**3GPP TSG-CT WG1 Meeting #123-eC1-202605**

**Electronic meeting, 16-24 April 2020**

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| *CR-Form-v12.0* |
| **CHANGE REQUEST** |
|  |
|  | **24.183** | **CR** | **0063** | **rev** | **1** | **Current version:** | **16.3.0** |  |
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| *For* [***HE******LP***](http://www.3gpp.org/3G_Specs/CRs.htm#_blank)*on using this form: comprehensive instructions can be found at* [*http://www.3gpp.org/Change-Requests*](http://www.3gpp.org/Change-Requests)*.* |
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| ***Proposed change affects:*** | UICC apps |  | ME | **X** | Radio Access Network |  | Core Network | **X** |

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| ***Title:***  | Use preconditions for CRS when terminating UE supports precondition |
|  |  |
| ***Source to WG:*** | Huawei, China Telecom, China Unicom, HiSilicon |
| ***Source to TSG:*** | C1 |
|  |  |
| ***Work item code:*** | eIMSVideo |  | ***Date:*** | 2020-04-16 |
|  |  |  |  |  |
| ***Category:*** | **B** |  | ***Release:*** | Rel-16 |
|  | *Use one of the following categories:****F*** *(correction)****A*** *(mirror corresponding to a change in an earlier release)****B*** *(addition of feature),* ***C*** *(functional modification of feature)****D*** *(editorial modification)*Detailed explanations of the above categories canbe found in 3GPP [TR 21.900](http://www.3gpp.org/ftp/Specs/html-info/21900.htm). | *Use one of the following releases:Rel-8 (Release 8)Rel-9 (Release 9)Rel-10 (Release 10)Rel-11 (Release 11)Rel-12 (Release 12)**Rel-13 (Release 13)Rel-14 (Release 14)Rel-15 (Release 15)Rel-16 (Release 16)* |
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| ***Reason for change:*** | Precondition mechanism is important for providing video CRS service. Because more resources are nceeded to play video CRS, if precondition is not used, clipping of video CRS media will be more obvious than audio CRS. To ensure video CAT user’s experience, precondition is recommended by some operators to be used for video CRS.In the part of AS actions for gateway model, there is no description for when the AS shall or may use precondition for CRS.We suggest TS 24.183 could take into account to clarify when the precondition shall or may be used for CRS. |
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| ***Summary of change:*** | Specify when the AS shall or may be used CRS. |
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| ***Consequences if not approved:*** | Unclear specification for using precondition for CRS. |
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| ***Clauses affected:*** | 2, 4.5.5.3.6 |
|  |  |
|  | **Y** | **N** |  |  |
| ***Other specs*** |  | **X** |  Other core specifications  | TS/TR ... CR ...  |
| ***affected:*** |  | **X** |  Test specifications | TS/TR ... CR ...  |
| ***(show related CRs)*** |  | **X** |  O&M Specifications | TS/TR ... CR ...  |
|  |  |
| ***Other comments:*** |  |
|  |  |
| ***This CR's revision history:*** |  |

\*\*\*\*\* Next change \*\*\*\*\*

# 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 TR 21.905: "Vocabulary for 3GPP Specifications".

[2] 3GPP TS 22.183: "Customized Ringing Signal (CRS) Requirements; Stage 1".

[3] 3GPP TS 24.229: "IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".

[4] RFC 3959: "The Early Session Disposition Type for the Session Initiation Protocol (SIP)".

[5] 3GPP TS 24.623: "Extensible Markup Language (XML) Configuration Access Protocol (XCAP) over the Ut interface for Manipulating Supplementary Services".

[6] 3GPP TS 24.238: "Session Initiation Protocol (SIP) based user configuration; Stage 3".

[7] RFC 6086 (January 2011): "Session Initiation Protocol (SIP) INFO Method and Package Framework".

[8] RFC 7462 (March 2015): "URNs for the Alert-Info Header Field of the Session Initiation Protocol (SIP)".

[9] RFC 4796: "The Session Description Protocol (SDP) Content Attribute".

[10] RFC 3960: "Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)".

[11] 3GPP TS 24.628: "Common Basic Communication procedures; Protocol specification".

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

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

\*\*\*\*\* Next change \*\*\*\*\*

##### 4.5.5.3.6 AS Actions for Gateway model

The AS performing the Gateway model shall follow the procedure as specified in RFC 3960 [10] and annex G in 3GPP TS 24.628 [11] with the additional procedures described in this subclause.

Upon receiving an initial INVITE request from the originating UE, the AS shall forward the initial INVITE request to the terminating UE after inserting an Alert-Info header field with an URN "urn:alert:service:crs".

Upon receiving the first reliable SIP 18x response to the initial INVITE request, the AS:

a) may contact the MRF to request CRS resource; and

b) shall forward the reliable SIP 18x response to the originating UE.

Upon receiving the PRACK request of the first reliable 18x response from originating UE, the AS shall forward the PRACK request to the terminating UE and contact the MRF to request CRS resource if it has not been previously requested.

When the video media feature tag is not included in the Contact header field of the previously received 18x response from terminating UE and there is no video description in the SDP answer included in the 18x response, the AS shall not request video CRS resource from MRF, and shall not apply video CRS media to the terminating UE.

After receiving 180 (Ringing) response or receiving a SIP 200 (OK) response to the PRACK request of the first reliable SIP 18x response from terminating UE, the AS shall update media of CRS service with terminating UE by UPDATE request as specified in RFC 3311 [12] with:

a) P-Early-Media header field with a "sendrecv" value or a "sendonly" value;

b) The SDP offer, which is based on the CRS information received from the MRF and includes a=content media-level attribute with a "g.3gpp.crs" value, the media types can be different from the media types required in the SDP answer of previous 18x response from terminating UE; and

c) precondition mechanism as specified in RFC 3312 [xx] if "precondition" option-tag is included in the Require header field of a received 18x response.

In the above UPDATE request, based on local policy, the AS may use the precondition mechanism as specified in RFC 3312 [xx] if "precondition" option-tag is not included in the Require header field of received 18x responses.

If the terminating UE requires the use of precondition mechanism, the AS shall not instruct the MRF to start applicable media for the CRS service before the terminating UE has indicated that preconditions are fulfilled. The point when the AS instruct the MRF to start applicable media for the CRS service is based on local policy.

Upon receiving a SIP 200 (OK) response to the INVITE request from the terminating UE, the AS shall instruct the MRF to stop media for the CRS service and update media for conversation. If the AS is going to update media with both originating side and terminating side, the AS shall:

a) send an offerless re-INVITE request to the terminating side;

b) upon receiving a SIP response to the re-INVITE request containing an SDP offer from the terminating side, generate an UPDATE request as specified in RFC 3311 [12] to send an SDP offer to the originating UE. The SDP offer shall only contain the media components which appeared both in the SDP offer contained in the SIP response to the re-INVITE request and the previously stored SDP offer in the initial INVITE request, and set the port number of the corresponding m-line to zero if it has been set to zero during previous SDP negotiation; and

c) upon receiving a 200 (OK) response to the UPDATE request from the originating side, generate an SDP answer to the terminating side, included in the ACK request associated with the re-INVITE request. The SDP answer shall be based on the SDP answer contained in the 200 (OK) response to the UPDATE request, and for the media components which do not appear in the SDP answer in the 200 (OK) response, set the port number of the corresponding m-line to zero.

Upon receiving a SIP 4xx, 5xx or 6xx response from the terminating UE, the AS shall:

a) instruct the MRF to stop the media for the CRS service; and

b) forward the final response to the originating UE.