Presentation of Technical Report to TSG

Presentation to:	**TSG RAN Meeting #28**
Document for presentation:	**TR25.862 RAB support for IMS, Version 2.0.0**
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Abstract of document:

This document captures an analysis of the RABs that can be used in Release 5 in support of IMS, and different optimisation proposals to improve the support of IMS in Release 6 or later.

The report may also serve as a basis to define test configurations to be incorporated in RAN5 for the test of IMS.

The report emphasises VoIP specifically, since it is where the optimisation is most needed when comparing a non optimised IMS speech call and a R99 CS speech call.

Changes since last presentation to TSG RAN:

Optimisation proposals have been analysed. Conclusions and recommendations have been made based on the analysis.

Outstanding Issues:

None

Contentious Issues:

None

3GPP TR 25.862 V2.0.0 (2005-05)

Technical Report

**3rd Generation Partnership Project;
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RAB support for IMS
(Release 6)**

Keywords

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Contents

Foreword

This Technical Report has been produced by the 3rd Generation Partnership Project (3GPP).

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version x.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

3 or greater indicates TSG approved document under change control.

y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the document.

1 Scope

This document captures an analysis of the RABs that can be used in Release 5 in support of IMS, and different optimisation proposals to improve the support of IMS in Release 6 or later.

The report may also serve as a basis to define test configurations to be incorporated in RAN5 for the test of IMS.

The report emphasises VoIP specifically, since it is where the optimisation is most needed when comparing a non optimised IMS speech call and a R99 CS speech call.

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 TS 25.322: "RLC Protocol Specification".

[2]   3GPP TS 25.323: "PDCP Protocol Specification".

[3]   3GPP TS 25.331: "Radio Resource Control (RRC); protocol specification".

[4]   IETF RFC 2507: "IP Header Compression".

[5]   3GPP TS 25.306: "UE Radio Access Capabilities"

[6]   IETF RFC 3095: "RObust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP, and uncompressed".

[7]   3GPP TS 34.108: "Common Test Environments for User Equipment"

[8]   3GPP TR 25.993: "Typical Examples of RABs and RBs Supported by UTRAN"

[9]   3GPP TR 21.877: "Radio optimisation impacts on PS architecture", v. 0.5.0

[10]   3GPP TS 26.236: "Packet switched conversational multimedia applications; Transport protocols"

[11]   3GPP TS 26.234: "Transparent end-to-end packet switched streaming service (PSS); Protocols and codecs"

[12]   IETF RFC3267: "Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs"

[13]   3GPP TR 26.937: "Transparent end-to-end packet switched streaming service (PSS); RTP usage model."

[14]   3GPP TS 26.235: "Packet switched conversational multimedia applications; Default codecs"

[15]   IETF RFC2793: "RTP Payload for Text Conversation"

[16]   3GPP TS23.228: "IP Multimedia Subsystem (IMS); Stage 2"

[17]   IETF RFC3550: "RTP: A Transport Protocol for Real-Time Applications"

[18] IETF RFC3556: "Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth"

3 Abbreviations and Terms

3.1 Abbreviations

For the purposes of the present document, the following abbreviations apply:

AMR	Adaptive MultiRate
DTX	Discontinuous Transmission
HC	Header Compression
IETF	Internet Engineering Task Force
I/B	Interactive / Background
IMS	IP Multimedia Subsystem
IP	Internet Protocol
PDCP	Packet Data Convergence Protocol
QoS	Quality of Service
RAB	Radio Access Bearer
RB	Radio Bearer
RLC	Radio Link Control
RNC	Radio Network Controller
ROHC	Robust Header Compression
RTP	Real-time Transport Protocol
RTCP	Real-time Transport Control Protocol
RTSP	Real-time Streaming Protocol
SIP	Session Initiation Protocol
SRB	Signalling Radio Bearer
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
UE	User Equipment
UEP	Unequal Error Protection

3.2 Terms

Bearer	Common term used to refer to RAB, RB, and/or SRB, when there is no need to distinguish between these terms.
Radio Access Bearer	Bearer terminating in CN.
Radio Bearer	User plane bearer on RAN level
Signalling Radio Bearer	RAN level bearer for RRC and NAS signalling. User plane signalling bearer (e.g., the bearer for SIP signalling) is not SRB, but RB.

Note: In [7] also the RAN level bearers are called as RABs. In order to maintain consistency with [7], the term RAB is used instead of RB also in this document in similar contexts as in [7].

4 Background and Introduction

4.1 General

IMS is considered to be crucial for the development of multimedia-based 3G networks. In order to make the deployment of IP based multimedia services economically viable in a 3G environment, it is necessary to ensure that the Radio Access Bearers used to support these services are optimised. RABs for IMS support are already defined in 3GPP Rel5 [7]. However, these RABs may need to be optimised, in order to ensure a commercially viable deployment of IMS services.

In general, IMS offers one of or combinations of, Real Time and Non Real Time Services e.g. Gaming, Voice, Video streaming, Audio Streaming, Instant Messaging, Chat, Web browsing etc.

In order to support these services, IMS requires one or combinations of the following RABs towards the PS domain:

- Conversational PS RABs for Voice & Video Services

- Streaming PS RABs for PS Streaming Services

- Interactive/Background PS RABs for Data Services

- Signalling RABs for SIP

While it can be argued that IMS can make use of most of the Interactive/Background PS RABs and Streaming PS RABs which have already been optimised and defined for R99/Rel4/Rel5, it is clear that the Conversational PS RABs and the Signalling RABs for SIP already defined for IMS could be optimised further.

4.2 Application Flows for support of Voice Calls over IMS

There are three distinctively different application flows needed to support voice calls over IMS as shown in the figure below:
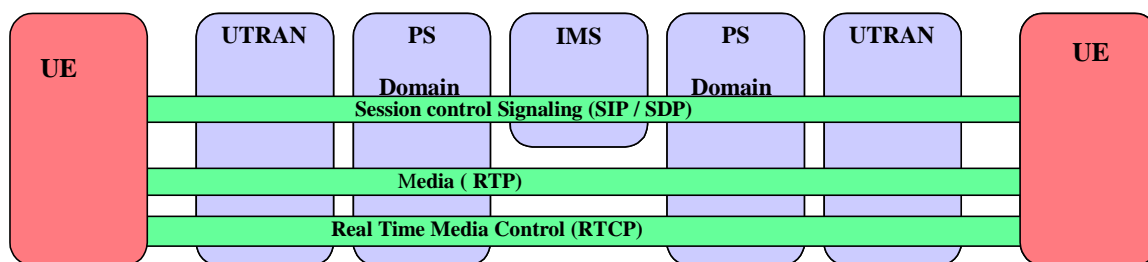
Figure 4.1: Application flows in voice over IMS

- SIP/SDP over UDP/IP for application control between the terminals. SIP signalling is used for multimedia session control.

- Voice payload (media) carried over RTP/UDP/IP between terminals. The coded speech is carried alongside the payload descriptor in the RTP payload. DTMF and SID packets are also carried alongside the speech packets.

- RTCP over UDP/IP for media control between the terminals. RTCP is used to feedback quality information to the source, for the media carried within the RTP flow, for exchanging CNAME in conference calls and for end-to-end proprietary applications.

4.2.1 Characteristics of the Application Flows

- **SIP/SDP**

  SIP/SDP signaling is used for multimedia session control. Some characteristics of SIP/SDP signaling are:

  - It is a request - response type of communication.

  - Low volume traffic with a low demand for average bandwidth.

  Some aspects of the service behavior depend on the QoS given to the SIP/SDP signaling. Hence, SIP/SDP signaling should get high priority and experience low delay. The requirements and the solution for supporting Signaling RAB for SIP is being studied separately in RAN3.

- **Media(RTP)**

The Media/RTP flow carries application data such as the coded speech. It also carries DTMF and SID packets. The overall packets for the media flow carrying speech are AMR/RTP/UDP/IPv6. The RTP payload size for AMR 12.2 Kbps coded speech is 32 Bytes. The characteristics of the flow are:

- Real Time/Conversational

- Consists of a fixed packet size of 92 Bytes including uncompressed RTP/UDP/IP header (60 Bytes) every 20 ms, for AMR 12.2 Kbps coded speech

A dedicated bearer with Conversational QoS class should be established for Media/RTP flow.

The SSD field can be used by the UE to specify the characteristics of the source of submitted SDUs. There are no other means specified for the SGSN to identify that the requested RAB is intended to carry speech packets. One should expect that for most applications using voice the SSD field indicates "speech". For Rel5 the setting of this parameter is optional. It is FFS whether the setting of the SSD field should be mandatory for Rel6.

- **RTCP**

    RTCP is carried over UDP/IP. It carries media control information between the terminals. The characteristics of the flow are:

    - Variable Packet Size and can be longer than the RTP payload.

    - Messages are transferred rather infrequently.

    RTCP, as specified in [17], defines different packet types – Sender Report, Receiver Report, Source Description, BYE and APP. [17] recommends that the fraction of the session bandwidth allocated to RTCP be 5% of the total bandwidth for the combined RTP/RTCP flow.

4.3 RAB Requirements for Voice Call over IMS

When the Radio Access Network receives a request to establish a voice call via IMS it needs to provide suitable RABs to transport the application flows defined in section above. From the NAS point of view, there is always a separate PDP Context and an associated RAB for SIP/SDP signalling. There is also a PDP Context and an associated RAB for RTP flow.

4.3.1 Bit rate requirements

The UE requests a maximum bit rate and a guaranteed bit rate for a voice call with AMR 12.2 Kbps over IMS, both for RTP and RTCP. There are the options of multiplexing RTP and RTCP on one RAB and on separate RABs.

The guaranteed bit rate according to TS 26.236 including uncompressed RTP/UDP/IPv4 headers would be 29 kbps without RTCP and 31 kbps with RTCP. The guaranteed bit rate including uncompressed RTP/UDP/IPv6 headers would be 37 kbps without RTCP and 39 kbps with RTCP. The maximum bitrate would be identical to the guaranteed bitrate. Robust header compression was not taken into account by SA4.

RTCP on a separate RAB has not been considered yet in TS 26.236. However, according to the numbers reported above, the bandwidth for RTCP (on a separate or multiplexed bearer) would amount to a guaranteed and maximum bitrate of 2 kbps for both IPv4 and IPv6. All the examples consider to use an RTCP bandwidth equal to the default 2.5% of the RTP session bandwidth. However, it must be noted that the RTCP bandwidth can be arbitrarily set to a value, which is not a function of the RTP bandwidth.

4.3.2 Handling of RTCP flow

When considering optimisation for IMS, for the handling of the associated RTCP flow, a number of options exist:

- RTCP flow multiplexed on the same PDP context (hence the RAB) and on the same Radio Bearer as that of RTP flow.

- RTCP flow on a separate PDP context (hence the RAB) to that of RTP flow

- RTCP flow multiplexed on the same PDP context (hence the RAB) as that of RTP flow but carried over separate Radio Bearers.

- The RTCP flow could be removed entirely

Note: The PDP context for SIP/SDP will be the primary and that for RTP/RTCP will be secondary.

In Rel-5 [16] identifies only multiplexing of RTCP and RTP on the same PDP context and hence the same Conversational PS RAB.

4.4 Issues to be addressed

It is identified that the following issues need to be addressed by RAN2, in order to be able to optimise the parameters for Conversational PS RABs to support Voice services over IMS:

4.4.1 RTCP Handling

As stated above, RTP and RTCP can be either

- Multiplexed over a Single RB, as currently defined in [16]

- Separated over different RBs

- The RTCP flow could be removed entirely

RTP consists of fixed size small packets, while RTCP consists of variable size packets. The size of the RTCP packet can be many times the size of the RTP packet. The RTP packets for IMS voice services are sensitive to delay variations, while RTCP packets are not. In order to make deployment of IMS voice service commercially viable it is necessary to identify a mechanism to efficiently handle these large and variable size RTCP packets.

4.4.2 RTP(RTCP)/UDP/IPv6 Header Handling

In order to optimise the Radio Bearer, ROHC will be used to perform compression of RTP (RTCP)/UDP/IPv6 header. However, there is a need for occasional transmission of full/partial RTP (RTCP)/UDP/IPv6 headers in order to maintain the header context integrity between peer entities over the radio interface. The header size can vary depending upon the ROHC profile, reducing the packet size by typically 63% over the full packet size, with the optimum header compression. Here also, there is a need to support variable size large packets along with small fixed size packets over the same radio bearer. It is therefore necessary to identify a mechanism for variable bandwidth handling to accommodate RTP (RTCP)/UDP/IPv6 headers.

4.4.3 Signaling RAB for SIP

For optimization of RABs also SIP traffic should be taken into account. SIP traffic is mainly generated at session start up and session release. However SIP traffic is also allowed during the session. The SIP RAB should allow the transmission of SIP packets without significant delay and without SIP packet loss. The allocated bandwidth should be as small as possible.

Please note that the signaling RAB for SIP is also the focus of a separate RAN3 investigation.

It can be expected that in most cases PDP contexts marked with the signalling flag over Iu interface transfer IMS related signalling traffic like SIP. Nevertheless, it is not guaranteed that only IMS related signalling traffic is transferred. Therefore in general UTRAN cannot rely on the "signalling flag".

There is an optional mechanism defined for the GGSN to enforce that only IMS related signalling traffic is transferred. There is no enforcement in the UE possible, i.e. in uplink direction any traffic would have to be transferred although later discarded by the GGSN. Currently the setting of this flag is optional for the UE. Furthermore, a GGSN could reject the setting of this flag.

4.4.4 RAB Parameters

Based on the agreement on the mechanism to address the above issues, it is required to define the optimum RAB/RB parameters. The key parameters being the following:

- Maximum Bit Rate

- ROHC parameters

- PDCP PDU Header Size

- RLC Mode

- RLC PDU Header Size

- Transport Block Size

- Transport Formats

The other RB parameters are expected to be similar to the one for the RBs already defined in [7].

4.4.5 Other Issues

The following issues are also identified:

- UEP

- Rate Control

UEP and Rate Control for Conversational services over IMS are likely to have some major impact on network elements/architecture impact on RAB attributes is FFS.

5   Overview of Optimisation proposals

5.2 RTCP flow multiplexed on the same PDP context and on the same Radio Bearer as RTP flow

This scheme is based on the RTP and RTCP flows to be multiplexed on a single RAB [16], and is characterised by:

- One PDP Context for both RTP and RTCP

- One RAB;

- One PDCP entity for that RB;

- One ROHC compressor/decompressor couple for that PDCP entity;

- Two Contexts for each ROHC instance, each associated to a flow and working with an appropriate ROHC Profile.
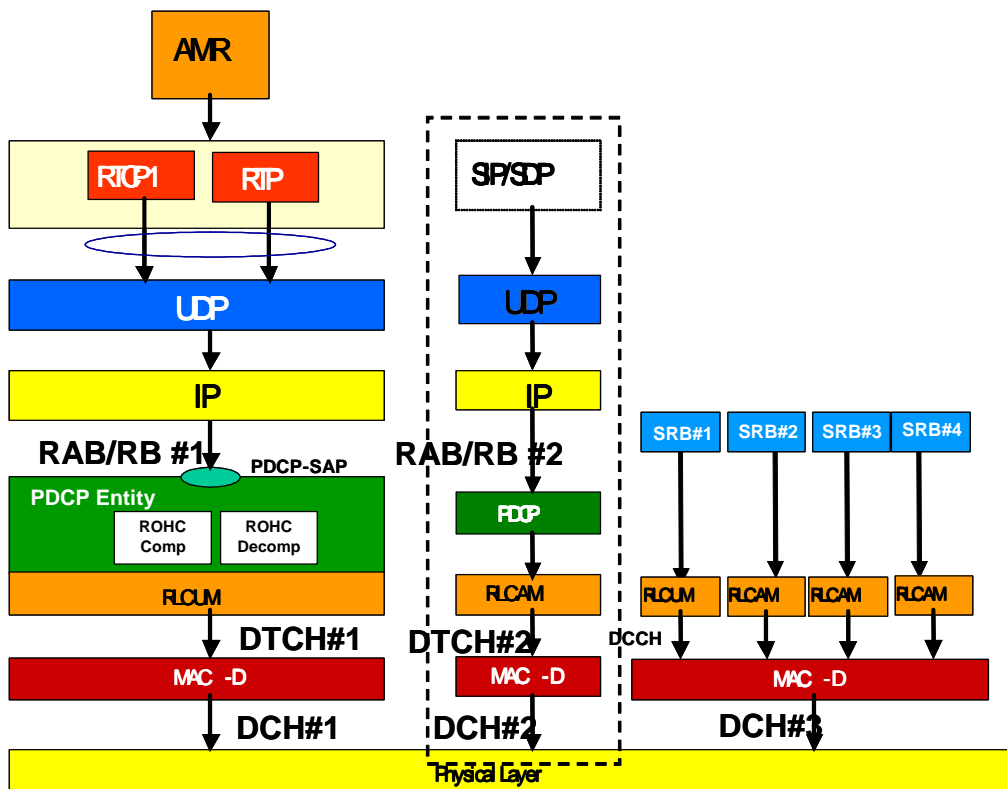
Figure 5.1: Protocol Architecture for Scheme with RTP/RTCP Multiplexed to Single Radio Bearer

For transmission, the RTP and RTCP packets are passed from the higher layers to the PDCP layer via the UDP/IP layer where they are multiplexed onto a single bearer. The PDCP entity distinguishes the two packet flows (RTP and RTCP) based on UDP port numbers, and routes the packets to the appropriate ROHC compressor. Following which, the compressed packets will be, if necessary, encapsulated in PDCP PDUs and passed via RLC for transmission by the lower layers.

When received, the compressed packets (PDCP SDUs) are passed up from RLC, to the PDCP entity, and subsequently to the ROHC de-compressor. After de-compression, the PDCP layer multiplexes the separate flows onto a single bearer for delivery to higher layers. Whereupon reaching the UDP/IP layer the RTP and RTCP flows are ultimately demultiplexed.

Due to the wide variation in the RTCP packet sizes there can be a problem in the management of variable bandwidth required on the air interface, with this multiplexed scheme. There are several mechanisms to address this problem, these are summarised as follows:

- **Frame Stealing**

Normal frame stealing consists in dropping packets carrying RTP when RTCP transmission is requested. This is equivalent to RTCP prioritisation over RTP packets. The application of this method is suggested by the required of minimisation of demanded bandwidth, and will lead to the use of an amount of bandwidth equal to that strictly needed for RTP flow with minimal delay whilst transfer of the RTCP packets takes place. The additional drawback is an increased FER at application level (AMR) due to discarded RTP packets.

The application of Voice Activity Detection (VAD) in a coordinated manner with this solution provides a complementary mechanism for obtaining minimum demanded bandwidth. It consists in RTP being prioritised over RTCP, with RTCP queuing and subsequent transmission during "silence" periods as detected by the VAD.

High level pros and cons of the frame stealing scheme is given below:

- The introduction of additional frame stealing introduces a higher FER requirement
- To achieve equivalent voice quality as the existing 12.2 kbps AMR CS call (BLER target of 1%)
    $\Rightarrow$ Required BLER target is 0.1%

      $\Rightarrow$ RTP bearer bandwidth is 16.4 kbps (UDP check sum applied)
- This scheme would require only one PDP context and RAB per voice call to support RTP and RTCP.

- May require changes to standards to address RTCP handling, for implementation of frame stealing.

- The consequence of IPsec has to be analysed. However, the current assumption is that IPsec is not used.

- This mechanism relies on the VoIP application generating silence periods in order that VAD may be applied as described.

- **Segmentation & Concatenation over the Radio Interface**

This mechanism is based on the segmentation and concatenation feature of RLC unacknowledged mode over the radio interface. However, this mechanism requires delaying of some of the RTP packets for the transmission of the RTCP packets to be completed, for the size of RTCP packet can be many times the size of the RTP packet. The net result is the additional delay and the delay variation (jitter) imposed on RTP(voice) packets, which is not desirable.

- **RB/TrCH/PhyCH Reconfiguration**

This mechanism relies on the radio interface to reconfigure the bearer used for IMS voice to allow higher bandwidth during the transport of RTCP packets. However such reconfiguration could take multiple 100s of milliseconds and such a large amount of delay imposed on voice service is also not desirable.

- **Allocation of Secondary scrambling code**

    NOTE: RAN WG1 and RAN WG4 should evaluate the benefits and drawbacks of this scheme.

By allocation of resources on the primary and on the secondary scrambling code extra bandwidth is available for transmission of RTCP packets in parallel to RTP packets as well as for uncompressed RTP packets. DTX bits are carried on the secondary scrambling code most of the time when only header compressed RTP packets are to be transmitted. This may keep additional interference relatively low. However since the size of RTCP packets can be many times the size of the RTP packet additional delay of RTP packets cannot be entirely avoided in case of multiplexed PDP contexts. This can only be avoided by separate PDP contexts.

This method is allowed in R99/R5 specifications with the two channelization codes having the same SF. The support of multiple DPCH channelization codes to be received simultaneously is a UE capability (e.g. in 25.306 the Maximum number of DPCH/PDSCH codes to be simultaneously received is 1 or 2 for UE class 32-128 kbps and 3 for UE class above 384kbps).

- **Multiple CCTrCHs of dedicated type with different SF**

Another solution to improve the radio resources efficiency using different spreading factors on different channelization codes is to support multiple CCTrCHs of dedicated type. After spreading, the first CCTrCH is scrambled with the Primary Scrambling Code, while the second CCTrCH is scrambled with a Secondary Scrambling Code. The two CCTrCHs may be spread using channelisation codes with different spreading factors.

This solution is currently not supported in RRC R99/R5 for FDD. The L1 specifications (i.e. 25.211 and 25.212) supported the use of multiple dedicated CCTrCHs until they were aligned to RRC by adding a note (CR 131 to 25.211 and 121 to 25.212 in Dec01) indicating that this configuration is not supported.

This type of solution assumes that there is a MAC functionality to map one logical channel (e.g. DTCH carrying the RTP/RTCP packets) towards two transport channels, each being mapped on a different CCTrCH. The RLC functionality is left unchanged. This solution proposes to extend the existing configuration of "DCH with an associated DSCH" to the case of a "DCH associated with another DCH". The first DCH would carry (i.e. is dimensioned) to the frequent traffic size while the associated DCH is used in case of overflow traffic (e.g. infrequent packets).

The physical layer model is showed in the next figure:
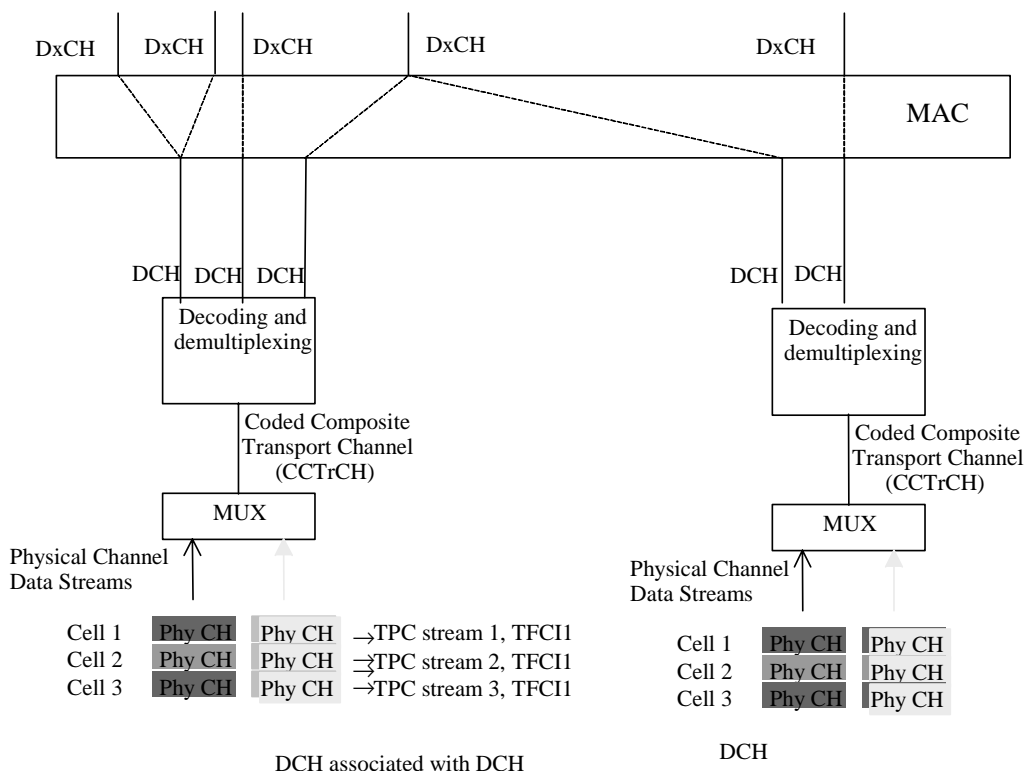
Figure 5.2: Example of physical layer model for associated DCHs mapped on different CCTrCHs

In case of non-acknowledged mode RLC, one change in MAC is needed in order to ensure in sequence delivery to RLC between the two TrCHs, e.g. MAC delivers the transport blocks to the first transport channel and then the remaining blocks are delivered to the associated transport channel in the same order as received from RLC.
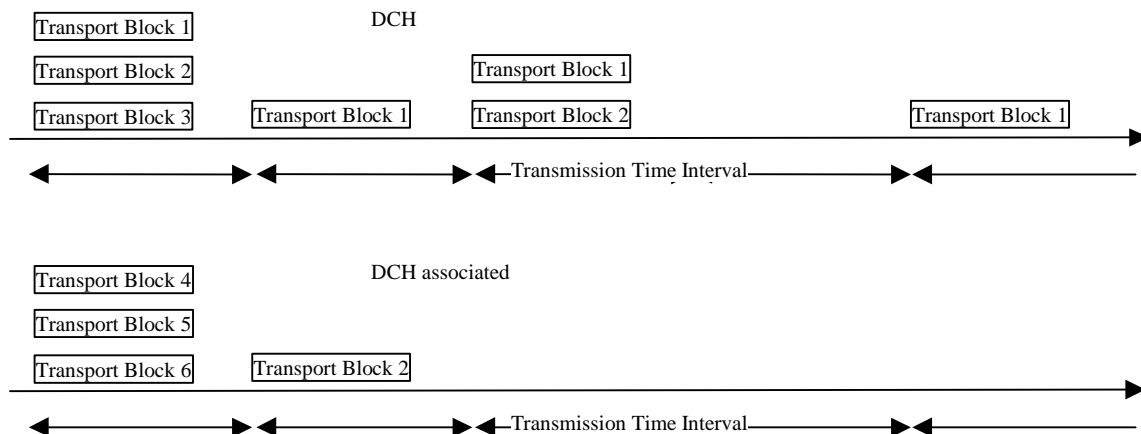
Figure 5.3: Example of data exchange between MAC and L1 for associated DCHs

• **Multiple CCTrCHs of dedicated type with different SF and time offset**

One drawback of previous solutions is that UE may need (depending on its implementation) to demodulate/despread the secondary scrambling code although most of the time there is only DTX.

One enhancement could be that the UE uses the TFCI sent in the first CCTrCH in order to know whether or not it needs to demodulate/despread the second CCTrCH. This solution may be viewed as an extension of DPCH/PDSCH timing.

Similar changes in MAC are needed as for the previous solution, e.g. MAC delivers the transport blocks to the first transport channel and then the remaining blocks are delivered in the next TTI to the associated transport channel in the same order as received from RLC.
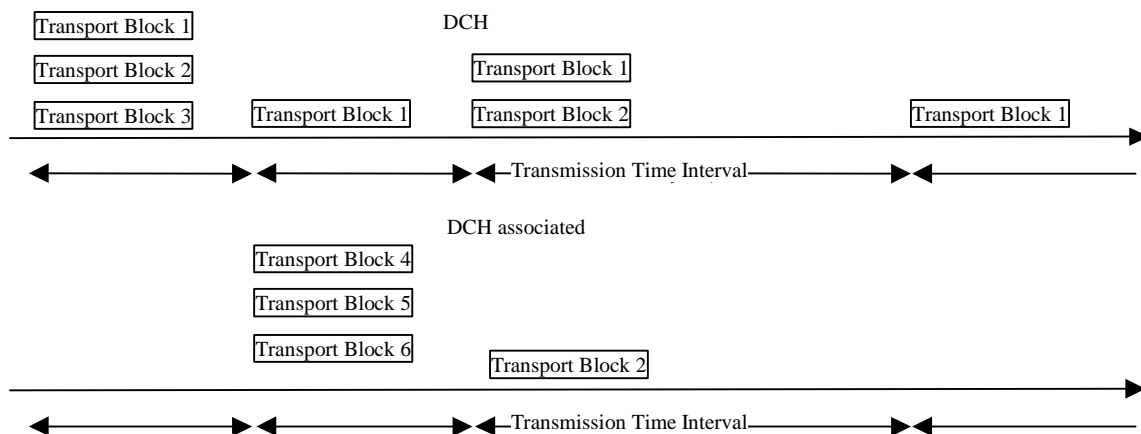
Figure 5.4: Example of data exchange between MAC and L1 for associated DCHs with one TTI time offset

This type of solution may have the drawback of introducing an additional delay of one TTI to deliver the packets to the application when large packets need to be transmitted. However this seems to be acceptable from the application point of view (e.g. VoIP).

5.3 RTCP flow and RTP flow separated on different PDP contexts

This scheme considers the RTP and RTCP flows are delivered to the Access Stratum in two separate flows, this is characterised by,

- Two RABs and two PDP contexts, one for RTP and one for RTCP

- Two RBs, one for each RAB;

- Two PDCP entities, one for each RB;

- One ROHC compressor/decompressor couple for each PDCP entity;

- One Context for each ROHC instance, working with an appropriate ROHC Profile.
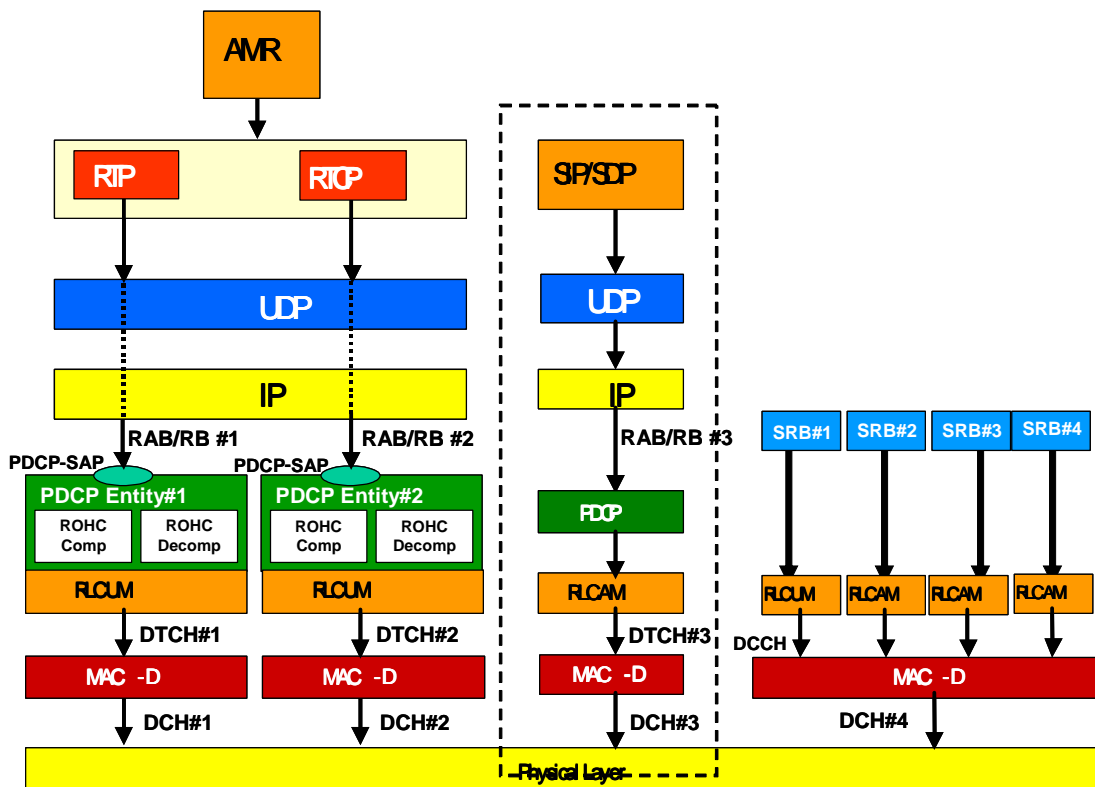
Figure 5.5: Protocol Architecture for Scheme with RTP/RTCP on Separate Bearers

For transmission, of the RTP and RTCP packets each flow is delivered to distinct PDCP entities, and each PDCP entity delivers packets to its ROHC compressor with an indication of the Context used.

Reception is analogous to the previous case, except that the delivery to the higher layers is provided on separate bearers and no de-multiplexing from a single bearer is required in the higher layers.

Because of the fact that RTCP packet transmission is not time critical the RTCP packets can be either transmitted when no RTP packets are to be transmitted, or RTCP is transmitted in parallel to RTP with either low bandwidth requirements, or with higher bandwidth on the secondary scrambling code.

In addition, a binding between the RABs associated to each of the separate RTP and RTCP flows is needed in the SRNC.

This is for the case of SRNS-Relocation. Moreover, RTCP RAB should not be terminated without terminating the RTP RAB and this may be an issue during busy/congested periods.

There is no additional mechanism foreseen in the RAN. However, a solution for separating RTP and RTCP into different PDP context is needed in the NAS layer.

The pros and cons of the scheme is given below:

- It requires a more stringent BLER target, with respect to the RTP/RTCP muxed solution, to obtain the same FER performances. To achieve equivalent voice quality as the existing 12.2 kbps AMR CS call (with BLER target of 1%), the required BLER target is 0.5%
- RTP bearer bandwidth is 16.4 kbps (UDP check sum applied)
- The average RTCP bearer bandwidth is 0.4 kbps
- This would require 2 PDP contexts and RABs per voice call to support RTP& RTCP and thus additional resources in network elements and in the UE.
- The consequence of IPsec has to be analysed. However, the current assumption is that IPsec is not used.

- Since the separation is proposed to be done in the higher layers, changes are required for CN/NAS specifications to split RTP and RTCP on different bearers. No additional changes to RAN standards are foreseen.

- Because of the separation of PDP contexts no delay of RTP packets is caused by RTCP transmission. However, a binding mechanism between the RABs associated to each of the separate RTP and RTCP flows is needed in the SRNC.

- In some cases RTCP is needed by the receiver for synchronisation between multiple RTP streams (for example audio and video), or for measurements of round trip delay. When RTCP is carried in a separate PDP context, it may not follow the same route as RTP and hence the information deduced from RTCP may not be accurate.

5.4 RTCP flow multiplexed on the same PDP context as that of RTP flow but carried over separate Radio Bearers

5.4.1 Using two PDCP entities

This scheme is based on the RTP and RTCP flows to be multiplexed on a single RAB but carried over separate Radio Bearers, and is characterised by:

- One PDP Context for both RTP and RTCP

- One RAB;

- Two Radio Bearers (RB);

- Two PDCP entities, one for each RB;

- One ROHC compressor/decompressor couple for each PDCP entity;

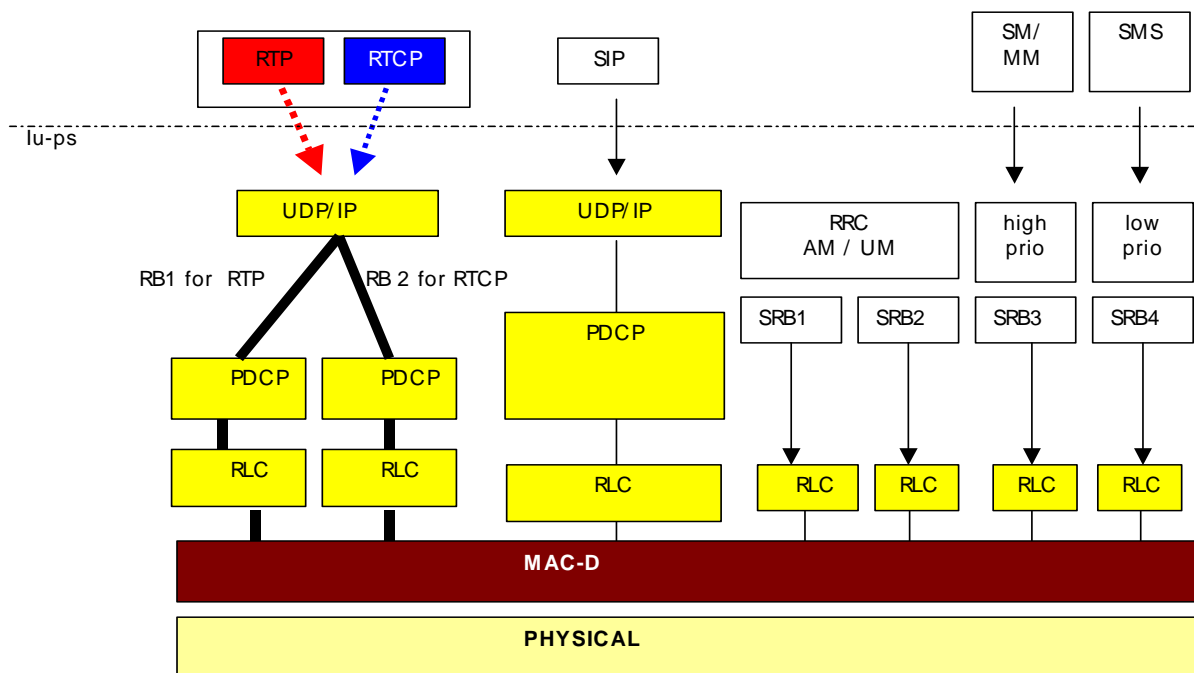- One Context for each ROHC instance, working with an appropriate ROHC Profile

Figure 5.6: Protocol Architecture for Scheme with RTP/RTCP on same PDP context and separate Radio Bearers using two PDCP entities

Globally, this solution is similar to the case "RTCP flow multiplexed on the same PDP context and on the same Radio Bearer as RTP flow" for the Iu interface, and similar to the case "RTCP flow and RTP flow separated on different PDP contexts" for the aspects related to the Radio Bearers.

For transmission, the RTP and RTCP flows are separated from the single RAB, each flow is delivered to distinct PDCP entities, and each PDCP entity delivers packets to its ROHC compressor with an indication of the Context used.

This means that a new functionality needs to be introduced that separates the different flows based on their UDP port number which needs to be given to the RNC in the RAB assignment procedure.

Reception is analogous to the case "RTCP flow and RTP flow separated on different PDP contexts".

Unlike the previous solution, there is no need for a binding between the RABs associated to each of the separate RTP and RTCP flows is needed in the SRNC.

The advantages of the scheme are given below:

- RTP and RTCP flows could experience different RoHC compression schemes, different RLC modes, different MAC-d priorities (possibly making frame stealing easy at MAC-d layer), and possibly different Channel Coding schemes if required. This solution gives a better flexibility for independent RLC and prioritisation at MAC-d layer compared to the "RTCP flow multiplexed on the same PDP context and on the same Radio Bearer as RTP flow" solution.

- No backward compatibility issue with Release 5, in which RTP and RTCP must use the same PDP Context:

    - TS 23.228 states it in clause 4.2.5.1 "All associated IP flows (such as e.g. RTP / RTCP flows) used by the UE to support a single media component are assumed to be carried within the same PDP context.",

    - LS from SA2 to CN3 (S2-022627 [2]) confirmed it very clearly: to the question on what should be the interpretation to use, SA2 answered: *" Interpretation 1) All RTCP flows shall be sent over the same PDP context as the associated RTP flows, Or Interpretation 2) That although it is assumed that in general RTCP packets will be sent over the same PDP context as the associated RTP flow, the mechanism shall also allow for the RTCP packets to be sent over different PDP contexts to the RTP data stream.*
    SA2 confirms that interpretation 1 is the correct interpretation for Rel-5 IMS."

- Because of the separation of RTP and RTCP on different radio bearers no delay of RTP packets is caused by RTCP transmission.

Comparison of the Iu aspects with the "separate PDP contexts solution":

- There is no need of a binding between the RABs associated to each separate RTP and RTCP flows in the SRNC. Instead the CN needs to provide the RNC with the UDP port number to enable the separation of the different flows.

- There is no need for the application in the UE to map RTP and RTCP flows onto the relevant PDP contexts. In the "separate PDP contexts solution", when a dedicated PDP context is used for RTCP flows, the application in the UE must be capable of multiplexing the RTCP flows into this dedicated PDP context.

- Unlike "separate PDP contexts solution", there is no need for the UE, the SGSN and the GGSN to be able to support a higher number of PDP contexts.

- In the "separate PDP contexts solution" the number of PDP contexts is increased drastically. If we consider that a multimedia session comprises potentially multiple media components, then either a pair of PDP contexts is required per component, or all RTCP flows for the different components are multiplexed into a single PDP context. This means that additional resources are introduced in the SGSN and GGSN for each RTP and RTCP PDP context.

- The authorization for RTCP and RTP will be done separately (for each PDP context) although there is a tight relationship between the QoS requirements for RTP and RTCP.

- In some cases RTCP is needed by the receiver for synchronisation between multiple RTP streams (for example audio and video), or for measurements of round trip delay. When RTCP is carried in a separate PDP context, it may not follow the same route as RTP and hence the information deduced from RTCP may not be accurate. This can be avoided if RTCP and RTP are carried in the same PDP context.

The disadvantages of the scheme are given below:

- The consequence of IPsec has to be analysed. However, the current assumption is that IPsec is not used.

- The UTRAN must be aware of the contents of the flows. New functionality needs to be introduced that separates the different flows.

- If the VoIP PDP context is upgraded to video, the bearer handling in the UTRAN and in the UE is more complex. (Assuming that the video would have only one bearer for RTP and RTCP, because only VoIP could require separate bearers for optimisation reasons.)
- The guaranteed bit rate, delay etc. QoS parameters are per PDP context only, and UTRAN must have some mechanism to calculate new values and select bearers based on these new values.
- Additional sub-layer on PDCP or RLC to separate/combine the RTP and RTCP bearers would be needed.
- Additional complexity and specification work for SRNS relocation.

5.4.2   Using single PDCP entity

This scheme is based on the RTP and RTCP flows to be multiplexed on a single RAB but carried over separate Radio Bearers, and is characterised by:

- One PDP Context for both RTP and RTCP;

- One RAB;

- Two Radio Bearers (RB);

- One PDCP entity, shared by two RBs;

- One ROHC compressor/decompressor couple in the PDCP entity;

- One ROHC Context Space for both RTP and RTCP, working with an appropriate ROHC Profile
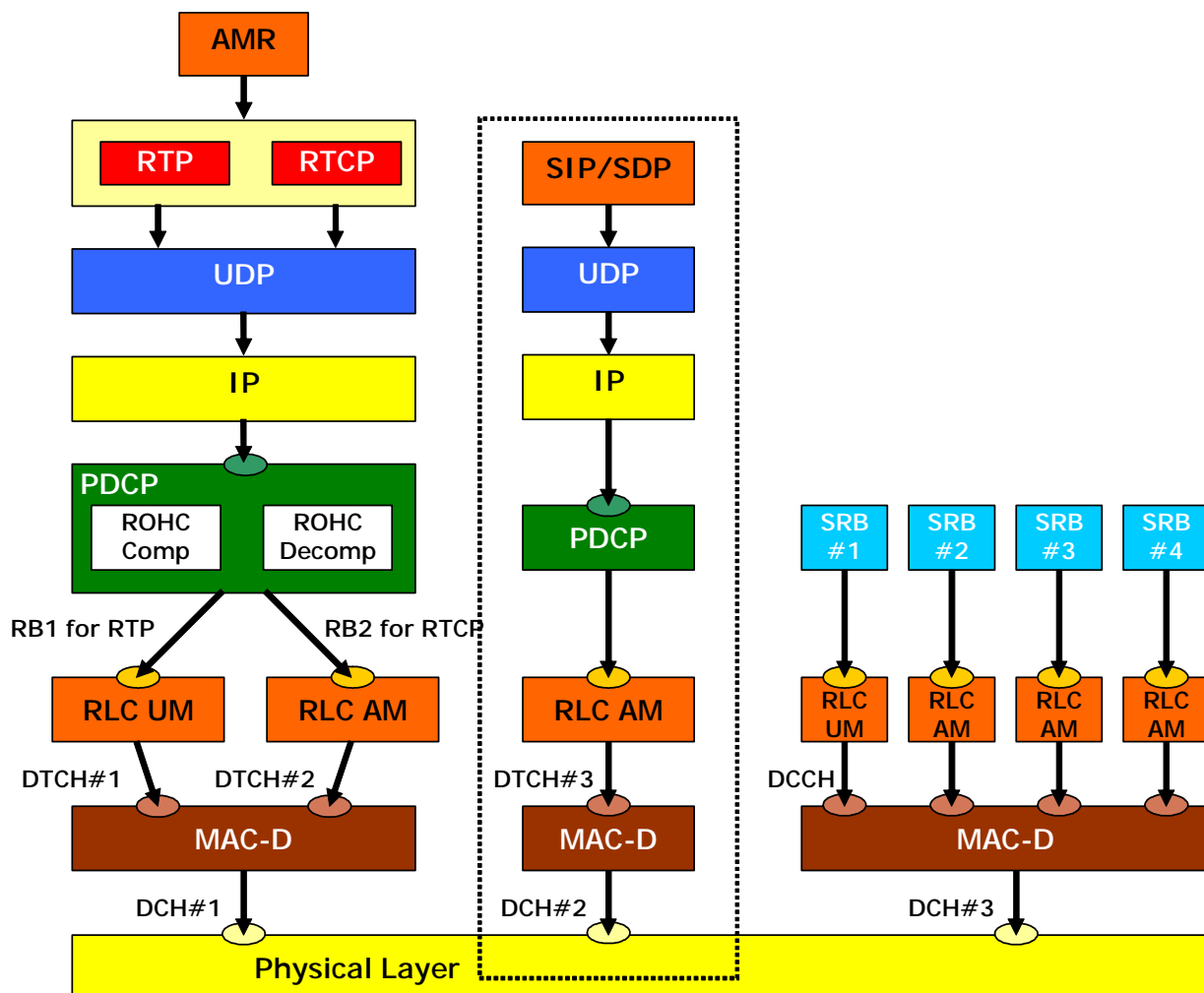
Figure 5.7: Protocol Architecture for Scheme with RTP/RTCP on same PDP context and separate Radio Bearers using a single PDCP entity

This solution is similar to the previous solution "RTCP flow multiplexed on the same PDP context as that of RTP flow but carried over separate Radio Bearers using two PDCP entities", except that it uses a single PDCP entity rather than two different PDCP entities.

For transmission, the RTP and RTCP packets are passed from the higher layers to the single PDCP entity, and the PDCP entity delivers packets to its ROHC compressor. Since the ROHC compressor is by nature able to differentiate between RTP and RTCP packets based on UDP port number, the separation is done by the ROHC compressor.

The ROHC compressor allocates different Context IDs to RTP and RTCP packets. After compression, the ROHC compressor delivers compressed packets to different RLC entities (i.e., RTP packets to UM RLC and RTCP packets to AM RLC) depending on the Context IDs. This means that a new functionality is needed for the delivery of the compressed packets to different RLC entities. This functionality could be introduced in either the ROHC compressor or the PDCP entity.

The receiving side operation is inverse to the transmitting side. A new functionality is also needed for the reception of the compressed packets from different RLC entities. This functionality could be introduced in either the ROHC decompressor or the PDCP entity. After decompression, the RTP and RTCP packets are combined to a single flow, and delivered to the higher layers.

Most features are similar to the case using two PDCP entities. The advantages and disadvantages of two PDCP case over other solutions are generally also applicable to single PDCP case.

The advantages of using a single PDCP entity over two PDCP entities are given below:

- ROHC compressor is able to differentiate between RTP and RTCP packets based on UDP port number. Thus, no new functionality is needed to separate the RTP and RTCP flows. Also, no new sub-layer is needed.

- A single PDCP entity is utilized for both RTP and RTCP flows. It means signalling load for PDCP configuration is smaller than two PDCP case.

- RTP and RTCP flows share the ROHC Context Space. It has benefits in memory usage.

The disadvantages of using a single PDCP entity over two PDCP entities are given below:

- New routing function would be needed to deliver/receive RTP and RTCP packets to/from different RLC entities. This functionality could be introduced in either the ROHC compressor/decompressor or the PDCP entity.

- Since two RBs share a single PDCP entity, new RB configuration needs to be defined. Currently, a PDCP entity is used for only one RB.

5.5 RTCP removal

This scheme considers removing RTCP for point-to-point voice calls, enabling IMS-voice frames to be transported using a 16.4 kbps channel. Certainly, while [17] states that RTCP "should be used in all environments", the use of RTCP is only mandated for IP Multicast. Furthermore, [18] states that turning off RTCP Receiver Reports ("while keeping the RTCP bandwidth for data senders –sender reports– non-zero") "may be appropriate for sessions that do not require feedback on quality of reception or liveness of receivers" (e.g. broadcast transmission). It is therefore recommended that UEs do not send RTCP packets for point-to-point voice calls. The required signalling for turning off RTCP uses the SDP bandwidth modifier (RFC 3556), which is set to zero, and may therefore be understood also by non 3GPP endpoints.

An extension to [18] should be provided by IETF (by liaison and the activation of the proper IETF dependency) to cope with bi-directional information exchange flows (e.g., voice, video, …).

In this scheme it is proposed to remove RTCP only for point-to-point voice calls, an environment in which the synchronisation capabilities are not required. In the event that user adds an associated RTP flow to an on-going voice call and both RTP flows need to be synchronized, or if transport feedback due to lack of end-to-end QoS guarantees is needed, then the media and bearers must be re-negotiated in which case RTCP should be turned on again. This same

mechanism can be used to cater for the situation where the end-user introduces a conference bridge, allowing RTCP to be used to control conference features.

The cons of RTCP removal are given below:

- There may be problems for endpoints or network elements that check link aliveness of other endpoints based on the reception of RTCP. In case of session-on-hold there is no way to verify if the remote party is alive or not.

5.6 Variable size RTP header handling

ROHC acc. to [6] is used for RTP packets in order to reduce the overhead due to large headers. Only a small compressed header of 1..3 bytes is required for regular packets. However, at the beginning of a connection and in case of severe errors on the air interface uncompressed headers have to be transmitted.

This means a large variation of the RTP data rate with an approx. packet size of 40…100 bytes.

Following solution is envisaged for an efficient transport of these data rates:

- **Allocation of Secondary scrambling code**

    NOTE: RAN WG1 and RAN WG4 should evaluate the benefits and drawbacks of this scheme.

For this solution resources are allocated on the primary and on the secondary scrambling code.

Two physical channels are allocated. The TFCS is chosen in a way, that compressed headers fit onto the primary SC with DTX bits carried on the secondary SC. So regular transmissions are done on the primary SC only. In the rare case of an uncompressed header, a large transport block is transmitted by means of both physical channels on the primary and secondary SC.

This means, that for the typical case of a compressed header the second DPDCH remains empty and will not be transmitted. In the unlikely case of an uncompressed header both DPDCHs will be filled with data.

This method has the advantage of immediately available extra bandwidth. The increase in interference may be relatively low, since uncompressed headers only occur during call set up and in very rare error situations.

This method is allowed in R99/R5 specifications with the two channelization codes having the same SF. The support of multiple DPCH channelization codes to be received simultaneously is a UE capability (e.g. in 25.306 the Maximum number of DPCH/PDSCH codes to be simultaneously received is 1 or 2 for UE class 32-128 kbps and 3 for UE class above 384kbps).

In order to evaluate the influence of SSC transmissions in terms of interference link level simulations have been performed. Details of these simulations are given in Annex A.

Typically, the physical channel under the SSC will experience a higher level of intracell interference from other transmissions under the PSC. To remedy this problem, an extra power offset, ρ is introduced between DPCH2 and DPCH1, that is set by the Node B. The setting of this power offset is left to the implementation.

In the simulations the geometry varied between –3dB and 15dB.

The simulation results show that the overall power increase due to using the SSC, calculated over both physical channels, varies between 0.6dB and 3dB. However, this is only an instantaneous increase, that occurs for a small fraction of time in practice. The SSC will only be needed as a short-term back-up for transmission capacity.

Since the usage of the SSC is restricted to very few transport format combinations that furthermore are not very frequent/likely during a call, the additional overall interference caused by SSC transmissions can be kept to a minimum.

- **Reconfiguration of RAB data rate**

It is here assumed that the radio bearer is operated in RLC UM mode. RLC TM mode may also be considered if the inbuilt segmentation capabilities of ROHC are used instead of the RLC segmentation. The use of RLC TM however requires futher work and the difference in overhead of the two variants is expected to be small.

The needed functionality is already supported in Rel-5 specifications and reuses existing radio bearers as much as possible.

Downlink RB realisation

At RAB setup, the RAB data rate is configured to accommodate the transmission of RTP packets with uncompressed ROHC headers. This RAB rate is used in the initial phase until ROHC can compress the headers. At that point the RAB data rate is reconfigured to a rate suitable for transmission of RTP packets with compressed ROHC headers. The reconfiguration may take a few hundred ms and during the reconfiguration data is transmitted with the previous RAB data rate, i.e. there is no interruption in the speech.

Since ROHC is very robust against packet losses, the transmission of full headers is typically not needed in normal operation. In the unlikely event where ROHC must send uncompressed headers, for example as a result of several (15-30) consecutive packet losses over the air, the call is anyway likely to be dropped due to those losses. In the case uncompressed headers need to be transmitted, the RAB data rate is again reconfigured and adapted to the transmission of uncompressed headers. During the reconfiguration, data can only be transmitted with the lower data rate which leads to that some RTP data needs to be either buffered or discarded in the transmitter. Potentially a combination can be used where the RLC discard functionality is used to buffer data until a max delay is reached. If the delay exceeds the max delay it is discarded. Instead of performing a reconfiguration to increase the data rate one of the other methods described in this document could be applied to transmit the full header.

During a call there may occasionally be a need to transmit dynamic header updates. This can be handled either by allocating a radio bearer rate that can carry these headers or by letting ROHC transmit only the headers and no payload in the cases the dynamic header updates are needed.

The needed functionality is already supported in Rel-5 specifications (in case the reconfiguration alternative is used to achieve higher data rates) and reuses existing radio bearers as much as possible. Note that the reduction in the rate that occurs after the ROHC contexts have been initialised, is only needed in case there is a code limitation. Otherwise the higher data rate can be used during a longer time.

Uplink RB realisation

Since the UE selects the spreading factor according to the transmitted transport format, separate transport formats can be configured adapted to a number of RTP payload sizes, including the minimum and maximum size of the packets. The UE autonomously selects the necessary TF to transmit the payload size and reduces the SF if an uncompressed header needs to be transmitted.

- **Handling of ROHC packet using RLC segmentation and concatenation**

A ROHC compressor transmits larger IR headers at the beginning of a flow, to initialize the decompressor context (typically 40 octets for IPv4 and 60 octets for IPv6). IR packets may also be sent in case of repeated decompression failure due to large consecutive losses. IR-DYN packets (typically 18 octets and up) may also be sent when some of the dynamic part of the context requires updating. The transmission of IR and IR-DYN packet are thus seldom.

It is here assumed that the radio bearer is operated in RLC UM mode. This mechanism is based on the segmentation and concatenation feature of RLC unacknowledged mode over the radio interface. The functionality needed is already supported in Rel-5 specifications and reuses existing radio bearers as much as possible. It is also assumed that the inbuilt segmentation capabilities of ROHC are not used.

This mechanism requires delaying of some of the smaller ROHC packets to allow for the completion of the transmission of larger ROHC packets. Note that in the case of AMR 12.2 kbps, the size of the largest ROHC packet (IR packet with AMR payload, IPv6 is ~92 octets) will be less than three times the size of the smallest ROHC packet (~35 octets).

The net result is some additional delay and the delay variation (jitter) imposed on RTP (voice) packets. However, the occurrence of larger IR packets is very seldom and their ratio compared to the smaller packet sizes for which the RAB would be configured for is relatively low. This implies that the additional delay may be reclaimed under DTX periods (when used) and/or by performing selective packet dropping at the RLC over a sufficiently long period of time. Using this mechanism, restricting the delay increase and the BLER increase to a minimum, it is thus possible to avoid any noticeable service degradation for the user while maintaining a low peak data rate and avoiding allocating additional resources for the larger packets in the flow.

Downlink RB realization

At RAB setup, the RAB data rate is configured to accommodate the transmission of RTP packets with small ROHC compressed headers.

Since ROHC is very robust against packet losses, the transmission of full headers is typically not needed in normal operation. In the unlikely event where ROHC must send uncompressed headers, for example as a result of several (15-30) consecutive packet losses over the air, the call is anyway likely to be dropped due to those losses.

In the case uncompressed headers need to be transmitted, SDUs (ROHC packets) larger than the maximum RLC PDU size defined for the bearer are transmitted over multiple TTIs using RLC segmentation; available RLC PDU space is filled using concatenation to maximize link utilization.

This method may require some buffering at the RLC layer when the instantaneous amount of SDU data overshoots the peak rate. Buffered data can be transmitted to exhaust any outstanding data with time by:

- Transmission during DTX periods (after a talk burst), if DTX is used (no data loss);

- Selective dropping of ROHC packets, if DTX is not used or has no silent period (some loss of data);

- A combination of the above.

Potentially a combination of the first two items above can be used based on DTX periods and dropping of less sensitive ROHC packet(s), where the RLC discard functionality is used to buffer data until a maximum delay is reached; if the delay exceeds a maximum delay, a non-IR/IR-DYN can be discarded.

Note that because ROHC has built-in robustness against packet losses, dropping packet types other than IR or IR-DYN will have no impact on ROHC; it could however produce a sporadic and possible audible service quality degradation to the user.

Uplink RB realisation

Using segmentation and concatenation in the downlink is not deemed to raise any issue for the uplink. In the uplink, the UE may use the same scheme but its use is less compelling than for the downlink since the UE may select the appropriate spreading factor as needed.

Since the UE selects the spreading factor according to the transmitted transport format, separate transport formats can be configured adapted to a number of ROHC packet sizes, including the minimum and maximum packet size. The UE autonomously selects the necessary TF to transmit the payload size and reduces the SF if an uncompressed header needs to be transmitted.

5.7 Signalling handling

An optimised solution for voice over IMS shall provide bandwidth efficient means for the transmission of IMS signalling information on Interactive/Background RABs in parallel to the conversational RABs.

6 Conclusions

At RAN2#43 it was decided not to proceed with the options 5.4.1 and 5.4.2 listed in Chapter 5.4 "RTCP flow multiplexed on the same PDP context as that of RTP flow but carried over separate Radio Bearers".

At RAN2#47 it was decided not to proceed with the Secondary scrambling code solution.

7 Recommendation

The optimised RABs for VoIMS should use the existing UM RLC segmentation and concatenation mechanism. In addition, RB reconfiguration can be used to accommodate e.g. the transmission of RTCP packets or RTP packets with uncompressed ROHC headers.

UM RLC PDU structure should be modified in Rel-6 to reduce the RLC overhead in VoIMS.

Annex A:

Link Level Simulations for Secondary Scrambling Code Solution:

A1. Background Information

A1.1 Requirements for SSC Solution

The RAB that needs to be defined shall allow for following transmissions:

- RTP packets with compressed header on the primary scrambling code
- RTP packets with uncompressed header on primary and secondary scrambling code
- RTCP packets on a separate RAB in silence periods, on primary scrambling code
- SIP packets at call set up on the primary scrambling code
- Mid-call and call release SIP signalling on the secondary (and possibly primary) scrambling code

The SF in DL should be as high as possible while puncturing is kept sufficiently low.

A1.2 Assumptions

A1.2.1 RTCP handling

Most likely, RTCP is not mandatory for point-to-point connections. For these cases RTCP might be turned off. At least for PtP conversational calls SA4 recommended to switch off RTCP [A8]. However, since there might still be applications that do require RTCP we propose a separate 16 kbps RAB for RTCP. With a proper transport format combination set in combination with standard priority handling in MAC, RTCP packets will automatically be transmitted in silence periods. We propose to transmit RTCP packets on a separate RAB, see [A2] and [A3]. However this is not a precondition for the SSC solution.

A1.2.2 SIP handling

SIP packets can be quite large (approx. 7500 byte for session set-up, [A4]) and therefore we propose a RAB of 16 kbps to reduce transmission delay. SIP packets are mainly transmitted at session set-up and session release. However, SIP messages might also occur within a session. By using the secondary scrambling code, SIP signalling can be transmitted with high data rate and low delay in parallel to compressed RTP packets without affecting speech transmission.

A1.2.3 RAB optimisation

The Siemens discussion paper [A5] presented at Ran2#41 shows proper RAB combinations, so that most of the time only the primary scrambling code will be used for transmission while on the secondary scrambling code nothing will be transmitted by using DTX. The usage of the SSC is limited to the rare cases of uncompressed headers and mid-call SIP signalling.

With the following assumptions

- IPv6 protocol
- ROHC reliable mode for RTP
- No ROHC for RTCP
- UDP checksum disabled
- PDCP No-header-PDU
- RTCP transmission on a separate 16 kbps RAB
- SIP transmission on a separate 16 kbps RAB
- SRB: 3.4 kbps
- AMR codec mode 12.2kbps
- Unacknowledged RLC mode

the uncompressed RTP/UDP/IPv6 packet has a size of 99 bytes and the compressed RTP/UDP/IPv6 packet 40 bytes.

Furthermore we assume an uncompressed RTCP packet size of 140 bytes (136 byte acc. to [A6] + 4 byte RLC overhead).

The RABs that fulfil the requirements mentioned above are defined in [A5].

- Stand-alone UL:3,4 kbps DL:3,4 kbps SRBs for DCCH, See [A7].

- Conversational UL:39.2 kbps DL:39.2 kbps PS RAB for RTP + I/B UL: 16kbps DL: 16kbps PS RAB for RTCP + I/B UL:16 kbps DL:16 kbps PS RAB for SIP + UL:3,4 kbps DL:3,4 kbps SRBs for DCCH.

TrCH parameters for RTP:

TF0, TF1 with 320 bits,TF2 with 792 bits, TC and TTI = 20ms

TrCH parameters for RTCP and SIP:

TF0, TF1 with 336 bits, TF2 with 2*336 bits, TC and TTI = 40ms.

In UL a minimum SF of 32 and a puncturing limit of 0.88 is needed.

In the more interesting DL case both for the DPCH on PSC and the DPCH on SSC a SF of 128 is sufficient.

A1.2.4 Extent of SSC transmissions

The following table shows to what extent the SSC is used for VoIMS transmission:

Required transmission	Transmitted on	Expected frequency of occurrence
Compressed RTP + SRB	PSC only	Typical transmission
Compressed RTP + SIP	PSC + 6%…36% of SSC (depending on SIP size)	Rarely
Compressed RTP + SRB + SIP	PSC + 25%…56% of SSC (depending on SIP size)	Rarely
Uncompressed RTP	PSC + 80% of SSC	At session start-up, unlikely during session
Uncompressed RTP + SRB (worst case)	PSC + 100% of SSC	At session start-up, unlikely during session
RTCP + SRB	PSC only	Every 5 seconds in average
RTCP + SIP	PSC + 0%…22% of SSC (depending on RTCP and SIP size)	Very rarely
RTCP + SRB + SIP	PSC + 0%…41% of SSC (depending on RTCP, SRB and SIP size)	Very rarely
SIP + SRB	PSC only	Rarely

It is assumed that the SSC will be used for not more than 2% of the session time. The activity level for the SSC can be estimated from the probabilities of the TFCs in the table above. For this short time however the interference will be increased.

A2. Simulation Results Summary

For details to the simulations, theoretical aspects and explanations see [A1].

The following table shows the TFC2 power requirements relative to TFC4, for 1% BLER target. The 'total' results have been adjusted for absence of DPCCH of DPCH2 (32/40) multiplier. The numbers are referenced to the Ior from the reference scenario TFC1.

TFC4: Transmission of SRB packets + uncompressed RTP packets on two codes on PSC (see [A1])

TFC2: Transmission of SRB packets + uncompressed RTP packets on both PSC and SSC (see [A1])

Ec_DPCH1_TFC4/Ior … Ec/Ior for DPCH1 (on PSC) for TFC4

Ec_DPCH1_TFC2/Ior - Ec_DPCH1_TFC4/Ior … Power increase on DPCH1 in case of transmission on PSC +SSC compared with the required DPCH1 power if transmission is done only on PSCs.

Ec_DPCH2_TFC2/Ior - Ec_DPCH1_TFC2/Ior …Power penalty on DPCH2 compared with DPCH1 for transmission on PSC+SSC.

Total Ec_TFC4/Ior …Transmitted power for DPCH1+DPCH2 on PSC divided by total NodeB transmit power

Total Ec_TFC2/Ior … Transmitted power for DPCH1+DPCH2 on PSC and SSC divided by total NodeB transmit power

Total Ec_TFC4/Ior - Total Ec_TFC2/Ior … Power penalty on DPCH1+DPCH2 in case of transmission on PSC+SSC compared to transmission on PSCs only.

	Geometry, dB			
	-3	3	9	15
Ec_DPCH1_TFC4/Ior (=Ec_DPCH2_TFC4/Ior) dB	-11.7	-15.6	-17.4	-18.0
Ec_DPCH1_TFC2/Ior - Ec_DPCH1_TFC4/Ior dB	0.3	0.5	0.6	0.7
Ec_DPCH2_TFC2/Ior - Ec_DPCH1_TFC2/Ior dB	0.7	2.0	3.7	4.0
total Ec_TFC4/Ior %	12.17	4.96	3.28	2.85
total Ec_TFC2/Ior %	14.05	7.01	6.01	5.60
total Ec_TFC4/Ior - total Ec_TFC2/Ior %	1.88	2.05	2.73	2.75

total Ec_TFC4/Ior - total Ec_TFC2/Ior dB	0.63	1.50	2.63	2.93

The simulation results provided in the table show the difference in Ec/Ior between the worst case, which is the case of transmission on PSC and complete SSC, and the reference case, in which 2 PSCs are used.

The simulations results are based on the assumption that 15 UEs, transmitting on PSCs, as well as common channels cause intracell interference. Thermal noise and intercell interference from other cells have been taken into account as well. Due to multipath propagation, which reduces orthogonality among PSCs, the interference caused by the SSC is only a part of the overall interference in the cell.

It can be seen that the additional power on the PSC in order to obtain the same SIR as without SSC is between 0.3 and 0.7 dB.

The additional power on the SSC required to obtain the same SIR on the SSC like for the reference case without SSC, is between 0.7 and 4 dB.

The last two rows indicate the total power increase, calculated over both physical channels, required by scenario TFC2, compared to scenario TFC4. It can be seen that this is between ~0.6 and 3 dB (last row).

It should be noted that, in practice, the SSC would be used during a small fraction of the time, therefore, on average, the increase in power consumption will likewise be small.

Furthermore the RTCP flow is not required at all for PtP conversational calls.

[A1] Siemens, R2-041064, IMS – Secondary Scrambling Code Simulations
[A2] Siemens, R2-032142
[A3] Siemens, R2-040046
[A4] 3, R2-032169, SIP RAB with SRBs Multiplexing for IMS Voice Service Optimisation
[A5] Siemens, R2-040531
[A6] Samsung, R2-040077
[A7] TS 34.108 v4.9.0
[A8] S4-040203

Annex B:

Change history

Document history		
Date	**Version**	**Comment**
2003-11	V 0.0.0	First proposal
2004-01	V 0.1.0	Updates based on R2-032319 and a Siemens text proposal.
2004-02	V 0.2.0	Updates from RAN2#40.
2004-03	V 1.0.0	Updates from RAN2#41. Raised to V1.0.0 to be presented for information at TSG RAN#23.

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<tr><td colspan="3" align="center">Document history</td></tr>
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Date	Version	Comment
2004-08	V 1.1.0	Updates from RAN2#42.
2004-08	V 1.2.0	Updates from RAN2#43.
2004-11	V 1.3.0	Removed text in 5.2 on RTP/RTCP differentiation, based on LS S4-040632/R2-042429.
2005-04	V 1.4.0	Added the text proposal in R2-050940 on using RLC segmentation and concatenation.
2005-05	V 1.5.0	Updates from RAN2#47: Added the conclusion not to proceed with the Secondary scrambling code solution. Added the recommendation that the optimised RABs for VoIMS should use the existing UM RLC segmentation and concatenation mechanism. Added the recommendation that the UM RLC PDU structure should be modified in Rel-6 to reduce the RLC overhead in VoIMS.
2005-05	V 2.0.0	Added the recommendation that the RB reconfiguration can also be used with the optimised RABs for VoIMS. Submitted for approval at TSG RAN#28.

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