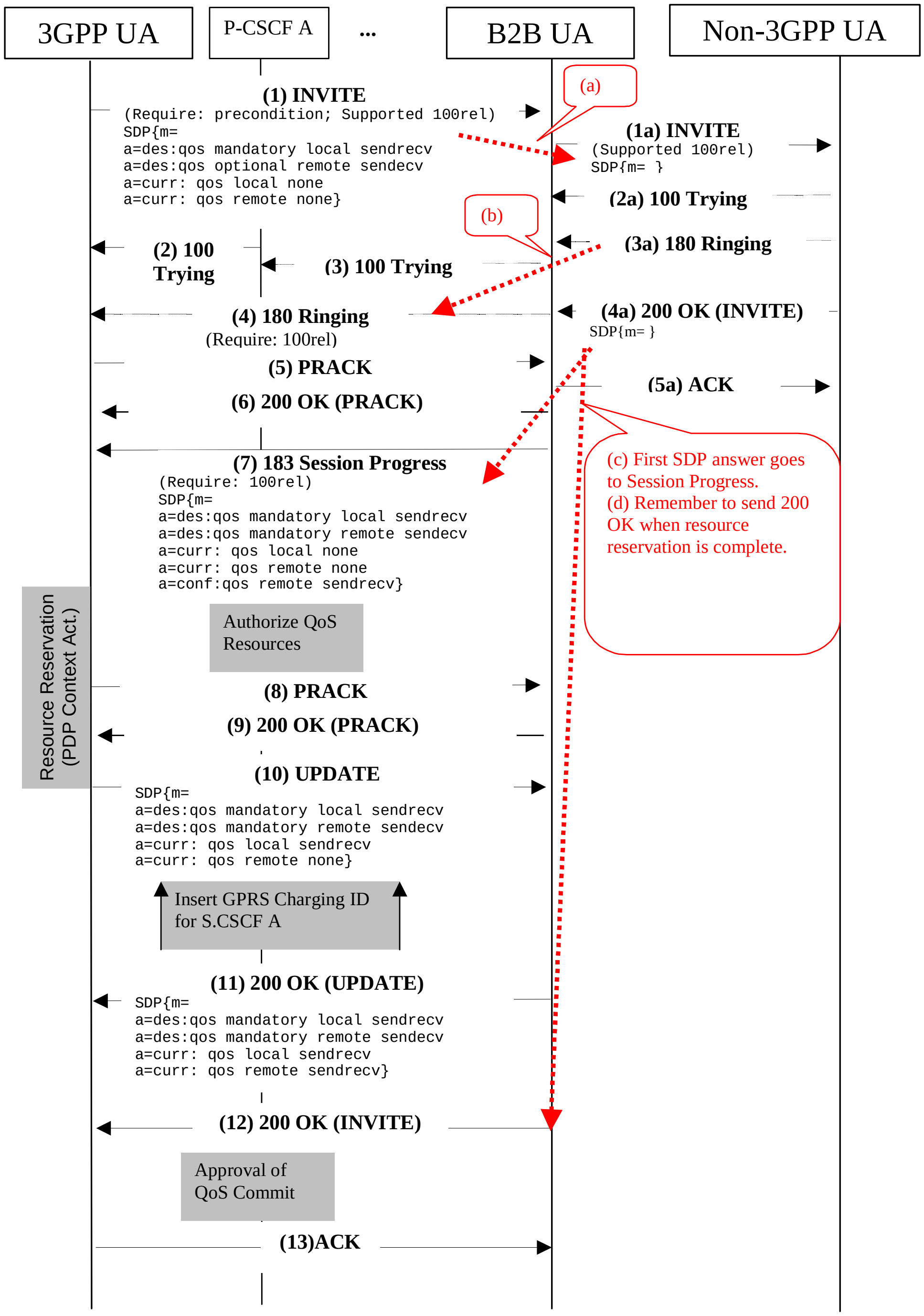

Figure 4.1.3.2.1.2/1: Functionality of ~~B2B UA~~B2BUA connecting ~~calling~~ an originating 3GPP UA to ~~called~~ a terminating non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension.
~~Calling~~ The originating UA sends no second SDP offer

There may be re-transmissions of the INVITE (1) by the 3GPP UA, which should be ~~pass~~forwarded transparently by the ~~B2B UA~~B2BUA, as indicated in interaction (a).

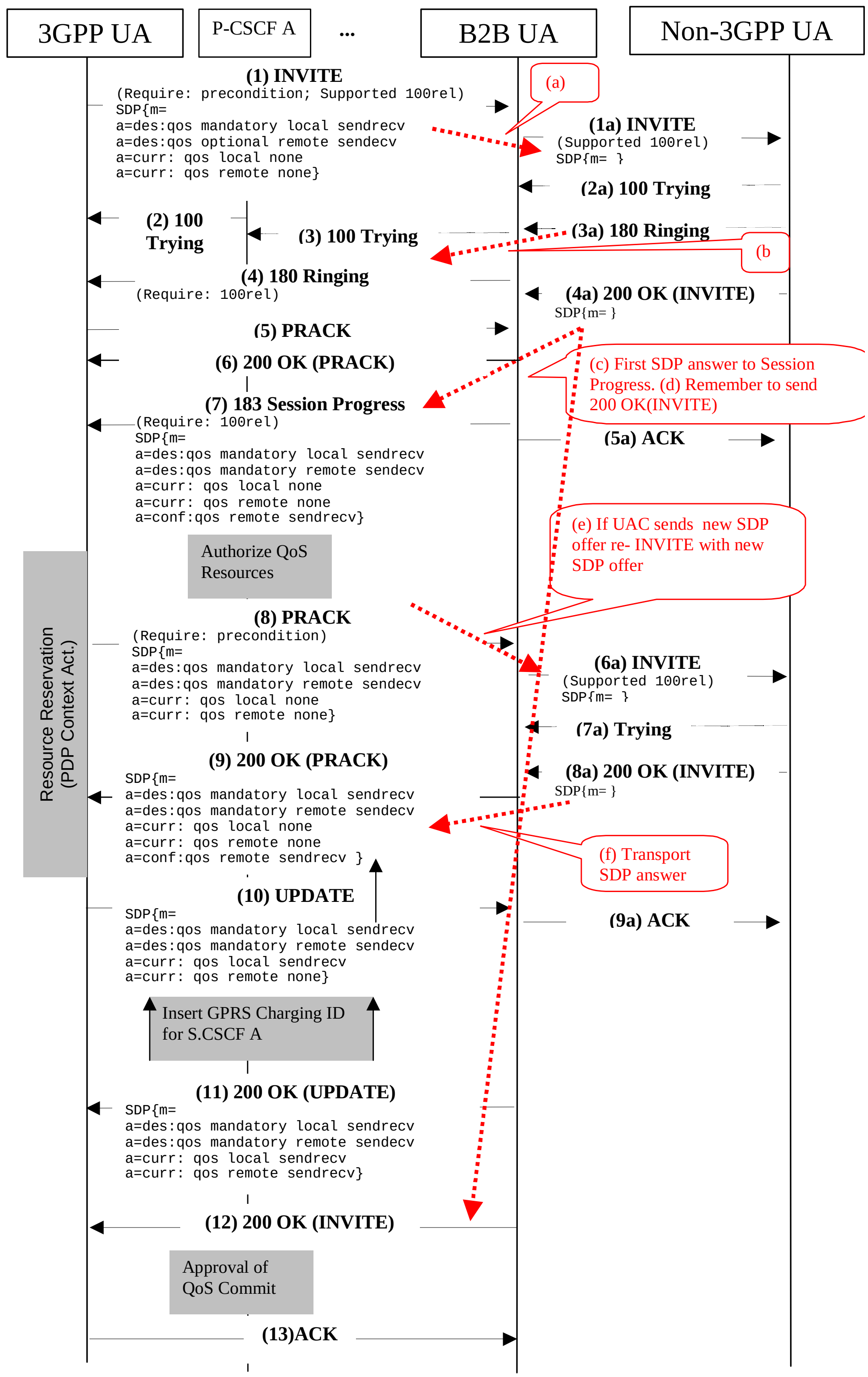

Figure 4.1.3.2.1.2/2: Functionality of ~~B2B UA~~B2BUA connecting ~~calling~~ an originating 3GPP UA to ~~called~~ a terminating non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension.
~~Calling~~ The originating UA sends second SDP offer

4.1.3.2.1.2.2 Advantages

General advantages of the ~~B2B UA~~B2BUA 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~~B2BUA 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 180 (Ringing) response~~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~~ originating 3GPP UA should (not shall) require preconditions in an initial INVITE request. The ~~calling~~ originating 3GPP UA may (re-)INVITE an external UA without requiring preconditions, e.g. if receiving a 420 (Bad Extension) ~~(precondition) error~~ 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 shall send a re-~~invite~~ 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~~ terminating non-3GPP UA ~~may~~ will accept ~~invites~~ INVITE requests not containing a ~~"Require(precondition)"~~"Require" header field with the "precondition" value.
3. (e.g. in TS 24.229) The ~~called~~ terminating non-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~~ terminating non-3GPP UA may also send a 200 (OK) response for an ~~(~~INVITE~~)~~ request before the 3GPP UA has complete the resource reservation ~~is complete~~, but ~~shall~~ will not send media, because it was requested in the SDP offer~~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) 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.
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) response for an ~~(~~INVITE~~) response~~request 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.

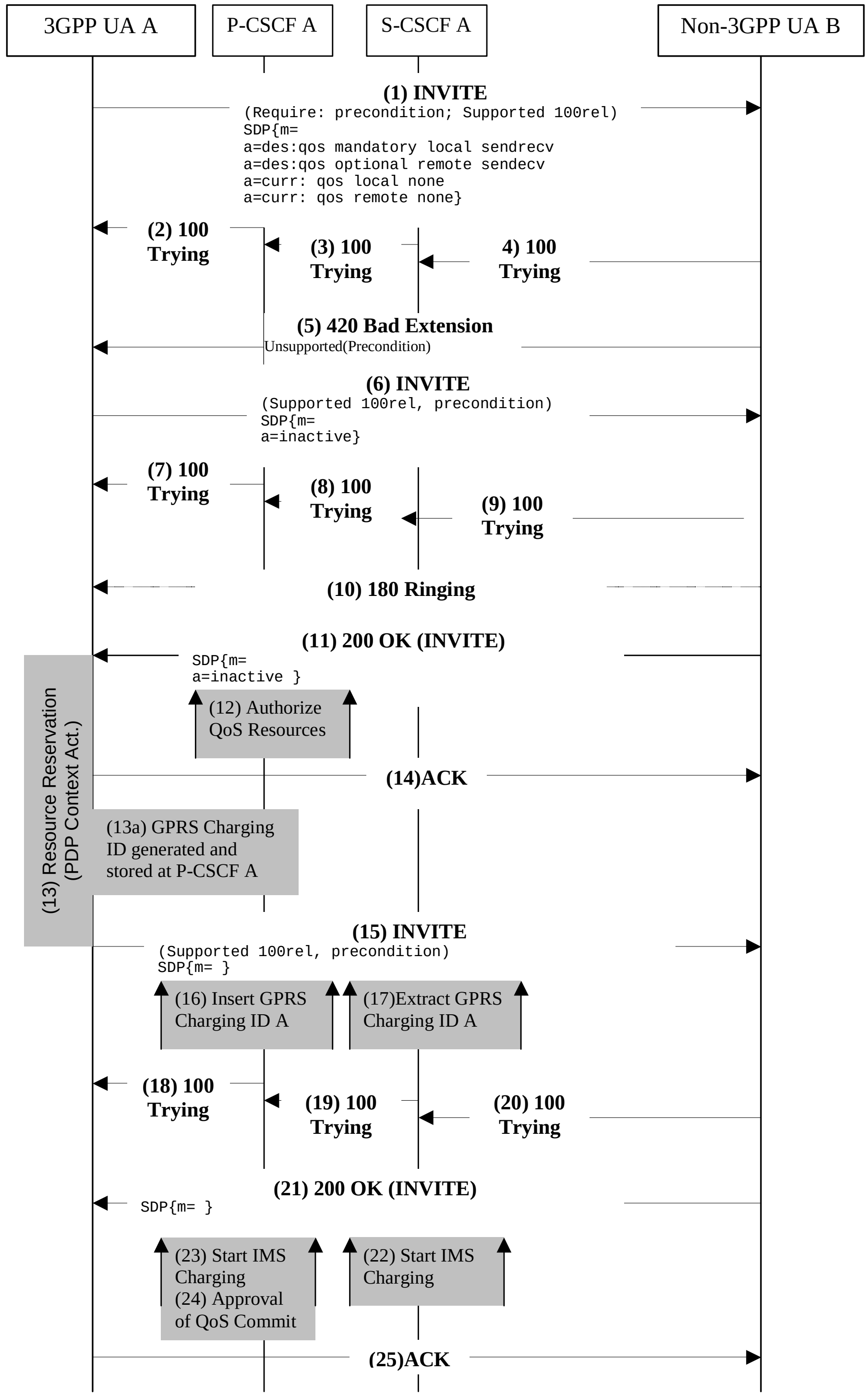

Figure 4.1.3.2.2.2/1: Using re-~~invite~~ INVITE to connect ~~calling~~ an originating 3GPP UA to ~~called~~ a terminating non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension.

4.1.3.2.2.2 Advantages

Only relatively minor changes to the 3GPP specifications 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

As the 3GPP UA requires the usage of the preconditions extension, and as the non-3GPP UA does not support such extenstion, t~~T~~he ~~call~~ session attemp 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~~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.

4.2.1.2 Proposed Resolutions to interworking issue

4.2.1.2.1 ~~B2B UA~~B2BUA

A ~~B2B UA~~B2BUA is used.

4.2.1.2.1.1 Insertion of ~~B2B UA~~B2BUA

How the ~~B2B UA~~B2BUA is inserted is discussed within Section 4.2.2.4.1.1.

4.2.1.2.1.2 Functionality of ~~B2B UA~~B2BUA

4.2.1.2.1.2.1 Description

The functionality of the ~~B2B UA~~B2BUA is as discussed in Section 4.2.2.4.1.2.1.

The ~~B2B UA~~B2BUA shall ~~pass~~forward additional UPDATE ~~messages~~requests, which are not related to the precondition extension, and related provisional acknowledgement (PRACK) ~~messages~~requests.

4.2.1.2.1.2.2 Advantages

General advantages of the ~~B2B UA~~B2BUA are discussed in Section 4.2.2.4.1.2.2.

The ~~calling~~ originating and the ~~called~~terminating UA may send UPDATE ~~messages~~ requests at various places within the call flow. Those ~~messages~~ 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 ~~messages~~ requests probably do not have harmful side effects.

4.2.1.2.1.2.3 Disadvantages

General disadvantages of the ~~B2B UA~~B2BUA 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

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

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.

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~~ requests 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~~ terminating 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

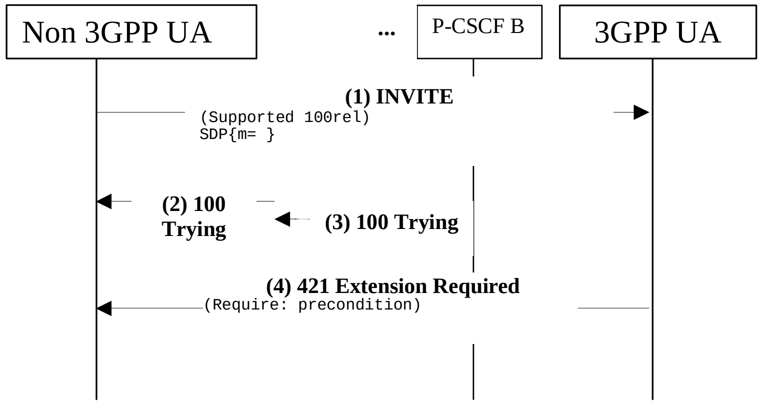

Figure 4.2.2.2/1: Non-3GPP SIP UA not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA.

4.2.2.3 Implications of Identified interworking issue

The ~~call~~ session attempt fails.

4.2.2.4 Proposed resolutions to interworking issue

4.2.2.4.1 ~~B2B UA~~B2BUA

A ~~B2B UA~~B2BUA is used.

4.2.2.4.1.1 Insertion of ~~B2B UA~~B2BUA

4.2.2.4.1.1.1 Static Insertion of ~~B2B UA~~B2BUA

4.2.2.4.1.1.1.1 Description

A ~~B2B UA~~B2BUA is permanently inserted at connection between IMS and a given external network. This ~~B2B UA~~B2BUA handles all calls, including calls where the call flows may be ~~pass~~forwarded without modification.

The ~~B2B UA~~B2BUA shall be inserted in the border of the ~~IMS~~ home network for all ~~calls~~ session attempts entering the ~~IMS~~ home network. Note that, even 3GPP to 3GPP session attempt could potentially byforward 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. ~~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~~B2BUA becomes active only when receiving an INVITE ~~message~~ request 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~~B2BUA ~~passes~~ forwards all SIP ~~messages~~ 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.

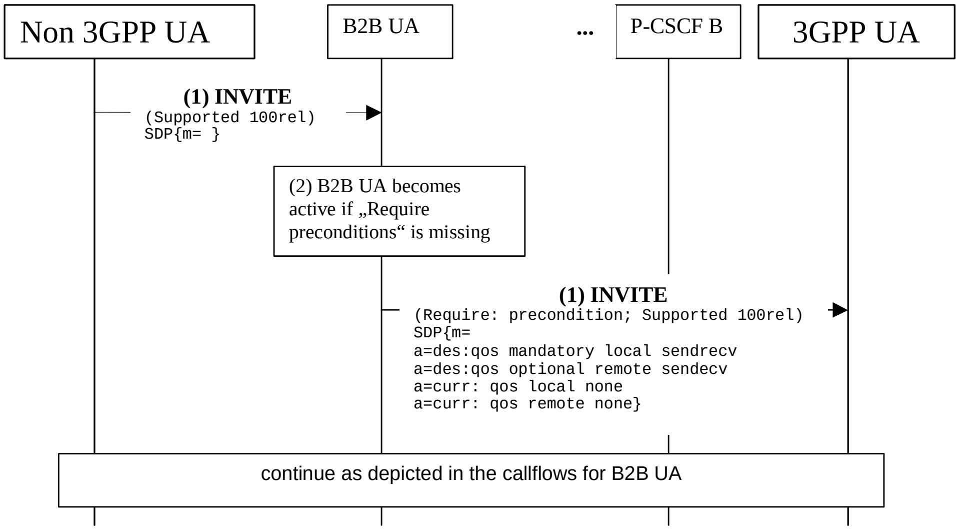

Figure 4.2.2.4.1.1.1.1/1: Activation of static ~~B2B UA~~B2BUA connecting Non-3GPP UA not indicating support of the SIP preconditions extension to 3GPP UA.

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~~B2BUA may be activated unnecessarily, if the Non-3GPP UA supports the precondition extension, but fails to indicate this in the INVITE ~~message~~request.

4.2.2.4.1.2 Functionality of ~~B2B UA~~B2BUA

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~~B2BUA shall apply the following rules:

1. The ~~B2B UA~~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 ~~B2B UA~~B2BUA shall also comply with the SIP 100rel and update extensions.

3. On the IMS side, the ~~B2B UA~~B2BUA shall also comply to and apply the SIP 100rel, update and precondition extensions.

4. The ~~B2B UA~~B2BUA shall ~~pass~~forward SIP requests arriving at one call leg to the call leg at the other side, with the exceptions below.

5. The ~~B2B UA~~B2BUA shall ~~pass~~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 ~~B2B UA~~B2BUA shall ~~pass~~forward SIP final responses arriving at one call leg to the call leg at the other side, with the exceptions below.

7. The ~~B2B UA~~B2BUA shall not require the SIP preconditions extension on the non-IMS side.

8. The ~~B2B UA~~B2BUA shall insert preconditions in SDP sent from non-3GPP UA to 3GPP UA.

9. The ~~B2B UA~~B2BUA should remove preconditions in SDP sent from 3GPP UA to non-3GPP UA

10. The ~~B2B UA~~B2BUA shall not ~~pass~~forward PRACK requests and 200 (OK) responses for the ~~(~~PRACK~~)~~ ~~messages~~ request.

11. The ~~B2B UA~~B2BUA shall inspect an INVITE ~~message~~ 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 ~~B2B UA~~B2BUA receives an SDP offer in a provisional response from the IMS side, the ~~B2B UA~~B2BUA shall send the SDP offer in a 200 (OK) response ~~(~~for an ~~invite~~INVITE request~~)~~ ~~message~~ at the non-IMS side. The ~~B2B UA~~B2BUA shall then forward the SDP answer received in the ACK ~~message~~ request from the non-IMS side to the PRACK ~~message~~ request 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~~B2BUA receives an SDP answer in a provisional response from the IMS side, the ~~B2B UA~~B2BUA shall send the SDP answer in a 200 (OK) response ~~(invite~~for an INVITE request~~)~~ ~~message~~ response at the non-IMS side.

14. For a re-~~Invite~~ INVITE request from the IMS side to the Non-IMS side, the ~~B2B UA~~B2BUA shall apply the rules in Section 4.1.2.4.1.2.1.

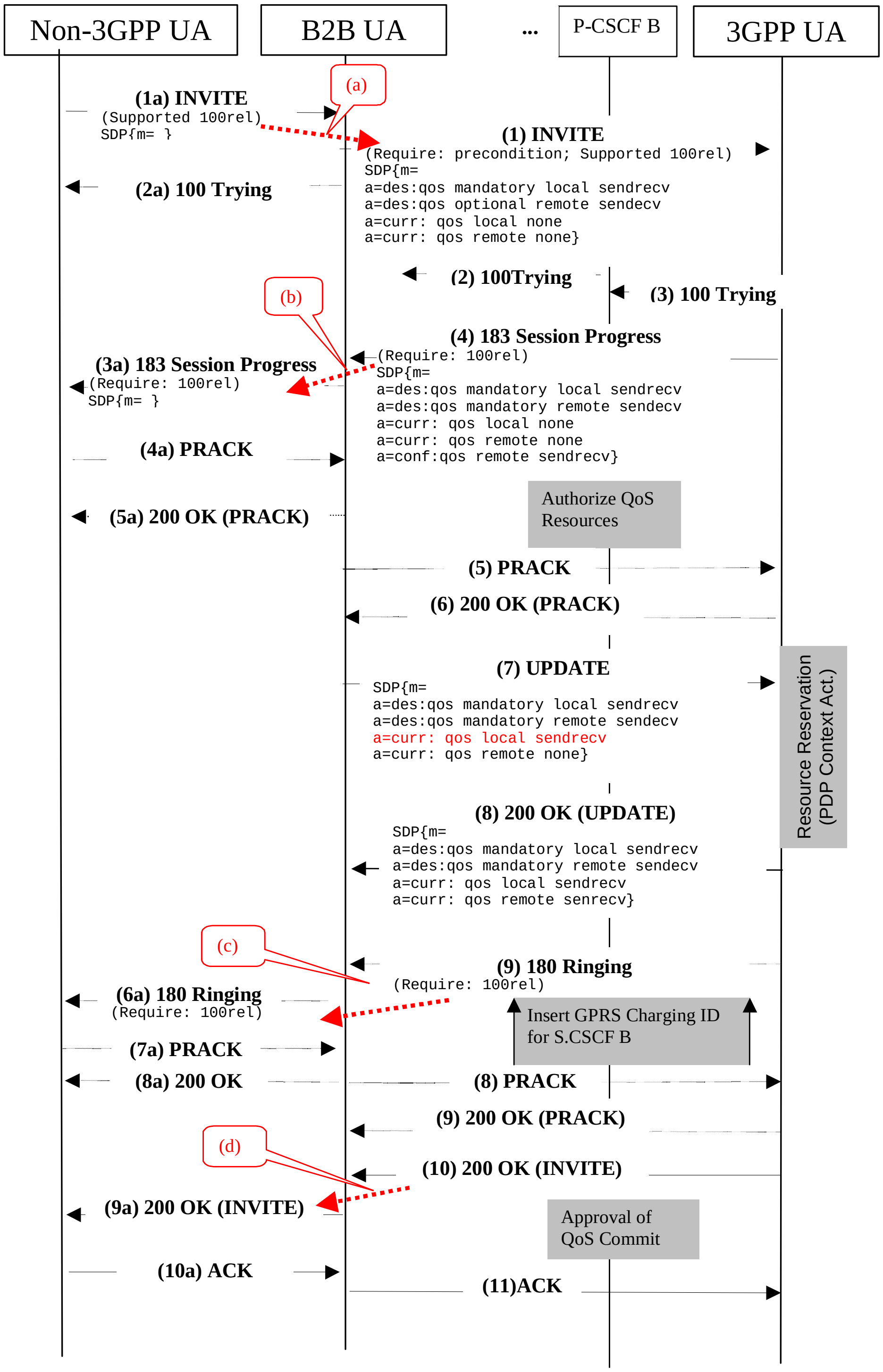

Figure 4.2.2.4.1.2.1/1: Functionality of ~~B2B UA~~B2BUA connecting ~~calling~~ an originating non-3GPP SIP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension, to ~~called~~ a terminating 3GPP UA.

4.2.2.4.1.2.2 Advantages

4.2.2.4.1.2.3 Disadvantages

The functionality and implementation of the ~~B2B UA~~B2BUA 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~~B2BUA. 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~~mandate the usage of the preconditions extension for terminating session attempts~~, 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.

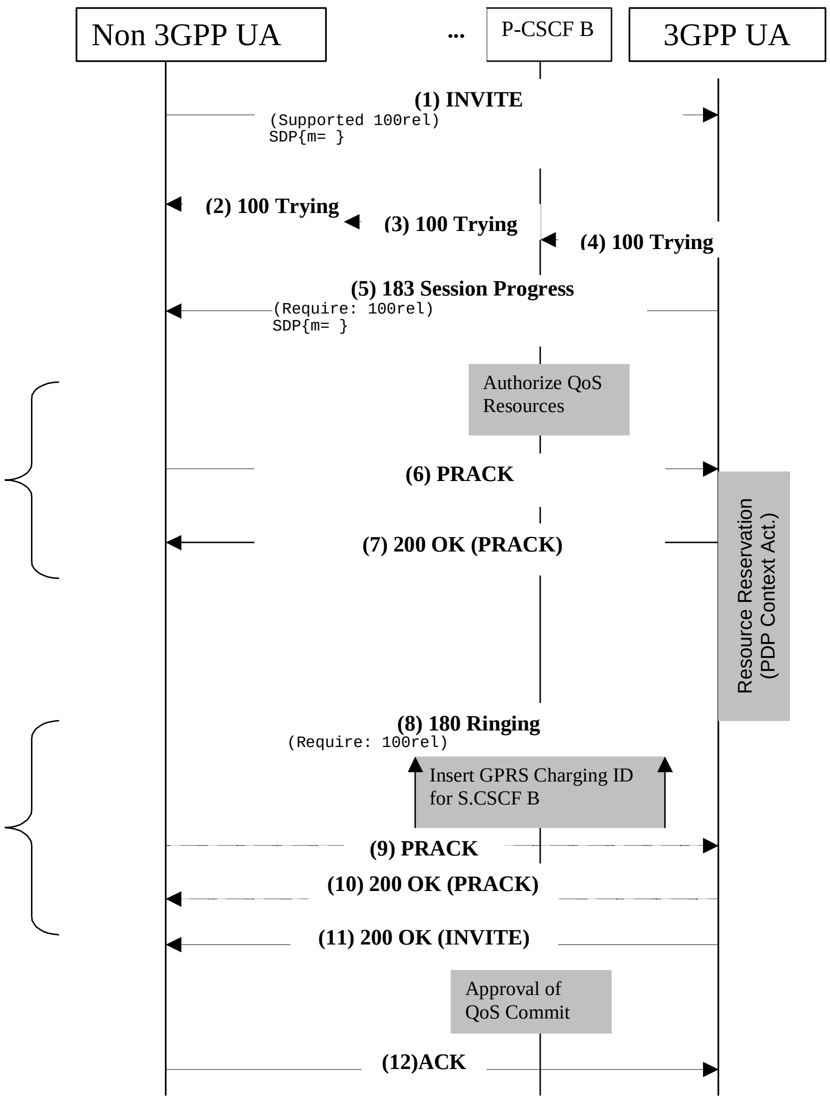

Figure 4.2.2.4.2.1/1: Modified end-to-end call flow for Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA. SDP offer in ~~Invite~~INVITE request.

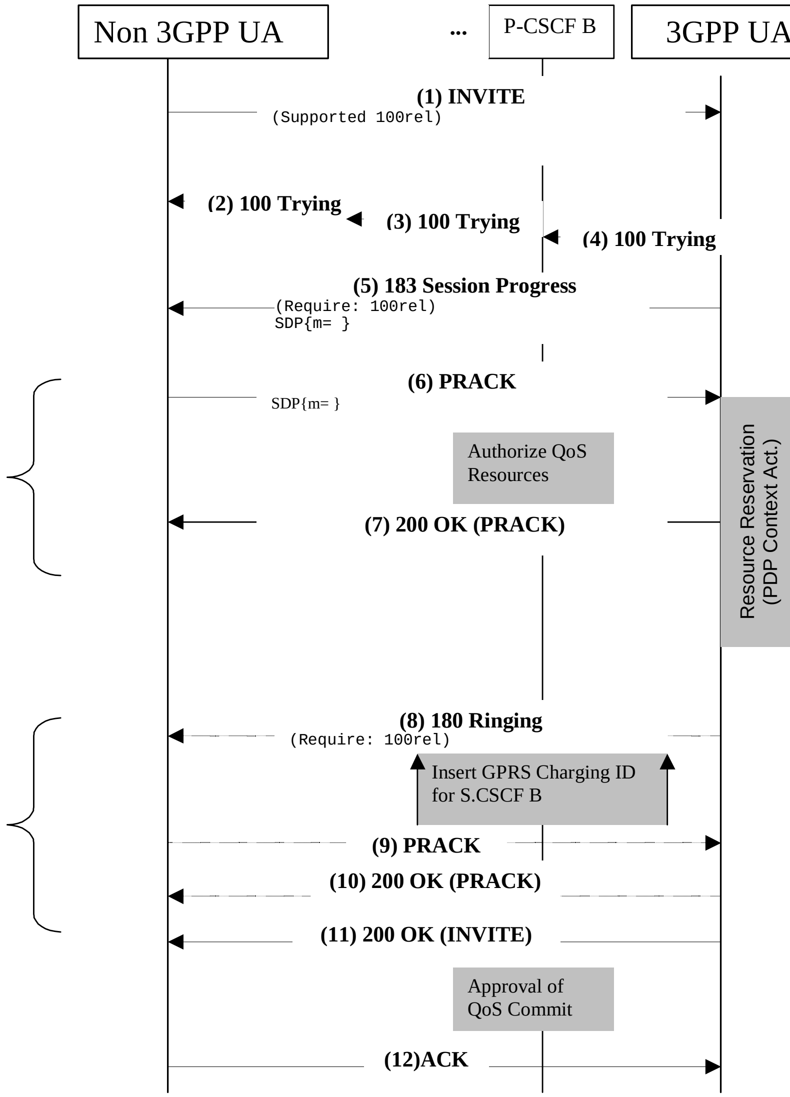

Figure 4.2.2.4.2.1/2: Modified end-to-end call flow for Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UA. No SDP offer in ~~Invite~~INVITE request.

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~~B2BUA

A ~~B2B UA~~B2BUA is used.

4.2.3.2.1.1 Insertion of ~~B2B UA~~B2BUA

4.2.3.3.1.1.1 Static Insertion of ~~B2B UA~~B2BUA

4.2.3.3.1.1.1.1 Description

A ~~B2B UA~~B2BUA is permanently inserted at connection between ~~IMS~~ the home operator and ~~a given external~~ any other network. This ~~B2B UA~~B2BUA handles all calls, including calls where the call flows may be ~~pass~~forwarded without modification.

The ~~B2B UA~~B2BUA shall be inserted in the border of the ~~IMS~~ home networ for all session attempts ~~calls~~ entering the ~~IMS the IMS from an external network (except for another IMS)~~home network. ~~.~~

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

The ~~B2B UA~~B2BUA becomes active only when receiving an INVITE ~~message~~ request 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~~B2BUA ~~passes~~ forwards all SIP ~~messages~~ requests and responses received at one side to the other side. Note that, even 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.

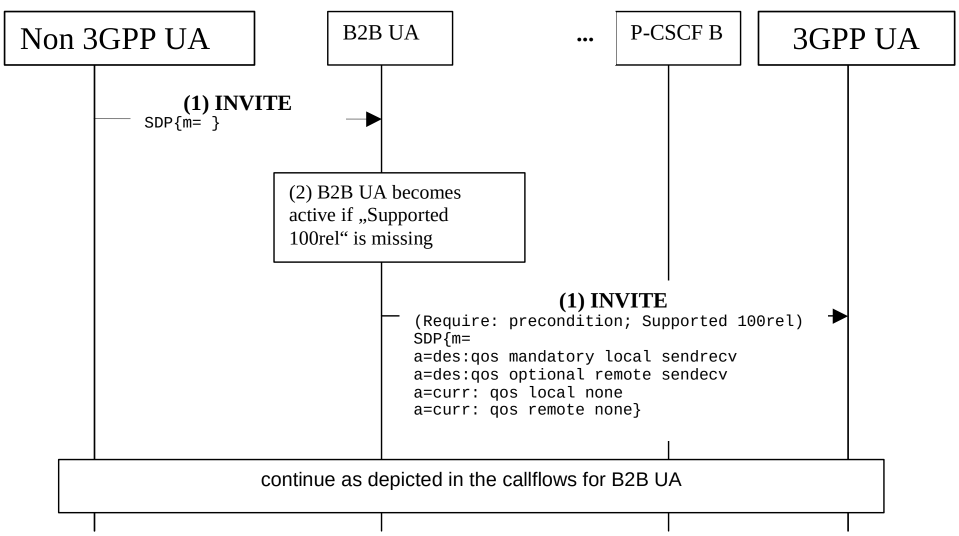

Figure 4.2.3.3.1.1.1.1/1: Activation of static B2B connecting Non-3GPP SIP UA not indicating support of the SIP preconditions extension to 3GPP UA.

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

4.2.3.2.1.2 Functionality of ~~B2B UA~~B2BUA

4.2.3.2.1.2.1 Description

The ~~B2B UA~~B2BUA shall apply the rules given in section 4.2.2.4.1.2.1.

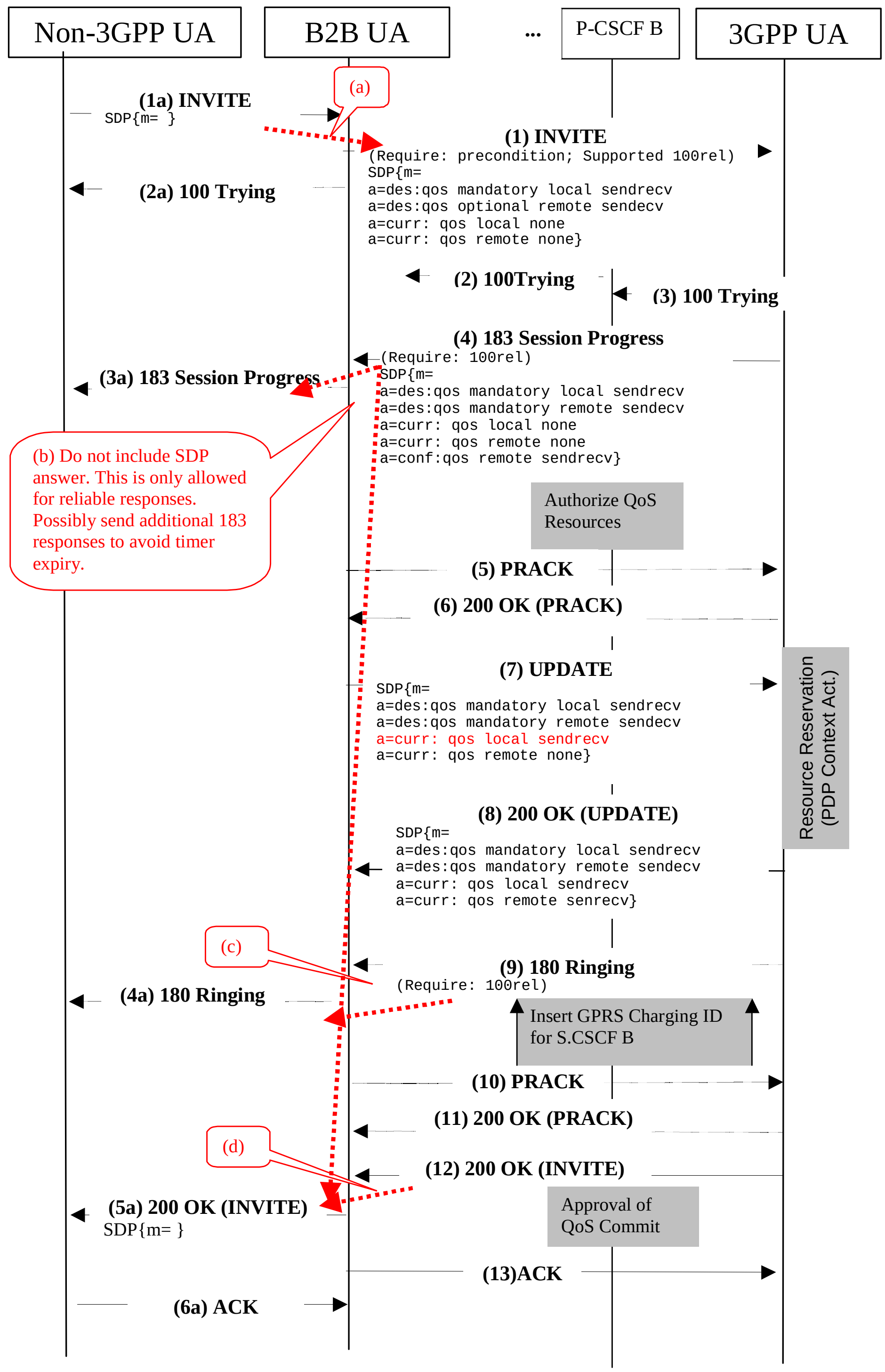

Figure 4.2.3.2.1.2.1/1: Functionality of ~~B2B UA~~B2BUA connecting ~~calling~~ an originating non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to ~~called~~ a terminating 3GPP UA. SDP offer in INVITE request.

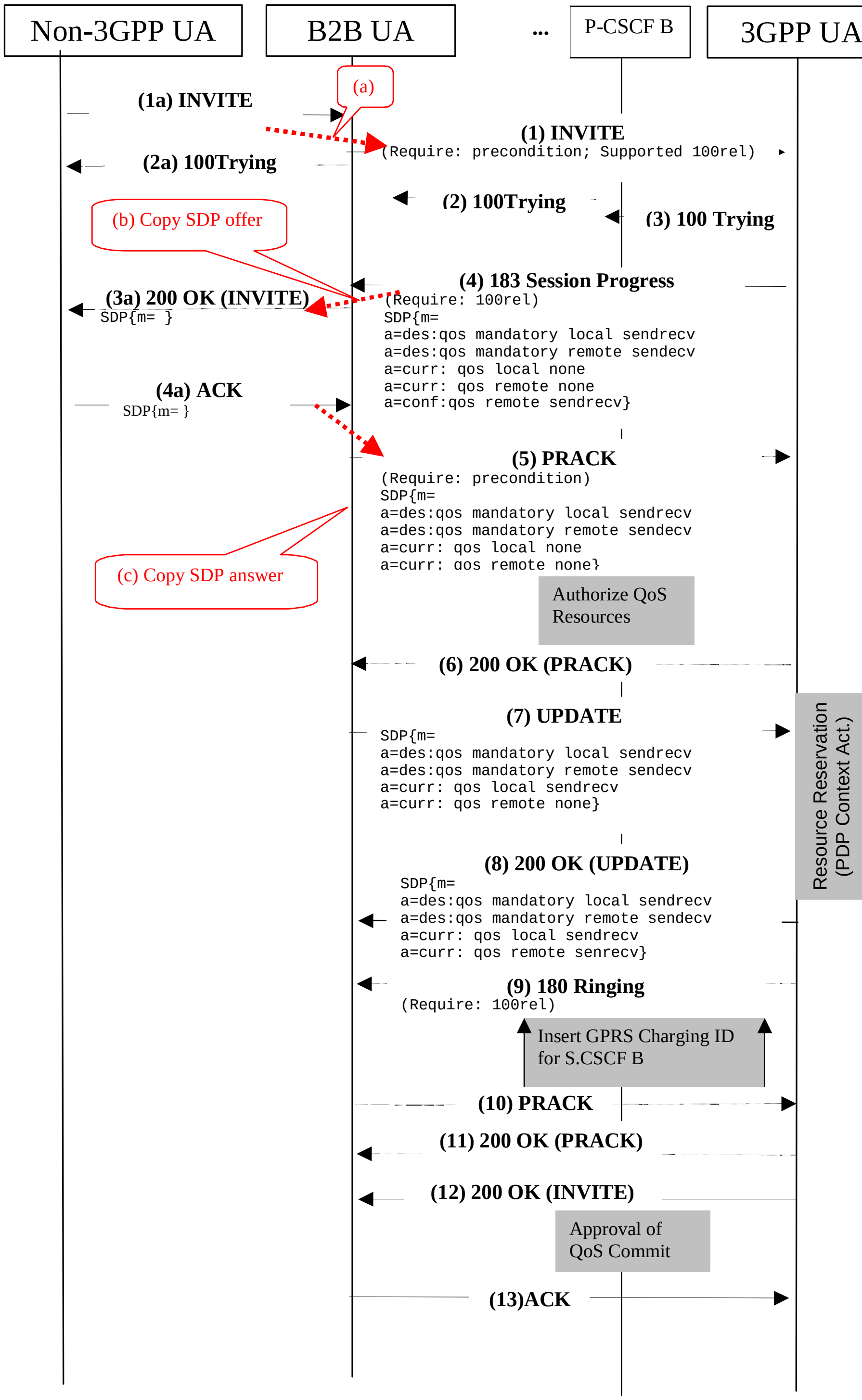

Figure 4.2.3.2.1.2.1/2: Functionality of ~~B2B UA~~B2BUA connecting ~~calling~~ an originating non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to ~~called~~ a terminating 3GPP UA. SDP offer in OK response.

4.2.3.2.1.2.2 Advantages

General advantages of the ~~B2B UA~~B2BUA 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~~B2BUA 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.

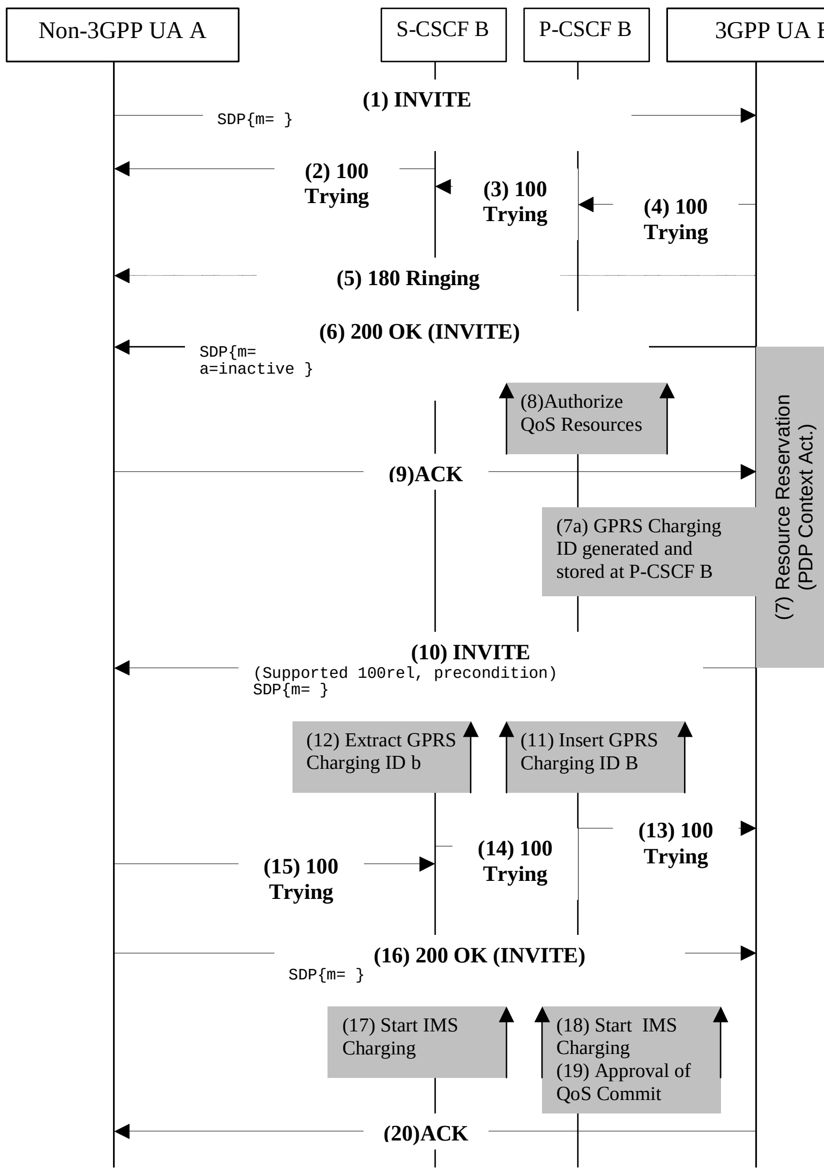

Figure 4.2.3.2.2.2/1: Using re-~~invite~~ INVITE to connect ~~calling~~ an originating non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension to ~~called~~ a terminating 3GPP UA. The INVITE request contains SDP.

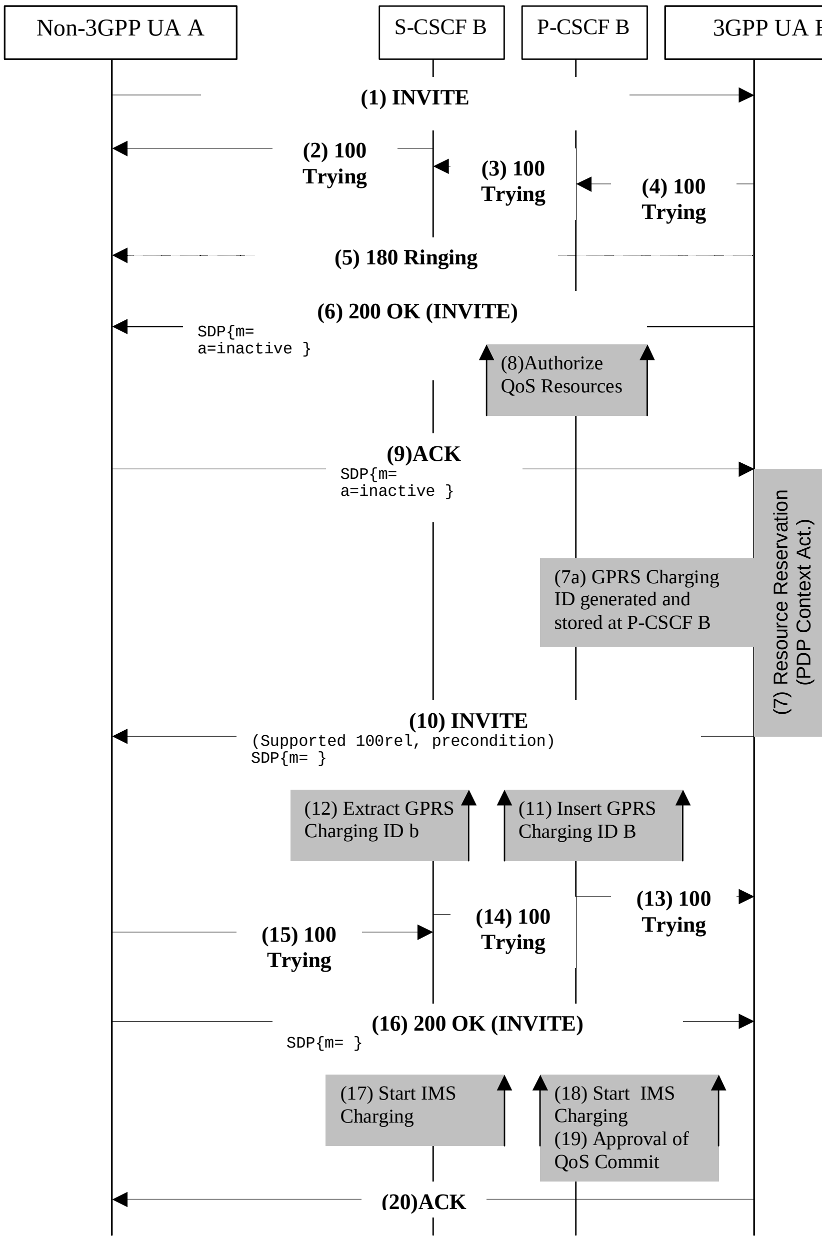

Figure 4.2.3.2.2.2/2: Using re-~~invite~~ INVITE to connect ~~calling~~ an originating non-3GPP UA not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension to ~~called~~ a terminating 3GPP UA. The INVITE request contains no SDP.

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

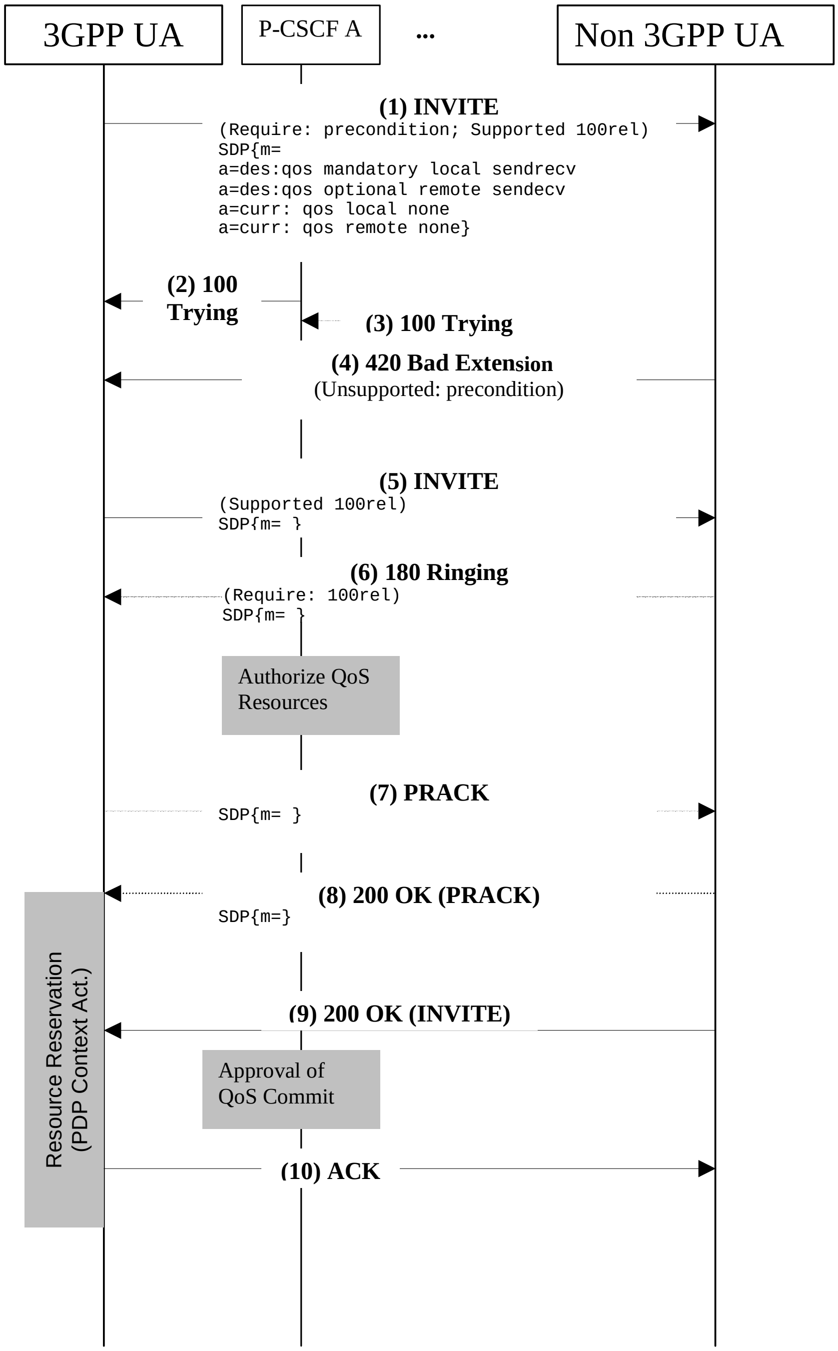

Figure C.1.2.2/1: rogue 3GPP UA to non-3GPP UA supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension

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

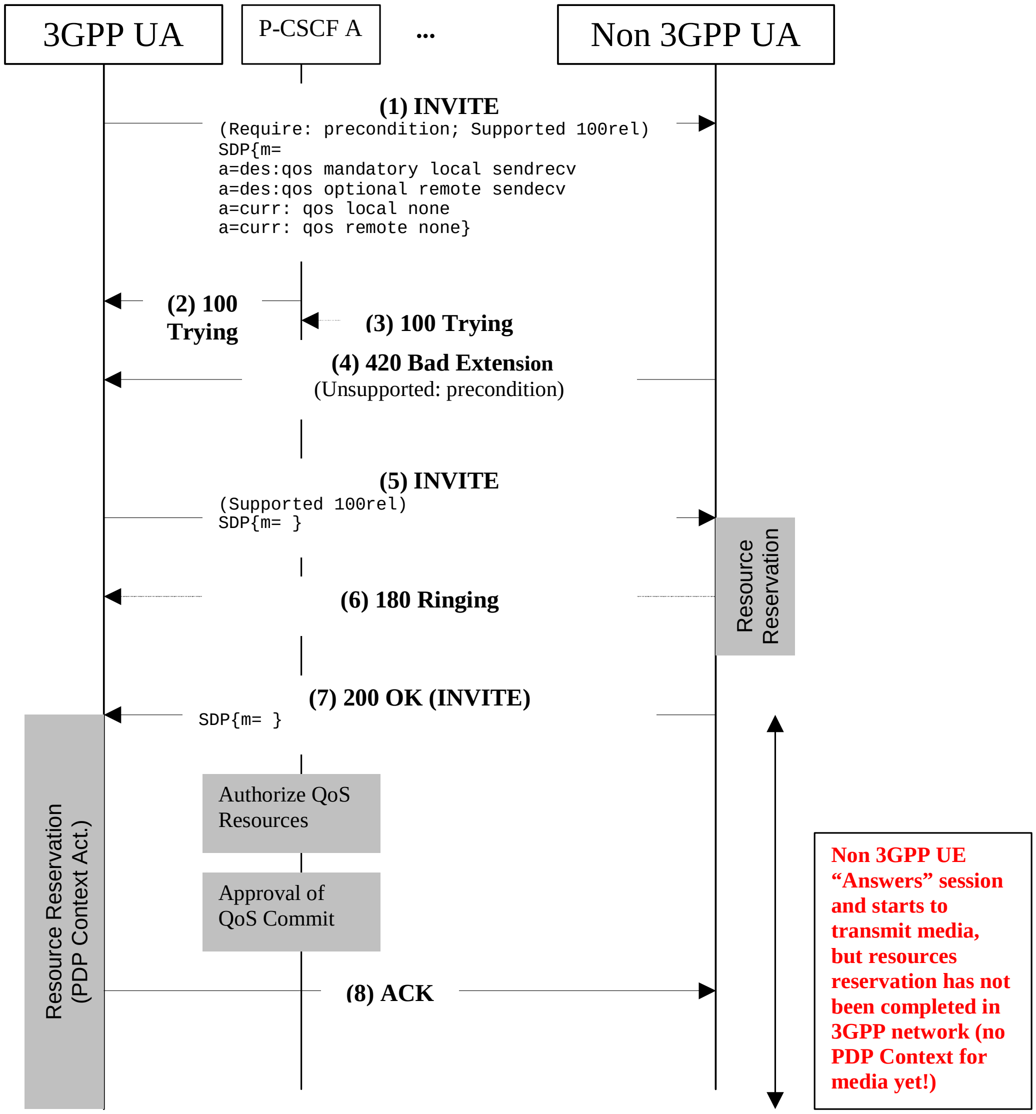

Figure C.1.3.2/1: 3GPP UA to non-3GPP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension.

(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

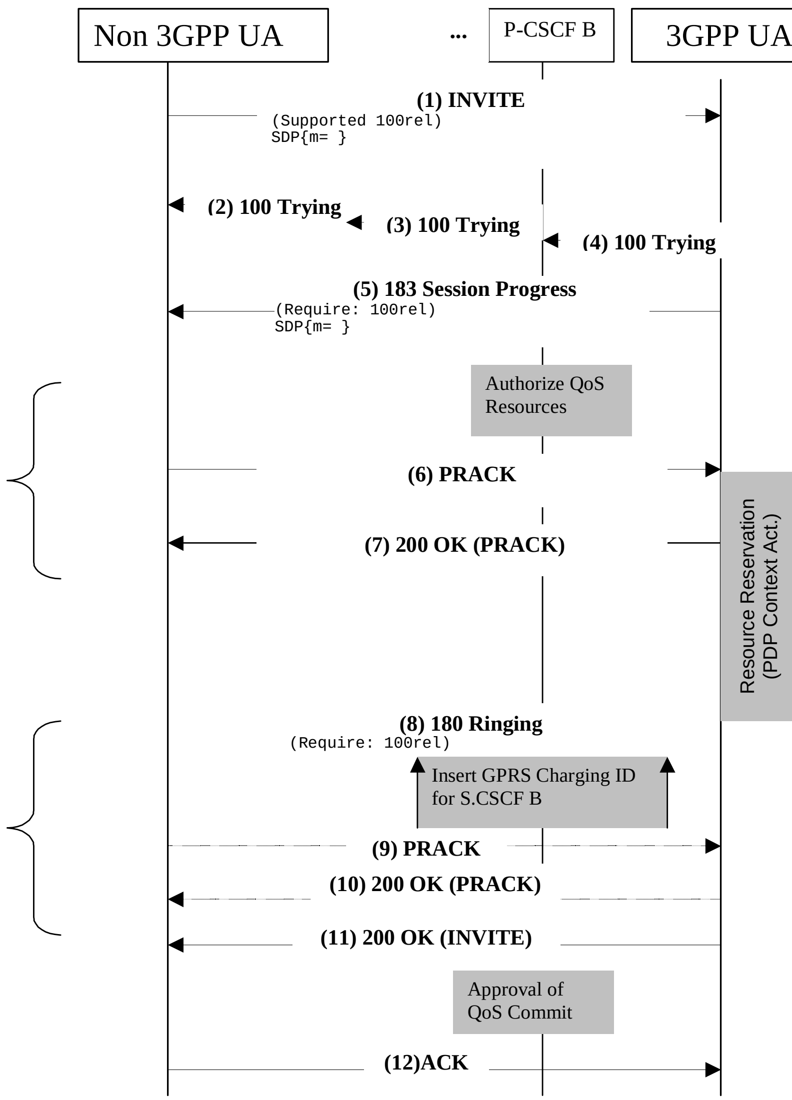

Figure C.2.2.2/1: Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in Invite.

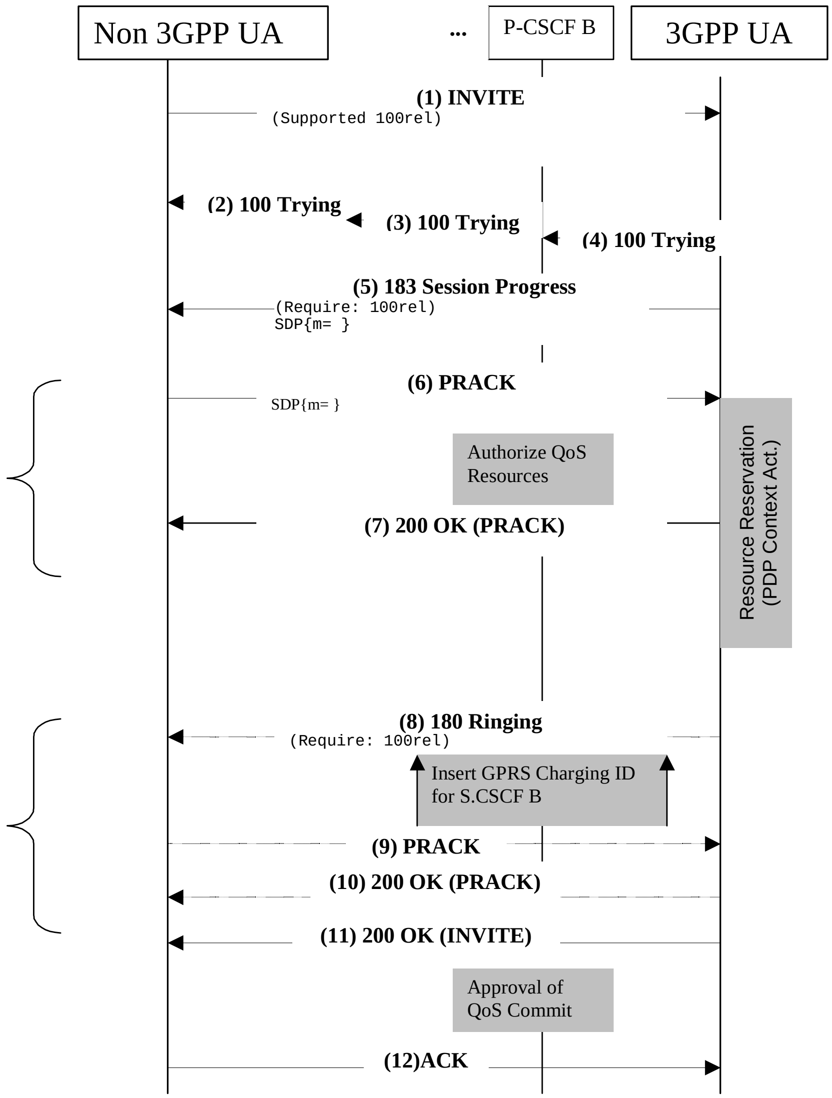

Figure C.2.2.2/2: Non-3GPP UA supporting the 100rel SIP extension, but not supporting the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. No SDP offer in Invite.

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

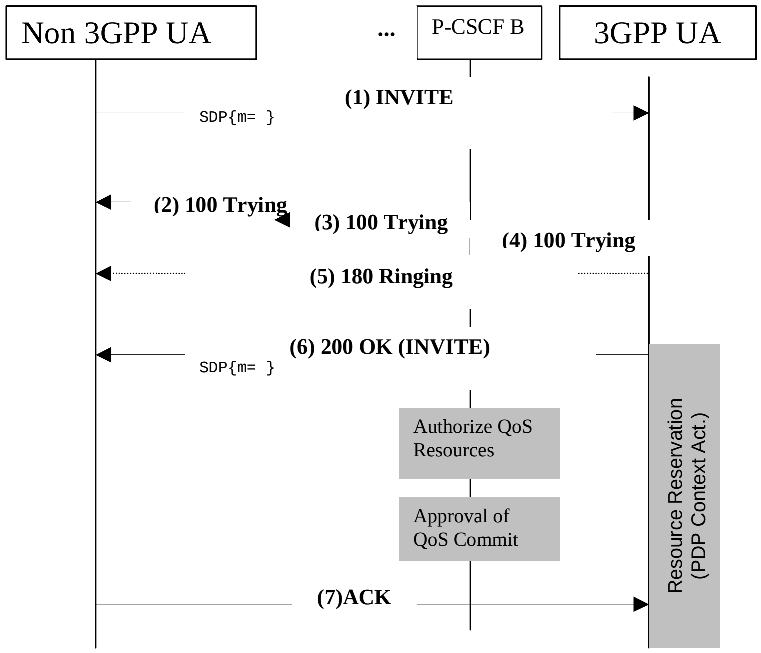

Figure C.2.3.2/1: Non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in INVITE request.

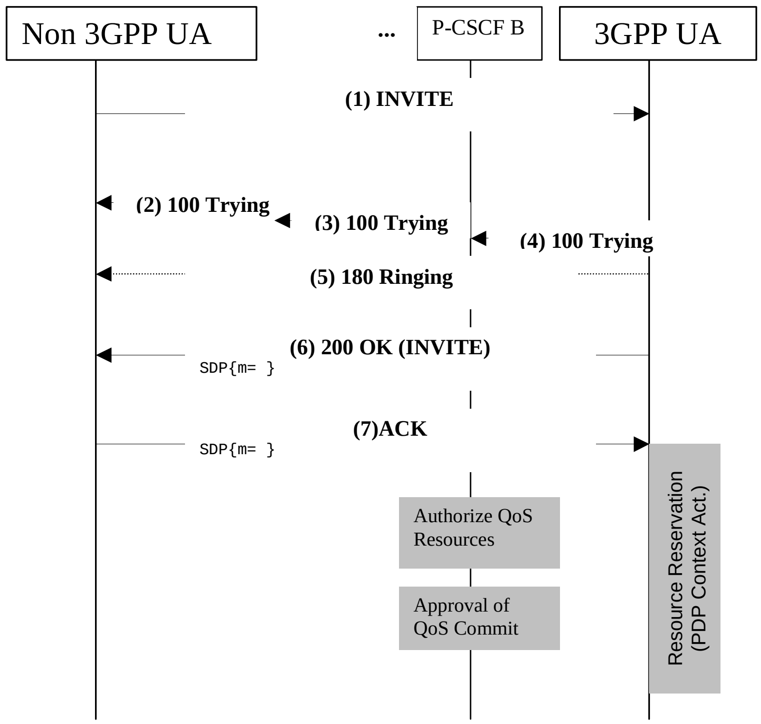

Figure C.2.3.2/2: Non-3GPP SIP UA not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UA. SDP offer in OK response.

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

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

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

NOTE 3: The 100 ~~TRYING~~ (Trying) response (2), (3), (4) to the INVITE ~~message~~ request (1) is ~~send~~ 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”~~"Require", ~~“supported”~~"Supported" and ~~“allowed”~~"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~~ 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”~~ 183 (Session Progress) response, is required to transport the SDP answer including the mandated “confirmation status” SDP attribute (Ref. [6], Section 6). Moreover, the ~~“180 Ringing” message~~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) ~~messages~~. ~~A calling~~An originating 3GPP UA will refrain from generating a new SDP offer within PRACK request (11), if it does not wish to further restrict the set of codecs selected within the first offer/answer pair.

NOTE 9: According to IETF Ref. 6, Section 5, the called UA should start the resource reservation (13) immediately after having send the SDP answer within of the ~~“183 Session Progress”~~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”~~183 (Session Progress) (7) provisional response.

NOTE 10: The use of the ~~“Update”~~ UPDATE ~~Request~~ 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”~~UPDATE request (14) ~~request~~ is not used, the subsequent ~~”~~ 200 (OK) response for an UPDATE request~~(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~~”~~ request (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.

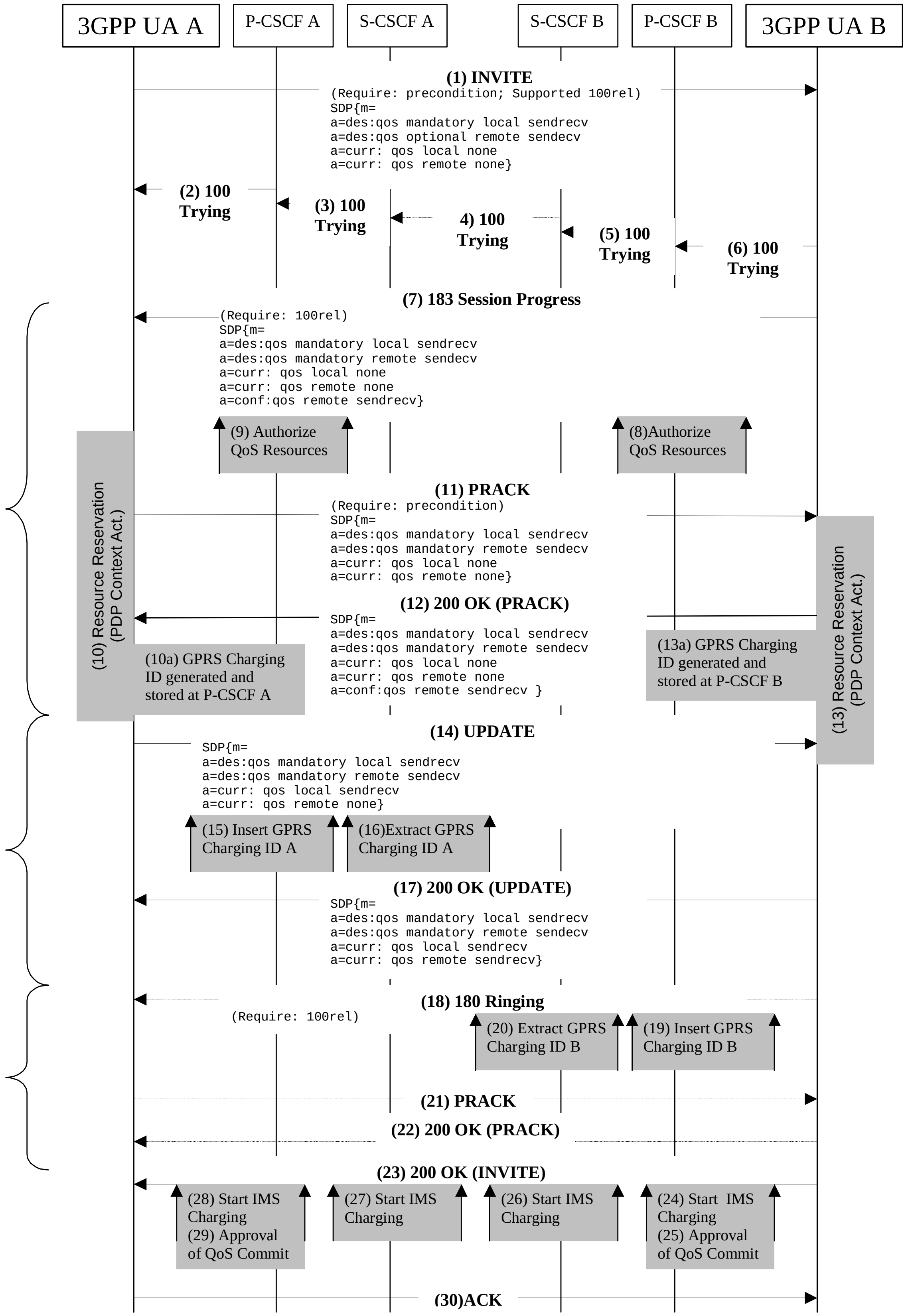

Figure ~~D~~C/1: 3GPP UA to 3GPP UA Call flow

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

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

Annex E: Scenarios without identified interworking issues

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

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

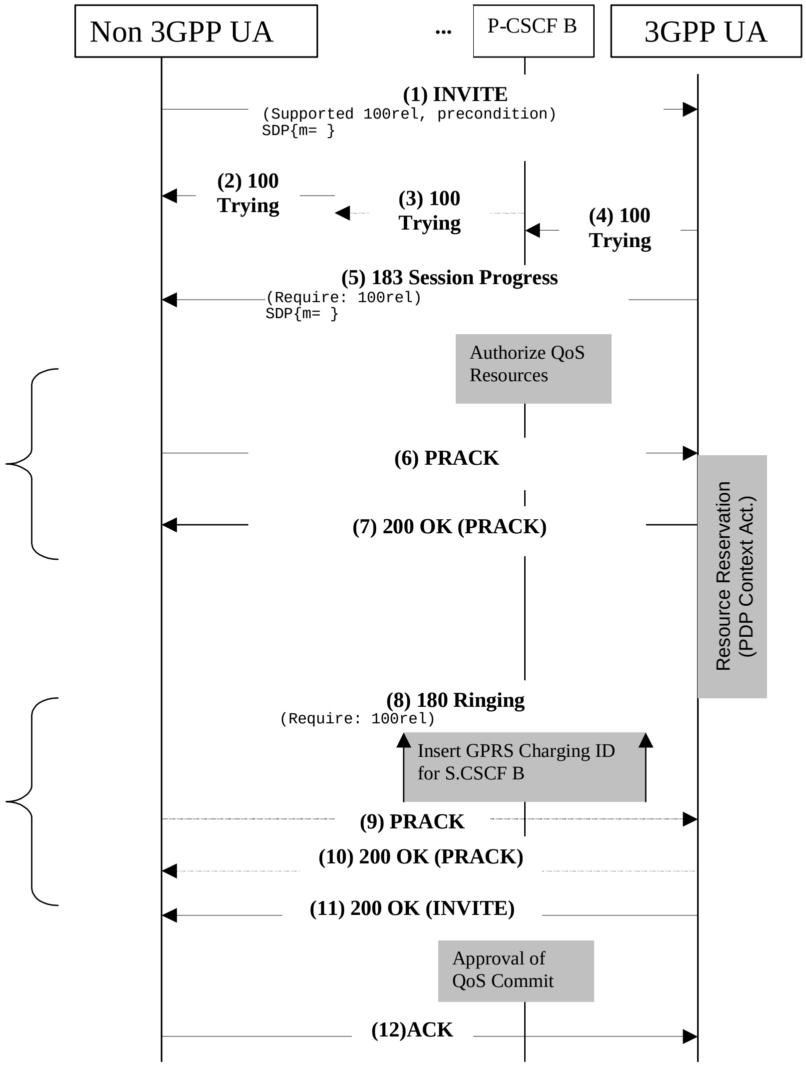

Figure E.1/1: Non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, and not performing QoS reservation, to 3GPP UA

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

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

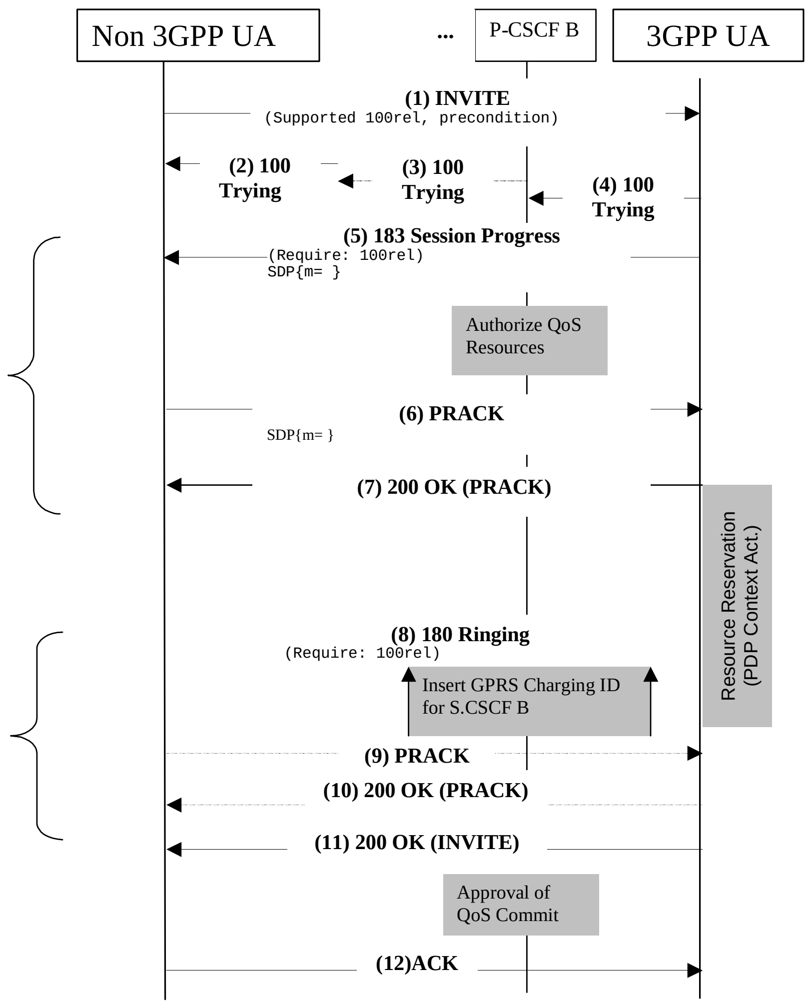

Figure E.2/1: Non-3GPP UA supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not including the SDP offer in the initial ~~invite~~INVITE request, to 3GPP UA

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 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 TSG-CN1 Meeting #29 ***Tdoc N1-030535***
Sophia-Antipolis, France, 31 March – 04 April 2003

Source:	Siemens
Title:	Proposal for restructuring TR 29.962
Agenda item:	8.6
Document for:	APPROVAL

TR 29.962 shall be modified in the following way:

- Headings shall be modified or deleted as indicated by changemarks below
- Corresponding figure captions shall be modified in the same way
- Sections shall be rearranged in the order indicted below
- Headings shall be renumbered as indicated below
- Figure Captions shall be renumbered accordingly
- Cross-References to Sections and Figures shall be updated accordingly
- Headings without a new number assigned shall be preserved as underlined text, if a sufficient amount of text fitting to them is available, and otherwise be removed
- Text within the sections shall be modified or added as indicated with changemarks in italics below
- The Abbreviation “B2B UA” shall be replaced “B2BUA” troughout the specification

Section Number

New Nbr	**Old Nbr**	**Section Heading Text / *proposed new text for Section***
		Foreword
1	1	Scope
2	2	References
3	3	Definitions, symbols and abbreviations
3.1	3.1	Definitions *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	3.2	Abbreviations
	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	4.1	Session Setup from Calling 3GPP UA towards Called non-3GPP UA *Each topic is contained in an own subsection with the structure defined in Annex A.* *The following scenarios are not considered, since they are not compliant with RFC 3312 [5], Section 11:* • *Session Setup ~~3GPP UA~~ towards non-3GPP UA ~~supporting~~ making use of the SIP precondition extension and the SIP update extension, but not ~~supporting~~ making use of the SIP 100rel extension.* • *Session Setup ~~3GPP UA~~ towards non-3GPP UA ~~supporting~~ making use of the SIP preconditions extension and the SIP 100rel extension, but not ~~supporting~~ making use of the SIP update extension.* • *Session Setup towards non-3GPP UA making use of the SIP precondition extension, but not making use of the SIP 100rel extension and the SIP update extension.*
4.1	4.1.3	~~3GPP UA~~Session Setup towards a non-3GPP UA not ~~supporting~~ making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

4.1.1	4.1.3.1	Description of interworking issue
	4.1.3.2	~~Proposed Resolutions to interworking issue~~
4.1.2	4.1.3.2.1	Proposed Resolution B2B~~-~~UA
	4.1.3.2.1.1	Insertion of B2B~~-~~UA
		move detailed description and implications from old 4.1.2.4.1.1.1 to this Section
	4.1.3.2.1.2	Functionality of B2B~~-~~UA
		move rules for B2BUA from old 4.1.2.4.1.2 to this Section
	4.1.3.2.1.2.1	Description
	4.1.3.2.1.2.2	implications of the above solution~~Advantages~~
	4.1.3.2.1.2.3	~~Disadvantages~~
4.1.3	4.1.3.2.2	Proposed Resolution Modified end-to-end call flow
	4.1.3.2.2.1	Description
	4.1.3.2.2.2	implications of the above solution~~Advantages~~
	4.1.3.2.2.3	~~Disadvantages~~
4.2	4.1.2	Session Setup ~~3GPP UA~~ towards a non-3GPP UA ~~supporting the SIP 100rel extension, but~~ not ~~supporting~~ making use of the SIP preconditions extension and the SIP update extension
4.2.1	4.1.2.1	Description of interworking issue
	4.1.2.2	~~Flow diagram~~
	4.1.2.3	~~Impact of Identified interworking issue~~
	4.1.2.4	~~Proposed resolutions to interworking issue~~
4.2.2	4.1.2.4.1	Proposed Resolution 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
	4.1.2.4.1.1.1.2	implications of the above solution~~Advantages~~
	4.1.2.4.1.1.1.3	~~Disadvantages~~
	4.1.2.4.1.2	Functionality of B2B~~-~~UA
	4.1.2.4.1.2.1	Description
	4.1.2.4.1.2.2	implications of the above solution~~Advantages~~
	4.1.2.4.1.2.3	~~Disadvantages~~
4.2.3	4.1.2.4.2	Proposed Resolution Modified end-to-end call flow
	4.1.2.4.2.1	Description
	4.1.2.4.2.2	implications of the above solution~~Advantages~~
	4.1.2.4.2.3	~~Disadvantages~~
4.3	4.1.1	Session setup ~~3GPP UA~~ towards a non-3GPP UA ~~supporting the SIP 100rel extension and the SIP update extension, but~~ not ~~supporting~~ making use of the SIP preconditions extension
4.3.1	4.1.1.1	Description of interworking issue
	4.1.1.2	~~Proposed Resolutions to interworking issue~~
4.3.2	4.1.1.2.1	Proposed Resolution B2B~~-~~UA
	4.1.1.2.1.1	Insertion of B2B~~-~~UA
	4.1.1.2.1.2	Functionality of B2B~~-~~UA
	4.1.1.2.1.2.1	Description
	4.1.1.2.1.2.2	implications of the above solution~~Advantages~~
	4.1.1.2.1.2.3	~~Disadvantages~~
4.3.3	4.1.1.2.2	Proposed Resolution Modified end-to-end call flow
	4.1.1.2.2.1	Description
	4.1.1.2.2.2	implications of the above solution~~Advantages~~
	4.1.1.2.2.3	~~Disadvantages~~
5	4.2	Session Setup from Calling non-3GPP UA towards Called 3GPP UA

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

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

- *Session Setup from ~~N~~non-3GPP UA ~~supporting~~ making use of the SIP precondition extension and the SIP update extension, but not ~~supporting~~ making use of the SIP 100rel extension~~, to 3GPP UA.~~*
- *Session Setup from ~~N~~non-3GPP UA ~~supporting~~ making use of the SIP preconditions extension and the SIP 100rel extension, but not ~~supporting~~ making use of the SIP update extension~~, to 3GPP UA~~.*
- *Session Setup from non-3GPP UA making use of the SIP preconditions extension, but not making use of the SIP update extension and the SIP 100rel extension..*

5.1	4.2.3	Session Setup from ~~Non~~non-3GPP SIP UA not ~~supporting~~ making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension~~, to 3GPP UA~~
5.1.1	4.2.3.1	Description of interworking issue
	4.2.3.2	~~Proposed Resolutions to interworking issue~~
5.1.2	4.2.3.2.1	Proposed Resolution B2B~~-~~UA
	4.2.3.2.1.1	Insertion of B2B~~-~~UA
		move detailed description and implications from old 4.2.2.4.1.1.1 to this Section
	4.2.3.3.1.1.1	Static Insertion of B2B~~-~~UA
	4.2.3.3.1.1.1.1	Description
	4.2.3.3.1.1.1.2	implications of the above solution~~Advantages~~
	4.2.3.3.1.1.1.3	~~Disadvantages~~
	4.2.3.2.1.2	Functionality of B2B~~-~~UA
	4.2.3.2.1.2.1	Description
	4.2.3.2.1.2.2	implications of the above solution~~Advantages~~
	4.2.3.2.1.2.3	~~Disadvantages~~
5.1.3	4.2.3.2.2	Proposed Resolution Modified end-to-end call flow
	4.2.3.2.2.1	Description
	4.2.3.2.2.2	Advantages
	4.2.3.2.2.3	Disadvantages
5.2	4.2.2	Session Setup from ~~Non~~non-3GPP SIP UA ~~supporting the SIP 100rel extension, but~~ not making use of ~~supporting~~ the SIP preconditions extension and the SIP update extension, to 3GPP UA
5.2.1	4.2.2.1	Description of interworking issue
	4.2.2.2	~~Flow diagram~~
	4.2.2.3	~~Implications of Identified interworking issue~~
	4.2.2.4	~~Proposed resolutions to interworking issue~~
5.2.2	4.2.2.4.1	Proposed Resolution B2B~~-~~UA
	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
	4.2.2.4.1.1.1.2	implications of the above solution~~Advantages~~
	4.2.2.4.1.1.1.3	~~Disadvantages~~
	4.2.2.4.1.2	Functionality of B2B~~-~~UA
	4.2.2.4.1.2.1	Description
	4.2.2.4.1.2.2	implications of the above solution~~Advantages~~
	4.2.2.4.1.2.3	~~Disadvantages~~
5.2.3	4.2.2.4.2	Proposed Resolution Modified end-to-end call flow
	4.2.2.4.2.1	Description
	4.2.2.4.2.2	implications of the above solution~~Advantages~~
	4.2.2.4.2.3	~~Disadvantages~~
5.3	4.2.1	Session Setup from ~~Non~~non-3GPP UA ~~supporting the SIP 100rel extension and the SIP update extension, but~~ not making use of ~~supporting~~ the SIP preconditions extension~~, to 3GPP UA~~
5.3.1	4.2.1.1	Description of interworking issue
	4.2.1.2	~~Proposed Resolutions to interworking issue~~
5.3.2	4.2.1.2.1	Proposed Resolution B2B~~-~~UA
	4.2.1.2.1.1	Insertion of B2B~~-~~UA
	4.2.1.2.1.2	Functionality of B2B~~-~~UA
	4.2.1.2.1.2.1	Description
	4.2.1.2.1.2.2	implications of the above solution~~Advantages~~
	4.2.1.2.1.2.3	~~Disadvantages~~
5.3.3	4.2.1.2.2	Proposed Resolution Modified end-to-end call flow
	4.2.1.2.2.1	Description
	4.2.1.2.2.2	implications of the above solution~~Advantages~~
	4.2.1.2.2.3	~~Disadvantages~~
6		Implications of the Proposed Solutions
		Editor´s Note: This section shall summarise the findings within the corresponding subsections within Sections 4 and 5.
6.1		B2BUA
6.2		Modified end-to-end call flow

A	Annex A:	Interworking topic template
x	4.x	Topic Name
x.1	4.x.1	Description of interworking issue
	4.x.2	~~Flow diagram~~
	4.x.3	~~Impact of Identified interworking issue~~
x.2	4.x.4	Proposed Resolution yy ~~s~~ ~~to interworking issue~~
	4.x.4.y	~~Suggestion yy~~
	4.x.4.y.1	Description
	4.x.4.y.1	Implications of the above solution~~Advantages~~
	4.x.4.y.1	~~Disadvantages~~
	4.x.5	~~Preferred Suggestion~~
B	Annex B:	Mechanisms allowing optional Additions within SIP
C	Annex C:	Impacts of Session Setup Call flows ~~between rogue 3GPP UA to non-3GPP UA, if~~where SIP extensions mandated by 3GPP are not applied. *According to TS 24.229, a 3GPP UA shall use the SIP 100rel, update, and precondition extensions. The 3GPP UA shall abort the call set-up in manners detailed in the main part of this TR, if the non-3GPP UA is not making use of~~does not support or use~~ these extensions.* *This annex aims to explain why TS 24.229 introduces these restrictions. It details the consequences if a~~, in case a rogue~~ 3GPP UA ~~does~~ would not behave according to TS 24.229 and ~~does~~ would 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	C.1	Impacts of Session Setup Callflows from Calling ~~Rogue~~ 3GPP UA ~~to Called non-3GPP UA~~
C.1.1	C.1.3	Session Setup ~~Rogue 3GPP UA~~ towards non-3GPP UA not ~~supporting~~ making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension
C.1.1.1	C.1.3.1	Description of interworking issue
	C.1.3.2	~~Flow diagram~~
C.1.1.2	C.1.3.3	Impacts of Identified interworking issue
	C.1.3.3.1	User interaction
	C.1.3.3.2	Charging and Billing Implications
	C.1.3.3.3	SIP Media authorisation
	C.1.3.3.4	SIP Media allocation
	C.1.3.3.5	Fraudulent and security risks
C.1.2	C.1.2	Session Setup ~~Rogue 3GPP UA~~ towards non-3GPP UA ~~supporting the SIP 100rel extension, but~~ not ~~supporting~~ making use of the SIP preconditions extension and the SIP update extension
C.1.2.1	C.1.2.1	Description of interworking issue
	C.1.2.2	~~Flow diagram~~
C.1.2.2	C.1.2.3	Impacts of Identified interworking issue
C.1.3	C.1.1	Session Setup ~~Rogue 3GPP UA~~ towards non-3GPP UA ~~supporting the SIP 100rel extension and the SIP update extension, but~~ not ~~supporting~~ making use of the SIP preconditions extension
C.2	C.2	Imacts of Session Setup ~~Calling non-3GPP UA~~ towards Called ~~Rogue~~ 3GPP UA
C.2.1	C.2.3	Non-3GPP SIP UA not ~~supporting~~ making use of the SIP 100rel extension, the SIP preconditions extension and the SIP update extension~~, to rogue 3GPP UA~~
C.2.1.1	C.2.3.1	Description of interworking issue
	C.2.3.2	~~Flow diagram~~
C.2.1.2	C.2.3.3	Impacts of Identified interworking issue
C.2.2	C.2.2	Non-3GPP SIP UA ~~supporting the SIP 100rel extension, but~~ not ~~supporting~~ making use of the SIP preconditions extension and the SIP update extension~~, to rogue 3GPP UA~~
C.2.2.1	C.2.2.1	Description of interworking issue
	C.2.2.2	~~Flow diagram~~
C.2.2.3	C.2.2.3	Impacts of Identified interworking issue
C.2.3	C.2.1	Non-3GPP UA ~~supporting the SIP 100rel extension and the SIP update extension, but~~ not ~~supporting~~ making use of the SIP preconditions extension~~, to rogue 3GPP UA~~
D	Annex D:	Reference Call Flow from 3GPP UA to 3GPP UA
E	Annex E:	Scenarios without identified interworking issues
E.1	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.

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030543***
Sophia-Antipolis, France, 31 March – 04 April 2003

Title: Reply LS on Early UE handling
Response to: LS (S2-030965, N1-030475) from SA2
Work Item: Early UE

Source: CN1
To: SA2
Cc: RAN3

Contact Person:
Name: Robert Zaus
E-mail Address: robert.zaus@siemens.com

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1. Overall Description:

CN1 would like to thank SA2 for their LS on early UE handling and the attached TS 23.195, V1.0.0.

CN1 understood that SA2 is studying two different solutions for the transfer of IMEISV via the Gs interface:
- the use of the MS information procedure and
- the use of the Location Update Request message.

Concerning the request to CN1:

> "To clarify if (and, optionally, why) there is any restriction on the use of the MS Information procedure that would prevent the use of the MS Information Request to transfer IMEISV",

CN1 confirms that actually there is a restriction in TS 29.018, subclause 14.2, which requests the VLR to postpone the initiation of the MS Information procedure until the location update is finished. This restriction was introduced to GSM 09.18, V6.0.0, with CR 09.18-A001r1, Removing GPRS's degradation of CAMEL's any time interrogation procedure. Unfortunately, the exact reason for introducing the restriction is not known (and no justification is given on the cover sheet of the CR).

CN1 would like to point out that this restriction does not necessarily prevent the use of the MS Information procedure in the context of early UE handling. Since the procedures on the Gs interface use connectionless SCCP, the MSC/VLR can start the MS Information procedure immediately after the transmission of the BSSAP+Location-Update-Accept message. As a rule the MS Information procedure will be finished by the time the MS initiates the first CS related transaction after the completion of the GPRS attach or routing area update.

2. Actions:

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3. Date of Next TSG-CN1 Meetings:

CN1_30	19th – 23rd May 2003	San Diego, USA
CN1_31	25th – 29th August 2003	Sophia-Antipolis, France

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030547***
Sophia-Antipolis, France, 31 March – 04 April 2003

Title: **Reply LS on Radio Access Bearer for PS conversational testing**

Response to: LS (S4-030260/NP-030125/N1-030336) on Radio Access Bearer for PS conversational testing from SA4

Source: CN1

To: SA4

Cc: RAN 2, GERAN 2

Contact Person:
Name: Miguel A. Garcia
Tel. Number: +358 9299 3553
E-mail Address: Miguel.A.Garcia@ericsson.com

Attachments: None

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1. Overall Description:

CN1 thanks SA4 for the LS on Radio Access Bearer for PS conversational testing (S4-030260/NP-030125/N1-030336)

Although the subject of the LS is not within the expertise of the CN1 Working Group, CN1 would like to comment that the example referred in the LS, part of 3GPP TS 26.236, is only covering IPv4 protocol. Although this example is valuable for a general PS service, it is not applicable to the IP Multimedia CN Subsystem (IMS), because IMS uses exclusively IPv6 protocol.

CN1 believes that including examples that consider IPv6 addresses would be a valuable information. Especially, CN1 is interested in examples of usage of the b line (bandwidth) in SDP in order to synchronize these values with those included in the SIP/SDP examples in 3GPP TS 24.228.

2. Actions:

To SA4 group.

ACTION: CN1 asks SA4 group to review the above statements, and if deemed appropriate, consider the addition of examples that take into account IPv6 addresses to Annex B of 26.236.

3. Date of Next TSG-CN1 Meetings:

CN1_30	19th – 23rd May 2003	San Diego, USA
CN1_31	25th – 29th August 2003	Sophia-Antipolis, France

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030548***
Sophia-Antipolis, France, 31 March – 04 April 2003

Title:	**LS on IPv6 DNS server discovery in release 99 and release 4**
Response to:	-
Source:	CN1
To:	SA2, CN3
Cc:	CN

Contact Person:

Name:	Atle Monrad
Tel. Number:	+47 372 93 665
E-mail Address:	atle.monrad@ericsson.com

Attachments: N1-030426

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1. Overall Description:

Over the past 2 years, 3GPP has made ways to align support of IPv6 in GPRS according to IETF standards. During this time, how to provide the UE with DNS server address(es) has been an issue not fully resolved or completed. Originally the possibilities considered for the 3GPP UE to get IPv6 DNS server address(es) were the following methods:

- Method 1; DNS Discovery (the progress of this work in IETF is uncertain).
- Method 2; DHCPv6 (this has been work in progress in IETF, now completed).
- Method 3; Manual or OTA configuration (the exact mechanism is out of scope of 3GPP standardisation).

Note also that the PPP option for DNS IPv6 discovery (as available in IPv4) is not available according to current IETF RFCs or stable Internet Drafts.

CN1 and CN3 have already introduced a GPRS specific generic method to introduce the IPv6 DNS server address to release 5. This generic mechanism is described in 24.008, 27.060 and 29.061. The reasons for proposing the same generic GPRS specific mechanism for release 99 and release 4 are similar to what have already been approved for release 5.

The following is proposed:

- The same GPRS specific mechanism for DNS IPv6 server address discovery as available in release 5 is introduced for release 99 and release 4. This means that the same method to download the IPv6 DNS server address will be available from the introduction of IPv6 for vendors and operators taking IPv6 into operation prior to release 5.
- Ericsson offers to submit CRs to 24.008, 27.060 and 29.061 to the next CN1 and CN3 meetings in May to continue the discussion in the topic to allow UEs and GGSNs to use the method already approved in release 5 in cases where IPv6 is introduced in release 99 or release 4.

2. Actions:

To SA2 group.

ACTION: CN1 kindly asks SA2 group to consider and comment on the above proposal.

3. Date of Next TSG-CN1 Meetings:

CN1 #30 19^{th} – 23^{rd} of May 2003 San Diego, USA

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030426***
Sophia-Antipolis, France, 31 March – 04 April 2003

Source: **Ericsson**

Title: **Support of IPv6 in pre-rel 5 networks**

Agenda item: **6.1**

Document for: **INFORMATION / APPROVAL**

Background

Over the past 2 years, 3GPP has made ways to align support of IPv6 in GPRS according to IETF standards. During this time, how to provide the UE with DNS server addresses has been an issue not fully resolved/completed.
Originally the possibilities considered for the 3GPP UE to get IPv6 DNS server addresses were:

- DNS Discovery (this has been a work in progress in IETF, now this seems to be discontinued)
- DHCPv6 (this as well a work in progress then in IETF, now completed)
- Manual configuration (the exact mechanism is out of scope of 3GPP standardisation)

As the way forward for the DNS Discovery option (draft-ietf-ipv6-dns-discovery) was a bit uncertain it was agreed, driven by the emerging IMS, to introduce the possibility for a UE to request the IPv6 DNS server address via the PCO IE from release 5 onwards. Now it is clear that the I-D for the DNS Discovery or similar approaches will not be completed anytime soon within IETF, leaving the terminals with only two options in Release 99 and Release 4: DHCPv6 or manual configuration.

Yet these two options are not optimal for the following reasons:

> Support of DHCP:
> Support of a DHCP client in the terminal has so far been seen as unnecessary complicated as indicated by 'draft-ietf-ipv6-cellular-host'.
>
> Manual configuration:
> The use of manual configuration is seen as non-user friendly and imposes operational constraints on the operator and the network (e.g. reconfiguring the service network requires reconfiguration of terminals). It also introduces a non-common mechanism to provide UE with the information.

In rel-5, the situation changes, as it is possible for the UE to request IPv6 DNS server addresses via the PCO IE, which is a simpler and more efficient method than manual configuration or use of DHCP for the GPRS architecture. However, in order to support pre-release 5 terminals in a release 5 network, the operator will still have to support either or both the manual and DHCP configuration, which adds to the administrative burden.

Ericsson is concerned that this situation, where a proper method for discovery of IPv6 addresses for DNS servers are not in place from the beginning may delay or complicate the introduction and use of IPv6, and propose the following:

Proposed way forward

Ericsson proposes that the possibility to allow the use of DNS IPv6 server address discovery via the PCO IE is introduced earlier, preferably from the general introduction of IPv6 starting from Release 99. The support in the UE shall be optional but in order for the feature to be useful, GGSN shall provide the DNS server addresses via PCO, when configured by the Operator."

The following is proposed:

- The use of the PCO-IE for DNS IPv6 server address discovery is introduced from rel-99.
- Ericsson provides CRs to 24.008, 27.060 and 29.061 to the next joint CN1/3 meeting in May to allow UEs and GGSNs to use the method already approved in rel-5.
- Operators take the situation into account when IPv6 networks are set into operation.

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030549***
Sophia-Antipolis, France, 31 March – 04 April 2003

Title:	**LS on change of IP address due to privacy**
Response to:	-
Source:	CN1
To:	SA2
Cc:	-

Contact Person:

Name:	Atle Monrad
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Attachments: -

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1. Overall Description:

For release 5, CN1 has discussed the requirements written in 23.228 subclause 4.3.1 'Address management':

> According to the procedures defined in TS 23.060 [23], when a UE is assigned an IPv6 prefix, it can change the global IPv6 address it is currently using via the mechanism defined in RFC 3041 [16a], or similar means. When a UE is registered in the IM CN Subsystem, any change to the IP address that is used to access the IM CN subsystem shall trigger automatic registration in order to update the UE's IP address.

CN1 has identified one solution for the above requirement. This solution requires the UE to de-register from IMS, succeeded by a new IMS registration. When de-registering from IMS, the UE must close all ongoing dialogs and reinitiate them on a need basis after registering back.

2. Actions:

To SA2 group.

ACTION: CN1 kindly asks SA2 group to consider and comment on the proposed solution.

3. Date of Next TSG-CN1 Meetings:

CN1 #30	19th – 23rd of May 2003	San Diego, USA
CN1 #31	25th – 29th of August 2003	Sophia-Antipolis, France

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030560***
Sophia-Antipolis, France, 31 March – 04 April 2003

was ***Tdoc N1-030498***

Title:	LS on duration of ICID at IMS registration
Response to:	not applicable
Release:	Release 5
Work Item:	IMS
Source:	CN WG1
To:	SA WG5
Cc:	SA WG2

Contact Person:

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Attachments: none

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1. Overall Description:

TS 24.229 specifies the points where the ICID is generated as part of the P-Charging-Vector header within the Session Initiation Protocol, and also defines when the contents of the P-Charging-Vector header is stored for future use. A number of comments have arisen as to when a new ICID is generated, as opposed to when an existing ICID is expected to be reused.

It is understood that all messages used in dialogs initiated by an INVITE use the same ICID, which is a new ICID created at the time of the INVITE request, because these messages all relate to the same session.

Question 1:

It is currently unclear to WG CN1 which of two options apply to the duration of use of the ICID created and used as a result of a REGISTER method.

OPTION A

The ICID is generated as a result of the REGISTER request and response itself, and all subsequent events (e.g. NOTIFY request, MESSAGE request) for that registration use a new ICID.

OPTION B

The ICID generated as a result of the REGISTER request is used for all events used by the user within that registration. Therefore, for example a NOTIFY request, or MESSAGE request would use the same ICID as the original REGISTER request.

WG CN1 believe that both of these options fulfil the requirements for charging, however:

- we are not convinced that it is desirable for both options to be allowed, especially as the option essentially gets implemented at the P-CSCF, which in many cases will be in the visited network, whereas any desire for one option or the other will really exist at the operator of the home network.

Question 2.

If the ICID exists beyond the duration of the REGISTER method itself, is an ICID created for each new registration?

Question 3.

We would also draw your attention to the fact that a single REGISTER request can register multiple public user identities. Therefore SA5 should be aware that there is not a one to one correspondence between ICID and public user identity even if option B is adopted. Additionally, public user identities that relate to the same private user identity, but registered in different REGISTER messages will use different ICIDs. Is this regarded as an issue?

2. Actions:

To SA WG5 group.

ACTION: CN WG1 asks SA WG5 group to identify the appropriate mechanisms in relation to the duration of the registration ICID, and respond to CN WG1 with the answer.

3. Date of Next TSG-CN1 Meetings:

CN1_30	19^{th} – 23^{rd} May 2003	San Diego, USA
CN1_31	25^{th} – 29^{th} August 2003	Sophia-Antipolis, France

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030566***
Sophia-Antipolis, France, 31 March – 04 April 2003

Title: LS on Protocols over the Mt interface
Response to: LS (S2-030999) on Protocols over the Mt interface from SA WG2
Release: Rel-6
Work Item: PRESNC / IMS2

Source: CN WG1
To: SA WG2
Cc: TSG CN, SA WG1

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Attachments: None

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1. Overall Description:

CN WG1 thanks SA WG2 for their liaison statement on protocols over the Mt interface.

TSG-CN#19 has tasked CN1 to start working on the stage-3 aspects of the Mt interface. During CN1#29 meeting it was agreed, that CN1 will start the work on Mt interface related signalling based on the ongoing work in IETF SIPPING and SIMPLE Working Groups.

CN WG1 sees the need for a standardized solution for Presence and possibly Conference related data management, due to e.g.

- the multiple terminal scenarios
  (Is there a requirement that the server notifies the user about any type of data manipulation that has happened to the data related to the user?)
- data manipulation / management integration into application
- interoperability between applications.

During the IETF Workshop in San Francisco, in the beginning of this year it was stated, that a pure web-based access would not be the proper protocol to fulfil such a requirement.

2. Actions:

To SA WG2 group.

ACTION: CN WG1 kindly asks SA WG2 to study whether notifications from the AS to the UE are needed, due to manipulation that occurred to the users data.

3. Date of Next TSG-CN1 Meetings:

CN1_30	19th – 23rd May 2003	San Diego, USA
CN1_31	25th – 29th August 2003	Sophia-Antipolis, France

3GPP TSG-CN1 Meeting #29 ***Tdoc N1-030567***
Sophia-Antipolis, France, 31 March – 04 April 2003

Title: LS on DRX parameters update
Response to: LS (S2-030958) on Liaison on DRX parameter from SA2
Release: Rel-5

Source: CN1
To: SA2
Cc: GERAN1, GERAN2, RAN1, RAN2, RAN3, T2

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Attachments: None.

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1. Overall Description:

CN1 thanks SA2 for their LS in S2-030958 regarding the addition of DRX parameter IE to Session Management procedures.

CN1 agrees that a mechanism for the UE to modify its DRX parameter in a flexible manner is useful for real time services and would like to remind SA2 that from Rel-5 onwards RAU can be used for this purpose.

CN1 would like to point out that DRX changes are critical since an un-synchronization between the UE and the network would lead to a situation when the UE is not pageable.

Both solutions, usage of RAU procedure and SM procedures, were discussed. At the moment RAU procedures seem more feasible than SM procedures from technical view point and also from specification impact.

Regarding the release where the changes would be applied, at least one company stated that for Rel-5 only the RAU procedure would be acceptable.

2. Actions:

To SA2 group.

ACTION: CN1 asks SA2 group to take this into account when making a decision.

3. Date of Next TSG-CN1 Meetings:

CN1_30	19th – 23rd May 2003	San Diego, USA
CN1_31	25th – 29th August 2003	Sophia-Antipolis, France