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PERFORMANCE OF RAB MAPPING AND ROHC FOR THE SUPPORT OF VOIP OVER UMTS

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ABSTRACT

In a UMTS network, the Radio Access Bearers (RAB) configuration has a direct impact on radio resource usage. The more suited the RAB is to data flows, the more efficient it is. This paper presents an optimal RAB combination for VoIMS (VoIP with IP Multimedia Subsystem in the Core Network). The proposed RAB makes an efficient use of ROHC (Robust Header Compression) which improves the physical layer QoS. The main advantage of the new RAB combination resides in adapting the throughput for SIP (Session Initiation Protocol) signalling and improving the call setup delays for a packet switched transmission. This is achieved via an efficient algorithm of Transport Format selection in conjunction with the flexible rate matching at the physical layer.

I. INTRODUCTION

Currently, intensive researches are targeting the Packet-Switched (PS) services performance over UTRAN (UMTS Terrestrial Radio Access Network), specifically those based on the Internet Protocol (IP). An IP-based UTRAN provides operators with flexibility for routing wireless traffic and decreases the cost of traffic delivery. With the expected introduction of IMS (IP Multimedia Subsystem) in the Core Network of UMTS mobile networks, the support of real time IP-based services as VoIP in UMTS with an adapted QoS will become possible. VoIP and VoIMS are equivalent in this paper.

For the support of VoIP over UMTS, 3GPP has adopted AMR (Adaptive Multi-Rate) speech codec [1] also used for the circuit-switched transmission. At the session level, VoIMS uses RTP (Real time Transport Protocol) in conjunction with RTCP (Real Time Transport Control Protocol) [2] to report feedbacks on the received quality. At the transport level, UDP (User Datagram Protocol) is used to avoid retransmissions, undesirable for delay sensitive services like VoIP. At the UTRAN level, RTP/UDP/IP and RTCP/UDP/IP packets are transmitted either on the same RAB as depicted in Fig.1 or via two RABs as shown in Fig. 2 [3]. A RAB corresponds to a given configuration of UTRAN layers including the PDCP (Packet Data Convergence Protocol) [4], RLC (Radio Link Control) [5], MAC (Medium Access Control) [6] and the physical layer. Hence, a RAB has a direct influence on the obtained QoS. As shown in Fig. 1 and Fig. 2, a VoIMS RAB combination includes 3 components. First, one (or two) PS Conversational RAB(s) conveys RTP and RTCP flows. Second, a PS Interactive RAB conveys

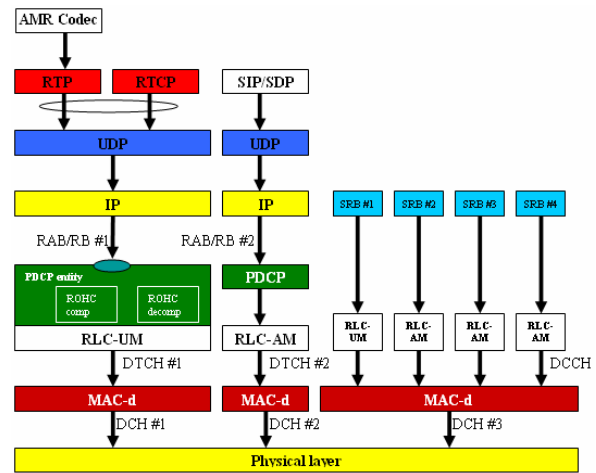


Figure 1: VoIMS RAB with multiplexed RTP/RTCP flows

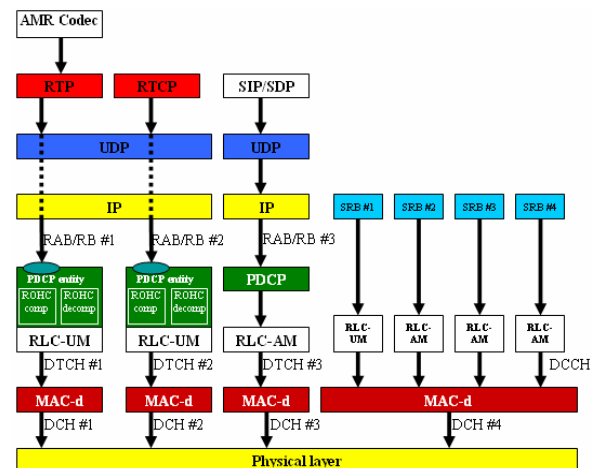


Figure 2: VoIMS RAB with RTP and RTCP on separate RABs

SIP signalling to manage VoIP sessions at IMS level. Third, Signalling Radio Bearers (SRB) deal with RRC (Radio Resource Control) and NAS (Non Access Stratum) signalling.

This paper proposes an optimal RAB combination for VoIMS that outperforms current 3GPP PS RABs [7]-[8]. This solution adapts the SIP signalling throughput and improves the setup delays of a PS call. It also makes a good use of ROHC for which the gains are evaluated.

The organisation of this paper is the following. Section II shows the challenges and issues for VoIMS. Section III analyses the reference VoIP 3GPP RAB configurations. Section IV shows the interest of using ROHC header compression in a VoIP context. Section V describes the optimal RAB combination for VoIMS and analyses its performance. Finally, conclusions are drawn in section VI.

II. CURRENT CHALLENGES AND ISSUES FOR VOIMS

A. Overhead and ROHC header compression

VoIP transmission is marked by an important overhead due to the higher layers encapsulation. As shown in Fig. 3, the protocol headers include a 40 bytes long IPv6 header, a 8 bytes long UDP header and a 12 bytes long (or more) RTP header. The total overhead is of at least 60 bytes. As the AMR 12.2kbps payload size is 32 bytes and the duration of TTI (Transmission Time Interval) for VoIP is 20ms, this leads to a necessary throughput of 36.8 kbps (92bytes/20ms) at IP level, which is much more significant than the current CS 12.2 AMR service throughput. Hence, a header compression is necessary to decrease the overhead. In this context, the Robust Header Compression (ROHC) algorithm was standardized by the IETF [9] and is an integral part of the 3GPP-UMTS [10]. ROHC is located in the PDCP layer to compress all the protocol headers starting from IP. ROHC defines different compression levels or states that are associated to various compression rates. The transient state performs no compression on packets. When static fields are no longer transmitted and therefore compression rates become higher, ROHC moves to the steady state. In this state, ROHC may reduce the overhead from 60 to 4 bytes (Fig. 3). The packet size is then 36 bytes and the corresponding throughput at IP level is cut down to 14.8kbps (36 bytes/ 20ms), which is near of 12.2 kbps associated to the current CS AMR service.

B. SIP signalling

Another challenge for VoIMS is to perform a fast and reliable transmission of SIP signalling. VoIP support [7]-[8] define non optimal RABs for SIP. Indeed, at session level, SIP signalling permits to initiate, modify and release a VoIP call and then, it is significant to provide it

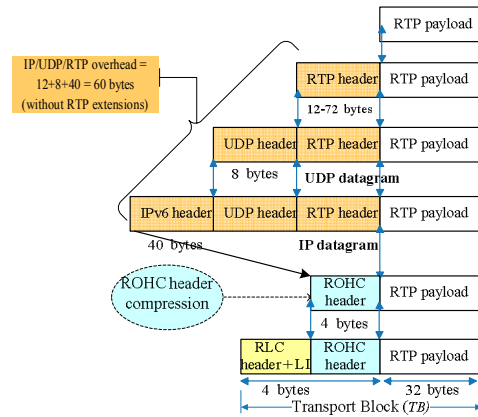


Figure 3: VoIP overhead and ROHC header compression

with enough radio resources to reduce call setup delays. The current 3GPP PS RAB combinations attribute low-throughputs for SIP (8/16kbps), which results in unacceptable call establishment durations.

However, the reservation of a significant amount of bandwidth for SIP during the whole VoIP call would waste radio resources. In fact, SIP signalling occurs mainly at call establishment and release. It is therefore important to have a good trade-off between the call setup delays and the allocated radio resources.

III. SYNTHESIS OF THE CURRENT PS RAB COMBINATIONS

Existing PS 3GPP RABs [7]-[8] are marked by their high diversity regarding to the way to convey RTP, RTCP and SIP streams, which makes their understanding as well as their comparison a quite complex operation. For this reason, an exhaustive analysis is made in this section and recapitulated in Table 1 to classify the current 3GPP PS RAB combinations in a methodical way. As depicted in Table 1, the performance of RAB mapping depends on the multiplexing of the different flows which impacts

Table 1: Performance analysis of 3GPP RAB configurations for VoIP

	config. 1	config. 2	config. 3	config. 4
Features	<ul style="list-style-type: none"> - RTP flow on a 40kbps PS Conversational RAB - RTCP flow not supported - SIP flow on a 16kbps PS Interactive RAB - SF in DL=32 	<ul style="list-style-type: none"> - RTP flow on a 40kbps PS Conversational RAB - RTCP flow on a dedicated PS RAB 16kbps - SIP flow on a dedicated PS Interactive RAB 16kbps - SF in DL=32 	<ul style="list-style-type: none"> - RTP flow on a 16kbps PS Conversational RAB - RTCP flow multiplexed with SIP on a PS Interactive RAB 8kbps - SF in DL=64 	<ul style="list-style-type: none"> - RTP and RTCP flows multiplexed on a PS Conversational RAB 40kbps - SIP flow on a PS Interactive RAB 16kbps - SF in DL=32
Advantages	<ul style="list-style-type: none"> - Only two PDP Context used simultaneously - Low delay for RTP flows as they are conveyed alone without RTCP 	<ul style="list-style-type: none"> - low delay for RTP flows as no multiplexing with RTCP - feedbacks on the quality of reception available with RTCP 	<ul style="list-style-type: none"> - Low delay for RTP flows as no multiplexing with RTCP - feedbacks on the quality of reception available with RTCP - Only two simultaneous PDP Contexts 	<ul style="list-style-type: none"> - Only two PDP Contexts used simultaneously - Better synchronization between RTCP feedbacks and RTP flow - feedbacks on the quality of reception available with RTCP
Drawbacks	<ul style="list-style-type: none"> - No feedback on the quality of reception is available to the sender - Unsuitable radio resource allocation for SIP during the VoIP call - Unsuitable radio resource allocation for RTP flows when RoHC is applied - Too long call setup duration 	<ul style="list-style-type: none"> - Hard synchronisation between RTP and RTCP flows - High number of PDP contexts used simultaneously - Unsuitable radio resource allocation for SIP during the VoIP call - Unsuitable radio resource allocation for RTP flows when RoHC is applied - Too long call setup duration 	<ul style="list-style-type: none"> - Hard synchronisation between RTP and RTCP flows - High number of PDP contexts used simultaneously - high delays for SIP and then too long call establishment duration - Too much Transport Formats for RTP flows (huge complexity of the treatment of the RB associated to RTP) 	<ul style="list-style-type: none"> - Seldom low delays for RTP flows when uncompressed RTP packets transmitted with RTCP packets. Correction of these delays during the following TTI - Unsuitable radio resource allocation for SIP during the call - Unsuitable radio resource allocation for RTP flows when RoHC is applied - Too long call setup duration

the RTP and SIP delays, the number of PDP contexts, the synchronisation between RTP and RTCP flows, etc.

The first configuration (config. 1) considers the transmission of RTP flows without RTCP. The second one (config. 2) consists in attributing a dedicated RAB for each flow: RTP, RTCP, SIP and SRBs. The third configuration (config. 3) suggested by Samsung (combination #7.1.119 in [7]) supports AMR 12.2 and conveys RTP and RTCP separately; RTCP is multiplexed with SIP. The fourth configuration (config. 4) considers a RAB where RTP and RTCP are multiplexed, a second one for SIP and a third for SRBs.

Another initiative [12] considers the multiplexing of RTP, RTCP and SIP in the same RAB, which implies high delays for SIP and RTP flows especially during the session establishment or when RTCP packets are sent.

IV. EVALUATION OF ROHC PERFORMANCE

The software used to evaluate the impact of ROHC as well as other results in this paper is based on an enhancement of OPNET UMTS simulator, where ROHC has been implemented in respect with the IETF standard [9]. In the framework of COSINUS project [13], an optimized ROHC parameterisation was defined to reduce the resource usage with acceptable error rates. It is based on the bidirectional optimistic mode chosen by 3GPP as the reference parameter for VoIP transmission [4]. In this part, we study the VoIPv6 transmission performance over UTRAN (UMTS R'99) respectively with and without ROHC. The multipath fading profile is ITU Vehicular A channel at 3 km/h.

Simulations show that ROHC allows a significant gain in QoS at the physical layer, in both directions: the UL (Uplink) and the DL (Downlink). This gain consists mainly in the reduction of signal over interference (represented here as the E_c/N_0) requirements to achieve a BLER target (Block Error Rate).

On the UL, the gain in E_c/N_0 for a BLER target of 10^{-2} is about 1.5 to 2 dB. Fig. 4 shows the UL VoIPv6 transmission performance for RTP/RTCP/SRB RABs. Tests show that the transmission with ROHC experiences a gain of about 2 dB over the transmission without ROHC. This is due to the increase of the spreading factor (SF) by a factor 2, which means a lower interference at the Node B when performing correlation to decode the mobile's signal and a better QoS.

On the DL, ROHC allows a significant gain of about 1.5 to 2.6 dB. Contrary to the UL, ROHC does not increase the SF. In fact, the DL SF is calculated with regard to the maximum bit rate of all the transport formats (TF) of a RAB including TFs for uncompressed flows. When radio resources are not completely used (e.g. during the ROHC steady state), the 3GPP standard imposes the insertion of DTX (Discontinuous Transmission) bits at the physical layer. These bits lead to gains in DL QoS (E_c/N_0 required vs target BLER) because they switch off transmission. Well interleaved these bits are mixed in the bit stream. Hence, less data

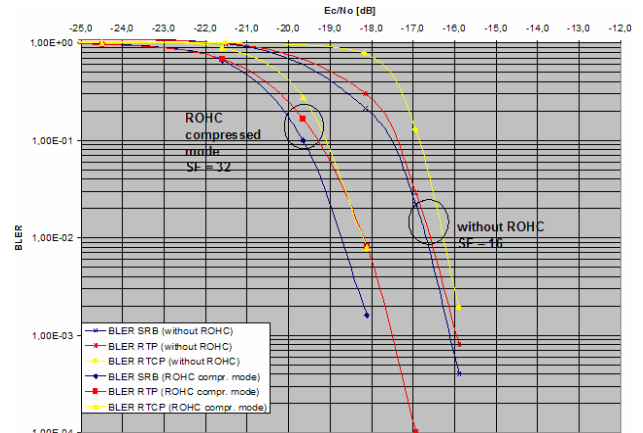


Figure 4: Comparison of UL IPv6 VoIP transmission with and without ROHC - VA 3 km/h

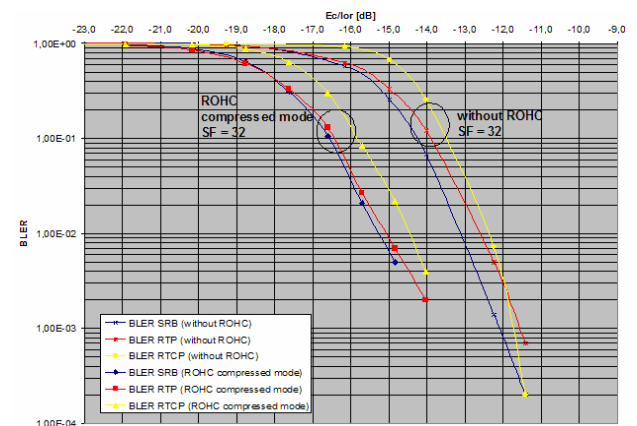


Figure 5: Comparison of DL IPv6 VoIP transmission with and w/o ROHC - VA 3 km/h

bits are found in a fading gap at once. Fig. 5 considers the same scenario as Fig. 4, but in the DL direction. We see that the use of ROHC leads to an interference gain of about 2 dB in the DL for a BLER target of 10^{-2} .

V. THE PROPOSED OPTIMIZED RAB COMBINATION

A. RAB combination description

The new VoIP RAB combination is based on the mapping architecture presented in Fig. 1 in which RTP/RTCP flows are multiplexed in the same RAB. Table 3 describes the different characteristics of the new RAB combination. The Radio Bearers dedicated to RTP/RTCP and SIP flows support a throughput of 27.6kbps and 24kbps, respectively.

The novelty of this combination is the optimal use of the flexible rate matching algorithm (as opposed to the classical fixed position algorithm) [14]. This algorithm calculates the number of bits after rate matching for a set of TFCs (Transport Format Combinations) excluding the combination with the highest TFs. The TFCS (TFC Set) is limited to TFCs that are adapted to the flows characteristics. In fact, RTP/RTCP and SIP flows do not have the same types of radio resource needs, which can be illustrated with six flow cases as shown in Fig. 6. It must be highlighted that for most of these cases, SIP and RTP/RTCP flows

Table 2: Optimized RAB combination

RAB configurations		RAB #1 for RTP and RTCP flows	RAB #2 for SIP flow	SRB #1	SRB #2	SRB #3	SRB #4	
				RRC	RRC	NAS-DT high priority	NAS-DT low priority	
RLC	Logical channel type	DTCH	DTCH	DCCH	DCCH	DCCH	DCCH	
	RLC mode	UM	AM	UM	AM	AM	AM	
	Payload sizes, bit	216, 296, 376, 552	320, 480, 540, 960	136	128	128	128	
	Max data rate, bps	27600	24000	3400	3200	3200	3200	
	UMD PDU header, bit	8	16	8	16	16	16	
MAC	MAC header, bit	0	0	4	4	4	4	
	MAC multiplexing	N/A	N/A	4 logical channel multiplexing				
Physical layer	TrCH type	DCH	DCH	DCH				
	TB sizes, bit	224, 304, 384, 560	336, 496, 656, 976	148				
	TFS	TF0, bits	0x560	0x976	0x148			
		TF1, bits	1x224	1x336	1x148			
		TF2, bits	1x304	1x496				
		TF3, bits	1x384	1x656				
		TF4, bits	1x560	1x976				
TTI, ms	20	40	40					

need resources at different times. Then, the combinations of TFs allocating a maximum throughput for SIP and RTP/RTCP simultaneously are not necessary. Forbidding these TFCs and using the flexible rate matching algorithm allow a considerable reduction of the resources reserved for the RAB combination. Table 3 depicts the set of retained TFCs. Note that no TFC combines TF4 of the RTP/RTCP RAB with TF4 of the SIP RAB which are the highest TFs. Fig. 7 illustrates the management of TFCs for the previous six flow cases. During the SIP session establishment, a throughput of 24kbps is provided to SIP flows, which reduces call establishment delays. During the call, a minimum throughput of 14,8kbps is guaranteed to RTP/RTCP flows: this value corresponds to the necessary throughput to transport the well compressed RTP flow when the used codec is the AMR Narrow Band in mode 12.2. If a higher throughput is necessary for the RTP/RTCP RAB (e.g. during the transmission of a RTCP packet or an uncompressed RTP packet), two throughputs are available to transmit RTP packets without critical delays:

- 27.6kbps if no SIP message is exchanged
- 18.5kbps if SIP messages are transmitted

Table 3: Limited TFCS for the new RAB combination

TFCs	TC #1	TC #2	TC #3	TFCs	TC #1	TC #2	TC #3
	RTP RTCP	SIP	SRB		RTP RTCP	SIP	SRB
TFC #0	TF0	TF0	TF0	TFC #15	TF1	TF2	TF1
TFC #1	TF0	TF0	TF1	TFC #16	TF1	TF3	TF0
TFC #2	TF0	TF1	TF0	TFC #17	TF1	TF3	TF1
TFC #3	TF0	TF1	TF1	TFC #18	TF2	TF0	TF0
TFC #4	TF0	TF2	TF0	TFC #19	TF2	TF0	TF1
TFC #5	TF0	TF2	TF1	TFC #20	TF2	TF1	TF0
TFC #6	TF0	TF3	TF0	TFC #21	TF2	TF1	TF1
TFC #7	TF0	TF3	TF1	TFC #22	TF2	TF2	TF0
TFC #8	TF0	TF4	TF0	TFC #23	TF2	TF2	TF1
TFC #9	TF0	TF4	TF1	TFC #24	TF3	TF0	TF0
TFC #10	TF1	TF0	TF0	TFC #25	TF3	TF0	TF1
TFC #11	TF1	TF0	TF1	TFC #26	TF3	TF1	TF0
TFC #12	TF1	TF1	TF0	TFC #27	TF3	TF1	TF1
TFC #13	TF1	TF1	TF1	TFC #28	TF4	TF0	TF0
TFC #14	TF1	TF2	TF0	TFC #29	TF4	TF0	TF1

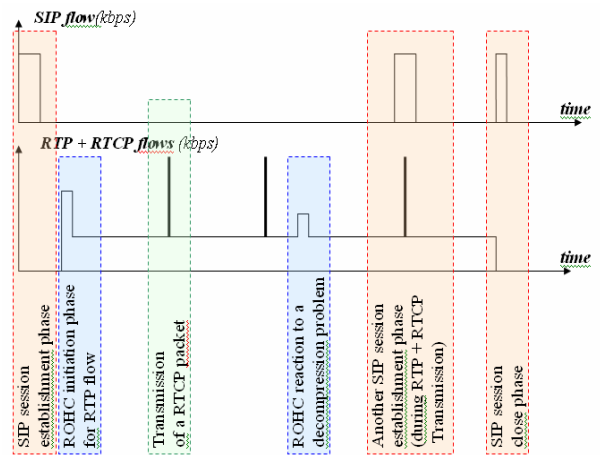


Figure 6: RTP/RTCP and SIP flows needs

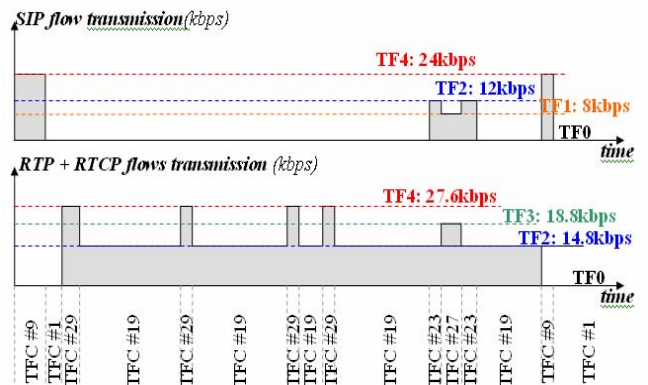


Figure 7: RTP/RTCP and SIP flows mapped on RABs

B. Evaluation of the optimized RAB combination

The evaluation of the optimal RAB combination compared to the current 3GPP configurations is based on the estimation of the resulting gains, mainly in terms of radio resource usage, traffic throughput and SIP delays.

First, the network capacity is increased with a lower interference. In fact, the DL SF of the new RAB (64) is higher than the SF for the majority of

configurations presented in table 1 except for config. 3 needing also a SF of 64.

Second, the throughput for RTP/RTCP is sufficient to support both the transient and steady states of ROHC. Contrary to config. 3 (table 1), no Radio Bearer (RB) modifications have to be performed when moving to the steady state of ROHC during the VoIP call. This is due to the fact, that the new RAB supports two TFs types: the first type is designed to support the transient state of ROHC (i.e. uncompressed headers), the second supports the steady state of ROHC.

Finally, the whole setup duration of a PS communication is decreased since the SIP delays are cut off. In order to estimate these delays our OPNET UMTS solution simulates the realistic support of SIP signalling over the UMTS R99 network, in conformance with 3GPP requirements. The processing and transmission delays introduced in the simulation chain are calibrated thanks to experimentation tests done on real UMTS networks. The management of radio resources and IP packets through the radio access network is simulated according to 3GPP specifications.

Simulation results are depicted in Table 4 giving the delay assessments for the current 3GPP RABs (8/16 kbps for SIP) and the optimal RAB (24kbps for SIP). The whole delay includes the core and the radio access networks delays. In the current RABs, SIP messages exchange can be assessed between 8.8s and 12.4s with a RAB of 8kbps and between 6.3s and 7.5s with a RAB of 16kbps. Moreover, the establishment of the PDP (Packet Data Protocol) Context for VoIP streams lasts about 3s in average. Consequently, the resulting call establishment duration for the current 3GPP RABs is too significant for the end-users (between 9.3s and 15.4s).

With the optimal RAB, the throughput for SIP reaches 24 kbps at the session establishment. The setup delays of a PS call are then reduced. It can be noticed that the gain is between 3.2s and 6s compared to the support over a PS RAB 8kbps and only between 0.8s and 1.2s compared to the support over a PS RAB 16kbps because the duration is also due to the interactions between SIP and UTRAN timers and mechanisms. The main gain is the adaptation of the SIP throughput to the VoIP call phases. At the session establishment, SIP is allocated larger throughputs (24kbps, 12kbps). During the VoIP call, the RAB for SIP switches to a lower throughput (8kbps) in favor of RTP/RTCP flows.

VI. CONCLUSION

This paper has proposed a new PS RAB combination to optimize the support of VoIMS. The novelty of this combination is the restriction of the TFCs conjointly with the flexible rate matching algorithm.

A better trade-off has been obtained between the call establishment delay and the allocated resources in a PS context. The new RAB has also allowed an efficient use of ROHC for which the gains have been evaluated

Table 4: Comparison of the call setup durations with the different RABs for SIP

SIP Message	Delays/RAB 8kbps	Delays/RAB 16kbps	Delays/RAB 24kbps
Primary PDP Context setup delay for SIP delay between SIP-Invite and SIP-Ringing	3	3	3
Total duration	11.81-15.41	9.35-10.55	8.55-9.35

in terms of physical layer QoS and interference.

The main limitation for manufacturers to adopt the optimized RAB combination concerns the necessity to introduce an enhanced TFS selection algorithm at RLC/MAC layer in order to choose at each TTI the most adapted TFC.

Finally, this study has focused on VoIP RAB mapping over UMTS dedicated channels. A future evolution would be the VoIP transmission over shared channels in a HSDPA context.

ACKNOWLEDGMENT

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