3GPP TSG CN Plenary Meeting #18 **NP-020607**
4th - 6th December 2002. New Orleans, USA.

Source: **TSG CN WG3**

Title: **LSs Approved and sent by CN3 since NP#17**

Agenda item: **6.3.1**

Document for: **INFORMATION**

This document contains 6 LSs that have been approved by TSG CN WG3 after CN#17, and are forwarded to TSG CN Plenary meeting #18 for information.

Tdoc #	Tdoc Title	LS to	LS cc	Attachment
N3-020838	Re. LS on CS data services for GERAN Iu-mode	SA2, GERAN2, CN1, CN4	-	N3-020786
N3-020860	LS on SCUDIF and Lawful Interception	SA3-LI	-	N3-020816
N3-020868	LS on SBLP control of DiffServ	SA2	SA5	-
N3-020878	LS on Interworking between SIP/SDP and BICC/ISUP	CN, ITU-T Ad Hoc	-	N3-020813
N3-020881	LS on Review of TR on 3GPP SIP Profile interworking	CN1	-	N3-020880
N3-021012	Reply LS on RTCP overhead in SDP bandwidth parameter	SA4	SA2	

3GPP TSG-CN WG3#25 **Tdoc N3-020786**
Miami, USA. 23rd - 27th September. 2002

Source: **SA2**

Title: **LS on CS data services for GERAN Iu-mode**

Agenda item: **7**

Document for: **INFORMATION**

3GPP TSG SA WG2 #26 ***S2-022625***
Toronto, Canada, 18th - 23rd August 2002

Response to: **(N3-020740/S2-022428) LS on CS data services for GERAN Iu-mode**

Source: **SA 2**
To: **CN3, GERAN 2, CN1**
Cc:

Contact Person:
Name: **Chris Pudney**
E-mail Address: **chris.pudney@vodafone.co.uk**

Attachments: **none**

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1. Overall Description:

S2 thanks CN3 for their LS on CS data services for GERAN Iu-mode and their efforts in the completion of the HSCSD support.

S2 accepts CN3's proposal to select:
- option 1 for transparent CS data services and
- option 3 for non-transparent CS data services

Despite the large size of CN 3's document, SA 2 note that many handover cases are not described. SA 2 guess that these handover cases will not cause fundamental problems to CN3's proposal, however, SA2 believe that the GERAN Iu mode standards will need to specify how the following handover scenarios are handled:

a) Intra BSC handover from an A/Gb mode cell to an Iu mode cell
b) Intra BSC handover from an Iu mode cell to an A/Gb mode cell
c) Intra MSC, inter BSC handover from an A/Gb mode cell to an Iu mode cell
d) Intra MSC, inter BSC handover from an Iu mode cell to an A/Gb mode cell
e) case "d" but in a relay MSC following a handover from an Iu mode anchor MSC
f) case "c" but in a relay MSC following a handover from a A/Gb mode anchor MSC
g) intra relay MSC handover from Iu mode BSC to Iu mode RNC following a handover from an A/Gb mode anchor MSC.

Further, does CN 3's proposal have any impact on the release 4 MSC server/MGW architecture? For example, does the MSC server - MGW interface signalling need upgrading?

2. Actions:

To CN 3 and GERAN 2:

a) To verify that the above handover cases do not cause problems.
b) To ensure that this MSC server - MGW interface supports CN3's proposal.

To CN 1 and GERAN 2:

To update their specifications according to these decisions.

3. Date of Next SA2 Meetings:

SA2#27	14th – 18th October 2002	Beijing

3GPP TSG-CN WG3#25 **Tdoc N3-020838**
Miami, USA. 23rd - 27th September. 2002

Title: **Reply LS on CS data services for GERAN Iu-mode**
Response to: **LS (N3-020786/ S2-022625) on CS data services for GERAN Iu-mode from SA2**

Source: **CN3**
To: **SA2, GERAN2, CN1, CN4**

Contact Person:
Name: Thomas Belling
Tel. Number: +49 89 722 47315
E-mail Address: Thomas.Belling@icn.siemens.de

Attachments: N3-020786 (LS on CS data services for GERAN Iu-mode from SA2)

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1. Overall Description:

CN3 would like to thank SA2 for the guidance provided in the LS on CS data services for GERAN Iu-mode.

SA2 suggested that a number of hand-over cases and the related impacts on the bearer independent architecture should be considered. However, CN3 is only responsible for user-plane aspects, and the listed hand-over cases and related impacts only affect the control plane.

CN3 would therefore like to suggest that the impacts of the hand-over cases are investigated in the working groups where the appropriate expertise resides, i.e. in GERAN2, CN1 and CN4.

2. Actions:

To GERAN2, CN1 and CN4 group.

ACTION: CN3 would like to ask GERAN2, CN1 and CN4 group to investigate possible impacts of the hand-over cases listed by SA2 on their specifications and inform SA2 about the results.

3. Date of Next CN3 Meetings:

CN3 #26 11th Nov. – 15th Nov. 2002 Bangkok, Thailand

3GPP TSG-CN WG3#25 **Tdoc N3-020881**
Miami, USA. 23rd - 27th September. 2002

Title: **LS on Review of TR on 3GPP SIP Profile interworking**

Release: **Release 6**

Source: **CN3**

To: **CN1**

Contact Person:

Name: Thomas Belling

Tel. Number: +49 89 722 47315

E-mail Address: Thomas.Belling@icn.siemens.de

Attachments: N3-020880 TR on 3GPP SIP Profile interworking.

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1. Overall Description:

CN3 would like to inform CN1 about its work on the TR on the "Signalling Interworking between the 3GPP Profile of SIP and non-3GPP SIP Usage", and would like to invite CN1 experts to review the TR.

The TR focuses on interworking scenarios between a SIP UA which complies to the 3GPP profile of SIP and therefore supports the SIP precondition, update and 100rel extensions, and an external SIP UA which does not support some or all of these extensions.

CN3 has agreed to use the present version of the TR as the basis for future contributions, and CN3 also agreed on the scope and structure of the document, but the proposed interworking solutions have not yet been discussed in detail.

In order to keep the work co-ordinated with CN1, CN3 would welcome CN1´s opinion, in particular with regard to the following questions:

- Are the scenarios considered within the TR valid?
- Is it correct that the 3GPP-compliant SIP UA will abort all calls in case of the considered scenarios?

CN3 would appreciate a review of the TR by CN1´s experts.

In order to accomplish this review in a timely manner, CN3 would like to propose the following procedure:

1. An agreed version of the TR is provided for a drafting session with CN1 experts prior to CN1´s Munich Rel.6 ad-hoc meeting.
2. The output of the drafting session is presented for endorsement by CN1 at CN1´s Munich Rel.6 ad-hoc meeting.
3. The updated version of the TR is presented for agreement by CN3 at the Bangkok meeting.

2. Actions:

To CN1 group.

ACTION: CN3 asks the CN1 group to discuss if CN1 is willing to participate within the suggested review, and in this case to either endorse the suggested procedure or suggest alternatives.

3. Date of Next CN3 Meetings:

CN3 #26	11th Nov. – 15th Nov. 2002	Bangkok, Thailand

3GPP draft TR ab.cde V0.2.0 (2002-07)

Technical Report

3rd Generation Partnership Project;
Technical Specification Group Core Network
Signalling Interworking between the 3GPP Profile of 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 callflows 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 considered SIP network entities external to the 3GPP network may feature different SIP capabilities, such as the support of arbitrary SIP packages

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

Preconditions, RFC 3312 [5]

Update, RFC 3311 [7]

100rel, RFC 3262 [6]The present document does not make any a-priory assumptions where a possible interworking is performed within the 3GPP network. Any SIP network entity within the 3GPP network may take part in the interworking. The network entities that may become involved in a certain interworking topic are identified for each of these topics separately.

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

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

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

It is forseen 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"

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]

User Agent Client (UAC): A user agent client is a logical entity that creates a new request, and then uses the client transaction state machinery to send it. The role of UAC lasts only for the duration of that transaction. In other words, if a piece of software initiates a request, it acts as a UAC for the duration of that transaction. If it receives a request later, it assumes the role of a user agent server for the processing of that transaction.
Within the present TR, UAC is always to be understood with respect to the "INVITE" request.

User Agent Server (UAS): A user agent server is a logical entity that generates a response to a SIP request. The response accepts, rejects, or redirects the request. This role lasts only for the duration of that transaction. In other words, if a piece of software responds to a request, it acts as a UAS for the duration of that transaction. If it generates a request later, it assumes the role of a user agent client for the processing of that transaction.
Within the present TR, UAS is always to be understood with respect to the "INVITE" request.

User Agent (UA): A logical entity that can act as both a user agent client and user agent server.

3.2 Abbreviations

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

UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
B2B UA	Back-to-back UA

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 3GPP UAC to 3GPP UAS

The interworking between 3GPP UAC to 3GPP UAS is as defined in accordance with 3GPP TS 24.228 and 3GPP TS 24.229. No interworking issues exist, but the flow diagram is depicted here for comparison.

4.1.1 Flow diagram

Notes:

1. The message flow between the 3GPP UEs is depicted.
2. SIP proxies are omitted with the exception of the P-CSCFs, which are depicted in this callflow but will be omitted in most subsequent callflows.
3. The TRYING response (2), (3), (4) to the INVITE message (1) is send hop-by-hop, as indicated in this flow diagram. It will be omitted in all subsequent flow diagrams for simplicity. All other messages are generated by the 3GPP UEs.
4. Most parts of the SIP messages are omitted for simplicity. Only the “require”, “supported” and “allowed” header fields are depicted.
5. Most parts of the SDP are omitted for simplicity. Only the SDP preconditions for one medium are depicted. There may be an arbitrary number of number of media, each with own preconditions
6. The P-CSCF inspects each SDP, in order to identify offer/answer pairs [8]. The P-CSCF may modify the QoS authorisation (6) when processing each SDP answer.
7. The use of the “183 Session Progress” (5) provisional response is optional according to IETF specifications. However, if the SIP precondition extension [6] is used and SDP with mandatory preconditions that the UAS is not capable of meeting unilaterally is included in the initial INVITE (1), a 101-199 provisional response, such as the “183 Session Progress”, is required to transport the SDP answer including the mandated “confirmation status” SDP attribute (Ref. [6], Section 6). Moreover, the “180 Ringing” message is not suitable because the user should not be alerted until the preconditions are met.
8. It is optional to convey a new SDP offer/answer within the PRACK (8) and OK(PRACK) (9) messages. A 3GPP UAC will refrain from generating a new SDP offer within PRACK (8), if it does not wish to further restrict the set of codecs selected within the first offer/answer pair.
9. According to IETF Ref. 6, Section 5, the UAS should start the resource reservation (10) immediately after having send the SDP answer within of the “183 Session Progress” (5) provisional response. However, a 3GPP UAS may expect a second SDP offer, and thus wait for the next message until starting resource reservation. The 3GPP UAS 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” (5) provisional response.

10. The use of the “Update” Request (11) 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 (5)) is reached or surpassed, the UA must send an SDP offer reflecting the new current status (Ref. [5], Section 7). The only suitable way to convey the new SDP offer/answer may be via an UPDATE request.

11. If the “UPDATE” (11) request is not used, the subsequent” OK(UPDATE)” (13) response is also not present.

12. The use of the “180 Ringing” provisional response (14) 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)” (18) response.

13. The “UPDATE” (11) request is used to convey the GPRS charging ID from P-CSCF A to S-CSCF-A. [1]

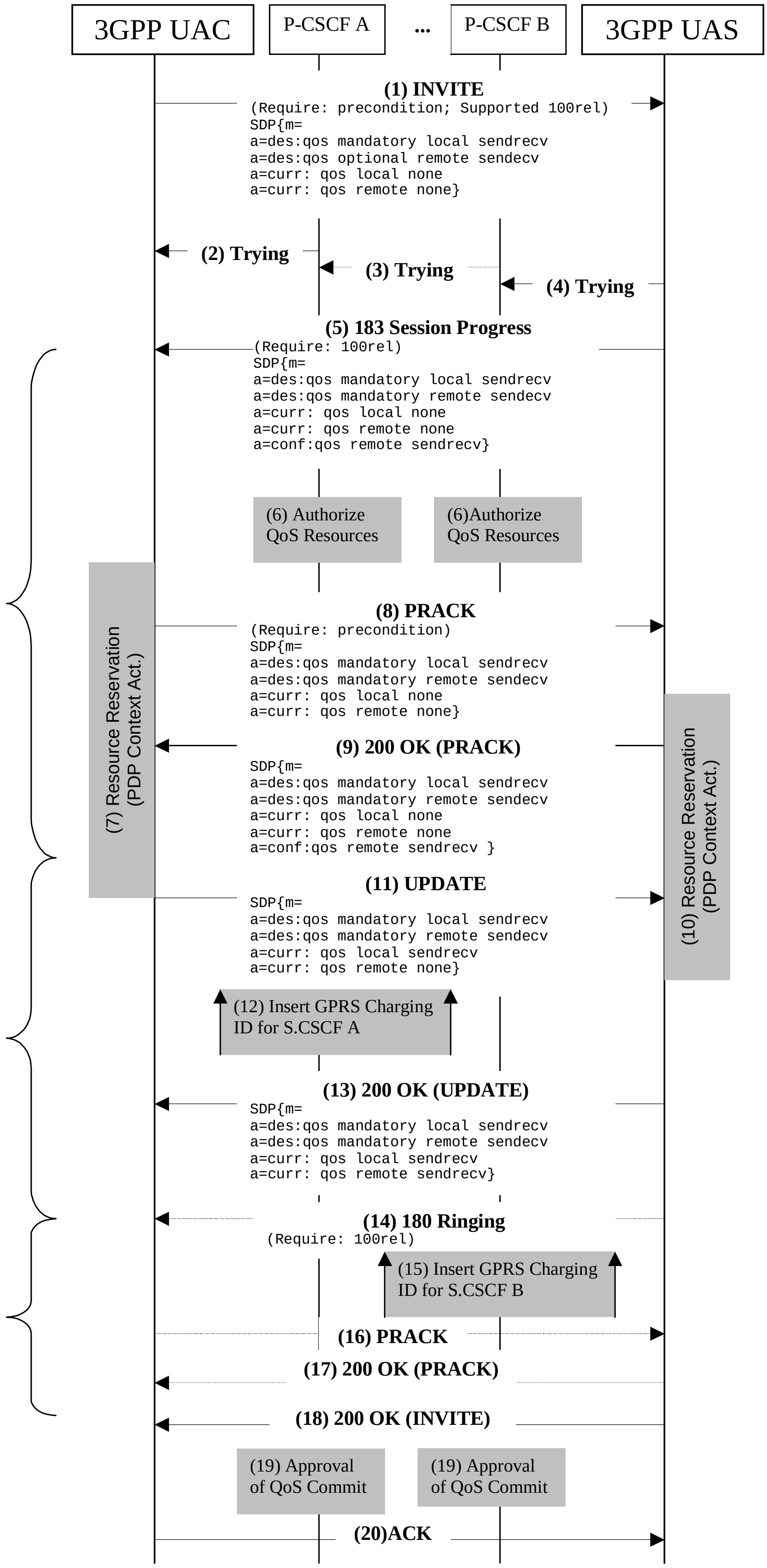

Figure 4.1.1/1: 3GPP UAC to 3GPP UAS Call flow

4.1.2 Impact of interworking issue

There is no interworking issue. This call flow is depicted for reference only.

4.2 3GPP UAC to non-3GPP-compliant UAS

4.2.1 3GPP UAC to non-3GPP-compliant UAS not supporting the SIP 100rel extension, but supporting the SIP precondition extension

According to the SIP preconditions specification [5], Section 10, this scenario is not possible.

4.2.2 3GPP UAC to non-3GPP-compliant UAS supporting the SIP 100rel extension and the SIP update extension, but not supporting the SIP preconditions extension

4.2.2.1 Description of interworking issue

The call fails, as detailed in Section 4.2.4.2

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

A fixed UE supporting the SIP update extension may use features of this extension for purposes not related to the SIP precondition extension. As a result, various extra messages may be inserted into the callflow.

4.2.2.2 Proposed Resolutions to interworking issue

A B2B UA is used.

4.2.2.2.1 Insertion of B2B UA

How the B2B UA is inserted is discussed within Section 4.2.4.4.1

4.2.2.2.2 Functionality of B2B UA

4.2.2.2.2.1 Pass Update Transparently

4.2.2.2.2.1.1 Description

The functionality of the B2B UA is as discussed in Section 4.2.4.4.2

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

4.2.2.2.2.1.2 Advantages

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

4.2.2.2.2.1.3 Disadvantages

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

4.2.3 3GPP UAC to non-3GPP-compliant UAS supporting the 100rel SIP extension and the SIP preconditions extension, but not supporting the SIP update extension

The call fails, as detailed in Annex C.2.3.

The usage of the precondition extension without the update extension is considered a very unrealistic scenario, and therefore no interworking solutions for this scenario are considered.

4.2.4 3GPP UAC to non-3GPP-compliant UAS supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension

4.2.4.1 Description of interworking issue

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

4.2.4.2 Flow diagram

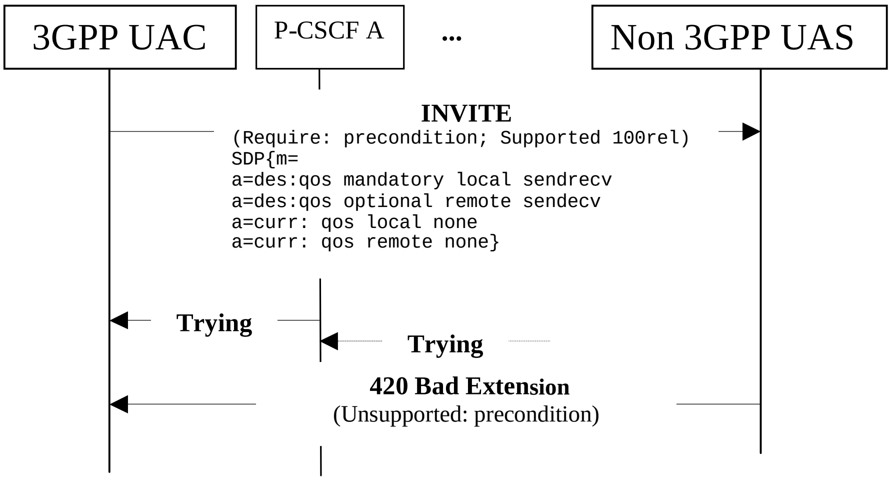

Figure 4.2.4.2/1: 3GPP UAC to non-3GPP-compliant UAS supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension

4.2.4.3 Impact of Identified interworking issue

The call fails.

4.2.4.4 Proposed resolutions to interworking issue

A B2B UA is used.

4.2.4.4.1 Insertion of B2B UA

4.2.4.4.1.1 Static Insertion of B2B UA

4.2.4.4.1.1.1 Description

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

4.2.4.4.1.1.2 Advantages

Functionality may be combined with other functionality required at edge of network, e.g. NAT

4.2.4.4.1.1.3 Disadvantages

Unnecessary processing load and hardware costs.

4.2.4.4.2 Functionality of B2B UA

4.2.4.4.2.1 Suggestion A

4.2.4.4.2.1.1 Description

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

The UAS may also send no “Session progress” message or “Ringing” message and include the SDP answer in the “200 OK(INVITE)” instead. This case is discussed in Section 4.2.5.2.2

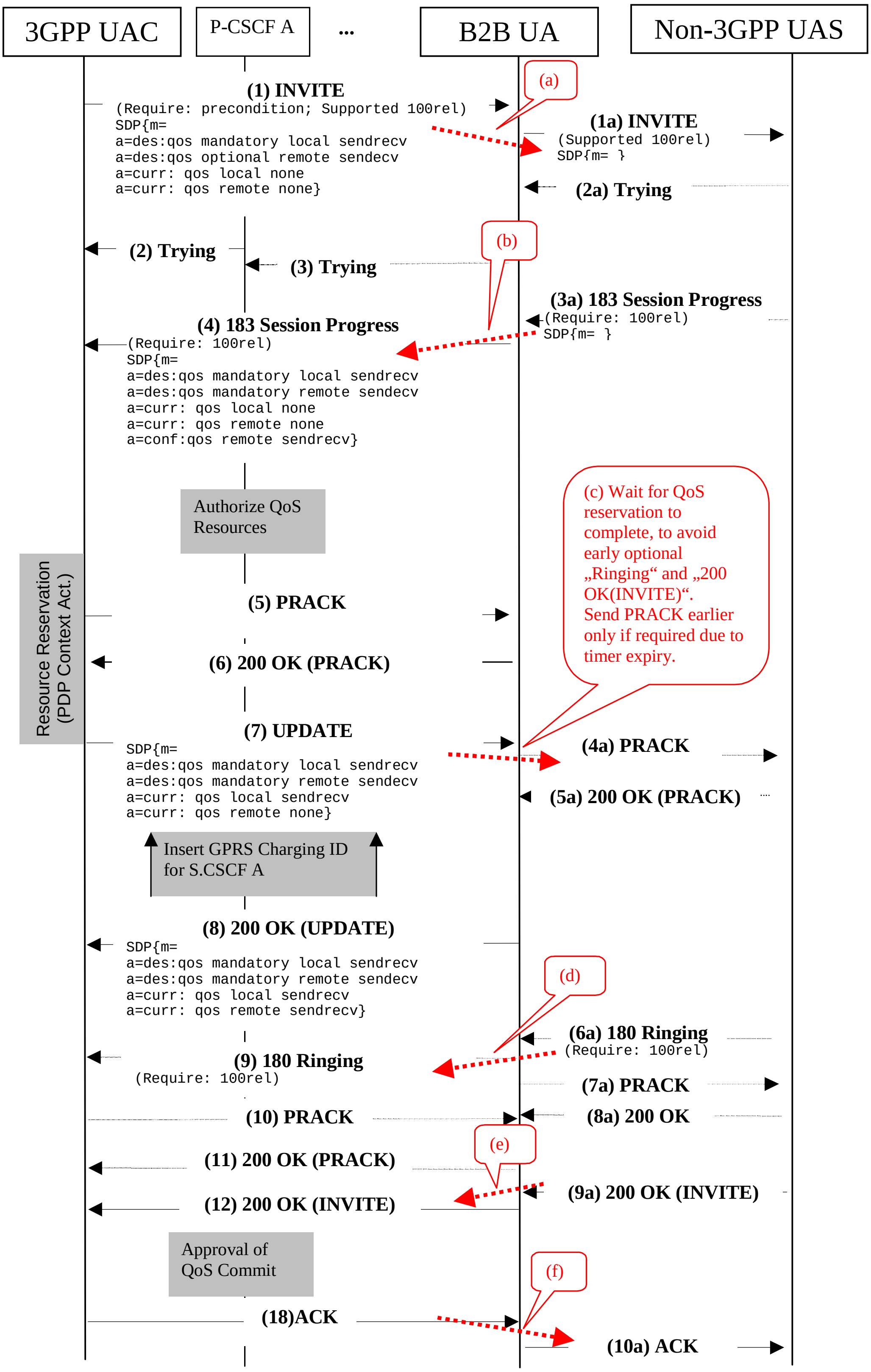

Figure 4.2.4.4.2.1/1: Functionality of B2B UA connecting 3GPP UAC to non-3GPP-compliant UAS supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. UAS includes SDP answer in 183 "Session Progress". UAC sends no second SDP offer

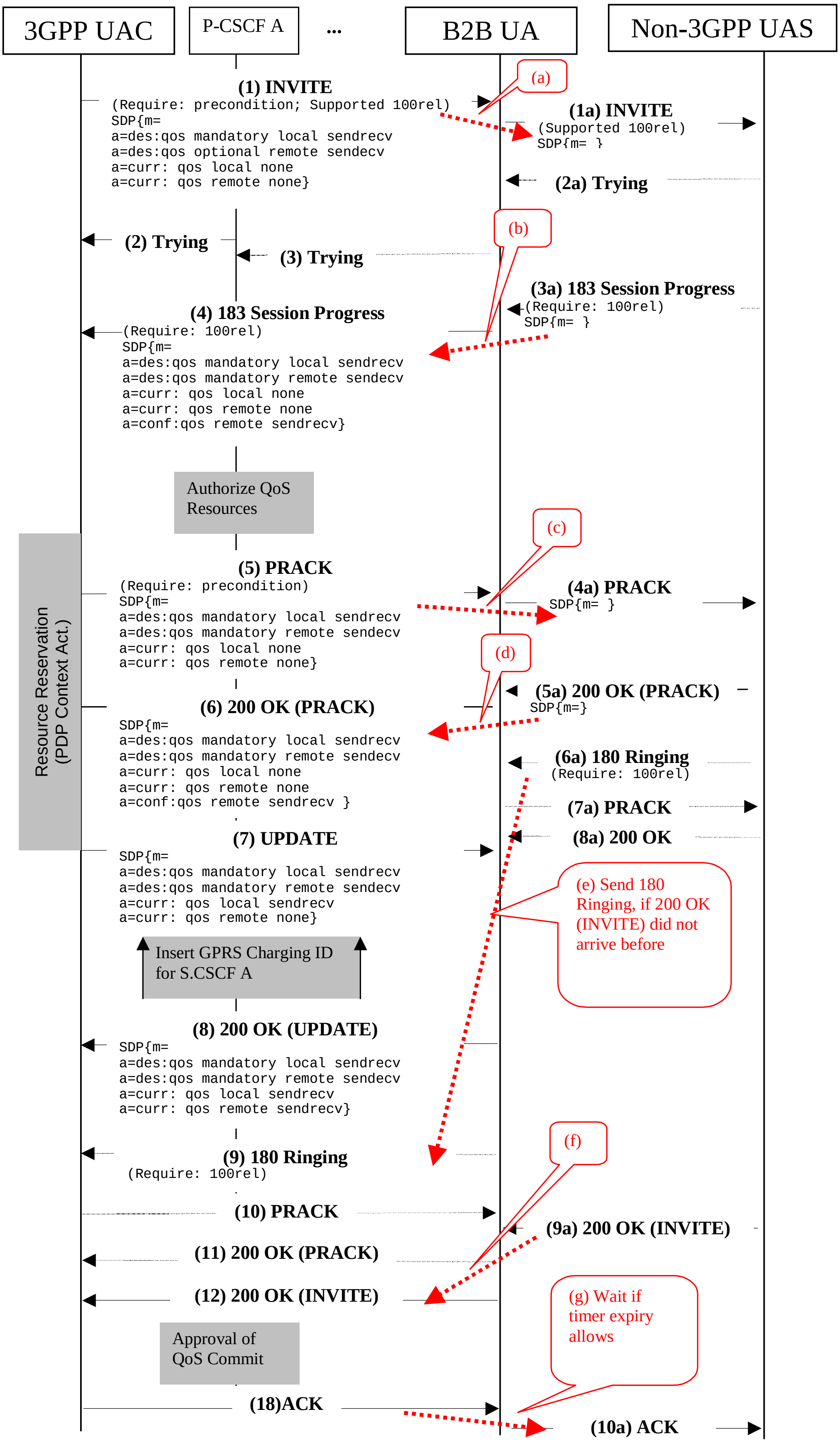

Figure 4.2.4.4.2.1/2: Functionality of B2B UA connecting 3GPP UAC to non-3GPP-compliant UAS supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. UAS includes SDP answer in 183 "Session Progress". UAC sends second SDP offer

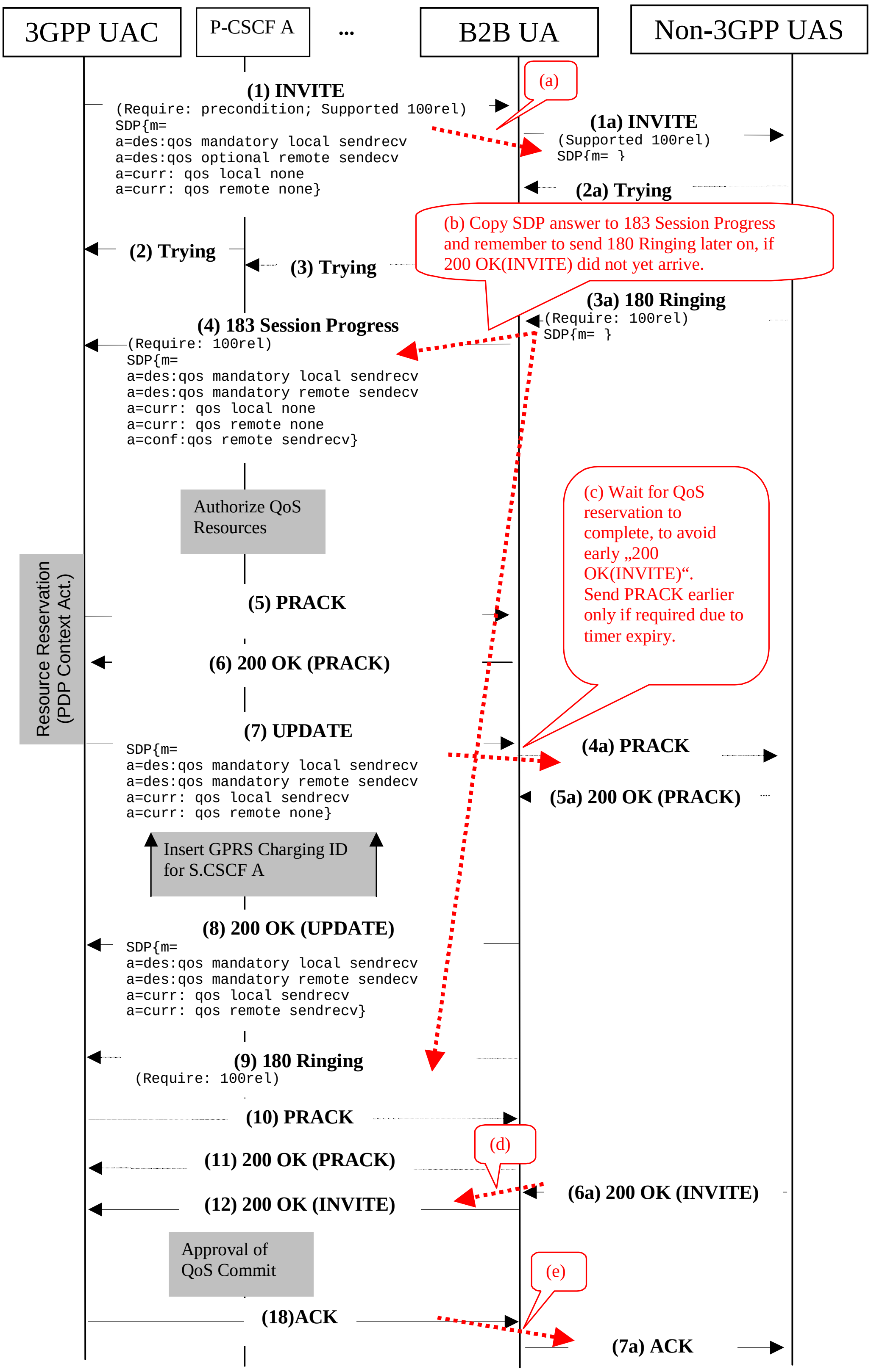

Figure 4.2.4.4.2.1/3: Functionality of B2B UA connecting 3GPP UAC to non-3GPP-compliant UAS supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. UAS includes SDP answer in 180 "Ringing". UAC sends no second SDP offer

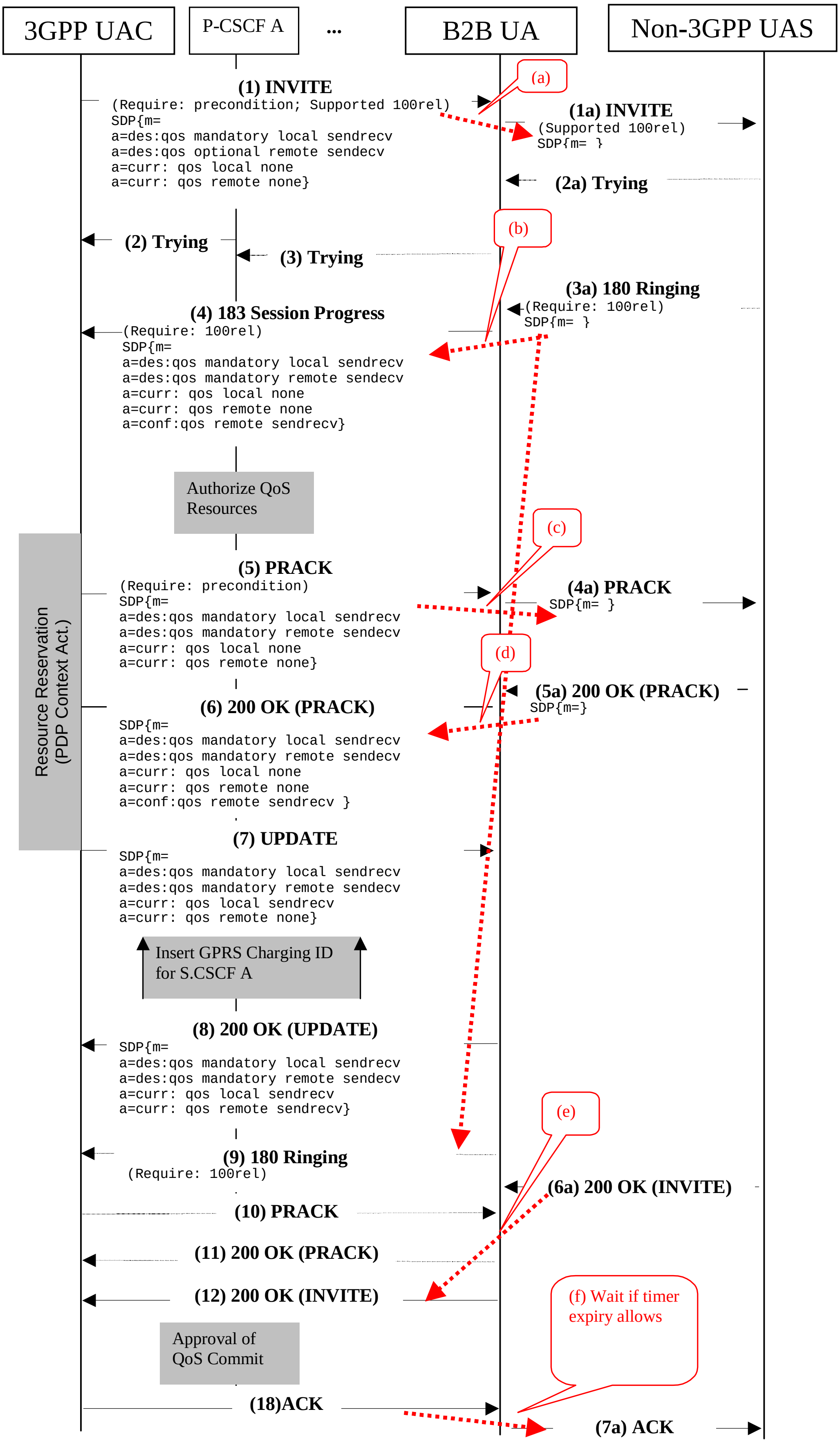

Figure 4.2.4.4.2.1/4: Functionality of B2B UA connecting 3GPP UAC to non-3GPP-compliant UAS supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension. UAS includes SDP answer in 180 “Ringing”. UAC sends second SDP offer

4.2.4.4.2.1.2 Advantages

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

4.2.4.4.2.1.3 Disadvantages

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

4.2.5 3GPP UAC to non-3GPP-compliant UAS not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

4.2.5.1 Description of interworking issue

The call fails, as detailed in Section 4.2.4.2

4.2.5.2 Proposed Resolutions to interworking issue

A B2B UA is used.

4.2.5.2.1 Insertion of B2B UA

How the B2B UA is inserted is discussed within Section 4.2.4.4.1

4.2.5.2.2 Functionality of B2B UA

4.2.5.2.2.1 Suggestion A

4.2.5.2.2.1.1 Description

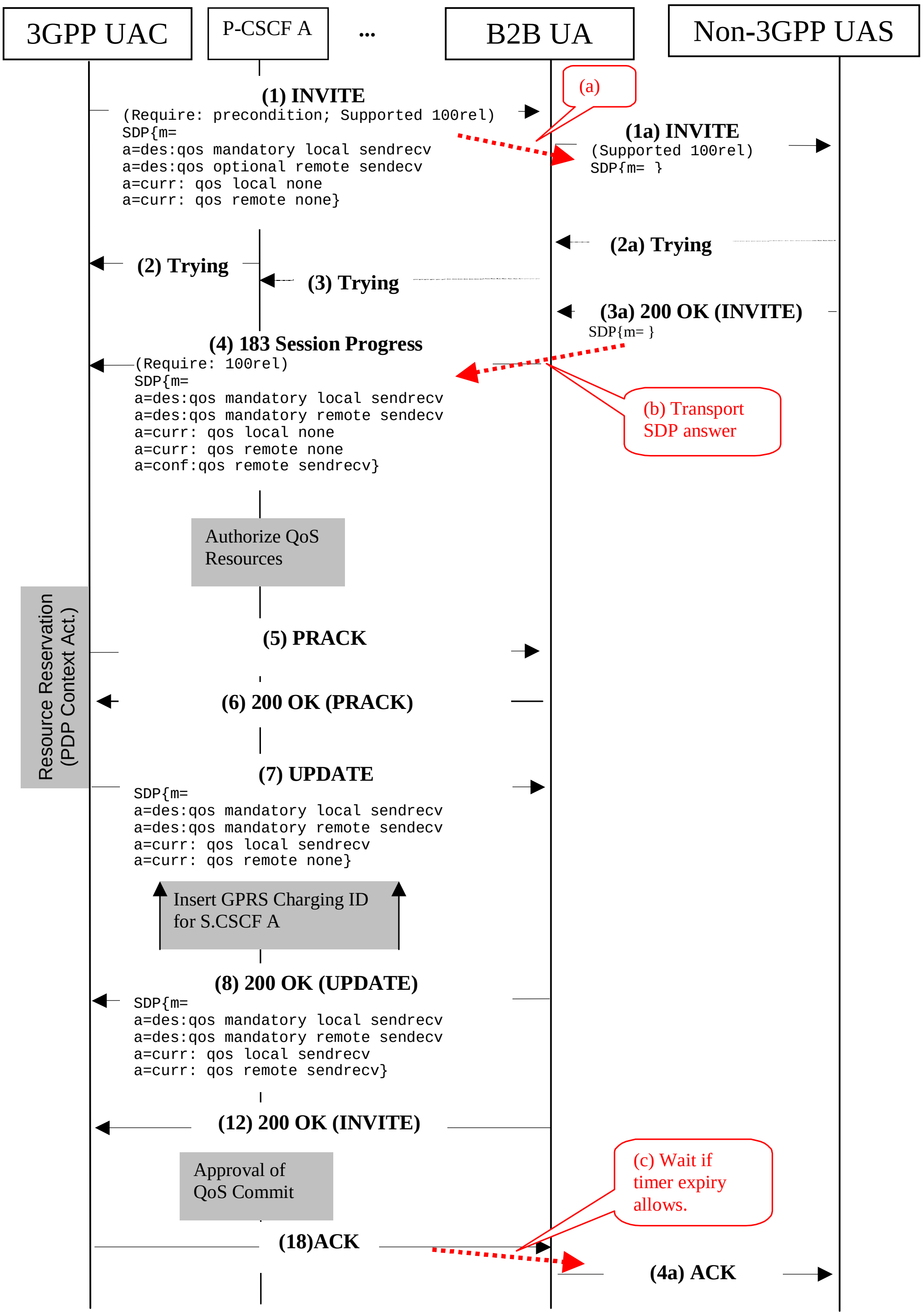

Figure 4.2.5.2.2.1/1: Functionality of B2B UA connecting 3GPP UAC to non-3GPP-compliant UAS not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension. UAC sends no second SDP offer

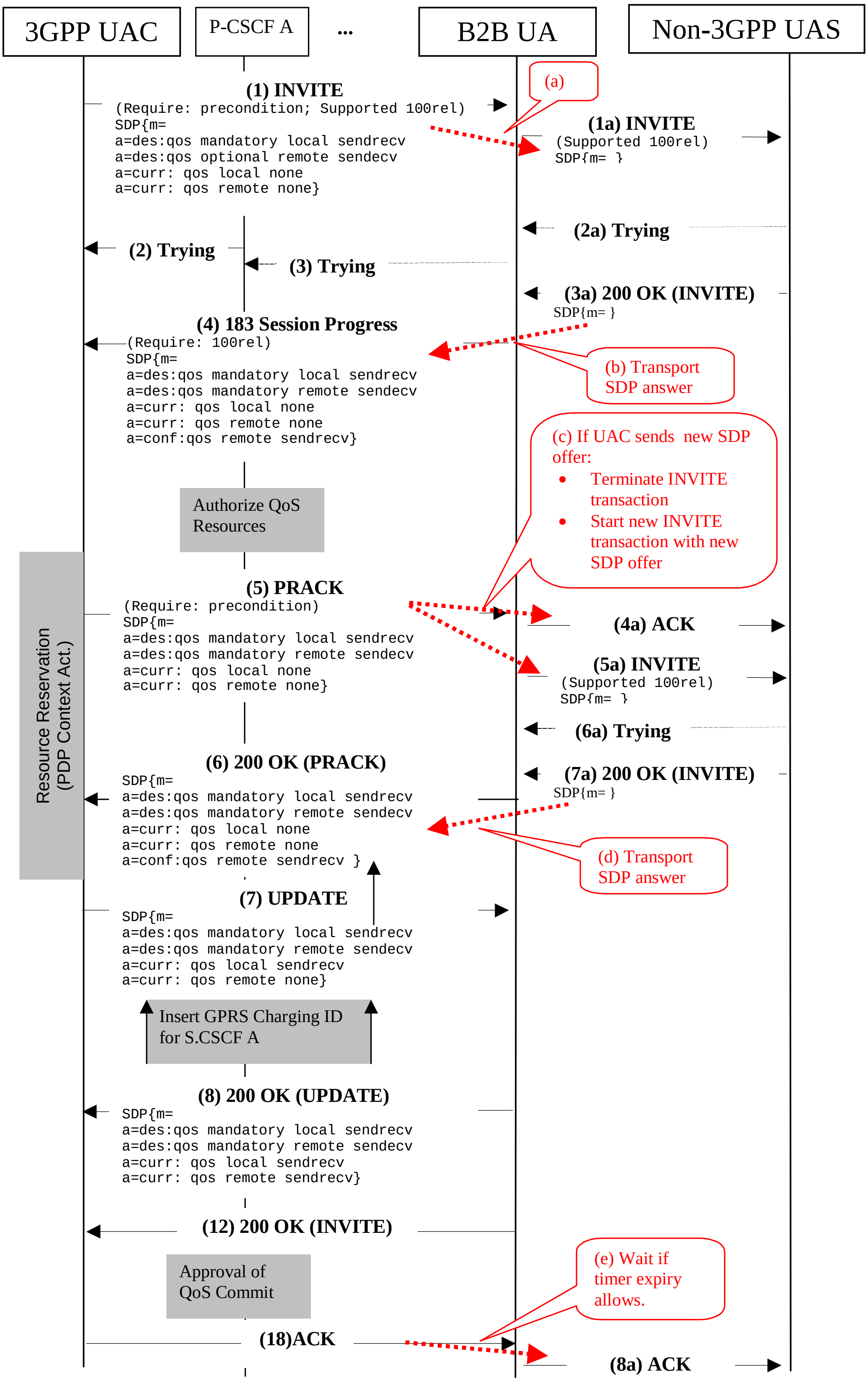

Figure 4.2.5.2.2.1/2: Functionality of B2B UA connecting 3GPP UAC to non-3GPP-compliant UAS not supporting the SIP preconditions extension, the SIP update extension and the SIP 100rel extension. UAC sends second SDP offer

4.2.5.2.2.1.2 Advantages

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

4.2.5.2.2.1.3 Disadvantages

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

4.3 Non-3GPP-compliant UAC to 3GPP UAS

4.3.1 Non-3GPP-compliant UAC not supporting the SIP 100rel extension, but supporting SIP preconditions extension, to 3GPP UAS

According to the SIP preconditions specification [5], Section 10, this scenario is not possible.

4.3.2 Non-3GPP-compliant UAC supporting the SIP 100rel extension and the SIP update extension, but not supporting the SIP preconditions extension, to 3GPP UAS

4.3.2.1 Description of interworking issue

The call fails, as detailed in Section 4.3.4.2

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

A fixed UE supporting the SIP update extension may use features of this extension for purposes not related to the SIP precondition extension. As a result, various extra messages may be inserted into the callflow.

4.3.2.2 Proposed Resolutions to interworking issue

A B2B UA is used.

4.3.2.2.1 Insertion of B2B UA

How the B2B UA is inserted is discussed within Section 4.3.4.4.1

4.3.2.2.2 Functionality of B2B UA

4.3.2.2.2.1 Pass Update Transparently

4.3.2.2.2.1.1 Description

The functionality of the B2B UA is as discussed in Section 4.3.4.4.2

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

4.3.2.2.2.1.2 Advantages

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

4.3.2.2.2.1.3 Disadvantages

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

4.3.3 Non-3GPP-compliant UAC supporting the 100rel SIP extension and the SIP preconditions extension, but not supporting the SIP update extension, to 3GPP UAS

The call fails, as detailed in Annex C.3.3.

The usage of the precondition extension without the update extension is considered a very unrealistic scenario, and therefore no interworking solutions for this scenario are considered.

4.3.4 Non-3GPP-compliant SIP UAC supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UAS

4.3.4.1 Description of interworking issue

Since the 3GPP UAS mandates, that the support of the SIP precondition extension is indicated in the SIP INVITE request, the call will be aborted.

4.3.4.2 Flow diagram

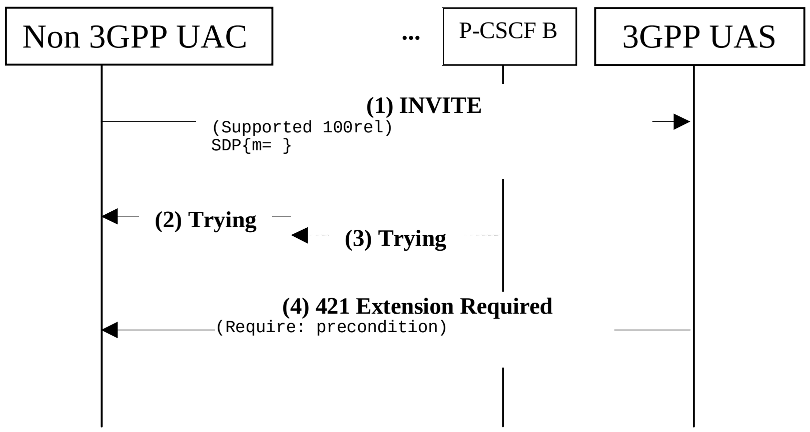

Figure 4.3.4.2/1: Non-3GPP-compliant SIP UAC not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UAS.

4.3.4.3 Implications of Identified interworking issue

The call fails.

4.3.4.4 Proposed resolutions to interworking issue

A B2B UA is used.

4.3.4.4.1 Insertion of B2B UA

4.3.4.4.1.1 Static Insertion of B2B UA

4.3.4.4.1.1.1 Description

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

4.3.4.4.1.1.2 Advantages

Functionality may be combined with other functionality required at edge of network, e.g. NAT

4.3.4.4.1.1.3 Disadvantages

Unnecessary processing load and hardware coasts.

4.3.4.4.2 Functionality of B2B UA

4.3.4.4.2.1 Suggestion A

4.3.4.4.2.1.1 Description

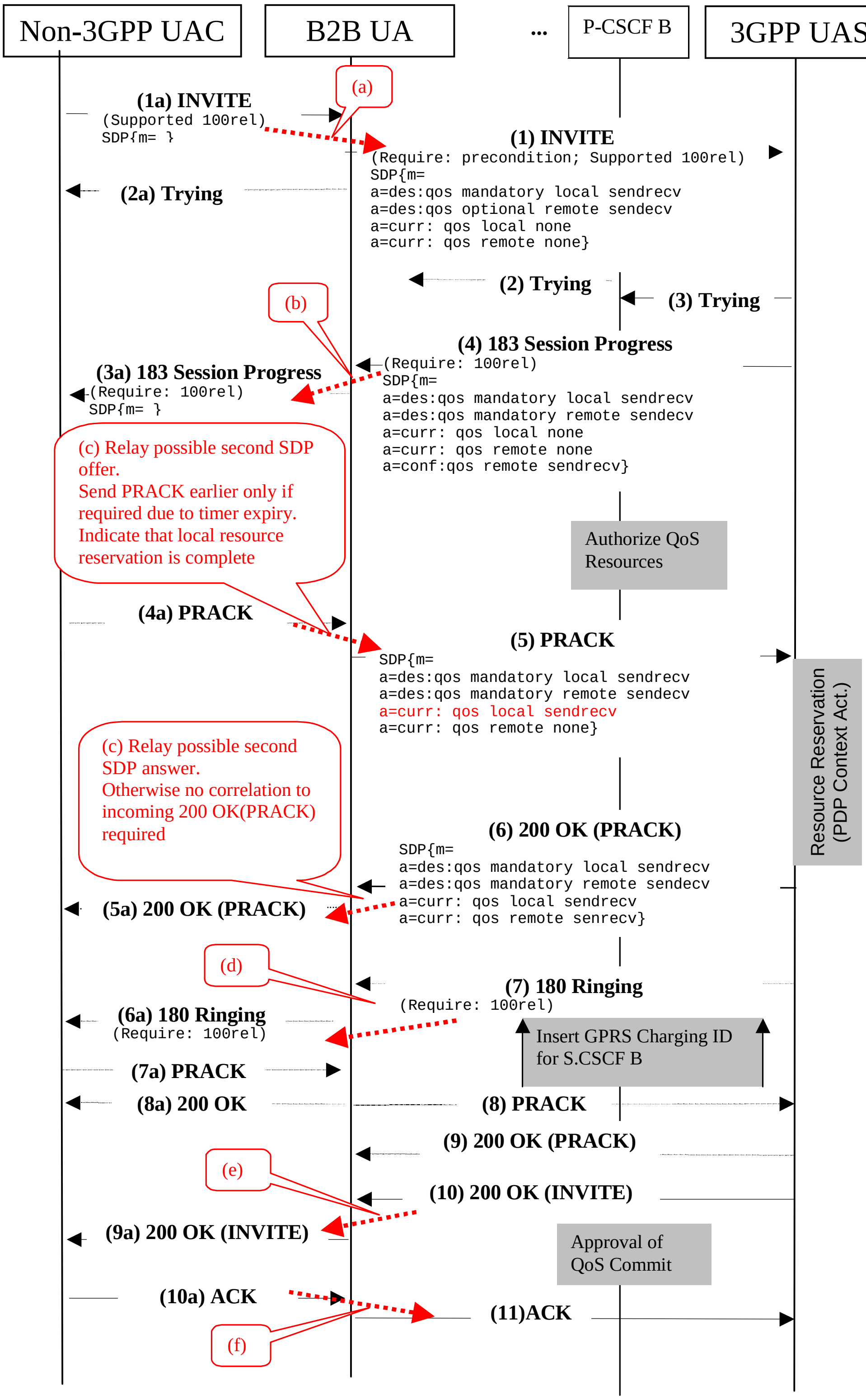

Figure 4.3.4.4.2.1/1: Functionality of B2B UA connecting non-3GPP-compliant SIP UAC not supporting the SIP preconditions extension and the SIP update extension, to 3GPP UAS.

4.3.4.4.2.1.2 Advantages

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

4.3.4.4.2.1.3 Disadvantages

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

4.3.5 Non-3GPP-compliant SIP UAC not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to 3GPP UAS

4.3.5.1 Description of interworking issue

The call fails, as detailed in Section 4.3.4.2

4.3.5.2 Proposed Resolutions to interworking issue

A B2B UA is used.

4.3.5.2.1 Insertion of B2B UA

How the B2B UA is inserted is discussed within Section 4.3.4.4.1

4.3.5.2.2 Functionality of B2B UA

4.3.5.2.2.1 Suggestion A

4.3.5.2.2.1.1 Description

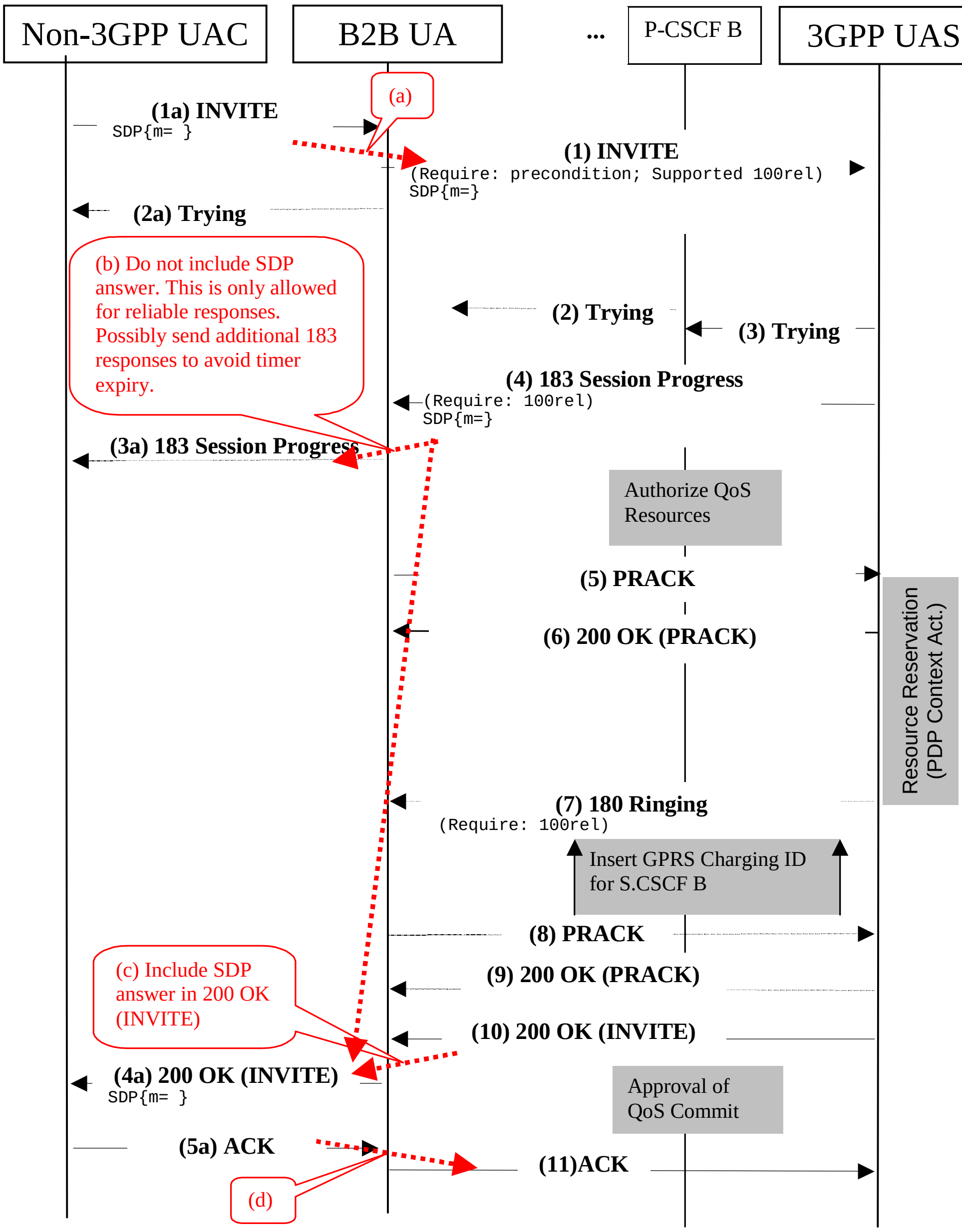

Figure 4.3.5.2.2.1/1: Functionality of B2B UA connecting non-3GPP-compliant SIP UAC not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to 3GPP UAS. SDP offer in INVITE request.

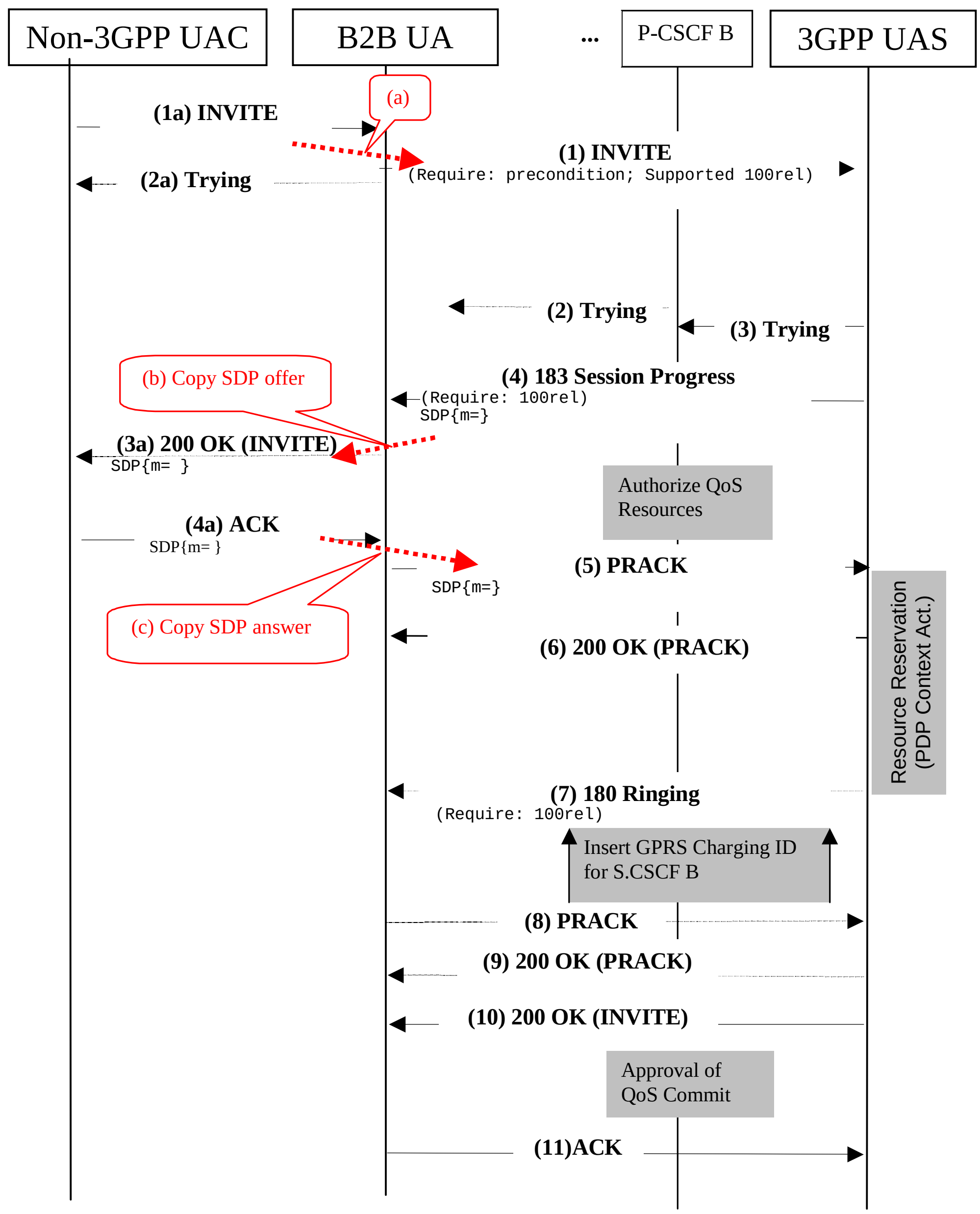

Figure 4.3.5.2.2.1/2: Functionality of B2B UA connecting non-3GPP-compliant SIP UAC not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to 3GPP UAS. SDP offer in OK response.

4.3.5.2.2.1.2 Advantages

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

4.3.5.2.2.1.3 Disadvantages

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

4.3.6 Non-3GPP-compliant UAC supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not performing QoS reservation, to 3GPP UAS.

4.3.6.1 Description of interworking issue

No interworking issues have been identified.

4.3.6.2 Flow diagram

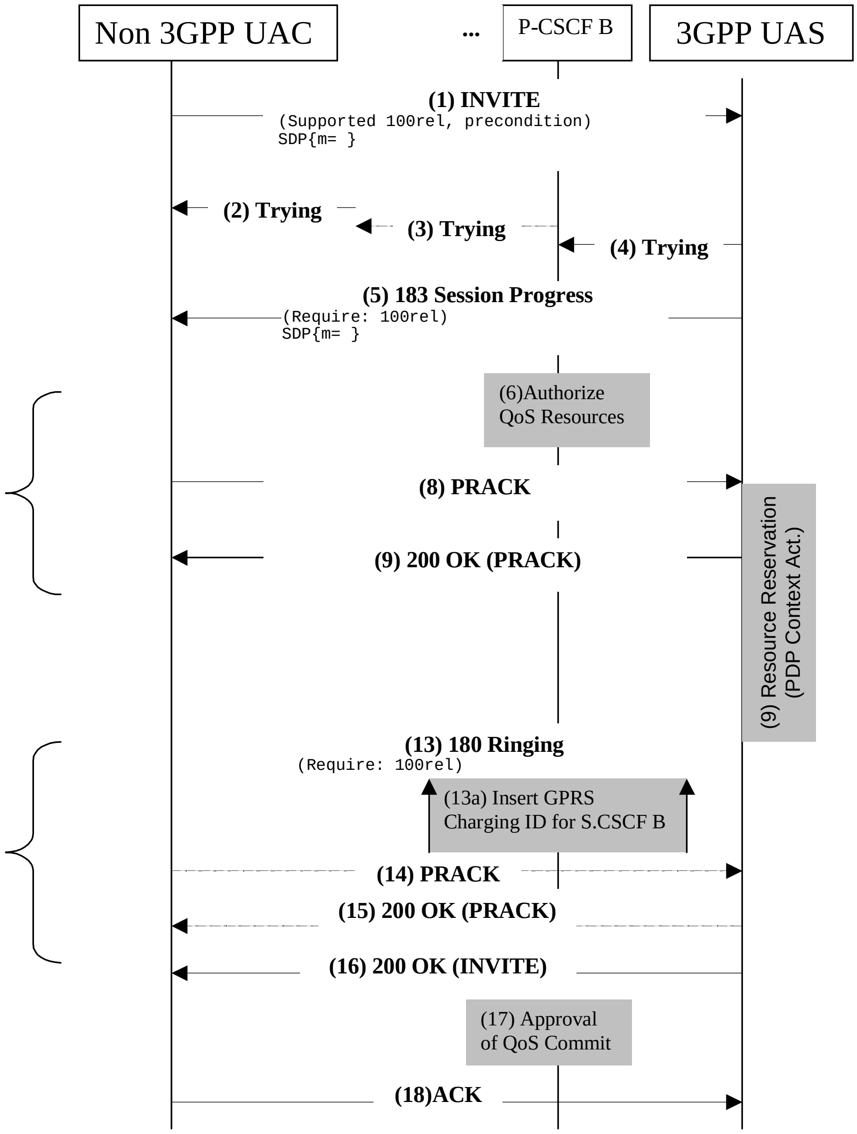

Figure 4.3.6.2/1: Non-3GPP-compliant UAC supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, and not performing QoS reservation, to 3GPP UAS

4.3.7 Non-3GPP-compliant UAC supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not including the SDP offer in the initial invite, to 3GPP UAS.

4.3.7.1 Description of interworking issue

No interworking issues have been identified.

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

4.3.7.2 Flow diagram

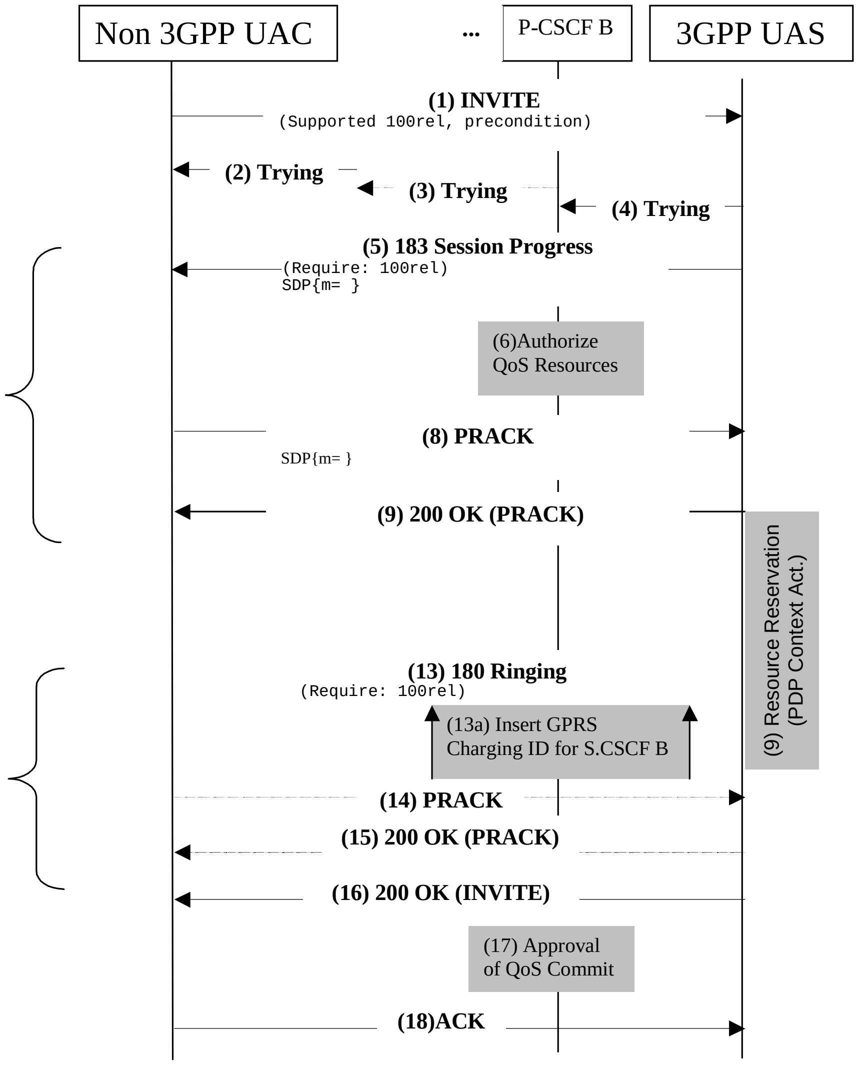

Figure 4.3.7.2/1: Non-3GPP-compliant UAC supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not including the SDP offer in the initial invite, to 3GPP UAS

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 the technical description of the possible interworking topic.

4.x.3 Impact of Identified interworking issue

Editor's Note: Identified interworking issues to be considered

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

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.3.y *Suggestion yy*

4.x.3.y.1 Description

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

4.x.3.y.1 Advantages

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

4.x.3.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,andFrom 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: Callflows between rogue 3GPP UA to non-3GPP-compliant UA, if SIP extensions mandated by 3GPP are not applied.

C.1 Introduction

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.

Note that callflows as presented in this Annex are likely to be disallowed by CSCFs.

C.2 rogue 3GPP UAC to non-3GPP-compliant UAS

C.2.1 rogue 3GPP UAC to non-3GPP-compliant UAS not supporting the SIP 100rel extension, but supporting the SIP precondition extension

According to the SIP preconditions specification [6], Section 10, this scenario is not possible.

C.2.2 rogue 3GPP UAC to non-3GPP-compliant UAS 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-compliant SIP UAS supports the SIP update extension, but does not use them, the situation is similar to Section C.2.4 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, e.g. to handle the “Heterogenous Error Response” forking problem.

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

- The UAS may handle repairable error conditions with a “155 Update Requested” provisional response. This shall not be discussed further here, since it is an error call flow.
- The UAS, may send UPDATE messages at various places within the callflow. Those messages may include additional SDP offers. Due to the large number of possibilities, such callflows 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.2.4 applies.

C.2.3 3GPP UAC to non-3GPP-compliant UAS supporting the 100rel SIP extension and the SIP preconditions extension, but not supporting the SIP update extension

C.2.3.1 Description of interworking issue

If 3GPP UAC mandates local QoS with the SIP 100rel extension, the UAS will suspend the session set-up until this precondition is met. However, UAC lacks means to inform UAS that the precondition is met. As a consequence, the UAS will never resume the session set-up, and the session set-up cannot be completed.

C.2.3.2 Flow diagram

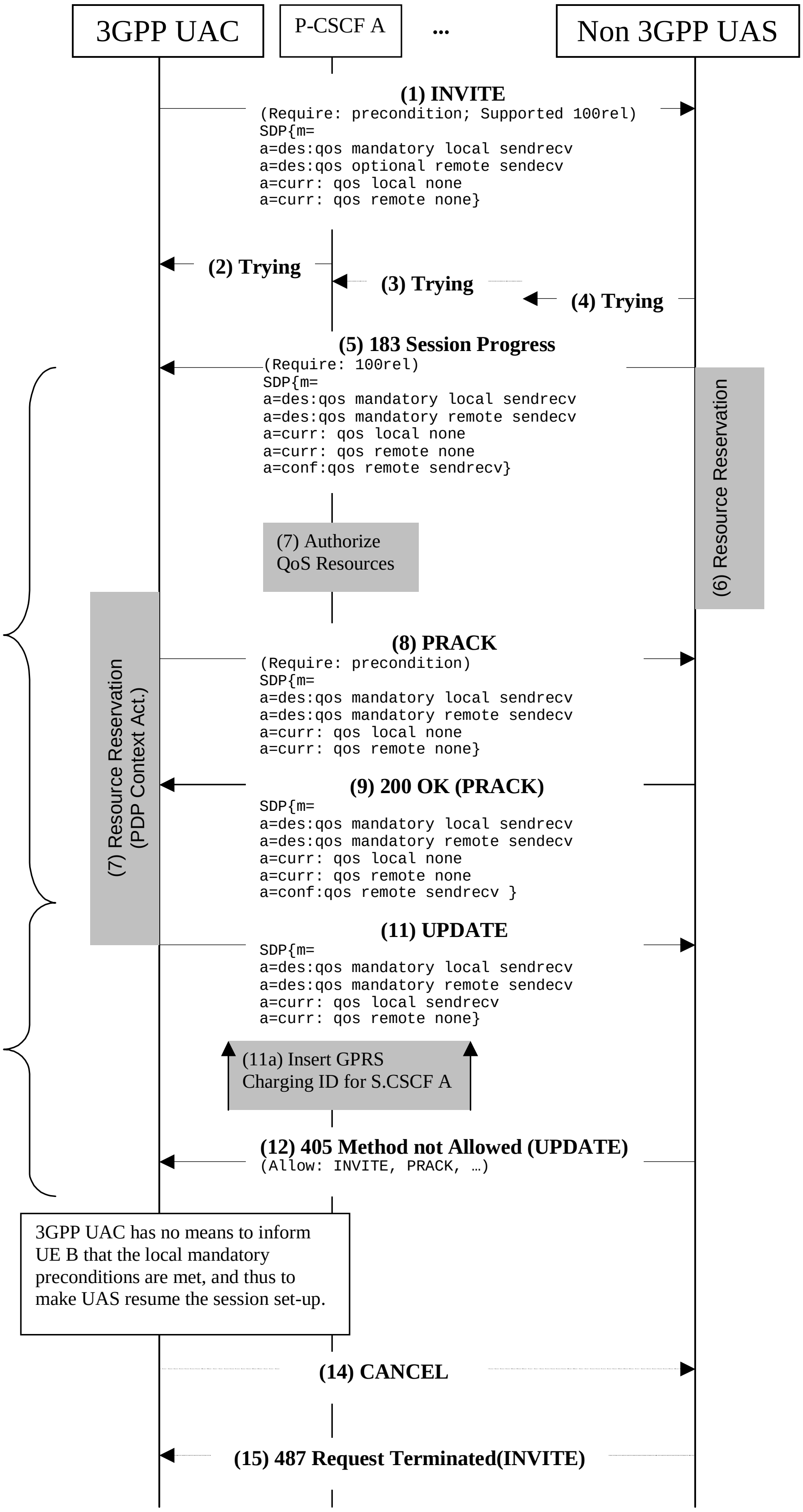

Figure C.2.3.2/1: 3GPP UAC to non-3GPP-compliant UAS supporting the 100rel SIP extension and the SIP preconditions extension, but not supporting the SIP update extension

C.2.3.3 Impacts of Identified interworking issue

The call fails.

C.2.3.4 Proposed resolutions to interworking issue

C.2.3.4.1 Suggestion “no mandatory local resource reservation”

C.2.3.4.1.1 Description

Rogue UAC does not demand any local preconditions in the INVITE message (1).

Thus, UAS does demand a confirmation in the 183 session progress message, and does not suspend the session set-up.

UAC is not required to send an unsupported UPDATE request.

The resulting callflow is discussed in Section C.3.0.

C.2.3.4.1.2 Advantages

The call succeeds.

C.2.3.4.1.3 Disadvantages

- The user of the UAS may be alerted before the required resources have been reserved.
- UAS may start to transmit media before the required resources have been reserved
- Charging may commence before a user plane connection is available

C.2.4 rogue 3GPP UAC to non-3GPP-compliant UAS supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension

C.2.4.1 Description of interworking issue

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

After this failure, the rogue 3GPP UAC may decide to invite the UAS 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 rogue 3GPP UAC 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.2.5 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.2.4.2 Flow diagram

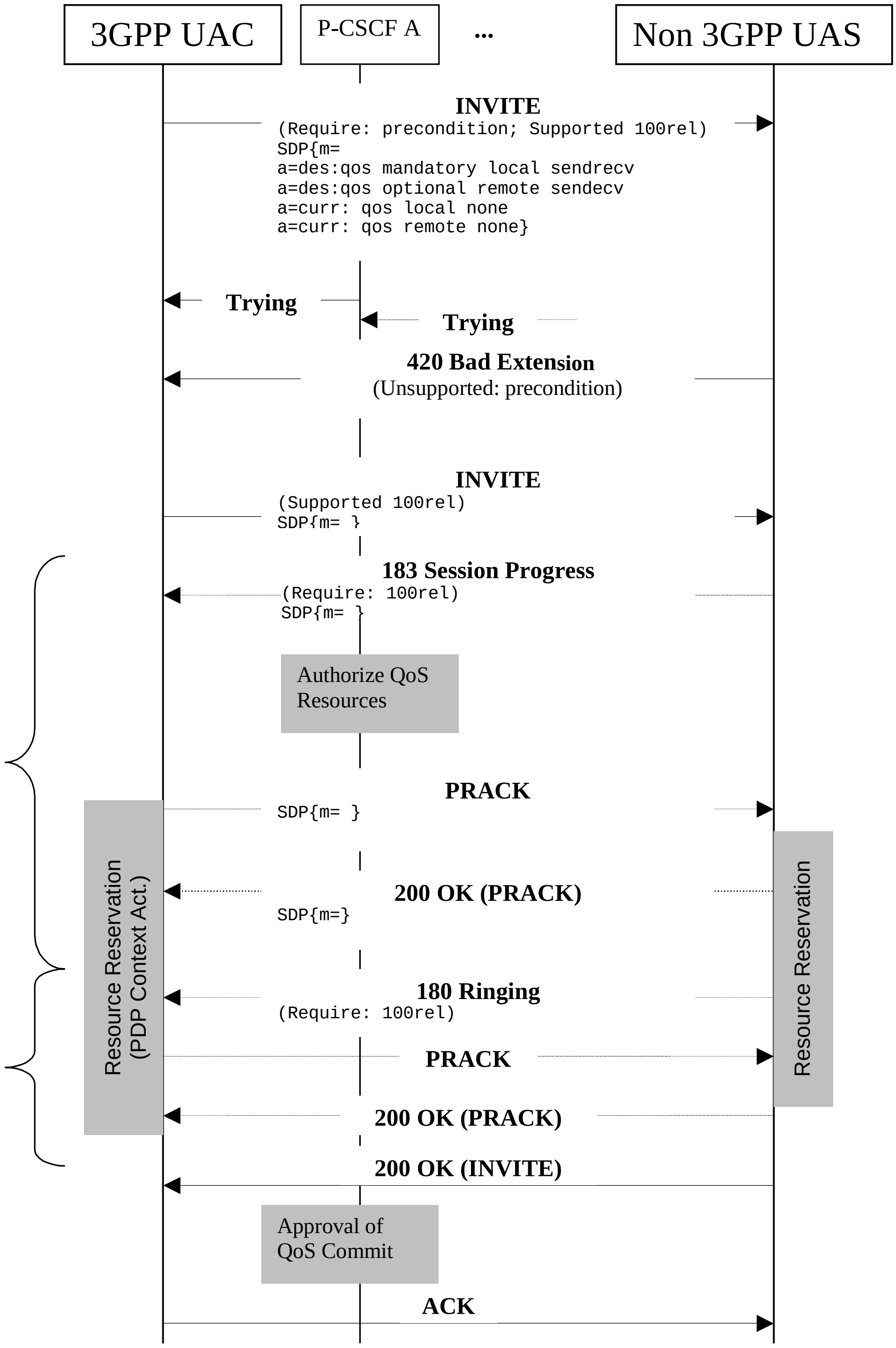

Figure C.2.4.2/1: rogue 3GPP UAC to non-3GPP-compliant UAS supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension

C.2.4.3 Impacts of Identified interworking issue

User at non GPP UAS is alerted before resource reservation at 3GPP UAC 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.2.5 3GPP UAC to non-3GPP-compliant UAS not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

C.2.5.1 Description of interworking issue

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

After this failure, 3GPP UAC may decide to invite UAS 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 UE B does not support the 100 rel 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 3GPP UAC and resource authorisation at P-CSCF will be triggered by this message.

C.2.5.2 Flow diagram

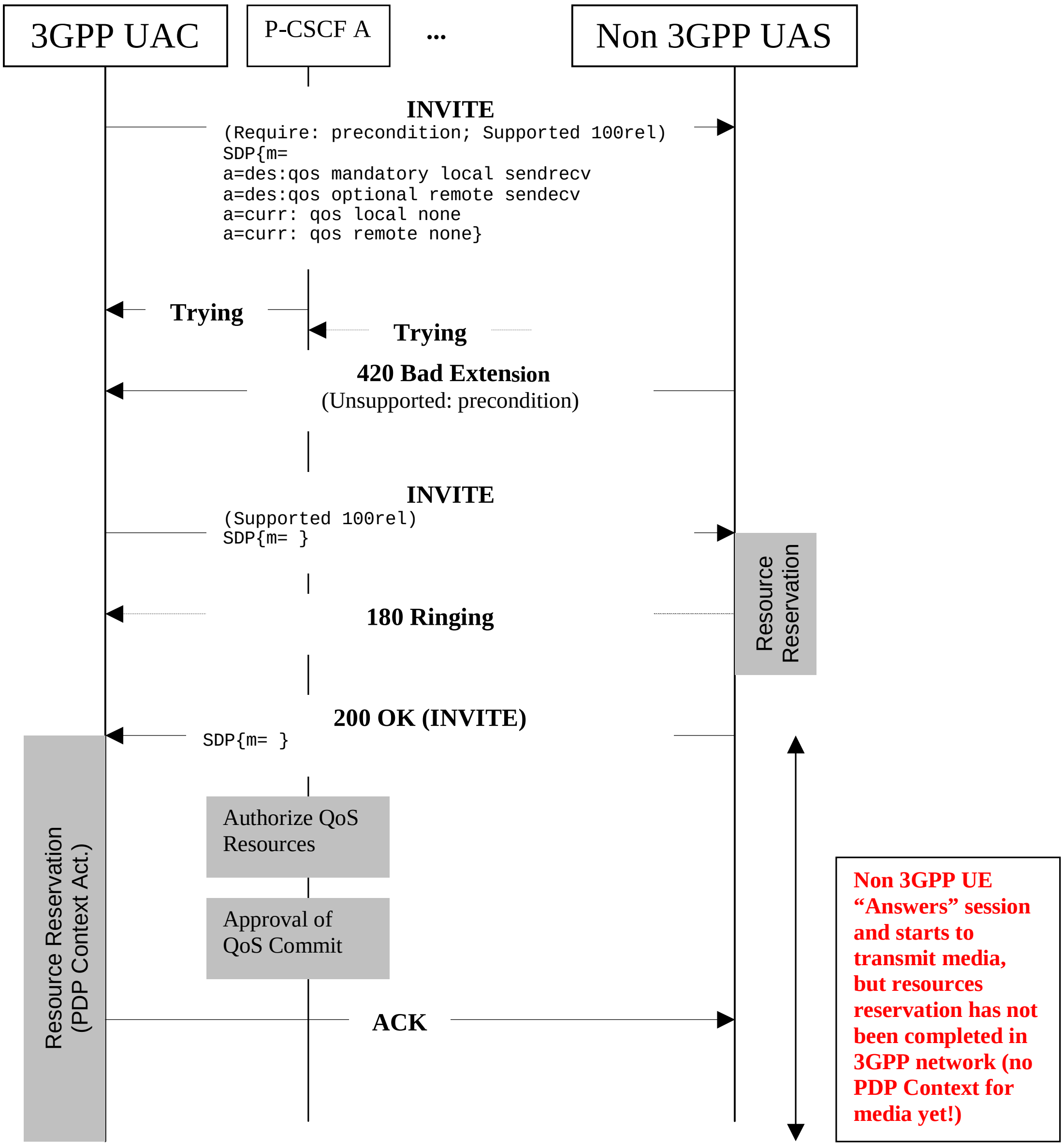

Figure C.2.5.2/1: 3GPP UAC to non-3GPP-compliant UAS not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension

1. **INVITE**

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

2. **180 Ringing**

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

3. **200 OK (Answer)**

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

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

4. **ACK**

   The 3GPP UE sends the “ACK” message to the Non 3GPP UE to acknowldege the 200 OK “final response” message.

C.2.5.3 Impacts of Identified interworking issue

C.2.5.3.1 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.2.5.3.2 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.2.5.3.3 SIP Media authorization

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

C.2.5.3.4 SIP Media allocation

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

C.2.5.3.5 Fraudulent and security risks

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

C.3 Non-3GPP-compliant UAC to rogue 3GPP UAS

C.3.0 Non-3GPP-compliant UAC supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, but not performing QoS reservation, to rogue 3GPP UAS.

C.3.0.1 Description of interworking issue

As outlined in Section C.1.1, Note 7, the “183 Session Progress” provisional response may be omitted, if 3GPP UE A 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.3.5 applies. Severe implications for IMS Charging have been identified.

Otherwise, no interworking issues have been identified. The callflow is depicted for reference only.

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

C.3.0.2 Flow diagram

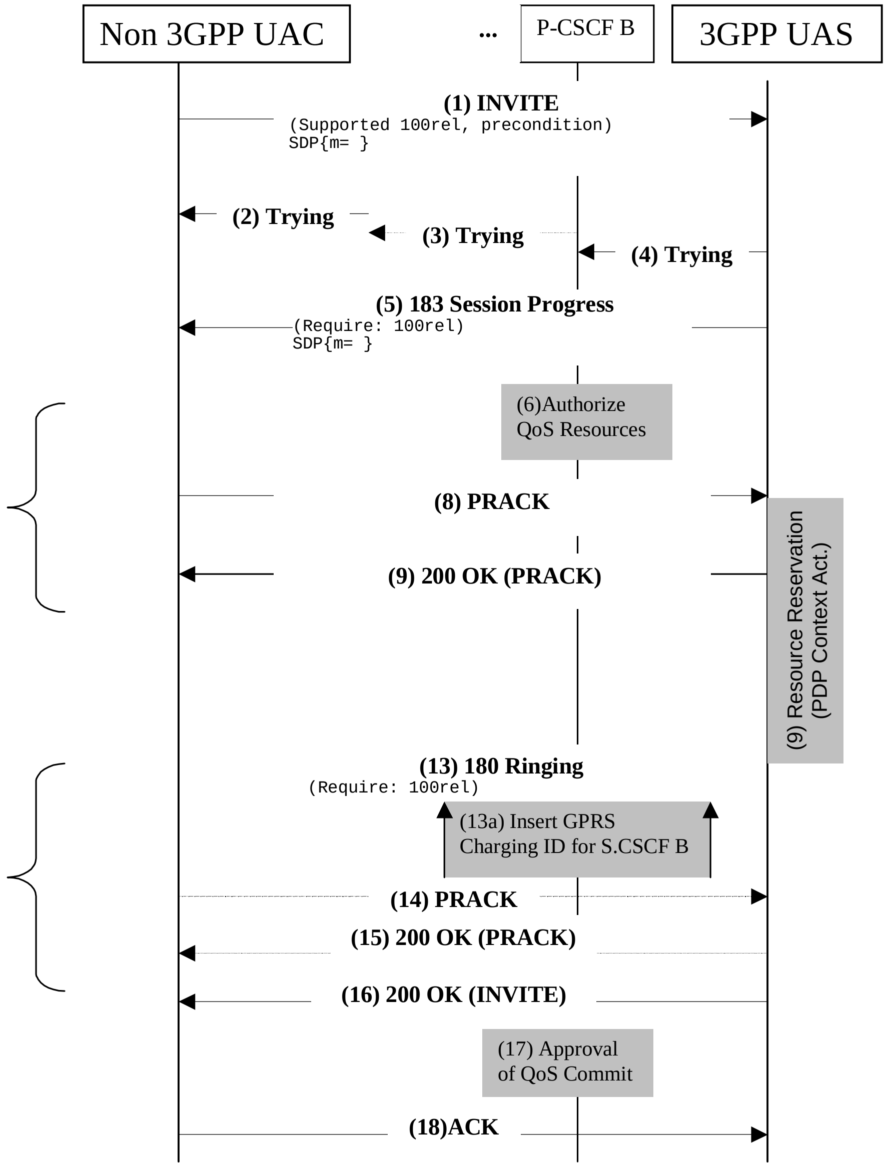

Figure C.3.0.2/1: Non-3GPP-compliant UAC supporting the 100rel SIP extension, the SIP preconditions extension and the SIP update extension, and not performing QoS reservation, to rogue 3GPP UAS

C.3.1 Non-3GPP-compliant UAC not supporting the SIP 100rel extension, but supporting SIP preconditions extension, to rogue 3GPP UAS

According to the SIP preconditions specification [5], Section 10, this scenario is not possible.

C.3.2 Non-3GPP-compliant UAC supporting the SIP 100rel extension and the SIP update extension, but not supporting the SIP preconditions extension, to rogue 3GPP UAS

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-compliant SIP UAC supports the SIP update extension, but does not use them, the situation is similar to Section C.3.4 and the discussion in this Section is applicable for the present scenario.

A fixed UE supporting the SIP update extension may use features of this extension for purposes not related to the SIP precondition extension. As a result, various extra messages may be inserted into the callflow. The UAC, may send UPDATE messages at various places within the callflow. Those messages may include additional SDP offers. Due to the large number of possibilities, such callflows 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.3.4 applies.

C.3.3 Non-3GPP-compliant UAC supporting the 100rel SIP extension and the SIP preconditions extension, but not supporting the SIP update extension, to rogue 3GPP UAS

C.3.3.1 Description of interworking issue

If UAC does not perform resource reservation, the flow diagram in Section C.3.0 applies.

In what follows, it is assumed that the UAC demands local QoS reservation. As a consequence, the call will fail because the UPDATE method would be required.

It could be argued that this scenario is not to realistic, because a UAC demanding mandatory local QoS reservation is likely to support the SIP update extension to avoid a call failure.

C.3.3.2 Flow diagram

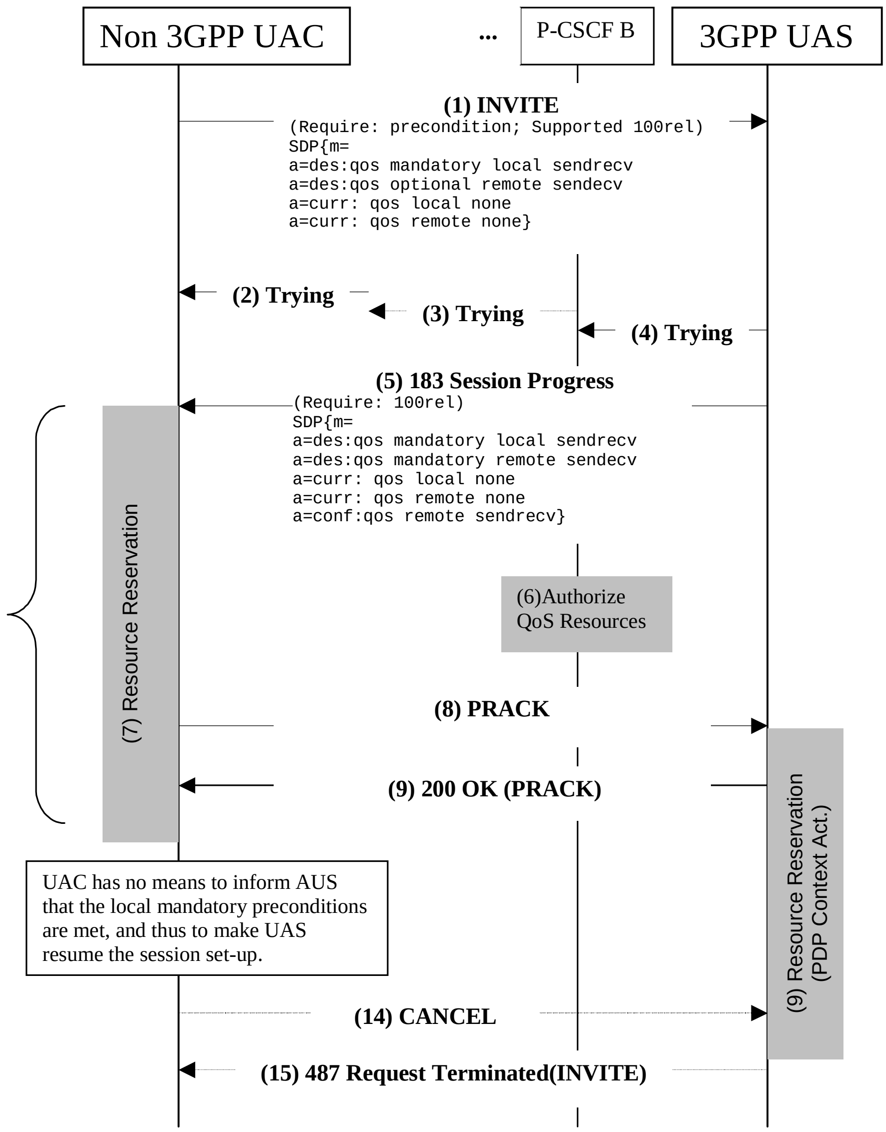

Figure C.3.3.2/1: Non-3GPP-compliant UAC supporting the 100rel SIP extension and the SIP preconditions extension, but not supporting the SIP update extension, and performing local QoS reservation, to 3GPP UAS

C.3.3.3 Impacts of Identified interworking issue

The call fails.

C.3.3.4 Proposed resolutions to interworking issue

C.3.3.4.1 Suggestion “no mandatory local resource reservation”

C.3.3.4.1.1 Description

UAC does not demand any local preconditions in the INVITE message (1).

Thus, UAS does demand a confirmation in the 183 session progress message, and does not suspend the session set-up.

UAC is not required to send an unsupported UPDATE request.

The resulting callflow is discussed in Section C.3.0.

C.3.3.4.1.2 Advantages

The call succeeds.

C.3.3.4.1.3 Disadvantages

- The user of the UAS may be alerted before the required resources have been reserved.
- UAS may start to transmit media before the required resources have been reserved
- Charging may commence before a user plane connection is available

C.3.4 Non-3GPP-compliant SIP UAC supporting the SIP 100rel extension, but not supporting the SIP preconditions extension and the SIP update extension, to rogue 3GPP UAS

The resulting scenario is similar to the one discussed in Section C.3.0.

C.3.5 Non-3GPP-compliant SIP UAC not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UAS

C.3.5.1 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.3.5.2 Flow diagram

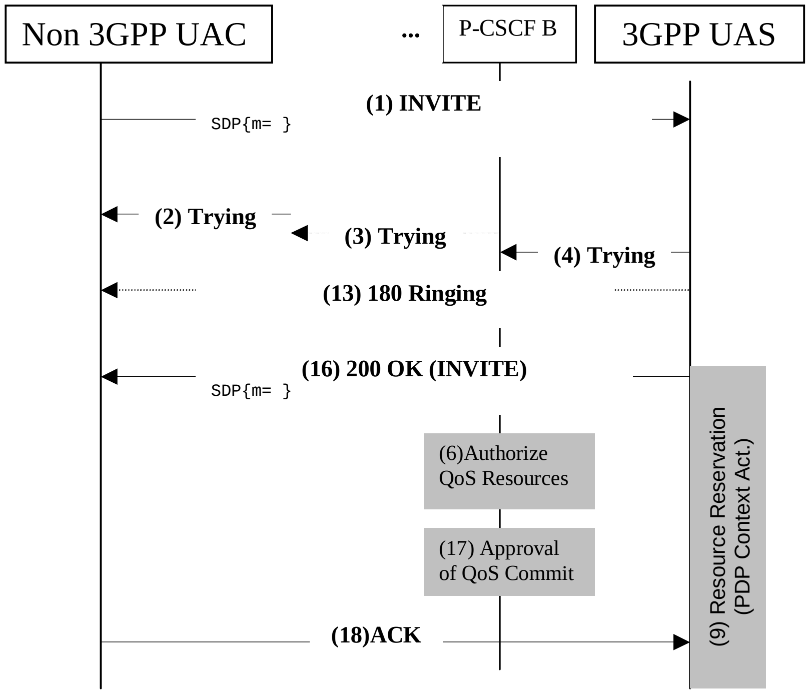

Figure C.3.5.2/1: Non-3GPP-compliant SIP UAC not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UAS. SDP offer in INVITE request.

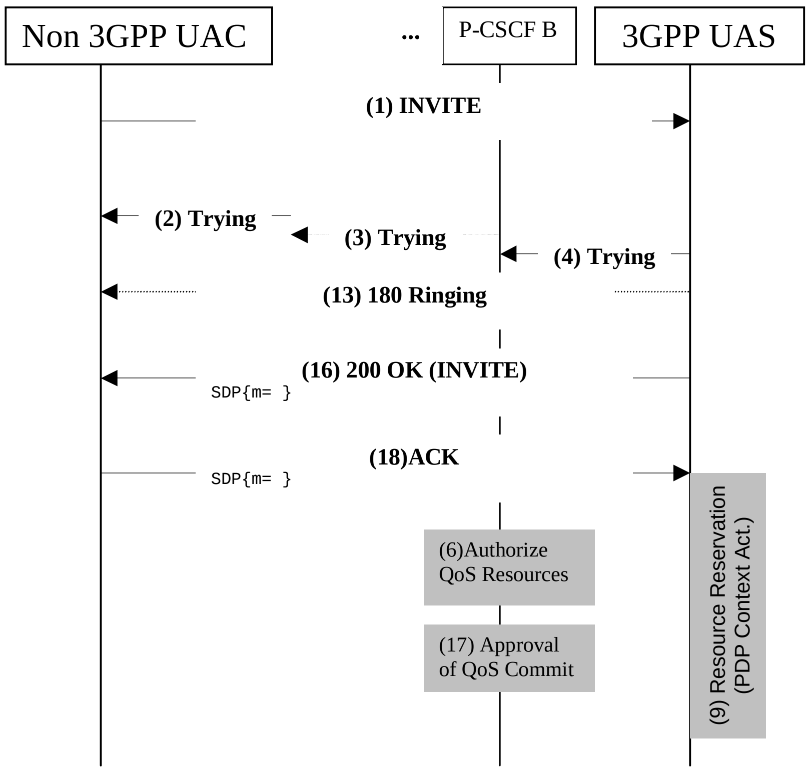

Figure C.3.5.2/2: Non-3GPP-compliant SIP UAC not supporting the SIP 100rel extension, the SIP preconditions extension and the SIP update extension, to rogue 3GPP UAS. SDP offer in OK response.

C.3.5.3 Impacts of Identified interworking issue

3GPP user may be alarmed 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.

If the SDP offer is transported in the INVITE Request, the ACK message would be a candidate to transport this information. However, this message is not transported reliably.

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

Annex D: Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2002-01	*CN3#21*				*Creation of document*	*-*	*0.1.0*
2002-07	*CN3#24*				*Include suggestions for B2BUA*	*0.1.0*	*0.2.0*

3GPP TSG-CN WG3#25 **Tdoc N3-020813**
Miami, USA. 23rd - 27th September. 2002

Source:	**Vodafone**
Title:	**Status report of work in ITU-T SG11**
Agenda item:	**IMS-CCR-IWCS**
Document for:	**Information and Discussion**

1 Introduction

ITU-T Study Group 11 are currently defining the interworking between SIP/SDP and ISUP/BICC at the Network-to-Network Interface. One of their deliverables is to define the interworking for ISUP/BICC to the 3GPP Profile of SIP which is specified in TS 24.229.

This contribution describes the status of the ITU-T SG11 work and considers the possible impacts on the CN3 work schedule

2 Overview

The CN3 work item, Interworking between IM CN Subsystem and CS networks (NP-020090), describes the requirements for interworking between CS networks and the 3GPP IM CN SS for the support of basic voice calls in terms of:

1. control plane interworking, for example, the mapping required between 3GPP Profile of SIP and ISUP/BICC protocols;
2. definition of the functionality of the MGW, together with aspects of the MGCF and SGW for the support of voice calls to and from CS networks;
3. user plane interworking, for example, between the AMR codec used in the IM CN subsystem and possibly other codec types used with in CS networks;
4. the description of transport protocols, transcoding and signalling issues for negotiation and mapping of bearer capabilities and QoS information.

During the CN3#19 meeting it was agreed that ITU-T, and in particular Study Group 11, is the most suitable forum to define the protocol interworking between SIP/SDP and ISUP/BICC, i.e. bullet 1.

It was also agreed that CN3 will continue to describe and specify the interworking in respect to bullets 2 to 4. More recently, it has been agreed that CN4 will describe the usage of the H.248 protocol between the MGCF and IM-MGW and the interactions between IMS-Mc signalling procedures and the related user plane procedures at the IM-MGW (see NP-020320).

3 Status of ITU-T SG11

3.1 Requirements definition

As part of the ITU BICC Capability Set 2 work, the ITU-T SG11 are tasked with defining the protocol interworking between SIP/SDP and ISUP/BICC. The requirements for this work originate from a number of external ITU external organisations which include:

- 3GPP;
- General internet community (IETF);
- Specific operators (such as AT&T, DT etc)

These requirements have been recognised by the Requirements group of SG 11 (Q.6 & 9/11) and has subsequently resulted in the definition of the interworking requirements for the three scenarios; each scenario involving a specific SIP Profile which fulfils different levels of functionality.

In summary, SG 11 Q6 & 9/11 have defined the requirements for the following scenarios:

- Profile A Defined specifically, but not precluding other users, for 3GPP
- Profile B Defined for interworking between SIP and ISUP/BICC

- Profile C Defined for interworking between SIP with MIME Encoding of ISUP and BICC/ISUP

For each scenario, considerations have been made to the scope of each SIP Profile. These considerations are:
- Applicability of particular Gateway types when Interworking SIP with ISUP BICC, e.g. Type 1 for the 3GPP;
- Interworking capabilities between BICC/ISUP and each SIP Profile;
- The mapping of Bearer control protocols, e.g. AAL2, ISUP, B-ISUP and IPBCP to SDP for Gateway type1.

Refer to Section 3.1 for a detailed description of the ITU-T Profiles.

The definition of the requirements within Q.6 & 9/11 are now becoming stable. The latest version of the ITU-T Technical Report, TRQ.BICC/ISUPSIP, is attached for information.

3.1 Protocol definition

In conjunction with the Requirements group the Protocol group of SG 11 (Q11 &12/11) continue to progress the definition of:

- Profile of the SIP/SDP for interworking between SIP/SDP and BICC/ISUP Recommendation (Q.SIPPROF);
- SIP/SDP to ISUP/BICC interworking Recommendation (Q.1912.SIP).

The intention of the Q.SIPPROF Recommendation is to supplement the Q.1912.SIP Recommendation by describing a minimum set of protocol features and extensions required in the SIP and SDP to support interworking at a “Network to Network Interface” (NNI) between a BICC or ISUP network and an originating or terminating SIP network. This document is not required for Profile A (3GPP) due to the work done in 3GPP SA2 and CN1.

The intention of Q.1912.SIP is to define the interworking between SIP/SDP to ISUP/BICC. It defines the interworking of all the mentioned scenarios and attempts to describe all the SIP Profiles without recognising the differences between the Profiles / scenarios and where they will be deployed.

The latest versions of Q.1912.SIP and Q.SIPPROF are attached for information. Both draft Recommendations are unstable and a number of technical and non-technical issues have been highlighted, which have caused a level of contention within SG11 resulting in not all inputs to the Ottawa meeting were handled and outputs of questionable agreement.

4 Issues for consideration

As described the Requirements group have detailed three scenarios:

Scenario / Profile	Preconditions	Overlap	ISUP Mime	MGW at interworking point
A (3GPP)	Yes	No	No	Yes
B	Yes/No	Yes/No	No	Yes/No
C	Yes/No	Yes/No	Yes	Yes/No

Note 1: Yes/No indicates that the case when the functionality is both included and not included must be described.

Note 2: These are high-level differences only, lower level differences also exist, such as offer/ answer is optional in Profiles B and C, while mandatory in Profile A.

Each scenario is supported by a several organisations. In order to meet the varying needs of the organisations looking towards the ITU, for example timing considerations and constraints, it is imperative to understand that functional content and network architecture are not common across these scenarios. This becomes obvious from the requirements work of Q.6&9/11.

From a 3GPP perspective, the Architectural definition, i.e. the IM CN SS, and the SIP Profile definition has taken several years to complete but has now been approved. This work essentially describes a "working" architectural model, which meets many of the Mobile Operator's requirements, e.g. Charging.

However, in considering the stability of SIP Overlap signalling and SIP-T (ITU-T appear to wants something different they call SIP-I) within the IETF and the absence a defined Architecture for each of these Profiles and the subsequent difficulty in defining a minimum set of SIP/SDP it is of particular concern how the interworking of

BICC/ISUP to abstract SIP based Architectures can in reality be achieved. From a "protocol" perspective, there are many "protocol" decisions, e.g. timing of sending a particular message, which will vary depending on each Architectural model.

These issues are under dispute in ITU-T and which will cause a long delay in the completion of the Interworking work. This delay will inadvertently delay in the completion of the 3GPP work in CN3.

In order to progress this work ITU-T should be remembering that timing, functional content and network architecture are not common to these scenarios. In considering possible impacts on the 3GPP completion date for the Interworking between IM CN Subsystem and CS networks it will be beneficial to have separate SIP-BICC/ISUP interworking Recommendations for each identified SIP Profile. This means that Q.1912.SIP should be split into 3 separate Recommendations. Again, this proposal is heavily disputed in ITU-T.

5 Conclusion

It is proposed to communicate this concern to ITU-T SG11 and encourage them to consider our needs and split the draft interworking document on a per Profile basis. Common parts can be referenced from one Recommendation to the other hence avoiding duplication of effort and divergence of the solution.

In considering the disputed issues and the possibility of a delayed output from the ITU-T Vodafone proposes that if there is no satisfactory decision made at the next full ITU-T meeting in Geneva after we have communicated with the ITU, i.e. a decision to split the definition of the interworking for the 3GPP scenario, then the ITU-T work applicable to the 3GPP scenario is ported back into CN3.

3GPP TSG-CN WG3#25 **Tdoc N3-020878**
Miami, USA. 23rd - 27th September. 2002

Title:	**LS on Interworking between SIP/SDP and BICC/ISUP**
Release:	**Release 6**
Work Item:	**IMS-CCR-IWCS**
Source:	**TSG CN WG3**
To:	**TSG CN, 3GPP TSG CN Ad-Hoc Group 1 (ITU_Co-ordination)**

Contact Person:
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Tel. Number: +44 16356 76684
E-mail Address: david.sanders@vodafone.co.uk

Attachments: N3-020813

1. Overall Description:

TSG CN WG3 is responsible for the specification of the Interworking between the 3GPP IP Multimedia Core Network Subsystem (IM CN SS) and Circuit Switched networks. An integral part of this specification is the definition of the Control Plane interworking, e.g. the mapping between 3GPP Profile of SIP, as defined in 3GPP TS 24.229, and the ISUP and BICC protocols.

As part of ITU-T BICC Capability Set 2, the ITU-T SG11 is tasked with defining the protocol interworking between SIP/SDP and ISUP/BICC at the Network-to-Network interface. The requirements for this work originate from a number of organisations external to the ITU, e.g. general Internet community and specific network operators.

During the TSG CN WG3 #19 meeting it was agreed that in order to avoid duplication, to ensure commonality and to avoid conflicting technical specifications it was agreed that the most suitable forum to define the protocol interworking would be the ITU-T. This decision was further supported whilst considering the level of ISUP and BICC technical expertise within ITU-T and in particular Study Group 11.

It was therefore agreed to utilise the work within the ITU-T SG11 and if necessary to provide 3GPP architectural and protocol specific requirements into the ITU-T SG11.

Status of the Interworking specification
During the TSG CN WG3 #25 meeting the status of the ITU-T SG11 specification, as described in N3-020813, for the Interworking between SIP/SDP and ISUP/BICC was discussed. This discussion focused on the progress of the work within ITU-T SG11 and the possible impacts on the 3GPP March 2003 deadline for completion of the specification of the Interworking between the 3GPP IM CN SS and CS networks.

In summary, the status of the ITU-T SG11 was noted as:
- The definition of the requirements within Q.6 & 9/11 are now becoming stable;
- Both draft Recommendations (Q.1912.SIP and Q.SIPPROF) are unstable and a number of issues have been highlighted, which have caused a level of contention within SG11. It is understood that these issues may result in a delay in completion of this work.

Issues for consideration
The stabilisation of the SG11 Technical Report now clearly describes the requirements in terms of individual scenarios; each scenario being supported by different organisations. However, it is also important to understand that that functional content and network architecture are not common across each of the defined scenarios. It is also important to note that, unlike the 3GPP scenario, several scenarios are technically unstable and will require further architectural development which will in turn enable the definition of a minimum set of SIP/SDP.

In considering that timing, functional content, network architecture and technical stability are not common across each of the these scenarios it is of particular concern how these issues will impact the ITU-T finalisation dates of their Recommendations and the subsequent impact on the 3GPP planned completion dates.

In considering the possible impacts on the 3GPP completion date for the Interworking between the IM CN SS and CS networks, TSG CN WG3 would encourage the ITU-T SG11 to consider the needs of 3GPP in terms of their time constraints and to consider the isolation of the 3GPP scenario from the other unstable scenarios.

2. Actions:

To: 3GPP TSG CN Ad-Hoc Group (ITU_Co-ordination)

ACTION: To consider the needs of the 3GPP in terms of their Release 6 time constraints, i.e. completion date of March 2003. The output from ITU-T SG 11 must be easy to reference and shall be in a form that avoids the need for 3GPP to produce additional specifications in order to identify the relevant parts applicable for 3GPP. 3GPP therefore asks ITU-T SG 11 to consider how the requirements, as defined in section 1, can best be accommodated.

3. Date of Next TSG CN WG3 Meetings:

CN3_26	11th - 15th November 2002	Bangkok.

3GPP TSG-CN WG3#25 **Tdoc N3-020868**
Miami, USA. 23rd - 27th September. 2002

Title:	**LS on SBLP control of DiffServ**
Response to:	**LS S2-022000 (N3-020557) on the Go Interface from SA2**
Release:	**Release 5 and 6**
Work Item:	**E2EQoS**
Source:	**CN3**
To:	**SA2**
Cc:	**SA5**

Contact Person:
Name: **Ragnar Huslende**
Tel. Number: +47 452 49237
E-mail Address: ragnar.huslende@eto.ericsson.se

Attachments: -

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1. Overall Description:

CN3 thank SA2 for the guidance provided in LS S2-022000. CN3 wish to advise SA2 that the conditions required for option A could not be met, so option B has been selected for release 5.

CN3 would also like to advise SA2 of a number of questions/issues which were identified during the discussion of option A. CN3 requests SA2 to consider these questions/issues in relation to this function for release 6.

1. **Issues related to the expected/delivered QoS Characteristics**
   With DiffServ classification based on PDP context parameters, the operator can use network engineering to control the QoS in order to deliver the QoS characteristics as specified by the UE in the QoS parameters according to TS 23.107.

   In comparison, there is no signalling for any per IP flow parameters (i.e. token bucket parameters), which may affect the QoS characteristics. How does the UE determine what per flow controls if any will be applied, considering both uplink/downlink directions, and the possible controls at both the local and remote interfaces? How is the end-to-end QoS affected in the different scenarios of per flow controls (e.g. applied at one end or both ends of a session)?

   Is the UE required to shape the traffic on a per IP flow basis in order to avoid these controls causing a reduction in the received QoS? If so, how is this performed in a UE that does not support an IP BS manager? How does such a UE derive the IP flow parameters?

   How can the operator determine the actual QoS that would have been delivered to the UE for an IP flow without some record of the IP flow parameters that were applied, or any measures of the effect of this control?

   The PDP context parameters impose requirements on the end-to-end QoS, which are assumed to apply in a mobile-to-mobile IMS scenario. How is it ensured that these requirements are met if the DSCP decision does not also consider the PDP context parameters?

   Until now, charging for the bearer service has been based on the PDP context parameters, from which the IP bearer service is derived. With introduction of an IP bearer service model in addition to the PDP context based service model, what are the impacts on the charging model? What information is required to support such a model? Also, what is the impact of the different scenarios (with/without per IP flow control at each end) on the charging models?

2. **QoS Management Issues**
   Policies for PDP context based QoS currently would be configured in the GGSN by the management

function. What is the required interaction between the managed bearer service layer in the GGSN, and the SBLP policies? Do the SBLP based policies control the scope for permitted service to be provided by the bearer layer, or does it override the policies from the bearer layer? For example, does the PCF control the maximum allowed DSCP (similarly to the control of the maximum traffic class), or does it define the specific DSCP to be applied?

If it controls the specific DSCP, then how does this policy ensure that the QoS requirements as defined from the PDP context are also met? How do you ensure compatibility between the per flow control policies and the PDP context based policies? This should further consider the aspects of access independence where the SBLP based policies are determined according to the service level, and should not be specific for the bearer service.

The QoS characteristics available from the SDP are very limited. The simple bandwidth information is the only parameter that is used to authorise the rate control for the PDP context. TS 23.207 refers to the derivation of token bucket parameters, but it has not been shown yet that these can be reliably determined based on the SDP.

What are the required parameters for the traffic profile of the per IP flow control, and how are each of these parameters to be derived from the limited information in the SDP?

3. **Other General Questions/Comments**
   The issues raised above are directly related to the actual development of the solution. In addition to the above, the following additional questions/concerns have been raised as to whether the function as proposed is actually the most appropriate mechanism to meet the requirements:

   The only action currently proposed for out-of-profile packets on a per IP flow basis is to discard them. Since such packets have already been transferred over the air interface (at least in the uplink direction), it is questioned whether this is the function that is actually required? It is also questioned whether other handling options such as accounting/charging have been considered?

   It is questioned whether the dynamic configuration of the DiffServ marking function by the PCF brings real additional benefits compared to existing operator configuration to mark the packets using operator configuration rules. For example, what is the benefit in re-marking and potentially downgrading the QoS of an IP flow, which was been already treated with higher QoS at the UMTS bearer level. Otherwise, if it has been given lower QoS over the radio, what would be the actual benefit on the end-to-end QoS of using a higher QoS across the backbone.

   Furthermore, the control of misbehaving IP flows inside one PDP context was identified as the major gain of this functionality. However, it is noted that when multiple flows are aggregated over a bandwidth constrained PDP context, even correctly behaving flows can interact unless they are properly managed by the UE, and a dedicated PDP context would provide sufficient control over a flow. It is thus questioned what are the actual control requirements for different session/media scenarios from which the specific solution requirements are derived.

2. Actions:

To SA2

ACTION 1:

CN3 asks SA2 to consider the above questions and issues for stage 2 specification of the function in release 6.

ACTION 2:

CN3 asks SA2 for feedback on the above questions and issues, with guidance on how to proceed with the stage 3 work for release 6 with minimum delay.

3. Date of Next CN3 Meetings:

CN3#26	11^{th} - 15^{th} November 2002	Bangkok.

3GPP TSG CN WG4 Meeting #16
Miami, USA, 23rd – 27th September 2002

N4-021816

CR-Form-v7

CHANGE REQUEST

⌘	**23.172**	**CR**	**001**	⌘ **rev**	**-**	⌘ Current version:	**5.0.0** ⌘

*For **HELP** on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

Proposed change affects: UICC apps⌘ [ ] ME [ ] Radio Access Network [ ] Core Network [X]

Title:	⌘	Lawful Interception For SCUDIF
Source:	⌘	LM Ericsson
Work item code:	⌘	SCUDIF — ***Date:*** ⌘ 12/09/02
Category:	⌘	**F** — ***Release:*** ⌘ Rel-5

Use one of the following categories:
F *(correction)*
A *(corresponds to a correction in an earlier release)*
B *(addition of feature),*
C *(functional modification of feature)*
D *(editorial modification)*
Detailed explanations of the above categories can be found in 3GPP TR 21.900.

Use one of the following releases:
2 (GSM Phase 2)
R96 (Release 1996)
R97 (Release 1997)
R98 (Release 1998)
R99 (Release 1999)
Rel-4 (Release 4)
Rel-5 (Release 5)
Rel-6 (Release 6)

Reason for change:	⌘	No requirements are defined for monitoring SCUDIF calls
Summary of change:	⌘	Introduce chapter to describe basic requirement.
Consequences if not approved:	⌘	Standards do not define how SCUDIF call is to be monitored

Clauses affected:	⌘	5

Other specs affected: ⌘

Y	N		
	X	Other core specifications	⌘
		Test specifications	
		O&M Specifications	

Other comments: ⌘ SA3-LI requested that we inform them of any functions that may need to be described for LI (attention was made to a similar entry in the TrFO 23.153 spec) but they have not made any provision in their specifications for this type of detail to be described and thus it was agreed that such a definition would remain until they had described the details in their specs, then we can remove it from our corresponding versions.

How to create CRs using this form:
Comprehensive information and tips about how to create CRs can be found at http://www.3gpp.org/specs/CR.htm. Below is a brief summary:

1) Fill out the above form. The symbols above marked ⌘ contain pop-up help information about the field that they are closest to.

2) Obtain the latest version for the release of the specification to which the change is proposed. Use the MS Word "revision marks" feature (also known as "track changes") when making the changes. All 3GPP specifications can be

downloaded from the 3GPP server under ftp://ftp.3gpp.org/specs/ For the latest version, look for the directory name with the latest date e.g. 2001-03 contains the specifications resulting from the March 2001 TSG meetings.

3) With "track changes" disabled, paste the entire CR form (use CTRL-A to select it) into the specification just in front of the clause containing the first piece of changed text. Delete those parts of the specification which are not relevant to the change request.

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 and/or edition number or version number) 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 TR 21.905: "Vocabulary for 3GPP Specifications".

[2] 3GPP TS 23.153: "Out of Band Transcoder Control; Stage 2".

[3] 3GPP TS 24.008: "Mobile Radio Interface Layer 3 specification; Core network protocols; Stage 3".

[4] 3GPP TS 26.103: "Speech Codec List for GSM and UMTS".

[5] 3GPP TS 27.001: "General on Terminal Adaptation Functions (TAF) for Mobile Stations (MS)".

[6] 3GPP TS 29.007: "General requirements on interworking between the Public Land Mobile Network (PLMN) and the Integrated Services Digital Network (ISDN) or Public Switched Telephone Network (PSTN)".

[7] 3GPP TS 29.205: "Application of Q.1900 series to bearer-independent circuit-switched core network architecture; Stage 3".

[8] 3GPP TS 22.101: "Service aspects; Service principles".

[9] 3GPP TS 33.106: "3GPP Security; Lawful Interception Requirements".

5 Lawful Interception

SCUDIF calls shall be monitored as for normal Circuit Switched data calls, for detailed requirements see [9]

3GPP TSG-CN WG3#25 **Tdoc N3-020860**
Miami, USA. 23^{rd} - 27^{th} September. 2002

Title:	**LS on SCUDIF and Lawful Interception**
Response to:	**-**
Release:	**Release 5**
Work Item:	**SCUDIF**
Source:	**CN3**
To:	**SA3-LI**
Cc:	**-**

Contact Person:
Name: **Phil Hodges**
Tel. Number: +61 3 93013414
E-mail Address: philip.hodges@ericsson.com.au

Attachments: N3-020816 (CR to TS 23.172)

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1. Overall Description:

CN3 has considered the Lawful Interception for the SCUDIF service as defined in TS 23.172 introduced in Rel5 and does not see a need to introduce any specific behaviour other than what is already defined for a CS data call. In light of this, CN3 has included such a statement in the TS for SCUDIF, see attached CR, it simply indicates that for this service no new handling is needed.

In response to an earlier LS from SA3-LI to inform them of any LI related issues considered by this WG, CN3 is hereby providing such notification.

2. Actions:

None expected.

3. Date of Next CN3 Meetings:

CN3_26 11^{th} - 15^{th} November 2002 Bangkok.

3GPP TSG-CN joint WG1 and WG3 meeting **Tdoc N3-021012**
Bangkok, Thailand, 11th - 15th November 2002.

Title:	**Reply LS on RTCP overhead in SDP bandwidth parameter**
Response to:	**LS (S4-020567) on RTCP overhead in SDP bandwidth parameter from SA4**
Release:	**Rel-5**
Work Item:	**IMS**
Source:	**CN1, CN3**
To:	**TSG-SA WG4**
Cc:	**TSG-SA WG2**

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Attachments: None

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1. Overall Description:

TSG-CN WG1 and WG3 thank TSG-SA WG4 for clarifying the RTCP overhead vs. SDP bandwidth issue (S4-020567). CN1 and CN3 accept SA4's interpretation that the *b=AS* parameter does not include RTCP bandwidth.

CN1 and CN3 welcome the adoption of the RFC titled "*SDP bandwidth modifiers for RTCP*" considered by SA4 for Rel-5. However, the adoption of this solution will have an impact on CN1's and CN3's specifications. Particularly, 3GPP TS 24.229 covers the SIP and SDP related procedures, and 3GPP TS 24.228 covers mostly example information flows. As such, any new requirement on the usage of SIP or SDP has to be documented in those specifications. The impact on CN3's specifications, e.g. 29.207 and 29.208, is under investigation.

If SA4 decides to adopt the considered solution, CN1 and CN3 will study the issue in detail and produce relevant Rel-5 CRs.

2. Actions:

To SA4 group.

ACTION: CN1 and CN3 ask SA4 to inform them about the possible adoption and details of the considered solution mentioned in S4-020567 by the next CN1 and CN3 meetings.

3. Date of Next CN1 and CN3 Meetings:

CN1, CN3 10^{th} - 14^{th} February 2002 Dublin/Ireland.