3GPP TSG CN Plenary Meeting #21 **NP-030343**
17th - 19th September 2003. Frankfurt, Germany.

Source: TSG CN WG3

Title: CRs on Rel-6 Work Item IMS-CCR-IWIP.

Agenda item: 9.12

Document for: APPROVAL

Introduction:

This document contains **1** CRs on **Rel-6 Work Item IMS-CCR-IWIP,** including the corresponding mirror CRs (as required).

These CRs have been agreed by TSG CN WG3 and are forwarded to TSG CN Plenary meeting for approval.

WG_tdoc	Title	Spec	CR	Rev	Cat	Rel	C_Ver
N3-030594	Editorial Corrections	29.962	001	1	D	Rel-6	6.0.0

3GPP TSG-CN WG3 Meeting #29 ***N3-030594***
Sophia Antipolis, France. 25th - 29th August 2003.

CR-Form-v7

CHANGE REQUEST

⌘ **29.962**	**CR 001**	⌘ **rev**	**-**	⌘ Current version:	**6.0.0** ⌘

*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 [X] Radio Access Network [ ] Core Network [X]

Title:	⌘	**Handling of SIP CANCEL Request**		
Source:	⌘	TSG_CN WG3 [Siemens]		
Work item code:	⌘	IMS-CCR-IWIP	***Date:*** ⌘	18/08/2003
Category:	⌘	D	***Release:*** ⌘	Rel-6
		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:	⌘	Some Editorial mistakes, e.g. one level one heading missing
Summary of change:	⌘	Editorial Corrections
Consequences if not approved:	⌘	Editorial Mistakes remain. TR does not comply with drafting rules.

Clauses affected:	⌘	2, 3, 4, 6, Annex D				
		Y	**N**			
Other specs affected:	⌘		X	Other core specifications	⌘	
			X	Test specifications		
			X	O&M Specifications		
Other comments:	⌘					

2 References

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

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.

- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document.*

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

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

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

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

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

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

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

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

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

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

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

3 Definitions and Abbreviations

3.1 Definitions

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

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

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

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

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

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

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

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

3.2 Abbreviations

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

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

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

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

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

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

Next modified Section

4.1.2 Proposed Resolution B2BUA

A B2BUA is used.

Insertion of B2BUA

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

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

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

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

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

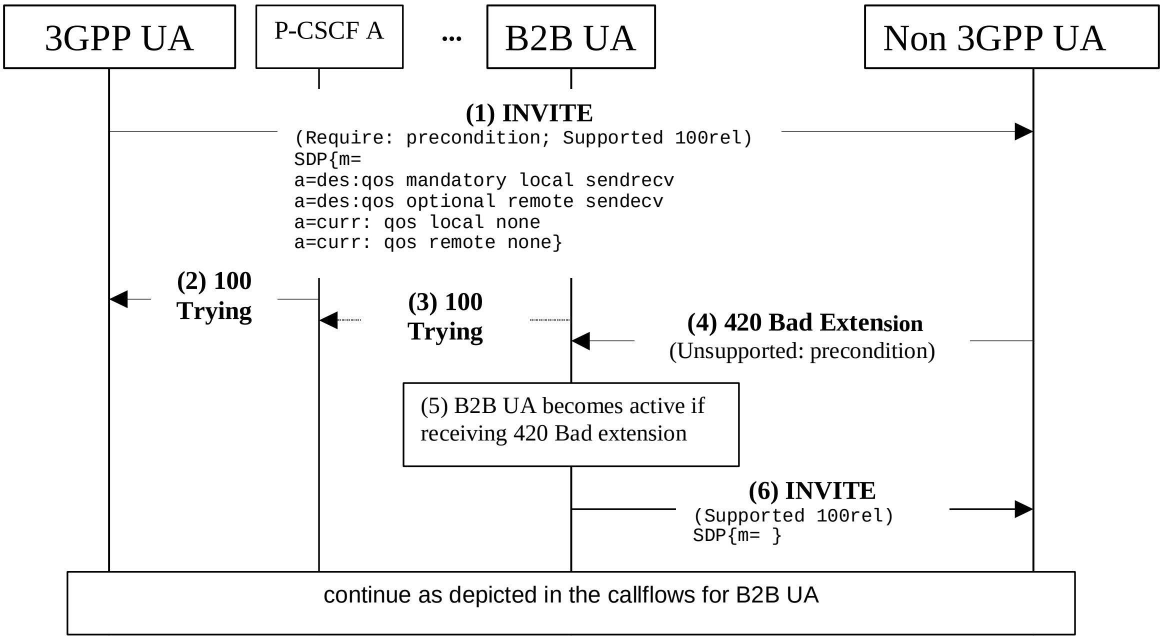

Figure 4.1.2/1: Activation of static B2BUA connecting 3GPP UA to non-3GPP UA not making use of the SIP preconditions extension

Functionality of B2BUA

The B2BUA shall apply the following rules:

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

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

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

Next modified Section

6 Implications of the Proposed Solutions

~~Editor's Note: This clause shall summarise the findings within the corresponding subclauses within Clauses 4 and 5.~~

6.1 B2BUA

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

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

Additional processing load and additional delay may result.

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

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

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

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

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

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

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

6.2 Modified end-to-end call flow

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

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

Changes have to be performed in various network entities.

Next modified Section

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

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

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

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

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

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

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

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

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

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

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

NOTE 9: According to RFC 3262[6], Clause 5, the called UA should start the resource reservation (13) immediately after having send the SDP answer within of the 183 (Session Progress) (7) provisional response. However, a called 3GPP UA may expect a second SDP offer, and thus wait for the next message until starting resource reservation. The called 3GPP UA can be certain to receive an new message soon, since it demands the PRACK message with the "Require 100rel" SIP header within the 183 (Session Progress) (7) provisional response.

NOTE 10: The use of the UPDATE request (14) is optional according to RFC 3312[5], RFC 3311[7], unless certain conditions make it mandatory. In particular, if a threshold specified in a previous SDP "confirm-status" attribute (e.g. in message (7)) is reached or surpassed, the UA must send an SDP offer reflecting the new current status (RFC 3312[5], Clause 7). The only suitable way to convey the new SDP offer/answer may be via an UPDATE request.

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

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

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

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

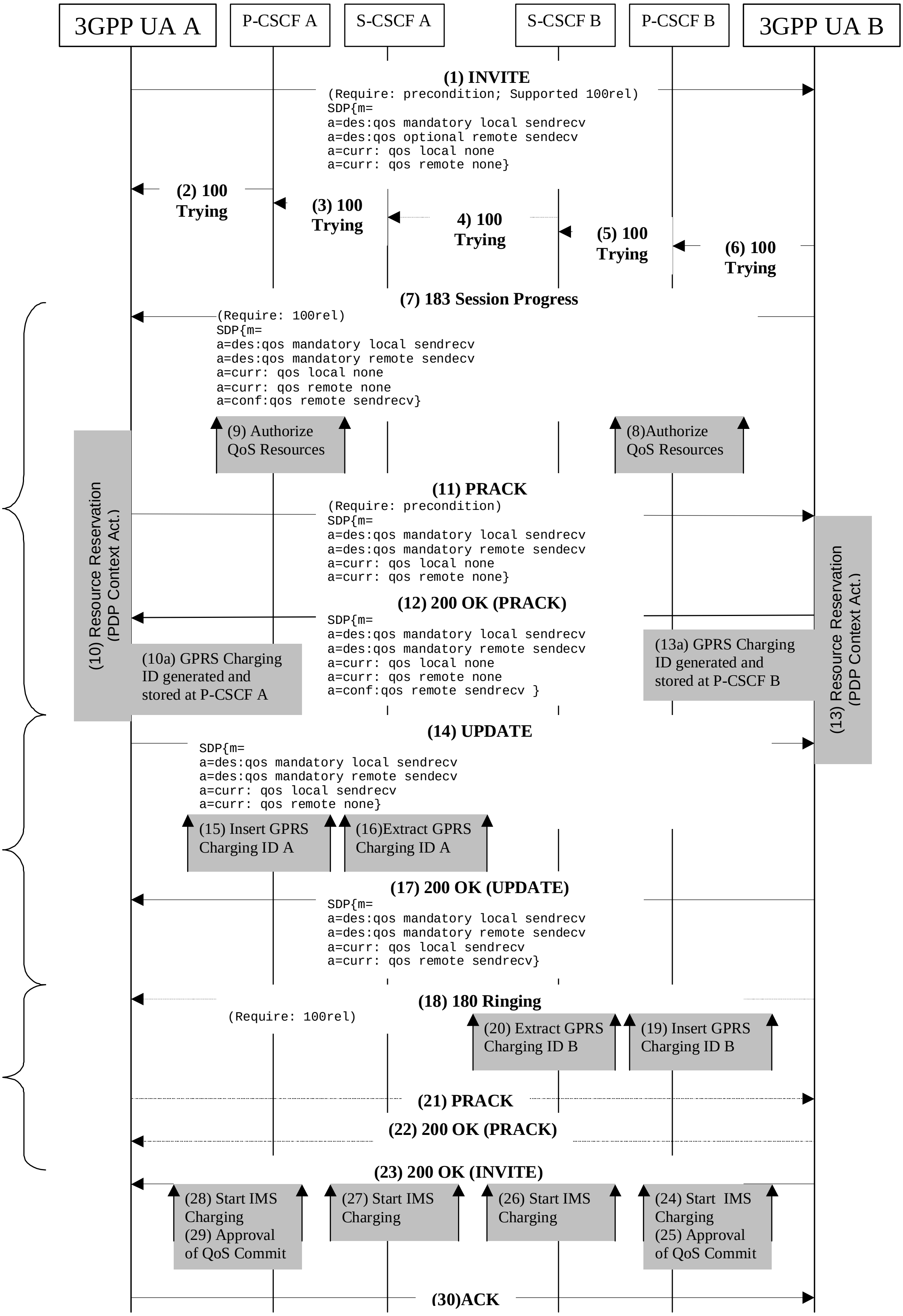

Figure D/1: 3GPP UA to 3GPP UA Call flow

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

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