Technical Specification Group Services and System Aspects **TSGS#23(04)0357**
Meeting #24, Seoul, South Korea
7-10 June 2004

Source: **TSG-SA WG4**

Title: **CRs TS 26.236 on "RTCP usage for IMS" (Release 5 and Release 6)**

Document for: **Approval**

Agenda Item: **7.4.3**

The following CRs, agreed at the TSG-SA WG4 meeting #31, are presented to TSG SA #24 for approval.

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.236	011	1	Rel-5	RTCP usage for IMS	F	5.4.0	S4	TSG-SA WG4#31	S4-040345
26.236	012		Rel-6	RTCP usage for IMS	A	5.4.0	S4	TSG-SA WG4#31	S4-040346

3GPP TSG-SA4 Meeting #31
Montreal, Canada, 17-21 May 2004

Tdoc ⌘S4-040345

CR-Form-v7

CHANGE REQUEST

⌘	**26.236**	**CR**	**011**	⌘ **rev**	**1**	⌘ Current version:	**5.4.0** ⌘

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

Title:	⌘	RTCP usage for IMS		
Source:	⌘	TSG SA WG4		
Work item code:	⌘	IMS-CODEC	***Date:*** ⌘	08/06/2004
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:	⌘	In point to point sessions involving only speech, RTCP might disrupt the RTP speech flow and cause impairment to speech quality. Also the SIP IETF reference is obsolete
Summary of change:	⌘	The implementation of RTCP is mandated. The use of RTCP is recommended in all cases except in point to point sessions involving only speech. The way to turn on and off RTCP is specified. The reference to SIP IETF specification has been updated to align with CN1.
Consequences if not approved:	⌘	VoIMS QoS can not be achieved.

Clauses affected:	⌘	2; 7.1, 7.3, 7.4				
		Y	**N**			
Other specs affected:	⌘		X	Other core specifications	⌘	
			X	Test specifications		
			X	O&M Specifications		
Other comments:	⌘					

How to create CRs using this form:

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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.

Change in Clause 2

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[2] IETF RFC 2327: "SDP: Session Description Protocol".

[3] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications", Schulzrinne H. et al, July 2003.

[4] IETF RFC 3551: "RTP Profile for Audio and Video Conferences with Minimal Control", Schulzrinne H. and Casner S., July 2003.

[5] 3GPP TS 26.235: "Packet switched conversational multimedia applications; Default codecs".

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[12] 3GPP TS 26.071: "Mandatory Speech Codec speech processing functions; AMR Speech Codec; General description".

[13] 3GPP TS 26.090: "AMR speech Codec; Transcoding Functions".

[14] 3GPP TS 26.073: "AMR speech Codec; C-source code".

[15] 3GPP TS 26.104: "ANSI-C code for the floating-point Adaptive Multi-Rate AMR speech codec".

[16] 3GPP TS 26.171 (Release 5): "AMR speech codec, wideband; General description".

[17] 3GPP TS 26.190 (Release 5): "Mandatory Speech Codec speech processing functions AMR Wideband speech codec; Transcoding functions".

[18] 3GPP TS 26.201 (Release 5): "AMR speech codec, wideband; Frame structure".

[19] 3GPP TS 26.235: "Packet switched conversational multimedia applications; Default codecs ". Annex B: "RTP payload format and storage format for AMR and AMR-WB audio".

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[22] ISO/IEC 14496-2 (1999): "Information technology - Coding of audio-visual objects - Part 2: Visual".

[23] IETF RFC 3016: "RTP Payload Format for MPEG-4 Audio/Visual Streams".

[24] ITU-T Recommendation H.263 (annex X): "Annex X: Profiles and levels definition".

[25] 3GPP TS 26.235: "Packet Switched Conversational Multimedia Applications; Default Codecs ". Annex C: "ITU-T H.263 MIME media type registration".

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[27] IETF RFC 2793: "RTP Payload for Text Conversation".

[28] IETF RFC 3556: "Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) bandwidth", Casner S., July 2003.

End of change in Clause 2

Change in Clause 7.1

7.1 Bandwidth

The bandwidth information of each media type shall be carried in SDP messages in both session and media type level during codec negotiation, session establishment and resource reallocation. Note that for RTP based applications, 'b=AS:' gives the RTP "session bandwidth" (including UDP/IP overhead) as defined in section 6.2 of [3].

The bandwidth for RTCP traffic shall be described using the "RS" and "RR" SDP bandwidth modifiers at media level, as specified by [28]. Therefore, a conversational multimedia terminal shall include the "b=RS:" and "b=RR:" fields in SDP, and shall be able to interpret them. There shall be a limit on the allowed RTCP bandwidth for a session signalled by the terminal. This limit is defined as follows:

- 4000 bps for the RS field (at media level);
- 3000 bps for the RR field (at media level).

If the session described in the SDP is a point-to-point speech only session (~~as described in~~ see ~~section~~ clause 7.4), the UE should request the deactivation of RTCP by setting its RTCP bandwidth modifier to zero.

If a UE receives SDP bandwidth modifiers for RTCP equal to zero from the originating UE, it should reply (via the SIP protocol) by setting its RTCP bandwidth using SDP bandwidth modifiers with values equal to zero.

End of change in Clause 7.1

Change in Clause 7.3

7.3 RTP receiver

The RTP receiver implementation shall also include an RTCP implementation.

The RTP receiver implementation and functionality including lost and delayed packet processing as well as jitter buffer is out of scope of the present document.

End of change in Clause 7.3

Change in Clause 7.4

7.4 RTP sender

The RTP sender implementation shall also include an RTCP implementation.

RTCP packets should be sent for all types of multimedia sessions except for point-to-point speech only sessions (i.e., using AMR and the AMR-WB codecs where synchronization with other RTP transported media or remote end-point aliveness information are not needed). For point-to-point speech only sessions, a UE should not send RTCP packets. Turning off RTCP can be done by setting to zero the SDP bandwidth modifiers (RR and RS) described in clause 7.1.

When RTCP is turned off (for point-to-point speech only sessions) and the media is put on hold, the terminal should re-negotiate the RTCP bandwidth with SDP bandwidth modifiers values greater than zero, and send RTCP packets to the other end, following the rules given below. This allows the remote end to detect link aliveness during hold. When media is resumed, the resuming terminal should turn off the RTCP sending again through a re-negotiation of the RTCP bandwidth with SDP bandwidth modifiers (as described in clause 7.1) equal to zero.

When RTCP is turned off (for point-to-point speech only sessions) and if sending of an additional associated RTP flow becomes required and both RTP flows need to be synchronized, or if transport feedback due to lack of end-to-end QoS guarantees is needed, a terminal should re-negotiate the bandwidth for RTCP by sending an SDP with the RS bandwidth modifier greater than zero.

Note: For speech sessions where RTCP is not turned off, to reduce the potential disruption of RTCP onto the RTP flow, it is beneficial to keep the RTCP bandwidth and the size of RTCP packets as small as possible. RTCP packet size can be minimized by only using the optional parts of RTCP (according to [3]) which are required by the application. A practical size limit for the RTCP sender is in the order of 2 to 5 times the RTP packet size. Additionally, the RTCP sender can attempt to schedule RTCP packets during speech inactivity periods. For example, if an RTCP packet is scheduled at a future time and a silence period starts, this RTCP packet could be sent immediately. The subsequent RTCP packets would be scheduled according to the normal rules (i.e. as if the previous packet was sent as originally scheduled).

End of change in Clause 7.4
End of document

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CHANGE REQUEST

⌘	**26.236**	**CR**	**012**	⌘ **rev**	**-**	⌘ Current version:	**5.4.0** ⌘

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Category:	⌘	**A**		***Release:*** ⌘	Rel-6
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