TSG-RAN Meeting #21 ***RP-030497***
Frankfurt, Germany, 16-19 September 2003

Title: **CR (Version 6.2.0 affecting the Rel-5) to TR 25.993.**

Source: **TSG-RAN WG2**

Agenda item: **7.3.5**

Spec	CR	Rev	Phase	Subject	Cat	Version-Current	Version-New	Doc-2nd-Level	Workitem
25.993	012	-	Rel-5	IMS RAB scenarios	F	6.2.0	6.3.0	R2-031954	TEI5

3GPP TSG-RAN WG2 Meeting #37
Budapest, Hungary, 25-29 August 2003

Tdoc ⌘R2-031953

CR-Form-v7

CHANGE REQUEST

⌘	**25.993**	**CR**	**012**	⌘ **rev**	**-**	⌘ Current version:	**6.2.0** ⌘

*For **HELP** on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

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

Title: ⌘	IMS RAB scenarios		
Source: ⌘	RAN WG2		
Work item code: ⌘	TEI5	***Date:*** ⌘	14/08/2003
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 TSG RAN#20 it was decided that TR25.893 will be discontinued.
Summary of change: ⌘	The relevant parts of TR25.893v0.1.0 are moved to Annexes in TR25.993.
Consequences if not approved: ⌘	IMS RAB scenarios are not decribed in 25.993.

Clauses affected: ⌘	2, 3, Annex A, Annex B			
	Y	**N**		
Other specs affected: ⌘		X	Other core specifications ⌘	
		X	Test specifications	
		X	O&M Specifications	
Other comments: ⌘	Due to ongoing discussion on RTP/RTCP and VoIP in SA2, the chapters A.1.1 and A.1.2, are left open. Consequently, the SIP and RTCP related paragraphs in “Bearer characteristics” and contents of “RAB scenarios” in Annex B, derived partly from A.1.1. and A.1.2, are excluded from this CR. These are to be included later as separate CRs when the items have been fixed for Rel-6. Also editor’s notes in 25.893 have been excluded.			

How to create CRs using this form:

Comprehensive information and tips about how to create CRs can be found at http://www.3gpp.org/specs/CR.htm. Below is a brief summary:

1) Fill out the above form. The symbols above marked ⌘ contain pop-up help information about the field that they are closest to.

2) Obtain the latest version for the release of the specification to which the change is proposed. Use the MS Word "revision marks" feature (also known as "track changes") when making the changes. All 3GPP specifications can be downloaded from the 3GPP server under ftp://ftp.3gpp.org/specs/ For the latest version, look for the directory name with the latest date e.g. 2001-03 contains the specifications resulting from the March 2001 TSG meetings.

3) With "track changes" disabled, paste the entire CR form (use CTRL-A to select it) into the specification just in front of the clause containing the first piece of changed text. Delete those parts of the specification which are not relevant to the change request

1 Scope

The present document provides a list of examples of RABs and RAB combinations which are supported by UTRA with examples of radio interface mapping for these RABs onto Radio Bearers and Signalling Radio Bearers.

This list of examples describes typical parameters, and should only be understood as possible configurations i.e. any other configuration supported by the Core Specifications and consistent with a given UE capability shall also be supported by this UE.

The present document addresses the FDD mode as well as the TDD mode.

This report is a release independent report. This means that the latest release applicable to 3GPP is the reference that this TR is defined upon, and contains information on all previous releases. Actual release where a given example applies is indicated in the relevant section.

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 34.108: "Common Test Environments for User Equipment (UE) Conformance Testing"

[2] 3GPP TS 23.107: "Quality of Service (QoS) concept and architecture".

[3] 3GPP TS 25.212: "Multiplexing and channel coding (FDD)".

[4] 3GPP TS 25.322: "RLC Protocol Specification".

[5] 3GPP TS 25.323: "PDCP Protocol Specification".

[6] 3GPP TS 25.331: "Radio Resource Control (RRC); protocol specification".

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

[8] 3GPP TS 25.306: "UE Radio Access Capabilities"

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

[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 RFC1889: "RTP: A Transport Protocol for Real-Time Applications."

[13] 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"

[14] 3GPP TR 26.937: "Transparent end-to-end packet switched streaming service (PSS); RTP usage model."

[15] 3GPP TS 26.235: "Packet switched conversational multimedia applications; Default codecs"

[16} IETF RFC2793: "RTP Payload for Text Conversation"

[17] 3GPP TR 21.877: "Radio optimisation impacts on PS architecture", v. 0.5.0

3 Abbreviations and Terms

3.1 Abbreviations

For the purposes of the present document, the abbreviations contained in TR 21.905 apply, as well as the following:

DL	Downlink.
HC	Header Compression
IETF	Internet Engineering Task Force
I/B	Interactive / Background
IP	Internet Protocol
kbps	kilo-bits per second.
RAB	Radio Access Bearer.
RB	Radio Bearer
RNC	Radio Network Controller
ROHC	Robust Header Compression
RT	Real-time.
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
UL	Uplink.

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 [1] also the RAN level bearers are called as RABs. In order to maintain consistency with [1], the term RAB is used instead of RB also in this document in similar contexts as in [1].

Annex A: Service scenarios

This chapter presents a selection of service scenarios, which are used as a basis for the RAB scenarios. Only the basic scenarios having impact on the lower layers are considered. Because the real time applications have the tightest connection with the lower layers, the real time scenarios are studied more in detail in this document. Other scenarios can be derived as combinations of these basic scenarios.

Even though these scenarios are for IMS, they are applicable also for non-IMS PS scenarios. The differences between IMS and non-IMS are small in RAN level: Usually, the difference is that in non-IMS cases the IMS signalling stream is left out or replaced by non-IMS signalling stream. Other differences are indicated later in the text, whenever necessary.

Table 1: Service scenarios

		IMS Signallingg	Speech (RTP)	Speech (RTCP)	Audio (RTP)	Audio (RTCP)	Video (RTP)	Video (RTCP)	Text (RTP)	Text (RTCP)	Data	Notes
1	Speech	X	X	X	-	-	-	-	-	-	O	
2	Audio	X	-	-	X	X	-	-	-	-	O	
3	Video	X	-	-	-	-	X	X	-	-	O	
4	Text	X	-	-	-	-	-	-	X	X	O	
5	Speech, Video	X	X	X	-	-	X	X	-	-	O	
6	Audio, Video	X	-	-	-	-	X	X	X	X	O	
7	Speech, Text	X	-	-	X	X	X	X	-	-	O	
8	Video, Text	X	X	X	-	-	-	-	X	X	O	
9	Speech, Video, Text	X	X	X	-	-	X	X	X	X	O	
10	Audio, Text	X	-	-	X	X	-	-	X	X	O	
11	Audio, Video, Text	X	-	-	X	X	X	X	X	X	O	

X = stream included in scenario

- = stream not included in scenario

O = stream optionally included in scenario

Note: In some 3GPP specifications (e.g., [10]) "audio" and "speech" are not separated, but handled under title "audio".

In most of the scenarios, the services can be either streaming or conversational. For PS streaming, there is no full IMS support in Rel 5. However, this does not have major impact on the items presented in this document.

The protocol layers of the scenarios are presented in Figure 1 for conversational and in Figure 2 for streaming services ([10], [11]).

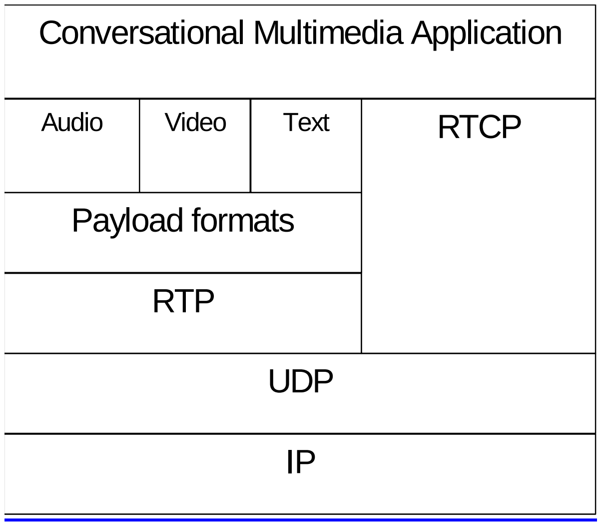

Figure 1 – User plane protocol stack for conversational multimedia terminal

The protocol layers for IMS signalling stream, not presented in the figure, are (SDP/)SIP/UDP/IP.

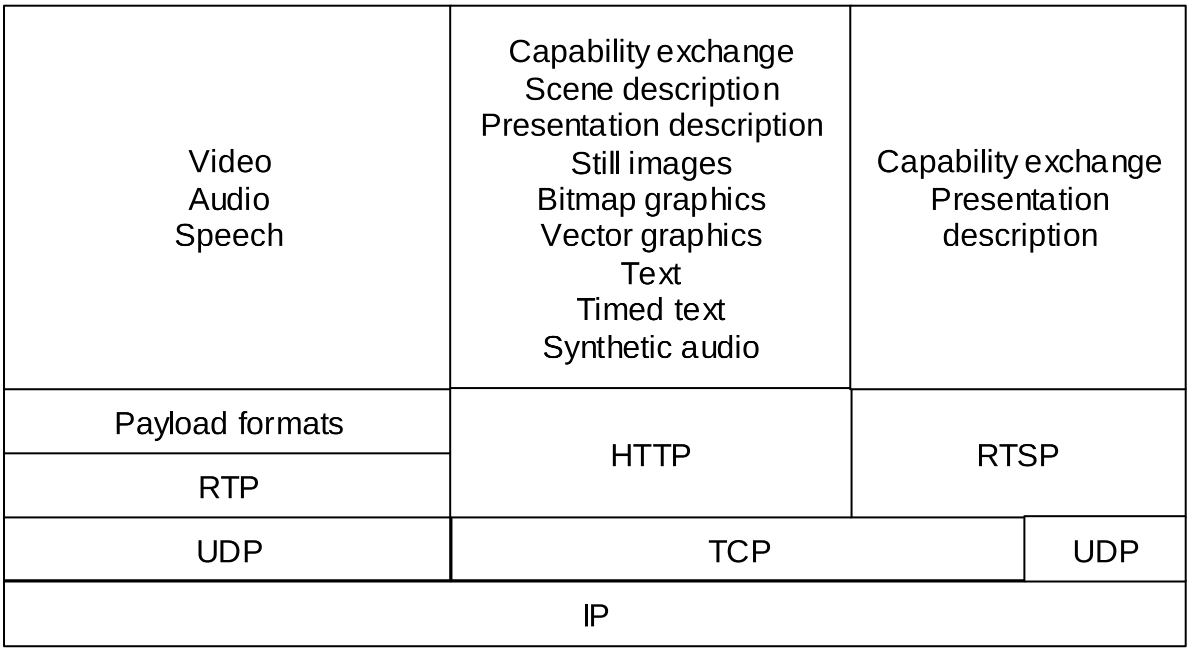

Figure 2: Protocol stack for PS streaming terminal

A.1 Common characteristics of scenarios

The characteristics of the streams in the next sub-chapters are common to all or most of the scenarios.

In scenarios, where the IP protocol header size or contents are relevant, it is assumed that IPv6 header without extension headers is used, i.e., the IP header size is 40 bytes. The UDP header size is 8 bytes.

A.1.1 RTP and RTCP streams

A.1.2 Signalling stream

A.1.3 Data stream

The data stream may be used to carry any background or interactive data. Examples on data are still images, graphics, and scene / presentation descriptions, shown in Figure 2 and [11], as well as web browsing and/or file download. Low delay is not guaranteed, and the data rates may vary between 0 kbps and the maximum bit rate of the context.

A.2 Scenarios

In each of the scenarios, there is also an additional PDP context for SIP or RTSP, and optionally one or more PDP contexts for data. Which PDP contexts are primary or secondary, is not relevant for RAB scenarios.

A.2.1 Speech

For the IMS speech service, the parameters that the transmitter should use (and the receiver shall at least support) are defined more precisely than for any other service in [10]. (Note: Speech is under term “audio” in [10]). Both AMR and AMR-WB are included. The parameters for speech are presented below, derived from [10] and [13]:

Table 2: Conversational IMS speech service parameters

		Selection or parameter value		**Notes**
	Nr of AMR / AMR-WB frames in RTP packet	One		Min. 20 ms packet interval RTP header adds 12 bytes
	AMR / AMR-WB payload mode	Bandwidth efficient		
	AMR , lowest and highest modes	AMR / AMR-WB mode	Payload bytes per frame	Payload bits include ARM data, payload header, table of contents and padding. Multi-channel session, interleaving or internal CRC not used. Size of SID frame is 7 bytes.
		4.75	14	
		12.2	32	
	AMR-WB, lowest and highest modes	6.6	18	
		23.85	61	

For non-IMS services, the above-mentioned restrictions are not applicable. However, it can be assumed that the parameters for conversational VoIP services do not usually deviate significantly from those given above.

For speech streaming, the codecs are the same as above (AMR and AMR-WB) [11]. In [14], examples on streaming services are presented. The most important difference to the conversational parameters is that the number of speech frames in one RTP packet may be much larger (e.g., 10). On the other hand, the payload mode can be different (octet aligned), CRCs included etc. (as in [14]), which gives larger payload presented in the table above.

A.2.2 Audio

"Audio" in this document refers to other than speech-based audio (music, combination of music and speech, etc.).

In [10] there is no distinction between audio and speech for conversational traffic. The default audio codecs for IMS are AMR and AMR-WB, hence the numbers of the previous chapter are applicable.

According to [11], MPEG-4 AAC-LC codec should be supported for audio streaming, and in addition, also MPEG-4 AAC-LTP may be supported. As for the speech streaming, the RTP packets contain of several audio frames, as presented in [14].

A.2.3 Video

The video codecs have a wide range of possible bit rates and packet sizes. For streaming and conversational video, the codecs are H.263 and MPEG 4 [11], [15]. RTP packet size is restricted in IMS conversational video to 512 bytes [10].

Examples on video streaming are presented in [14]. There is a wide range of RTP packet rates, depending on various factors, e.g., codec rate or packetization.

A.2.4 Text

According to [16], the data rate of T.140 text telephony over RTP is low: "The rate of character entry is usually at a level of a few characters per second or less. Therefore, the expected number of characters to transmit is low. Only one or a few new characters are expected to be transmitted with each packet." Hence, large part of the traffic consists of the overhead, i.e., RTP/UDP/IP headers and RTCP/UDP/IP packets. The data rate is mostly less than 1 kbps. Whenever the delay has to be guaranteed, the context cannot be of interactive or background traffic class, but e.g., streaming class has to be used.

It should be noted that text telephony does not include document viewing or other similar use, but only situations where the text is entered by human users in the both ends. For example, the "Text" service in Figure 2 does not refer to text telephony.

A.2.5 Speech and video

There are basically two different alternatives, depending on whether audio and video streams are on the same or different PDP contexts. The former case is basically similar to the scenario in the chapter A.2.3. The latter latter case has different implications on lower layers. For streaming case with speech and video over the same context, there is an example in [14].

A.2.6 Audio and video

The difference in this scenario to the previous one is that the audio/speech coded may be different. On lower layers, this can be handled as the previous scenario.

A.2.7 Video, audio, or speech with text

The additional text telephony stream adds a low bit rate PDP context. Whenever there is a requirement to synchronize the text with the voice or video stream, the text telephony context delay parameters have to be aligned with those of the others (i.e., the delay requirement may be stricter than for stand-alone text telephony).

Annex B: Mapping of service scenarios to Radio Access Bearers

B.1 Common requirements

The bearers in this document shall be based entirely on existing 25-series specifications. That is, no requirement on RABs that is not in line with existing RAN specifications, shall be presented.

In this chapter, the main principles for selecting the parameters are presented.

B.2 Bearer characteristics

The following table lists general characteristics of the bearers in the scenarios.

	Parameter	**Typical selection or parameter value**	**Notes**
PDCP	PDCP header, bits	8	8 bit PDCP header is the default in the scenarios. (For lossless SRNS relocation support, PDCP header can also contain sequence number of 16 bits.)
	Header compression	RFC 3095 (ROHC)	ROHC assumed to compress [RTP/]UDP/IP (and ESP/IP) traffic. No ROHC context identifier needed: PID field (5 bits) of PDCP header is sufficient to indicate all ROHC contexts in the given scenarios. The most common header (shortest 2nd order header) is 3 bytes when UDP checksum is present (with IPv6); see RLC payload sizes. ROHC feedback packets transmitted in opposite direction, interspersed with main flow packets. Segmentation of ROHC not in use, because only non-transparent RLC modes in these scenarios.
		RFC 2507	For TCP/IP compression (even though any IP headers, also those in UDP/IP could be compressed by RFC 2507). TCP/IP used in interactive and background, therefore no impact on RLC payload sizes
RLC			

	RLC mode	UM or AM	TM not possible because no a priori information on (compressed) IP packets, and no mechanism specified for negotiating ROHC packet sizes parameters. UM used for conversational traffic class, AM for all other classes.
	Payload sizes, bit		Number of different payload sizes to be limited so that max size of TFCS is reasonably low. In some scenarios, one of payload sizes is IP payload with shortest ROHC header. For AM, default payload size is 320 bits
	Max data rate, kbps		The actual data rate on IP layer is somewhat different from this nominal figure, due to: • PDCP header • Length indicator part of RLC header • Retransmissions (in AM) • Header compression
	UMD/AMD PDU header, bit	8 / 16	8 for UM, 16 for AM

Table 3: Common characteristics of L2

In the scenarios, the RABs for data stream are not presented. Each of the scenarios may or may not have one or more RABs for data stream. The RABs can be selected from the interactive/background RABs.

B.3 RAB Scenarios

Note: The following RAB combinations are only examples on possible implementations of the scenarios. Due to flexibility in RAN specifications (and in PDP context parameters) there is a large number of other possible RABs and their combinations that could implement the scenarios. There are also other RAB combinations applicable for other scenarios, not listed below.

This chapter concentrates on the basic scenarios of the previous chapter, thus excluding most of the combinations of multiple sessions.