3GPP TSG-CN Meeting #24 **NP-040193**
2nd – 4th June 2004. Seoul, Korea.

Source: **TSG CN WG 1**

Title: **CR to Rel-6 on Work Item IMS2 towards 24.229, CR#644r3**

Agenda item: **9.1**

Document for: **APPROVAL**

Introduction:

This document contains **1** CR, **Rel-6** Work Item **"IMS2"**, that have been agreed by **TSG CN WG1 in CN1#34 meeting**, and are forwarded to TSG CN Plenary meeting #24 for approval.

Spec	CR	Rev	Phase	Subject	Cat	Version-Current	Doc-2nd-Level
24.229	644	3	Rel-6	Session initiation without preconditions	B	6.2.0	N1-041096

3GPP TSG-CN1 Meeting #34 **Tdoc N1-041096**
Zagreb, Croatia 10 – 14 May 2004 ***was tdoc N1-040873, N1-041035***

CR-Form-v7

CHANGE REQUEST

⌘ 24.229	CR 644	⌘ rev 3 ⌘	Current version: 6.2.0 ⌘

*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: ⌘	Session initiation without preconditions		
Source: ⌘	Nokia		
Work item code: ⌘	IMS2	***Date:*** ⌘	16/04/04
Category: ⌘	**B** *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.	***Release:*** ⌘	Rel-6 *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: ⌘	This CR alignes 3GPP TS 24.229 with the approved CR to 23.228 (CR 337, tdoc SP-030538) regarding the possibility to allow a UE, upon getting an indication that the remote terminal does not support the required capabilities (e.g., preconditions), to re-try the INVITE without requiring the support for unsupported extension.
Summary of change: ⌘	• It is allowed that a UE, upon receiving a response from the remote UA indicating that the precondition extension is not supported, re-initiates the session without requiring this extension. The media streams are set to inactive mode • When resource reservation is complete, the UE resumes the media streams previously set to active mode. • The procedures at the P-CSCF to insert the access network charging info are generalized by making them dependent on the availability of the information and the first SIP request or response that traverses the P-CSCF, rather than specific SIP messages. • The SDP procedures to request preconditions are made dependent on the existance of the preconditions tag in the SIP Require header.
Consequences if not approved: ⌘	• Missalignment between stage 2 and stage 3 specifications. • The behaviour of the UE receiving a 420 (Bad Extension) response causes misalignment with the IETF specifications.

Clauses affected: ⌘	5.1.3, 5.1.4, 5.2.7, 6.1
	Y N

Other specs affected:	⌘		X	Other core specifications	⌘
			X	Test specifications	
			X	O&M Specifications	
Other comments:	⌘				

First proposed change

5.1.3 Call initiation - mobile originating case

5.1.3.1 Initial INVITE

Upon generating an initial INVITE request, the UE shall:

- indicate the support for reliable provisional responses and specify it using the Supported header mechanism;
- indicate the requirement of precondition and specify it using the Require header mechanism.

The UE may also indicate that the proxies should not fork the INVITE request by including a "no-fork" directive within the Request-Disposition header in the initial INVITE request as described in draft-ietf-sip-callerprefs-10 [56B].

NOTE 1: Table A.4 specifies that UE support of forking is required in accordance with RFC 3261 [26]. The UE may accept or reject any of the forked responses, for example, if the UE is capable of supporting a limited number of simultaneous transactions or early dialogs.

When a final answer is received for one of the early dialogues, the UE proceeds to set up the SIP session. The UE shall not progress any remaining early dialogues to established dialogs. Therefore, upon the reception of a subsequent final 200 (OK) response for an INVITE request (e.g., due to forking), the UE shall:

1) acknowledge the response with an ACK request; and
2) send a BYE request to this dialog in order to terminate it.

If the UA receives a 503 (Service Unavailable) response to an initial INVITE request containing a Retry-After header, then the UE shall not automatically reattempt the request until after the period indicated by the Retry-After header contents.

If the UE receives a 488 (Not Acceptable Here) response to an initial INVITE request, the UE should send a new INVITE request containing SDP according to the procedures defined in subclause 6.1.

NOTE 2: An example of where a new request would not be sent is where knowledge exists within the UE, or interaction occurs with the user, such that it is known that the resulting SDP would describe a session that did not meet the user requirements.

If the UE receives a 420 (Bad Extension) response to an initial INVITE request with "precondition" option-tag in the Unsupported header field, the UE shall either:

a) abort the session attempt and shall not resend this INVITE request without "precondition" option-tag in the Require header; or

b) try to complete the session by relaxing the requirement on the usage of the "integration of resource management in SIP" extension as described in RFC 3312 [30] and proceed with the procedures described in subclause 5.1.3.2 and subclause 6.1.

~~NOTE: An example of where a new request would not be built is where knowledge exists within the UE, or interaction occurs with the user, such that it is known that the resultant SDP would describe a session that did not meet the user requirements.~~

5.1.3.2. INVITE request not requiring "integration of resource management in SIP"

This procedure is initiated upon the reception of a first 420 (Bad Extension) response to an initial INVITE request, the response containing the "precondition" option-tag in the Unsupported header field value.

The UE may create another INVITE request addressed to the same destination the initial INVITE was sent. In creating this new initial INVITE request, the UE shall:

1) populate the From, To, Call-ID headers and the Request-URI as per the initial INVITE request;

2) include a Supported header that contains the "preconditions" and "100rel" option-tag, in addition to other supported option-tags;

3) set each of the media streams in inactive mode in SDP as described in draft-ietf-mmusic-sdp-new [39] and subclause 6.1 in this specification; and

4) forward the INVITE request as per regular procedures.

Upon receiving a provisional response or final response containing the remote SDP, the UE shall:

1) answer, if needed, the SIP response as per regular SIP procedures defined in RFC 3261 [26] and RFC 3262 [27]; and

2) initiate the regular resource reservation mechanism, as described in subclause 9.2.5.

When the above INVITE transaction is successfully completed, and when the local resource reservation procedure is complete, the UE shall create and forward a re-INVITE request including:

1) the From, To, Call-ID headers as per a re-INVITE request;

2) SDP in which the media streams previously set in inactive mode are set to active mode, according to the procedures described in draft-ietf-mmusic-sdp-new [39] and subclause 6.1 in this specification.

5.1.4 Call initiation - mobile terminating case

5.1.4.1 Initial INVITE

5.1.4.1.1 General

Upon receiving an initial INVITE request without containing either Supported: precondition or Require: precondition header values, the UE shall either follow the procedures described in subclause 5.1.4.1.2 or follow the procedures described in subclause 5.1.4.1.3~~generate a 421 (Extension Required) response indicating the required extension in the Require header field~~.

NOTE: Table A.4 specifies that UE support of forking is required in accordance with RFC 3261 [26].

Editor's Note: The above note needs further investigation.

5.1.4.1.2 Preconditions and reliable provisional responses required

Upon receiving an initial INVITE request without the "precondition" option-tag in either the Supported or Require header field values, if the UE is configured to require the usage of the "integration of resource management in SIP" extension as described in RFC 3312 [30], the UE shall generate a 421 (Extension Required) response indicating the required extension in the Require header field value.

Upon generating the first response to the initial INVITE request, the UE shall indicate the requirement for reliable provisional responses and specify it using the Require header mechanism.The UE shall send the 200 (OK) response to the initial INVITE request only after the local resource reservation has been completed.

~~NOTE: Table A.4 specifies that UE support of forking is required in accordance with RFC 3261 [26].~~

5.1.4.1.3 Preconditions and / or reliable provisional responses not required

Upon receiving an initial INVITE request without containing the "precondition" option-tag in either the Supported or Require header field values, if the UE is configured to not use either the "integration of resource management in SIP" extension as described in RFC 3312 [30] or the reliable provisional responses extension defined in RFC 3262 [27], the UE shall:

1) if the INVITE request includes the "100rel" option-tag in the Supported header field value, answer with a 183 (Session Progress) response and include the "100rel" option-tag in the Require header field in the response; or

2) if the INVITE request does not include the "100rel" option-tag in the Supported header field value, providing that the user accepts the session establishment, answer with a 200 (OK) response; and

3) in any of the above cases, set each of the media streams in inactive mode in SDP as described in draft-ietf-mmusic-sdp-new [39] and subclause 6.1 in this specification; and

4) initiate the regular resource reservation mechanisms, as described in subclause 9.2.5.

When the above INVITE transaction has successfully complete, and when the local resource reservation procedure has complete, the UE shall create and forward a re-INVITE request which shall include:

1) the From, To, Call-ID headers as per a re-INVITE request;

2) a Supported header containing the "preconditions" and "100rel" option-tags, in addition to other supported option-tags; and

3) SDP in which the media streams previously set in inactive mode are set to active mode, according to the procedures described in draft-ietf-mmusic-sdp-new [39] and subclause 6.1 in this specification.

Second proposed change

5.2.7.2 Mobile-originating case

The P-CSCF shall respond to all INVITE requests with a 100 (Trying) provisional response.

Upon receiving a response (e.g. 183 (Session Progress), 200 (OK)) to the initial INVITE request, the P-CSCF shall:

- if a media authorization token is generated by the PDF as specified in RFC 3313 [31] (i.e. when service-based local policy control is applied), insert the P-Media-Authorization header containing that media authorization token.

NOTE: Typically, the first 183 (Session Progress) response contains an SDP answer including one or more "m=" media descriptions, but it is also possible that the response does not contain an SDP answer or the SDP does not include at least an "m=" media description. However, the media authorization token is generated independently of the presence or absence of "m=" media descriptions and sent to the UE in the P-Media-Authorization header value. The same media authorization token is used until the session is terminated. For further details see 3GPP TS 29.207 [12].

~~When the P-CSCF sends the UPDATE request towards the S-CSCF, t~~The P-CSCF shall also include the access-network-charging-info parameter in the P-Charging-Vector header in the first request originated by the UE that traverses the P-CSCF, as soon as the charging information is available in the P-CSCF, e.g., after the local resource reservation is complete. Typically, this first request is an UPDATE request if the remote UA supports the "integration of resource management in SIP" extension or a re-INVITE request if the remote UA does not support the "integration of resource management in SIP" extension. See subclause 5.2.7.4 for further information on the access network charging information.

5.2.7.3 Mobile-terminating case

When the P-CSCF receives an initial INVITE request destined for the UE, it will contain the URI of the UE in the Request-URI, and a single preloaded Route header. The received initial INVITE request will also have a list of Record-Route headers. Prior to forwarding the initial INVITE to the URI found in the Request-URI, the P-CSCF shall:

- if a media authorization token is generated by the PDF as specified in RFC 3313 [31] (i.e. when service-based local policy control is applied), insert the P-Media-Authorization header containing that media authorization token.

NOTE: Typically, the initial INVITE request contains an SDP offer including one or more "m=" media descriptions, but it is also possible that the INVITE request does not contain an SDP offer or the SDP does not include at least an "m= media description. However, the media authorization token is generated independently of the presence or absence of "m=" media descriptions and sent to the UE in the P-Media-Authorization header value. The same media authorization token is used until the session is terminated. For further details see 3GPP TS 29.207 [12].

In addition, the P-CSCF shall respond to all INVITE requests with a 100 (Trying) provisional response.

~~When the P-CSCF sends 180 (Ringing) or 200 (OK) (to INVITE) towards the S-CSCF, t~~The P-CSCF shall also include the access-network-charging-info parameter in the P-Charging-Vector header in the first request or response originated by the UE that traverses the P-CSCF, as soon as the charging information is available in the P-CSCF e.g., after the local resource reservation is complete. Typically, this first response is a 180 (Ringing) or 200 (OK) response if the remote UA supports the "integration of resource management in SIP" extension, or a re-INVITE request if the remote UA does not support the "integration of resource management in SIP" extension. See subclause 5.2.7.4 for further information on the access network charging information.

Third and last proposed change

6 Application usage of SDP

6.1 Procedures at the UE

Usage of SDP by the UE:

1. In order to authorize the media streams, the P-CSCF and S-CSCF have to be able to inspect the SDP payloads. Hence, the UE shall not encrypt the SDP payloads.

2. An INVITE request generated by a UE shall contain SDP payload. The SDP payload shall reflect the calling user's terminal capabilities and user preferences for the session. The UE shall order the SDP payload with the most preferred codec listed first.

3. If the SIP request includes a "precondition" option-tag in the Require header field value~~In addition~~, the calling user shall indicate the desired QoS for the session, using the segmented status type. In an initial INVITE request the UE shall indicate that it mandates local QoS and that this precondition is not yet satisfied, i.e. the UE shall include the following preconditions:

   a=des: qos mandatory local sendrecv

   a=curr: qos local none

   If the SIP request does not include the "precondition" option-tag in the Require header, the UE shall not indicate that it mandates local QoS. The UE may indicate its desire for optional local QoS, by including the following preconditions:

   a=des:qos optional local sendrecv

~~3~~4. Providing that the INVITE request received by the UE contains an SDP offer including one or more "m=" media descriptions, the first 183 (Session Progress) provisional response that the UE sends, shall contain the answer for the SDP received in the INVITE. The said SDP answer shall reflect the called user's terminal capabilities and user preferences.

   If the SIP "integration of resource management in SIP" extension as described in RFC 3312 [30] is not used, in the first SDP answer the UE sends (as described in subclause 5.1.4.1.2), the UE shall set the each media streams in inactive mode by setting including an "a=inactive" line, according to the procedures described in draft-ietf-mmusic-sdp-new [39] and subclause 5.1.4.1.2 in this specification.

   If the UE is setting one or more media streams in active mode, it shall apply the procedures described in draft-ietf-mmusic-sdp-new [39] with respect to setting the direction of media streams.

~~4~~5. When the UE sends a 183 (Session Progress) response with SDP payload including one or more "m=" media descriptions, if the SIP "integration of resource management in SIP" extension as described in RFC 3312 [30] is supported by the calling UE, the called UE ~~it~~ shall request confirmation for the result of the resource reservation at the originating end point.

~~5~~6. During session establishment procedure, SIP messages shall only contain SDP payload if that is intended to modify the session description.

~~6~~7. For "video" and "audio" media types that utilize the RTP/RTCP, the UE shall specify the proposed bandwidth for each media stream utilizing the "b=" media descriptor and the "AS" bandwidth modifier in the SDP.

If the media line in the SDP indicates the usage of RTP/RTCP, in addition to the "AS" bandwidth modifier in the media-level "b=" line, the UE shall include two media-level "b=" lines, one with the "RS" bandwidth modifier and the other with the "RR" bandwidth modifier as described in RFC 3556 [56] to specify the required bandwidth allocation for RTCP.

For other media streams the "b=" media descriptor may be included. The value or absence of the "b=" parameter will affect the assigned QoS which is defined in 3GPP TS 29.208 [13].

NOTE 1: In a two-party session where both participants are active, the RTCP receiver reports are not sent, therefore, the RR bandwidth modifer will typically get the value of zero.

~~7~~8. The UE shall include the MIME subtype "telephone-event" in the "m=" media descriptor in the SDP for audio media flows that support both audio codec and DTMF payloads in RTP packets as described in RFC 2833 [23].

~~8~~9. The UE shall inspect the SDP contained in any SIP request or response, looking for possible indications of grouping of media streams according to RFC 3524 [54] and perform the appropriate actions for IP-CAN bearer establishment for media according to IP-CAN specific procedures (see subclause B.2.2.5 for IP-CAN implemented using GPRS).

~~9~~10. If an IP-CAN bearer is rejected or modified, the UE shall, if the SDP is affected, update the remote SIP entity according to RFC 3261 [26] and RFC 3311 [29].

1~~0~~1. If the UE builds SDP for an INVITE request generated after receiving a 488 (Not Acceptable Here) response, as described in subclause 5.1.3.1, the UE shall include SDP payload containing a subset of the allowed media types, codecs and other parameters from the SDP payload of all 488 (Not Acceptable Here) responses related to the same session establishment attempt (i.e. a set of INVITE requests used for the same session establishment). The UE shall order the codecs in the SDP payload according to the order of the codecs in the SDP payload of the 488 (Not Acceptable Here) response.

NOTE 2: The UE may be attempting a session establishment through multiple networks with different policies and potentially may need to send multiple INVITE requests and receive multiple 488 (Not Acceptable Here) responses from different CSCF nodes. The UE therefore takes into account the SDP contents of all the 488 (Not Acceptable Here) responses received related to the same session establishment when building a new INVITE request.