3GPP TSG-CN Meeting #24 **NP-040164**
2nd – 4th June 2004. Seoul, Korea.

Source: **CN1**

Title: **Interworking with non-IMS SIP UEs (precondition fallback)**

Agenda item: **5.1**

Document for: **INFORMATION**

3GPP TSG-CN1 Meeting #34 **Tdoc N1-041097**
Zagreb, Croatia 10 – 14 May 2004

Release: Rel-6
Work Item: IMS2

To: SA2
Cc: CN, CN3

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Attachments: N1-041096 (agreed CR on precondition fallback), N1-040962 (alternative proposal)

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1. Overall Description:

At their last meeting, CN1 has agreed the attached CR against 24.229 on interworking with non-IMS SIP UEs (tdoc N1-041096). The solution proposed in this CR is aligned with the end-to-end call flow scenarios as described in TR 29.962 ("Signalling interworking between the 3GPP profile of the Session Initiation Protocol (SIP) and non-3GPP SIP usage"). This TR was produced by CN3 and SA2 based interworking related decisions on its content.

<u>Solution as proposed in the CR / TR 29.962:</u>

```
calling SIP UA                                              called SIP UA
         ---------INVITE (preconditions)---------------------------->
         <--------------------420 (Bad Extension)--------------------

         ---------INVITE (no preconditions, media=inactive)---->
         <-------------200 (OK, media-authorization token)---------
```

<start of resource reservation>
<resource reservation successful>

```
         ---------re-INVITE (media=active)------------------------->
         <--------------------200 (OK)-------------------------------
```

During the presentation of the CR concerns were raised against the solution. The main concerns were:

- the called non-IMS UE would need to handle the "inactive" indication in the SDP, although it is not a common procedure to receive this indication in the first SDP Offer. One foreseeable scenario is, that the called user picks up the phone before the resources have been reserved at the calling side. In this case anything the called user would speak into the phone would not reach the calling side until the resources have been allocated; and

- that there is a possibility that the called non-IMS UE would not support the SDP "inactive" flag, as it is not part of the SDP RFC 2327 ("old" SDP). This flag is part of basic SDP Offer / Answer behaviour (RFC 3264) and needs to be supported by any implementation that claims compliance to RFC 3261.

An alternative solution was proposed and is also attached, which in summary proposes the following interworking signalling flow:

Alternative solution:

calling SIP UA called SIP UA

```
---------INVITE (preconditions)---------------------------->
<---------------------420 (Bad Extension)--------------------
```

<start of resource reservation>
<resource reservation successful>

```
---INVITE (no precondition, media=active)-------------->
<---------------------200 (OK)----------------------------------
```

Also this proposal was discussed and it was raised that SBLP at the calling side will not work with this solution as the calling UE has no media authorization token at the time of resource reservation. Therefore it was proposed that SBLP shall not be used in this scenario. Furthermore this proposal may lead to media PDP context modifications due to media negotiation taking place during the second INVITE flow. Some delegates were of the opinion that such a media PDP context modification might not be needed, as the reserved bandwidth should stay unchanged even if this would be overprovisioning of bandwidth for the selected media.

CN1 agreed to go forward with the solution as outlined in TR 29.962 as long as SA2 does not indicate any necessity to change the scenario.

2. Actions:

To SA2 group.

ACTION: CN1 kindly asks SA2 to study whether the shortcomings of the solution in the agreed CR (N1-041096, which is according to TR 29.962) are acceptable. If they are not acceptable, CN1 asks SA2 whether the alternative proposal should be considered. SA2 is kindly asked to indicate this fact towards CN plenary, as the accepted CR against 24.229 (N1-041096) could then there be rejected.

3. Date of Next TSG-CN1 Meetings:

CN1_35	16th – 20th August 2004	Sophia Antipolis, France (ETSI)
CN1_36	15th – 19th November 2004	Asia?

3GPP TSG-CN1 Meeting #34 **Tdoc N1-040962**
Zagreb, Croatia 10 – 14 May 2004

CR-Form-v7

CHANGE REQUEST

⌘	**24.229**	**CR**	**CR 622**	⌘ **rev**	**-1**	⌘ 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 [ ]

Title:	⌘	Interworking with non-IMS SIP clients		
Source:	⌘	Lucent Technologies		
Work item code:	⌘	IMS2	***Date:*** ⌘	20/03/2004
Category:	⌘	**B**	***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:	⌘	The document TS 23.228 states: " It is possible that the external SIP client does not support one or more of the SIP extensions required for IMS end points to set up IMS sessions (e.g. Preconditions, Update, 100Rel) as described in 3GPP TS 24.229 [10a], then the UE or other SIP user agents within the IMS should be able to fall back to SIP procedures which allow interworking towards the external client."
Summary of change:	⌘	The procedure and associated text that describes "the fall back procedures which allow interworking towards the external client" is provided.
Consequences if not approved:	⌘	Noncompliance with TS 23.228

Clauses affected:	⌘	5.1.3.1 and 5.1.4.1				
		Y	**N**			
Other specs affected:	⌘			Other core specifications	⌘	
				Test specifications		
				O&M Specifications		
Other comments:	⌘	This CR was originally submitted to the CN1#33bis meeting in Sophia Antipolis, France, March 30 - April 2, 2004. It was postponed at the meeting				

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.

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.

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.

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) send a new initial INVITE request without "precondition" option-tag in the Require and Supported headers, and addressed to the same destination that the initial INVITE was sent. Prior to sending the new initial INVITE request, the UE will reserve local resources for each media stream offered in the SDP offer that will be included in the new initial INVITE request. When the local resource reservation procedure is successfully completed (i.e. the UE is ready to receive and send the offered media streams), the UE shall in the new initial INVITE request:

   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 "100rel" tag, in addition to other supported tags; and

   3) forward the new initial INVITE request containing the SDP offer as defined in RFC 3261 [26] and RFC 3262 [27].

Upon receiving a 1xx or 2xx response containing the SDP answer, the UE shall:

   1) relinquish all local resources that were reserved for the media streams and were rejected as indicated in the SDP answer; and

   2) answer the SIP response as per regular SIP procedures defined in RFC 3261 [26] and RFC 3262 [27].

If the UE receives a second 420 (Bad Extension) response for the new initial INVITE request, the UE shall abort the session attempt.

NOTE 2: 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.

5.1.4 Call initiation - mobile terminating case

5.1.4.1 Initial INVITE

Upon receiving an initial INVITE request without containing either Supported: precondition or Require: precondition header values, the UE shall either:

a) generate a 421 (Extension Required) response indicating the required extension in the Require header field;~~.~~ or

b) reserve local resources for each media stream that it has accepted, and it will indicate in the SDP answer that will be included in the response to the initial INVITE request. When reserving the local resources, the UE may utilise the authorization token if received in the initial INVITE request; and

forward the response to the initial INVITE request without either the Supported: precondition or Require: precondition header values, and containing the SDP answer.

~~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 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].

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	⌘

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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 aligne s 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.