Source: **TSG SA WG2**
Title: **CRs on 23.107 v 3.0.0**
Agenda Item: **5.2.3**

The following CRs have been approved by TSG SA WG2 and are requested to be approved by TSG SA plenary #6.

On 23.107 v.3.0.0

TDoc #	CR #	spec	Title	cat
S2-99C21	001r1	23.107	GPRS/UMTS QoS Parameter Mapping	
S2-99E25	003r1	23.107	Update of Transfer delay attribute definition	
S2-99D33	004	23.107	Clarification on rate control, asymetry and error ratios attributes	
S2-99E90	005 draft	23.107	Clarification on parameter value ranges	F
S2-99D35	006r1	23.107	Clarification of Maximum bit rate attribute	C
S2-99E28	007r1	23.107	Mapping of QoS profiles between R97/98 and R99	
S2-99D38	008	23.107	Generation of QoS parameters for CS data services for call setup and interworking UMTS-CS	
S2-99F37	010r2	23.107	Rules for the Comparison of QoS Profiles	
S2-99F25	011	23.107	Editorial changes to 23.107	D

3GPP TSG SA2 Meeting #9
London, United Kingdom, 25-29.10.1999

***Document* S2-99C21**
e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

CHANGE REQUEST

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23.107	CR	001r1	Current Version:	3.0.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	SA#6	for approval	X	strategic		(for SMG use only)
list expected approval meeting # here ↑		for information		non-strategic		

Form: CR cover sheet, version 2 for 3GPP and SMG *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects: *(at least one should be marked with an X)*

(U)SIM	X	ME	X	UTRAN / Radio	X	Core Network	X

Source: Nokia, Alcatel, Motorola, Ericsson, Siemens, Lucent, Fujitsu, T-Mobile

Date: 22.10.1999

Subject: GPRS/UMTS QoS Parameter Mapping

Work item:

Category: *(only one category shall be marked with an X)*

			Release:	
F	Correction			Phase 2	
A	Corresponds to a correction in an earlier release			Release 96	
B	Addition of feature	X		Release 97	
C	Functional modification of feature			Release 98	
D	Editorial modification			Release 99	X
				Release 00	

Reason for change: Mapping between GPRS 99/UMTS QoS parameters and GPRS 97/98 QoS parameters is required for interworking purposes. This CR introduces an informative annex into 23.107 which proposes a baseline for discussion. It will be necessary to later introduce such mapping rules based on Annex C into subclause 9.1 as normative information.

Clauses affected: Section 9.1.3, Addition of Annex C

Other specs affected:

Other 3G core specifications		→ List of CRs:
Other GSM core specifications		→ List of CRs:
MS test specifications		→ List of CRs:
BSS test specifications		→ List of CRs:
O&M specifications		→ List of CRs:

Other comments:

help.doc

<--------- double-click here for help and instructions on how to create a CR.

9.1.2 UMTS-GPRS

Note: The mapping between UMTS and GPRS release 97/98 is currently under discussion. Annex C provides a baseline for further discussions in this area.

~~Note: Part of GPRS phase 1 QoS are vaguely defined. This chapter has to be updated according to CRs to GPRS phase 1 QoS parameters. Definition of GPRS phase 2 is starting and it has to be taken into account in here and vice versa.~~

~~GPRS has more QoS parameters than GSM CS thus requiring more complex mapping rules. Below an example of mapping GPRS phase 1 QoS parameters to UMTS traffic classes is presented.~~

~~**Conversational Class**~~

~~Conversational class services are mainly for conversational real time use. An example of conversational real time application is video telephony.~~

~~An appropriate use of GPRS parameters:~~

- ~~Mean Throughput Class = Peak bit rate (constant bit rate)~~

~~Or Mean Throughput Class < Peak bit rate (variable bit rate)~~

- ~~Reliability: 4 or 5 (no retransmissions)~~
- ~~Precedence:1-3~~
- ~~Delay Class: 1 (real time)~~

~~Streaming Class~~

~~Streaming class services are mainly appropriate for streaming real time applications, e.g. video downloading. Some variation in delay can be tolerated because of application level buffering.~~

~~An appropriate use of GPRS parameters:~~

- ~~Mean Throughput Class = Peak bit rate (constant bit rate)~~

~~Or Mean Throughput Class < Peak bit rate (variable bit rate)~~

- ~~Reliability: 3 (light retransmissions)~~
- ~~Precedence: 1-3~~
- ~~Delay Class: 1 (real time)~~

~~Interactive Class~~

~~Interactive class services are mainly for interactive services requiring a variable guaranteed throughput: specialised applications (banking, plane reservation, …), interactive WWW, Telnet etc.~~

~~An appropriate use of GPRS parameters:~~

- ~~Mean Throughput Class has no meaning in UMTS~~
- ~~Reliability: 1-2~~
- ~~Precedence: 1-3~~
- ~~Delay class: 2-4~~

~~Background Class~~

~~Background services are mainly for best effort services: background download, emails, calendar, event etc.~~

~~An appropriate use of GPRS parameters:~~

~~- Mean Throughput Class has no meaning in UMTS~~

~~- Reliability: 2~~

~~- Precedence: 1~~ 3

~~- Delay class: 4 (best effort)~~

Annex C: Mapping between UMTS and GPRS Release 97/98 QoS Parameters (informative)

GPRS Release 99 QoS attributes shall be equivalent to the UMTS QoS Attributes. For interworking purposes between different releases, mapping rules between GPRS Release 97/98 and GPRS Release 99 as well as UMTS have to be defined. These inter-release mapping rules are defined in this annex.

Mapping shall occur whenever both network elements using GPRS Release 97/98 and GPRS release 99 or UMTS are involved in a PDP context activation procedure. Scenarios in which such a mapping occur will have to be determined and studied in the future.

It is not within the scope of this annex to determine if any value combinations for GPRS Release 97/98 parameters can not be supported. This means that complete mapping rules are defined here, and if the user requests a QoS profile which the network may not be able to support (e.g. a low delay and a high reliability), the decision if such a parameter combination can be supported is left to admission control functionality within the PDP context activation procedure, and the QoS for such a profile may be renegotiated by the network based on the available resources.

C.1 Mapping of GPRS Release 97/98 to UMTS QoS Parameters

The tables below show the mapping of the GPRS Release 97/98 QoS parameters to the UMTS QoS parameters.

The UMTS traffic class is determined by the delay class. However, for the delay classes 2 and 3, the mean throughput class is also considered to determine the UMTS traffic class. This mapping is shown in Table 1~~Table 1~~.

Table 1: Determining the UMTS Traffic Class and Transfer Delay

Delay class	**Mean Throughput class**	**Resulting UMTS Traffic Class**	**Resulting UMTS Transfer Delay**
1	Any	Conversational	100ms
2	1 to 18	Streaming	500ms
	31 (Best effort)	Interactive	N/A
3	1 to 18	Streaming	1s
	31 (Best effort)	Interactive	N/A
4	Any	Background	N/A

The mapping of the other GPRS Release 97/98 QoS parameters to the UMTS QoS parameters depends on the traffic class chosen according to Table 1~~Table 1~~ and is defined in Table 3~~Table 2~~.

Table 3~~2~~: Application of Pre-GPRS99 QoS Parameters to UMTS Traffic Classes

Pre-GPRS99 Parameters	**UMTS Traffic Class**

	Conversational	Streaming	Interactive	Background
Delay (1-4) **(See table 1)**	1	2 and 3	2 and 3	4
Peak Throughput (1-9)	UMTS Maximum Bit Rate	UMTS Maximum Bit Rate	UMTS Maximum Bit Rate	UMTS Maximum Bit Rate
Mean Throughput **(1-18, 31=Best Effort)**	UMTS Guaranteed Bit Rate (<= UMTS Maximum Bit Rate)	UMTS Guaranteed Bit Rate (<= UMTS Maximum Bit Rate)	N/A	N/A
Reliability (1-5) **(See Table 5~~Table 3~~ and Table 7~~Table 4~~)**	UMTS Delivery of erroneous SDUs: 5=Yes, all other values = No	UMTS Delivery of erroneous SDUs: 5=Yes, all other values = No	UMTS Delivery of erroneous SDUs: 5=Yes, all other values = No	UMTS Delivery of erroneous SDUs: 5=Yes, all other values = No
Precedence (1-3) **(value is directly mapped to the following parameters)**	-	-	UMTS Traffic handling priority	-
	UMTS Allocation/ Retention priority	UMTS Allocation/ Retention priority	UMTS Allocation/ Retention priority	UMTS Allocation/ Retention priority
Max N-PDU (1500 octets)	Max SDU = 1500 octets	Max SDU = 1500 octets	Max SDU = 1500 octets	Max SDU = 1500 octets

The UMTS Delivery Order parameter is directly derived from the PDP type in the mapping procedure. There is however no mapping for the UMTS SDU format information

The mapping from the Reliability Class in GPRS Release 97/98 to the SDU error ratio values and residual bit error ratios in Release 99 and UMTS is shown in Table 5~~Table 3~~ and Table 7~~Table 4~~.

Table 5~~3~~: Mapping of the Reliability Class to SDU Error Ratio Values

Reliability Class	Conversational Class	Streaming Class	Interactive Class	Background Class
1	10^{-5}	10^{-5}	10^{-6}	10^{-6}
2	10^{-4}	10^{-4}	10^{-6}	10^{-6}
3	10^{-3}	10^{-3}	10^{-4}	10^{-4}
4	10^{-2}	10^{-2}	10^{-3}	10^{-3}
5	10^{-2}	10^{-2}	10^{-3}	10^{-3}

Table 74: Mapping of the Reliability Class to Residual Bit Error Ratio Values

Reliability Class	**Conversational Class**	**Streaming Class**	**Interactive Class**	**Background Class**
1	10^{-2}	10^{-2}	10^{-5}	10^{-5}
2	10^{-2}	10^{-2}	10^{-5}	10^{-5}
3	10^{-2}	10^{-2}	10^{-5}	10^{-5}
4	10^{-2}	10^{-2}	10^{-5}	10^{-5}
5	$5*10^{-2}$	$5*10^{-2}$	$4*10^{-3}$	$4*10^{-3}$

C.2 Mapping of UMTS QoS to GPRS Release 97/98 Parameters

Table 9~~Table 5~~ shows a mapping of UMTS bearer attributes to GPRS Release 97/98 QoS parameters. The table entries show the applicable GPRS Release 97/98 QoS parameters and any value restrictions.

Table 9~~5~~: Mapping of UMTS Bearer Attributes to Pre-GPRS99 QoS Parameters

UMTS QoS Bearer Attributes	**UMTS Traffic Class**			
Traffic class	Conversational	Streaming	Interactive	Background
Maximum bit rate **see note (4)**	GPRS Peak Throughput	GPRS Peak Throughput	GPRS Peak Throughput	GPRS Peak Throughput
Delivery order	Linked to PDP Type	Linked to PDP Type	Linked to PDP Type	Linked to PDP Type
Maximum SDU size	GPRS Max N-PDU Size, see note (1)	GPRS Max N-PDU Size, see note (1)	GPRS Max N-PDU Size, see note (1)	GPRS Max N-PDU Size, see note (1)
SDU format information	N/A	N/A	N/A	N/A
Delivery of erroneous SDUs	GPRS Reliability Class 1-4=No, 5=Yes	GPRS Reliability Class 1-4=No, 5=Yes	GPRS Reliability Class 1-4=No, 5=Yes	GPRS Reliability Class 1-4=No, 5=Yes
Transfer delay **see note (3)**	GPRS delay = 1 (realtime)	GPRS delay = 2 (realtime)	GPRS delay = 3 (non-realtime)	GPRS delay = 4 (best effort)
Guaranteed bit rate **see note (4)**	Bit rate of GPRS Mean Throughput Class	Bit rate of GPRS Mean Throughput Class	GPRS Mean Throughput Class = 31 (Best effort)	GPRS Mean Throughput Class = 31 (Best effort)
Traffic handling priority **see note (2)**	-	-	GPRS Service Precedence	-
Allocation/Retention priority	GPRS Service Precedence	GPRS Service Precedence	-	GPRS Service Precedence

Table notes:

(1) PDP-PDUs are transferred between the MS and the GGSN via N-PDUs which are limited in size to 1500 octets. The handling of PDP-PDUs larger than this size is implementation dependent. See GSM 03.60.

(2) The Traffic handling priority is only applicable to the Interactive traffic class.

(3) The actual UMTS Transfer Delay is not used in the mapping. The UMTS Traffic Class is used to map to the GPRS Delay Class.

(4) The throughput classes are to be determined by comparing the throughput values which correspond with the throughput classes with the UMTS guaranteed and maximum bitrate values respectively and choosing the next lower throughput value.

The GPRS Release 97/98 reliability class is chosen by performing a reversed mapping based on Table 5~~Table 3~~ and Table 7~~Table 4~~. The UMTS SDU error ratio is compared with the values in Table 5~~Table 3~~ to determine the reliability class. If the matching is ambiguous, then in addition to that, the UMTS residual bit error ratio is compared to the values in Table 7~~Table 4~~. Here, the value in the table shall be chosen which is closest to the UMTS residual bit error ratio. If the matching is still ambiguous, the highest of all matching reliability classes shall be chosen.

3GPP TSG SA2

Abiko , Japan, 29 Nov – 3 Dec 1999

Document S2-99E25

3G CHANGE REQUEST

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23.107	**CR**	**003 draft**	Current Version:	3.0.0

3G specification number ↑ *↑ CR number as allocated by 3G support team*

For submision to TSG SA2# *list TSG meeting no. here ↑*

for approval [X]
for information [ ]

(only one box should be marked with an X)

Form: 3G CR cover sheet, version 1.0 *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/3GCRF-xx.rtf*

Proposed change affects: *(at least one should be marked with an X)* USIM [ ] ME [X] UTRAN [X] Core Network [X]

Source: Ericsson **Date:** Nov 26, 1999

Subject: Update of Transfer delay attribute definition

3G Work item: End-to-end QoS

Category: *(only one category shall be marked with an X)*

F	Correction	[ ]	**Release:**	Phase 2
A	Corresponds to a correction in a 2G specification	[ ]		Release 96
B	Addition of feature	[ ]		Release 97
C	Functional modification of feature	[X]		Release 98
D	Editorial modification	[ ]		Release 99 X
		[ ]		Release 00

Reason for change: In version 3.0.0 of 23.107, the definition of the Transfer delay attribute is not yet finalized. The current draft definition includes an indicative delay definition based on delay for typical predefined SDU sizes. The problem with this indicative approach is the difficulty to decide on these predefined sizes (they are still FFS), and that there is no delay bound for other SDU sizes. Hence, an application producing SDU sizes not predefined has no means to express a delay requirement through the transfer delay attribute. A more relevant and exact definition is needed.
Moreover, in the current draft definition, the "exact statistical transfer delay definition is FFS". This CR proposes such a statistical definition.

Clauses affected: 6.4.3.1, 6.4.4.1

Other specs affected:

Other 3G core specifications	[ ]	→ List of CRs:
Other 2G core specifications	[ ]	→ List of CRs:
MS test specifications	[ ]	→ List of CRs:
BSS test specifications	[ ]	→ List of CRs:
O&M specifications	[ ]	→ List of CRs:

Other comments:

6.4.3 UMTS Bearer Service Attributes

6.4.3.1 List of attributes

Note: The text within square brackets explaining the purpose of each attribute can be excluded later if that information is given elsewhere in the technical report.

Traffic class [‘conversational’, ’streaming’, ’interactive’, ’background’]
Definition: type of application for which the UMTS bearer service is optimised
[Purpose: By including the traffic class itself as an attribute, UMTS can make assumptions about the traffic source and optimise the transport for that traffic type.]
Maximum bitrate [kbps]
Definition: maximum number of bits delivered by UMTS at a SAP within a period of time, divided by the duration of the period. The traffic is conformant with Maximum bitrate as long as it follows a token bucket algorithm where token rate equals Maximum bitrate and bucket size equals Maximum SDU size.
The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.
[Purpose: Maximum bitrate can be used to make code reservations in the downlink of the radio interface. Its purpose is to limit the delivered bitrate to applications or external networks with such limitations]
Guaranteed bitrate [kbps]
Definition: guaranteed number of bits delivered by UMTS at a SAP within a period of time (provided that there is data to deliver), divided by the duration of the period. The traffic is conformant with the guaranteed bitrate as long as it follows a token bucket algorithm where token rate equals Guaranteed bitrate and bucket size equals k*Maximum SDU size. For release 99, k=1. A value of k greater than one Maximum SDU size may be specified in future releases to capture burstiness of sources. Signalling to specify the value of k may be provided in future releases.
The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.
[Purpose: Guaranteed bitrate may be used to facilitate admission control based on available resources, and for resource allocation within UMTS. Quality requirements expressed by e.g. delay and reliability attributes only apply to incoming traffic up to the guaranteed bitrate.]
Delivery order [y/n]
Definition: indicates whether the UMTS bearer shall provide in-sequence SDU delivery or not.
[Purpose: the attribute is derived from the user protocol [PDP type] and specifies if out-of-sequence SDUs are acceptable or not. This information cannot be extracted from the traffic class. Whether out-of-sequence SDUs are dropped or re-ordered depends on the specified reliability]
Maximum SDU size [bits]
Definition: the maximum allowed SDU size
[Purpose: The maximum SDU size is used for admission control and policing.]
SDU format information [bits]
Definition: list of possible exact sizes of SDUs
[Purpose: UTRAN needs SDU size information to be able to operate in transparent RLC protocol mode, which is beneficial to spectral efficiency and delay when RLC re-transmission is not used. Thus, if the application can specify SDU sizes, the bearer is less expensive.]
SDU error ratio
Definition: Indicates the fraction of SDUs lost or detected as erroneous. SDU error ratio is defined only for conforming traffic.
Note that by reserving resources, SDU error ratio performance is independent of the loading conditions, whereas without reserved resources, such as in Interactive and Background classes, SDU error ratio is used as target value.
[Purpose: Used to configure the retransmission protocol on layer 2 and the error detection coding on layer 1.]
Residual bit error ratio
Definition: Indicates the undetected bit error ratio in the delivered SDUs. If no error detection is requested, Residual bit error ratio indicates the bit error ratio in the delivered SDUs.
[Purpose: Used to configure channel coding and error detection coding on layer 1.]
Delivery of erroneous SDUs (y/n/-)
Definition: Indicates whether SDUs detected as erroneous shall be delivered or discarded.

Note: 'yes' implies that error detection is employed and that erroneous SDUs are delivered together with an error indication, 'no' implies that error detection is employed and that erroneous SDUs are discarded, and '-' implies that SDUs are delivered without considering error detection.

[Purpose: Used to decide whether frames with failed CRC on layer 1 shall be forwarded or not.]

Transfer delay [s]

Definition: Indicates maximum delay for 95th percentile of the distribution of delay for all delivered SDUs during the lifetime of a bearer service, where delay for an SDU is defined as the time from a request to transfer an SDU at one SAP to its delivery at the other SAP.

~~Definition: time between request to transfer an SDU at one SAP to its delivery at the other SAP. Transfer delay is specified for one or more fixed SDU sizes. Exact statistical transfer delay definition and fixed SDU sizes are FFS.~~

[Purpose: used to specify the delay tolerated by the application. It allows UTRAN to set transport formats and ARQ parameters.]

Note: Transfer delay of an arbitrary SDU is not meaningful for a bursty source, since the last SDUs of a burst may have long delay due to queuing, whereas the meaningful response delay perceived by the user is the delay of the first SDU of the burst.

Traffic handling priority

Definition: specifies the relative importance for handling of all SDUs belonging to the UMTS bearer compared to the SDUs of other bearers.

[Purpose: Within the interactive class, there is a definite need to differentiate between bearer qualities. This is handled by using the traffic handling priority attribute, to allow UMTS to schedule traffic accordingly. By definition, priority is an alternative to absolute guarantees, and thus these two attribute types cannot be used together for a single bearer.]

Allocation/Retention Priority

Definition: specifies the relative importance compared to other UMTS bearers for allocation and retention of the UMTS bearer.

[Purpose: Priority is used for differentiating between bearers when performing allocation and retention of a bearer, and the value is typically related to the subscription.

6.4.4 Radio Access Bearer Service Attributes

Radio Access Bearer Service Attributes shall be applied to both CS and PS domains.

6.4.4.1 List of attributes

Note: The text within square brackets explaining the purpose of each attribute can be excluded later if that information is given elsewhere in the technical report.

Traffic class ['conversational', 'streaming', 'interactive', 'background']

Definition: type of application for which the Radio Access Bearer service is optimised

[Purpose: By including the traffic class itself as an attribute, UTRAN can make assumptions about the traffic source and optimise the transport for that traffic type. In particular, buffer allocation may be based on traffic class.]

Maximum bitrate [kbps]

Definition: maximum number of bits delivered by UTRAN at a SAP within a period of time, divided by the duration of the period. The traffic is conformant with the Maximum bitrate as long as it follows a token bucket algorithm where token rate equals Maximum bitrate and bucket size equals Maximum SDU size. The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.

[Purpose: to limit the delivered bitrate to applications or external networks with such limitations]

Guaranteed bitrate [kbps]

Definition: guaranteed number of bits delivered at a SAP within a period of time (provided that there is data to deliver), divided by the duration of the period. The traffic is conformant with the Guaranteed bitrate as long as it follows a token bucket algorithm where token rate equals Guaranteed bitrate and bucket size equals k Maximum SDU size. For Release 99, k = 1. A value of k greater than one Maximum SDU size may be specified in future releases to capture burstiness of sources. Signalling to specify the value of k may be

provided in future releases.
The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.
[Purpose: Guaranteed bitrate may be used to facilitate admission control based on available resources, and for resource allocation within UTRAN. Quality requirements expressed by e.g. delay and reliability attributes only apply to incoming traffic up to the guaranteed bitrate. The guaranteed bitrate at the RAB level may be different from that on UMTS bearer level, for example due to header compression.]
Delivery order [y/n]
Definition: indicates whether the UMTS bearer shall provide in-sequence SDU delivery or not.
[Purpose: specifies if out-of-sequence SDUs are acceptable or not. This information cannot be extracted from the traffic class. Whether out-of-sequence SDUs are dropped or re-ordered depends on the specified reliability]
Maximum SDU size [bits]
Definition: the maximum allowed SDU size
[Purpose: The maximum SDU size is used for admission control and policing.]
SDU format information [bits]
Definition: list of possible exact sizes of SDUs. If unequal error protection shall be used by a Radio Access Bearer service, SDU format information defines the exact subflow format of the SDU payload.
Note: SDU format information is used by UTRAN to define which bits of the payload that belongs to each subflow. Exact syntax of SDU format information attribute is the task of RAN WG3

[Purpose: UTRAN needs SDU format information to be able to operate in transparent RLC protocol mode, which is beneficial to spectral efficiency and delay when RLC re-transmission is not used. Thus, if the application can specify SDU sizes, the bearer is less expensive. Moreover, in case of unequal error protection, UTRAN needs to know the exact format of SDU payload to be able to demultiplex the SDU onto different radio bearer services.]
SDU error ratio
Definition: Indicates the fraction of SDUs lost or detected as erroneous. SDU error ratio is defined only for conforming traffic. In case of unequal error protection., SDU error ratio is set per subflow and represents the error ratio in each subflow. SDU error ratio is only set for subflows for which error detection is requested.
Note: By reserving resources, SDU error ratio performance is independent of the loading conditions, whereas without reserved resources, such as in Interactive and Background classes, SDU error ratio is used as target value.

[Purpose: Used to configure the retransmission protocol on layer 2 and the error detection coding on layer 1]
Residual bit error ratio
Definition: Indicates the undetected bit error ratio for each subflow in the delivered SDUs. For equal error protection, only one value is needed. If no error detection is requested for a subflow, Residual bit error ratio indicates the bit error ratio in that subflow of the delivered SDUs.
[Purpose: Used to configure channel coding and error detection coding on layer 1. For services requiring unequal error protection, residual bit error ratio is given for each subflow.]
Delivery of erroneous SDUs (y/n/-)
Definition: Indicates whether SDUs with detected errors shall be delivered or not. In case of unequal error protection, the attribute is set per subflow.
Note: ‘yes’ implies that error detection is employed and that erroneous SDUs are delivered together with an error indication, ‘no’ implies that error detection is employed and that erroneous SDUs are discarded, and ‘-‘ implies that SDUs are delivered without considering error detection.
In case of unequal protection, different subflows may have different settings. Whenever there is a detected error in a subflow with ‘no’, the SDU is discarded, irrespective of settings in other subflows. For an SDU with multiple subflows with a ‘yes’ setting, there may be one error indication per subflow, or, if there is only one error indication per SDU, it indicates that an error was detected in at least one of these subflows. Exact definitions are the task of RAN3.
[Purpose: Used to decide whether frames with failed CRC on layer 1 shall be forwarded or discarded.]
Transfer delay [s]
Definition: Indicates maximum delay for 95th percentile of the distribution of delay for all delivered SDUs during the lifetime of a bearer service, where delay for an SDU is defined as the time from a request to transfer an SDU at one SAP to its delivery at the other SAP.
~~Definition: time between request to transfer an SDU at one SAP to its delivery at the other SAP. Transfer delay is specified for one or more fixed SDU sizes. Exact statistical transfer delay definition and fixed SDU sizes are FFS.~~

[Purpose: specifies the UTRAN part of the total transfer delay for the UMTS bearer. It allows UTRAN to set transport formats and ARQ parameters.]

Traffic handling priority

Definition: specifies the relative importance for handling of all SDUs belonging to the radio access bearer compared to the SDUs of other bearers.

[Purpose: Within the interactive class, there is a definite need to differentiate between bearer qualities. This is handled by using the traffic handling priority attribute, to allow UTRAN to schedule traffic accordingly. By definition, priority is an alternative to absolute guarantees, and thus these two attribute types cannot be used together for a single bearer.]

Allocation/Retention Priority

Definition: specifies the relative importance compared to other Radio access bearers for allocation and retention of the Radio access bearer.

[Purpose: Priority is used for differentiating between bearers when performing allocation and retention of a bearer, and the value is typically related to the subscription.

Source statistics descriptor ['speech'/'unknown']

Definition: specifies characteristics of the source of submitted SDUs.

[Purpose: Conversational speech has a well-known statistical behaviour (or the discontinuous transmission (DTX) factor). By being informed that the SDUs for a RAB are generated by a speech source, UTRAN may, based on experience, calculate a statistical multiplex gain for use in admission control on the radio and Iu interfaces.]

3GPP TSG SA2 *Document* ***S2-99D33***

Abiko , Japan, 29 Nov – 3 Dec 1999

3G CHANGE REQUEST

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23.107	**CR**	**004 draft**	Current Version:	3.0.0

3G specification number ↑ *↑ CR number as allocated by 3G support team*

For submision to TSG SA2# *list TSG meeting no. here ↑*

for approval [X] for information [ ] *(only one box should be marked with an X)*

Form: 3G CR cover sheet, version 1.0 *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/3GCRF-xx.rtf*

Proposed change affects: *(at least one should be marked with an X)* USIM [ ] ME [X] UTRAN [X] Core Network [X]

Source: Ericsson **Date:** Nov 26, 1999

Subject: Clarifications on rate control, asymmetry and error ratio attributes

3G Work item: End-to-end QoS

Category: *(only one category shall be marked with an X)*

F	Correction	[ ]	**Release:**	Phase 2	
A	Corresponds to a correction in a 2G specification	[ ]		Release 96	
B	Addition of feature	[ ]		Release 97	
C	Functional modification of feature	[ ]		Release 98	
D	Editorial modification	[X]		Release 99	X
		[ ]		Release 00	

Reason for change: A recent LS from TSG RAN3, R3-99G18, brings up a number of issues related to TS 23.107. Some of the comments indicate that editorial clarifications to 23.107 are needed. This CR clarifies the use of rate control, asymmetric bearers and reliability attributes.

Clauses affected: 6.4.1, 6.4.3, 6.4.4

Other specs affected:

Other 3G core specifications	[ ]	→ List of CRs:
Other 2G core specifications	[ ]	→ List of CRs:
MS test specifications	[ ]	→ List of CRs:
BSS test specifications	[ ]	→ List of CRs:
O&M specifications	[ ]	→ List of CRs:

Other comments:

6.4.1 Asymmetric Bearers

Uni-directional and bi-directional bearer services shall be supported. For bi-directional bearer services, ~~t~~The attributes Maximum bitrate and Guaranteed bitrate~~parameters related to throughput/bitrate~~ should be possible to set separately~~d~~ for uplink/downlink in order to support asymmetric bearers.

6.4.3 UMTS Bearer Service Attributes

6.4.3.1 List of attributes

Note: The text within square brackets explaining the purpose of each attribute can be excluded later if that information is given elsewhere in the technical report.

Traffic class [‘conversational’, ’streaming’, ’interactive’, ’background’]
Definition: type of application for which the UMTS bearer service is optimised
[Purpose: By including the traffic class itself as an attribute, UMTS can make assumptions about the traffic source and optimise the transport for that traffic type.]
Maximum bitrate [kbps]
Definition: maximum number of bits delivered by UMTS at a SAP within a period of time, divided by the duration of the period. The traffic is conformant with Maximum bitrate as long as it follows a token bucket algorithm where token rate equals Maximum bitrate and bucket size equals Maximum SDU size.
The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.
[Purpose: Maximum bitrate can be used to make code reservations in the downlink of the radio interface. Its purpose is to limit the delivered bitrate to applications or external networks with such limitations]
Guaranteed bitrate [kbps]
Definition: guaranteed number of bits delivered by UMTS at a SAP within a period of time (provided that there is data to deliver), divided by the duration of the period. The traffic is conformant with the guaranteed bitrate as long as it follows a token bucket algorithm where token rate equals Guaranteed bitrate and bucket size equals k*Maximum SDU size. For release 99, k=1. A value of k greater than one Maximum SDU size may be specified in future releases to capture burstiness of sources. Signalling to specify the value of k may be provided in future releases.
The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.
[Purpose: Guaranteed bitrate may be used to facilitate admission control based on available resources, and for resource allocation within UMTS. Quality requirements expressed by e.g. delay and reliability attributes only apply to incoming traffic up to the guaranteed bitrate.]
Delivery order [y/n]
Definition: indicates whether the UMTS bearer shall provide in-sequence SDU delivery or not.
[Purpose: the attribute is derived from the user protocol [PDP type] and specifies if out-of-sequence SDUs are acceptable or not. This information cannot be extracted from the traffic class. Whether out-of-sequence SDUs are dropped or re-ordered depends on the specified reliability]
Maximum SDU size [bits]
Definition: the maximum allowed SDU size
[Purpose: The maximum SDU size is used for admission control and policing.]
SDU format information [bits]
Definition: list of possible exact sizes of SDUs
[Purpose: UTRAN needs SDU size information to be able to operate in transparent RLC protocol mode, which is beneficial to spectral efficiency and delay when RLC re-transmission is not used. Thus, if the application can specify SDU sizes, the bearer is less expensive.]
SDU error ratio
Definition: Indicates the fraction of SDUs lost or detected as erroneous. SDU error ratio is defined only for conforming traffic.
Note that by reserving resources, SDU error ratio performance is independent of the loading conditions, whereas without reserved resources, such as in Interactive and Background classes, SDU error ratio is used as target value.
*[Purpose: Used to configure ~~the retransmission~~ protocol*s*, *algorithms* ~~on layer 2~~ and ~~the~~ error detection schemes, primarily within UTRAN~~coding on layer 1~~.]*
Residual bit error ratio

Definition: Indicates the undetected bit error ratio in the delivered SDUs. If no error detection is requested, Residual bit error ratio indicates the bit error ratio in the delivered SDUs.
[Purpose: Used to configure radio interface protocols, algorithms ~~channel coding~~ and error detection coding ~~on layer 1~~.]
Delivery of erroneous SDUs (y/n/-)
Definition: Indicates whether SDUs detected as erroneous shall be delivered or discarded.
Note: 'yes' implies that error detection is employed and that erroneous SDUs are delivered together with an error indication, 'no' implies that error detection is employed and that erroneous SDUs are discarded, and '-' implies that SDUs are delivered without considering error detection.

[Purpose: Used to decide whether error detection is needed and whether frames with detected errors ~~failed CRC on layer 1~~ shall be forwarded or not.]
Transfer delay [s]
Definition: time between request to transfer an SDU at one SAP to its delivery at the other SAP. Transfer delay is specified for one or more fixed SDU sizes. Exact statistical transfer delay definition and fixed SDU sizes are FFS.
[Purpose: used to specify the delay tolerated by the application. It allows UTRAN to set transport formats and ARQ parameters.]
Note: Transfer delay of an arbitrary SDU is not meaningful for a bursty source, since the last SDUs of a burst may have long delay due to queuing, whereas the meaningful response delay perceived by the user is the delay of the first SDU of the burst.

Traffic handling priority
Definition: specifies the relative importance for handling of all SDUs belonging to the UMTS bearer compared to the SDUs of other bearers.
[Purpose: Within the interactive class, there is a definite need to differentiate between bearer qualities. This is handled by using the traffic handling priority attribute, to allow UMTS to schedule traffic accordingly. By definition, priority is an alternative to absolute guarantees, and thus these two attribute types cannot be used together for a single bearer.]
Allocation/Retention Priority
Definition: specifies the relative importance compared to other UMTS bearers for allocation and retention of the UMTS bearer.
[Purpose: Priority is used for differentiating between bearers when performing allocation and retention of a bearer, and the value is typically related to the subscription.

6.4.4 Radio Access Bearer Service Attributes

Radio Access Bearer Service Attributes shall be applied to both CS and PS domains.

6.4.4.1 List of attributes

Note: The text within square brackets explaining the purpose of each attribute can be excluded later if that information is given elsewhere in the technical report.

Traffic class ['conversational', 'streaming', 'interactive', 'background']
Definition: type of application for which the Radio Access Bearer service is optimised
[Purpose: By including the traffic class itself as an attribute, UTRAN can make assumptions about the traffic source and optimise the transport for that traffic type. In particular, buffer allocation may be based on traffic class.]
Maximum bitrate [kbps]
Definition: maximum number of bits delivered by UTRAN at a SAP within a period of time, divided by the duration of the period. The traffic is conformant with the Maximum bitrate as long as it follows a token bucket algorithm where token rate equals Maximum bitrate and bucket size equals Maximum SDU size. The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.
[Purpose: to limit the delivered bitrate to applications or external networks with such limitations]
Guaranteed bitrate [kbps]
Definition: guaranteed number of bits delivered at a SAP within a period of time (provided that there is data to deliver), divided by the duration of the period. The traffic is conformant with the Guaranteed bitrate as

long as it follows a token bucket algorithm where token rate equals Guaranteed bitrate and bucket size equals k Maximum SDU size. For Release 99, k = 1. A value of k greater than one Maximum SDU size may be specified in future releases to capture burstiness of sources. Signalling to specify the value of k may be provided in future releases.
The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.
[Purpose: Guaranteed bitrate may be used to facilitate admission control based on available resources, and for resource allocation within UTRAN. Quality requirements expressed by e.g. delay and reliability attributes only apply to incoming traffic up to the guaranteed bitrate. The guaranteed bitrate at the RAB level may be different from that on UMTS bearer level, for example due to header compression.]
Delivery order [y/n]
Definition: indicates whether the UMTS bearer shall provide in-sequence SDU delivery or not.
[Purpose: specifies if out-of-sequence SDUs are acceptable or not. This information cannot be extracted from the traffic class. Whether out-of-sequence SDUs are dropped or re-ordered depends on the specified reliability]
Maximum SDU size [bits]
Definition: the maximum allowed SDU size
[Purpose: The maximum SDU size is used for admission control and policing.]
SDU format information [bits]
Definition: list of possible exact sizes of SDUs. If unequal error protection shall be used by a Radio Access Bearer service, SDU format information defines the exact subflow format of the SDU payload.
Note: SDU format information is used by UTRAN to define which bits of the payload that belongs to each subflow. Exact syntax of SDU format information attribute is the task of RAN WG3

[Purpose: UTRAN needs SDU format information to be able to operate in transparent RLC protocol mode, which is beneficial to spectral efficiency and delay when RLC re-transmission is not used. Thus, if the application can specify SDU sizes, the bearer is less expensive. Moreover, in case of unequal error protection, UTRAN needs to know the exact format of SDU payload to be able to demultiplex the SDU onto different radio bearer services.]
SDU error ratio
Definition: Indicates the fraction of SDUs lost or detected as erroneous. SDU error ratio is defined only for conforming traffic. In case of unequal error protection., SDU error ratio is set per subflow and represents the error ratio in each subflow. SDU error ratio is only set for subflows for which error detection is requested.
Note: By reserving resources, SDU error ratio performance is independent of the loading conditions, whereas without reserved resources, such as in Interactive and Background classes, SDU error ratio is used as target value.

[Purpose: Used to configure ~~the retransmission~~ protocol<u>s, algorithms</u> ~~on layer 2~~ and ~~the~~ error detection <u>schemes, primarily within UTRAN.</u>~~coding on layer 1~~]
Residual bit error ratio
Definition: Indicates the undetected bit error ratio for each subflow in the delivered SDUs. For equal error protection, only one value is needed. If no error detection is requested for a subflow, Residual bit error ratio indicates the bit error ratio in that subflow of the delivered SDUs.
[Purpose: Used to configure <u>radio interface protocols, algorithms</u> ~~channel coding~~ and error detection coding ~~on layer 1~~. For services requiring unequal error protection, residual bit error ratio is given for each subflow.]
Delivery of erroneous SDUs (y/n/-)
Definition: Indicates whether SDUs with detected errors shall be delivered or not. In case of unequal error protection, the attribute is set per subflow.
Note: ‘yes’ implies that error detection is employed and that erroneous SDUs are delivered together with an error indication, ‘no’ implies that error detection is employed and that erroneous SDUs are discarded, and ‘-‘ implies that SDUs are delivered without considering error detection.
In case of unequal protection, different subflows may have different settings. Whenever there is a detected error in a subflow with ‘no’, the SDU is discarded, irrespective of settings in other subflows. For an SDU with multiple subflows with a ‘yes’ setting, there may be one error indication per subflow, or, if there is only one error indication per SDU, it indicates that an error was detected in at least one of these subflows. Exact definitions are the task of RAN3.
[Purpose: Used to decide whether <u>error detection is needed and whether</u> frames with <u>detected errors</u> ~~failed CRC on layer 1~~ shall be forwarded or discarded.]
Transfer delay [s]
Definition: time between request to transfer an SDU at one SAP to its delivery at the other SAP. Transfer

delay is specified for one or more fixed SDU sizes. Exact statistical transfer delay definition and fixed SDU sizes are FFS.
[Purpose: specifies the UTRAN part of the total transfer delay for the UMTS bearer. It allows UTRAN to set transport formats and ARQ parameters.]
Traffic handling priority
Definition: specifies the relative importance for handling of all SDUs belonging to the radio access bearer compared to the SDUs of other bearers.
[Purpose: Within the interactive class, there is a definite need to differentiate between bearer qualities. This is handled by using the traffic handling priority attribute, to allow UTRAN to schedule traffic accordingly. By definition, priority is an alternative to absolute guarantees, and thus these two attribute types cannot be used together for a single bearer.]
Allocation/Retention Priority
Definition: specifies the relative importance compared to other Radio access bearers for allocation and retention of the Radio access bearer.
[Purpose: Priority is used for differentiating between bearers when performing allocation and retention of a bearer, and the value is typically related to the subscription.
Source statistics descriptor ['speech'/'unknown']
Definition: specifies characteristics of the source of submitted SDUs.
[Purpose: Conversational speech has a well-known statistical behaviour (or the discontinuous transmission (DTX) factor). By being informed that the SDUs for a RAB are generated by a speech source, UTRAN may, based on experience, calculate a statistical multiplex gain for use in admission control on the radio and Iu interfaces.]

6.4.4.2 Attributes discussed per class

Conversational class
If the RAB carries a speech service, **Source statistics descriptor** can be set, which allows UTRAN to calculate a statistical multiplexing gain on radio and Iu interfaces and use that for admission control.
Unequal error protection can be supported in conversational class. In case unequal error protection is requested for a given RAB, the attributes Delivery of erroneous SDUs, Residual bit error ratio and SDU error ratio are specified per subflow**. Delivery of erroneous SDUs** determines whether error detection shall be used and, if so, whether SDUs with error in a certain subflow shall be delivered or not. **Residual bit error ratio** specifies the bit error ratio for undetected delivered bits. **SDU error ratio** specifies the fraction of SDUs with detected error in each subflow. It is only set for subflows for which error detection is requested.

In case of unequal error protection the payload of the user data SDU, transported by the Radio Access Bearer Service, must conform to a SDU format defined with possible exact sizes. The payload bits are statically structured into subflows. The **SDU format information** attribute defines the exact subflow format of SDU payload.
UTRAN includes a rate control protocol, making it able of controling the rate of sources requesting this, provided that they are periodic and that **SDU format information** is specified. UTRAN is allowed to control the rate between **Guaranteed bitrate** and **Maximum bitrate**. Each of these two rates must correspond to an SDU format specified in **SDU format information**.
Streaming class
If the RAB carries streaming speech, **Source statistics descriptor** can be set, which allows UTRAN to calculate a statistical multiplexing gain on radio and Iu interfaces and use that for admission control.
Unequal error protection can be supported in streaming class. In case unequal error protection is requested for a given RAB, the attributes Delivery of erroneous SDUs, Residual bit error ratio and SDU error ratio are specified per subflow**. Delivery of erroneous SDUs** determines whether error detection shall be used and, if so, whether SDUs with error in a certain subflow shall be delivered or not. **Residual bit error ratio** specifies the bit error ratio for undetected delivered bits. **SDU error ratio** specifies the fraction of SDUs with detected error in each subflow. It is only set for subflows for which error detection is requested.
In case of unequal error protection the payload of the user data SDU, transported by the Radio Access Bearer Service, must conform to a SDU format defined with possible exact sizes. The payload bits are statically structured into subflows. The **SDU format information** attribute defines the exact subflow format of SDU payload.
UTRAN includes a rate control protocol, making it able of controling the rate of sources requesting this, provided that they are periodic and that **SDU format information** is specified. UTRAN is allowed to control the rate between **Guaranteed bitrate** and **Maximum bitrate**. Each of these two rates must correspond to an SDU format specified in **SDU format information**.
Other classes

The RAB attribute sets and their use in, interactive and background classes are identical to those of UMTS bearer services (Section 6.4.2.2).

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***Document* S2-99E90**
e.g. for 3GPP use the format TP-99xxx
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23.107	CR	005 draft	Current Version:	3.0.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to: SA#6	for approval	X	strategic		(for SMG use only)
list expected approval meeting # here ↑	for information		non-strategic		

Form: CR cover sheet, version 2 for 3GPP and SMG *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects: *(at least one should be marked with an X)* (U)SIM [ ] ME [X] UTRAN / Radio [X] Core Network [X]

Source: Ericsson **Date:** 1999-11-26

Subject: Clarification on parameter value ranges

Work item: End-to-end QoS

Category: *(only one category shall be marked with an X)*

F	Correction	X
A	Corresponds to a correction in an earlier release	
B	Addition of feature	
C	Functional modification of feature	
D	Editorial modification	

Release:

Phase 2	
Release 96	
Release 97	
Release 98	
Release 99	X
Release 00	

Reason for change: The interpretation of the values and value ranges in the tables of chapter 6.5 need to be clarified. Editorial modifications in chapter 6.5.

Clauses affected: 6.5

Other specs affected:

Other 3G core specifications		→ List of CRs:
Other GSM core specifications		→ List of CRs:
MS test specifications		→ List of CRs:
BSS test specifications		→ List of CRs:
O&M specifications		→ List of CRs:

Other comments:


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6.5 Parameter Value Ranges

For UMTS Bearer service and Radio Access Bearer services a list of finite attribute values or the allowed value range is defined for each attribute. The value list/value range define the values that are possible to be used for an attribute considering every possible service condition for release 1999. When a service is defined as a combination of attributes, further limitations may apply; for example the shortest possible delay may not be possible to use together with the lowest possible SDU error ratio. Service requirements, i.e. required QoS and performance for a given UMTS service is defined in the service requirement specifications (22.1xx). The aspect of future proof coding (beyond release 1999) of attributes in protocol specifications is not considered in the defined value list/value range tables.

6.5.1 Ranges of UMTS Bearer Service Attributes

The following table lists the value ranges of the UMTS bearer service attributes. The value ranges reflect the capability of UMTS network.

Table 4: Value ranges for UMTS Bearer Service Attributes

Traffic class	Conversational class	Streaming class	Interactive class	Background clas
Maximum bitrate [kbps]	<2000 (1) (2)	<2000 (1) (2)	< 2000 - overhead (2) (3)	<2000 - overhead (2) (3)
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size [octets]	<1500 (4)	<1500 (4)	<1500 (4)	<1500 (4)
SDU format information	(5)	(5)		
Delivery of erroneous SDUs	Yes/No/- (6)	Yes/No/- (6)	Yes/No/- (6)	Yes/No/- (6)
Residual BER	$5*10^{-2}$, 10^{-2}, 10^{-3}, 10^{-4} (7)	$5*10^{-2}$, 10^{-2}, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6} (7)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (8) (7)	$4*10^{-3}$, 10^{-5}, $6*10^{-}$ (8) (7)
SDU error ratio	10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} (7)	10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} (7)	10^{-3}, 10^{-4}, 10^{-6} (7)	10^{-3}, 10^{-4}, 10^{-6} (7)
Transfer delay [ms]	100 – maximum value(7)	500 – maximum value (7)		
Guaranteed bit rate [kbps]	<2000 (1) (2)	<2000 (1) (2)		
Traffic handling priority			1,2,3 (9)	
Allocation/Retention priority	1,2,3 (9)	1,2,3 (9)	1,2,3 (9)	1,2,3 (9)

1) Bitrate of 2000 kbps requires that UTRAN operates in transparent RLC protocol mode, in this case the overhead from layer 2 protocols is negligible.

2) The granularity of the bit rate parameters must be studied. Although the UMTS network has capability to support a large number of different bitrate values, the number of possible values must be limited not to unnecessarily increase the complexity of for example terminals, charging and interworking functions. Exact list of supported values shall be defined together with S1, N1, N3 and R2.

3) Impact from layer 2 protocols on maximum bitrate in non-transparent RLC protocol mode shall be estimated.

4) Maximum SDU size shall at least allow UMTS network to support external PDUs having as high MTU as Internet/Ethernet (1500 octets). The need for higher values must be investigated by N1, N3, S1, R2, R3.

5) Definition of possible values of exact SDU sizes for which UTRAN can support transparent RLC protocol mode, is the task of RAN WG3.

6) If *Delivery of erroneous SDUs* is set to 'Yes' error indications can only be provided on the MT/TE side of the UMTS bearer. On the CN Gateway side error indications can not be signalled outside of UMTS network in release 1999.

7) Values are indicative. Exact values on Residual BER, SDU error ratio and transfer delay shall defined together with S1, N1, N3 and R2.

8) Values are derived from CRC lengths of 8, 16 and 24 bits on layer 1.

9) Number of priority levels shall be further analysed by S1, N1 and N3.

6.5.2 Ranges of Radio Access Bearer Service Attributes

The following table lists the value ranges of the radio access bearer service attributes. The value ranges reflect the capability of UTRAN.

Table 5: Value ranges for Radio Access Bearer Service Attributes

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate [kbps]	<2000 (1) (2)	<2000 (1) (2)	< 2000 - overhead (2) (3)	<2000 - overhead (2) (3)
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size [octets]	<1500 (4)	<1500 (4)	<1500 (4)	<1500 (4)
SDU format information	(5)	(5)		
Delivery of erroneous SDUs	Yes/No/-	Yes/No/-	Yes/No/-	Yes/No/-
Residual BER	$5*10^{-2}$, 10^{-2}, 10^{-3}, 10^{-4} (6)	$5*10^{-2}$, 10^{-2}, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6} (6)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (6) (7)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (6) (7)
SDU error ratio	10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} (6)	10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} (6)	10^{-3}, 10^{-4}, 10^{-6} (6)	10^{-3}, 10^{-4}, 10^{-6} (6)
Transfer delay [ms]	80 – maximum value(6)	500 – maximum value (6)		
Guaranteed bit rate [kbps]	<2000 (1) (2)	<2000 (1) (2)		
Traffic handling priority			1,2,3 (8)	
Allocation/Retention priority	1,2,3 (8)	1,2,3 (8)	1,2,3 (8)	1,2,3 (8)
Source statistic descriptor	Speech/unknown	Speech/unknown	~~Speech/unknown~~	~~Speech/unknown~~

1) Bitrate of 2000 kbps requires that UTRAN operates in transparent RLC protocol mode, in this case the overhead from layer 2 protocols is negligible.

2) The granularity of the bit rate parameters must be studied. Although the UMTS network has capability to support a large number of different bitrate values, the number of possible values must be limited not to unnecessarily increase the complexity of for example terminals, charging and interworking functions. Exact list of supported values shall be defined together with S1, N1, N3 and R2.

3) Impact from layer 2 protocols on maximum bitrate in non-transparent RLC protocol mode shall be estimated.

4) Maximum SDU size shall at least allow UMTS network to support external PDUs having as high MTU as Internet/Ethernet (1500 octets). The need for higher values must be investigated by N1, N3, S1, R2, R3.

5) Definition of possible values of exact SDU sizes for which UTRAN can support transparent RLC protocol mode, is the task of RAN WG3.

6) Values are indicative. Exact values on Residual BER, SDU error ratio and transfer delay shall defined together with S1, N1, N3 and R2.

7) Values are derived from CRC lengths of 8, 16 and 24 bits on layer 1.

8) Number of priority levels shall be further analysed by S1, N1 and N3.

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or for SMG, use the format P-99-xxx

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23.107	CR	006 Draft	Current Version:	3.0.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	SA#6	for approval	X	strategic		(for SMG use only)
list expected approval meeting # here ↑		for information		non-strategic		

Form: CR cover sheet, version 2 for 3GPP and SMG *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects: *(at least one should be marked with an X)* (U)SIM [ ] ME [X] UTRAN / Radio [X] Core Network [X]

Source: Ericsson **Date:** 1999-11-26

Subject: Clarification of Maximum bit rate attribute

Work item: End-to-end QoS

Category: *(only one category shall be marked with an X)*

F	Correction		**Release:**	Phase 2	
A	Corresponds to a correction in an earlier release			Release 96	
B	Addition of feature			Release 97	
C	Functional modification of feature	X		Release 98	
D	Editorial modification			Release 99	X
				Release 00	

Reason for change: The use of Maximum bit rate as the users wanted maximum bit rate is added as purpose of the parameter. This facilitates mapping to non transparent CS data, as *wanted air interface user rate* is a GSM parameter for non transparent CS data.

Clauses affected: 6.4.3.1 and 6.4.4.1

Other specs affected:

Other 3G core specifications		→ List of CRs:
Other GSM core specifications		→ List of CRs:
MS test specifications		→ List of CRs:
BSS test specifications		→ List of CRs:
O&M specifications		→ List of CRs:

Other comments:


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6.4.3 UMTS Bearer Service Attributes

6.4.3.1 List of attributes

Note: The text within square brackets explaining the purpose of each attribute can be excluded later if that information is given elsewhere in the technical report.

Traffic class [‘conversational’, ’streaming’, ’interactive’, ’background’]

Definition: type of application for which the UMTS bearer service is optimised

[Purpose: By including the traffic class itself as an attribute, UMTS can make assumptions about the traffic source and optimise the transport for that traffic type.]

Maximum bitrate [kbps]

Definition: maximum number of bits delivered by UMTS and to UMTS at a SAP within a period of time, divided by the duration of the period. The traffic is conformant with Maximum bitrate as long as it follows a token bucket algorithm where token rate equals Maximum bitrate and bucket size equals Maximum SDU size.

The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.

[Purpose: Maximum bitrate can be used to make code reservations in the downlink of the radio interface. Its purpose is 1) to limit the delivered bitrate to applications or external networks with such limitations 2) to allow maximum wanted user bitrate to be defined for applications able to operate with different rates (e.g. non transparent circuit switched data)]

6.4.4 Radio Access Bearer Service Attributes

Radio Access Bearer Service Attributes shall be applied to both CS and PS domains.

6.4.4.1 List of attributes

Note: The text within square brackets explaining the purpose of each attribute can be excluded later if that information is given elsewhere in the technical report.

Traffic class [‘conversational’, ’streaming’, ’interactive’, ’background’]

Definition: type of application for which the Radio Access Bearer service is optimised

[Purpose: By including the traffic class itself as an attribute, UTRAN can make assumptions about the traffic source and optimise the transport for that traffic type. In particular, buffer allocation may be based on traffic class.]

Maximum bitrate [kbps]

Definition: maximum number of bits delivered by UTRAN and to UTRAN at a SAP within a period of time, divided by the duration of the period. The traffic is conformant with the Maximum bitrate as long as it follows a token bucket algorithm where token rate equals Maximum bitrate and bucket size equals Maximum SDU size.

The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.

[Purpose: 1) to limit the delivered bitrate to applications or external networks with such limitations, 2) to allow maximum wanted RAB bitrate to be defined for applications able to operate with different rates (e.g. non transparent circuit switched data)]

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3G CHANGE REQUEST

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23.107	CR	001r3	Current Version:	3.0.0

3G specification number ↑ *↑ CR number as allocated by 3G support team*

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for approval [X] for information [ ] *(only one box should be marked with an X)*

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Proposed change affects: *(at least one should be marked with an X)* USIM [ ] ME [X] UTRAN [X] Core Network [X]

Source: Ericsson **Date:** Nov 30, 1999

Subject: Mapping of QoS profiles between R97/98 and R99

3G Work item:

Category: *(only one category shall be marked with an X)*

F	Correction	[ ]	**Release:**	Phase 2
A	Corresponds to a correction in a 2G specification	[ ]		Release 96
B	Addition of feature	[ ]		Release 97
C	Functional modification of feature	[X]		Release 98
D	Editorial modification	[ ]		Release 99 X
		[ ]		Release 00

Reason for change: The section 9.1.2 on UMTS-GPRS interworking has an editors not that the section needs to be updated concerning the mapping between QoS profiles of R97/98 and R99. This CR presents the necessary mapping rules to provide interoperation.

Clauses affected: 9.1.2

Other specs affected:

Other 3G core specifications	[ ]	→ List of CRs:
Other 2G core specifications	[ ]	→ List of CRs:
MS test specifications	[ ]	→ List of CRs:
BSS test specifications	[ ]	→ List of CRs:
O&M specifications	[ ]	→ List of CRs:

Other comments:

9.1.2 UMTS-GPRS

~~Note: Part of GPRS phase 1 QoS are vaguely defined. This chapter has to be updated according to CRs to GPRS phase 1 QoS parameters. Definition of GPRS phase 2 is starting and it has to be taken into account in here and vice versa.~~

~~GPRS has more QoS parameters than GSM CS thus requiring more complex mapping rules. Below an example of mapping GPRS phase 1 QoS parameters to UMTS traffic classes is presented.~~

~~**Conversational Class**~~

~~Conversational class services are mainly for conversational real time use. An example of conversational real time application is video telephony.~~

~~An appropriate use of GPRS parameters:~~

- ~~Mean Throughput Class = Peak bit rate (constant bit rate)~~

~~Or Mean Throughput Class < Peak bit rate (variable bit rate)~~

- ~~Reliability: 4 or 5 (no retransmissions)~~
- ~~Precedence:1-3~~
- ~~Delay Class: 1 (real time)~~

~~Streaming Class~~

~~Streaming class services are mainly appropriate for streaming real time applications, e.g. video downloading. Some variation in delay can be tolerated because of application level buffering.~~

~~An appropriate use of GPRS parameters:~~

- ~~Mean Throughput Class = Peak bit rate (constant bit rate)~~

~~Or Mean Throughput Class < Peak bit rate (variable bit rate)~~

- ~~Reliability: 3 (light retransmissions)~~
- ~~Precedence: 1-3~~
- ~~Delay Class: 1 (real time)~~

~~Interactive Class~~

~~Interactive class services are mainly for interactive services requiring a variable guaranteed throughput: specialised applications (banking, plane reservation, …), interactive WWW, Telnet etc.~~

~~An appropriate use of GPRS parameters:~~

- ~~Mean Throughput Class has no meaning in UMTS~~
- ~~Reliability: 1-2~~
- ~~Precedence: 1-3~~
- ~~Delay class: 2-4~~

~~Background Class~~

~~Background services are mainly for best effort services: background download, emails, calendar, event etc.~~

~~An appropriate use of GPRS parameters:~~

- ~~Mean Throughput Class has no meaning in UMTS~~
- ~~Reliability: 2~~
- ~~Precedence: 1-3~~
- ~~Delay class: 4 (best effort)~~

This section covers primarily the mapping of QoS attributes that are necessary across standardised interfaces. In addition to these, there are cases when mapping of QoS attributes are needed internal to a node.

GPRS Release 99 (R99) QoS attributes shall be equivalent to the UMTS QoS Attributes. For interworking purposes between different releases, mapping rules between GPRS Release 97/98 (R97/98) and GPRS Release 99 (R99) as well as UMTS are defined. Mapping shall occur whenever the MS, the SGSN, the GGSN and the HLR nodes are of different releases R97/98 or R99. The mapping is required in PDP context

activation and modification procedures and when a R99 HLR Insert Subscriber Data towards a R97/98 SGSN.

It is not within the scope of this document to determine if any value combinations of attribute values can not be supported. This means that complete mapping rules are defined here, and if the user requests a QoS profile which the network is not able to support (e.g. a low delay and a high reliability), the decision if such a parameter combination can be supported is left to admission control functionality within the PDP context activation procedure, and the QoS for such a profile may be renegotiated by the network based on the available resources.

The overall principle for the mapping between two profiles is that the two profiles, applied in their respective network releases, give the same or at least similar QoS. The GPRS R97/98 equipment will not be able to provide realtime service corresponding to the R99 conversational and streaming traffic classes. Therefore, the mapping is always to the non-realtime interactive and background traffic classes.

9.1.2.1 General rules

Air interface Session Management and GTP messages of R99 shall contain the R99 attributes as an extension of the R97/98 QoS Information Element thus unnecessary mapping can be avoided. When a R97/98 MS is visiting a GPRS R99 or UMTS SGSN and the GGSN is of R97/98 or R99, the visited SGSN shall not perform any mapping of QoS attributes. In case of GGSN R99, the GTP version 1 (R99) QoS profile only contains the R97/98 QoS attributes. It can be noted that for this PDP Context a Traffic Flow Template (TFT) can not be requested.

When a R99 MS is visiting a GPRS R99 or UMTS SGSN (or serving PLMN) and the GGSN (or home PLMN) is of R97/98, the visited SGSN (or visited PLMN) must be capable of providing bearers having QoS support according to R99. When a PDP Context is activated (mobile or network initiated) mapping takes place in the serving SGSN.

For MS initiated PDP Context Activations as well as network initiated PDP Context Activations, the home R97/98 GGSN will respond to the activation request by returning a the QoS Negotiated Profile, which contain the accepted and changed R97/98 attributes. A mapping of the changed attributes into R99 attributes will be done in serving SGSN and signalled to the mobile station in the Activate PDP Context Accept message.

It is a general mapping rule that returned and unchanged attributes during negotiation procedures shall not be mapped a second time by serving SGSN, i.e. the unchanged R99 attributes received in the Create PDP Context Response message will be sent to MS in QoS Negotiated Profile of the Activate PDP Context Accept message.

MAP message of R99 shall also contain the R99 attributes as an extension of the R97/98 QoS Information Element when Insert Subscriber Data message is sent to a R99 SGSN. In the case when a R99 HLR send a Insert Subscriber Data message to a R97/98 SGSN, the message shall contain the R97/98 QoS attributes. A R99 SGSN shall use the R99 attributes of subscribed QoS profile when a R99 MS requests to use subscription data in the PDP Context Activation. The R99 SGSN shall use the R97/98 attributes of subscribed QoS profile when a R97/98 MS requests to use subscription data in the PDP Context Activation.

9.1.2.2 Determining R99 attributes from R97/98 attributes

This mapping is applicable in the following cases:

- Hand over of PDP Context from GPRS R97/98 SGSN to GPRS R99 or UMTS SGSN.
- PDP Context Activation in a serving R99 SGSN with a R97/98 GGSN. When GGSN respond to the PDP Context Activation, mapping of the changed R97/98 QoS attributes received from the GGSN to R99 QoS attributes is performed in the serving SGSN.

Resulting R99 Attribute		**Derived from R97/98 Attribute**	
Name	**Value**	**Value**	**Name**
Traffic class	Interactive	1, 2, 3	Delay class
	Background	4	
Traffic handling priority	1	1	Delay class
	2	2	
	3	3	
SDU error ratio	10^{-6}	1, 2	Reliability class
	10^{-4}	3	
	10^{-3}	4, 5	
Residual bit error ratio	10^{-5}	1, 2, 3, 4	Reliability class
	$4*10^{-3}$	5	
Delivery of erroneous SDUs	'no'	1, 2, 3, 4	Reliability class
	'yes'	5	
Maximum bitrate [kbps]	8	1	Peak throughput class
	16	2	
	32	3	
	64	4	
	128	5	
	256	6	
	512	7	
	1024	8	
	2048	9	
Allocation/Retention priority	1	1	Precedence class
	2	2	
	3	3	
Delivery order	yes'	yes'	Reordering Required (Information in the SGSN and the GGSN PDP Contexts)
	'no'	'no'	
Maximum SDU size	1500 octets	(Fixed value)	

Table 1. Rules for determining R99 attributes from R97/98 attributes.

9.1.2.3 Determining R97/98 attributes from R99 attributes

This mapping is applicable in the following cases:

- PDP Context is handed over from GPRS R99 or UMTS to GPRS R97/98.
- When a R99 MS perform a PDP Context Activation in a serving R99 SGSN while the GGSN is of R97/98. In this case the SGSN shall perform mapping of the R99 QoS attributes to the R97/98 QoS attributes.
- A R99 HLR may need to map the stored subscribed QoS attributes in the HLR subscriber data to R97/98 QoS attributes that are going to be sent in the Insert Subscriber Data message from the R99 HLR to the R97/98 and R99 SGSN. It is an implementation issue if the R97/98 QoS attributes are stored in the HLR in addition to the R99 QoS attributes.

Resulting R97/98 Attribute		**Derived from R99 Attribute**	
Name	**Value**	**Value**	**Name**
Delay class	1	conversational	Traffic class
	1	streaming	Traffic class

	1	Interactive	Traffic class
		1	Traffic handling priority
	2	Interactive	Traffic class
		2	Traffic handling priority
	3	Interactive	Traffic class
		3	Traffic handling priority
	4	Background	Traffic class
Reliability class	2	<= 10^{-5}	SDU error ratio
	3	10^{-5} < x <= $5*10^{-4}$	SDU error ratio
	4	> $5*10^{-4}$	SDU error ratio
		<= $2*10^{-4}$	Residual bit error ratio
	5	> $5*10^{-4}$	SDU error ratio
		> $2*10^{-4}$	Residual bit error ratio
Peak throughput class	1	< 16	Maximum bitrate [kbps]
	2	16 <= x < 32	
	3	32 <= x < 64	
	4	64 <= x < 128	
	5	128 <= x < 25	
	6	256 <= x < 512	
	7	512 <= x < 1024	
	8	1024 <= x < 2048	
	9	>= 2048	
Precedence class	1	1	Allocation/retention priority
	2	2	
	3	3	
Mean throughput class	Always set to 31	-	
Reordering Required (Information in the SGSN and the GGSN PDP Contexts)	yes'	yes'	Delivery order
	'no'	'no'	

Table 2. Rules for determining R97/98 attributes from R99 attributes.

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Abiko, Japan, 29 Nov - 03 Dec 1999

***Document* S2-99D38**
e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

CHANGE REQUEST

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23.107	CR	008 draft	Current Version:	3.0.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	SA#6	for approval	X	strategic		(for SMG use only)
list expected approval meeting # here ↑		for information		non-strategic		

Form: CR cover sheet, version 2 for 3GPP and SMG *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects: *(at least one should be marked with an X)* (U)SIM [ ] ME [ ] UTRAN / Radio [ ] Core Network [X]

Source: Ericsson **Date:** 1999-11-26

Subject: Generation of QoS parameters for CS data services for call setup and interworking UMTS- CS

Work item: End-to-end QoS

Category: *(only one category shall be marked with an X)*

F	Correction	
A	Corresponds to a correction in an earlier release	
B	Addition of feature	
C	Functional modification of feature	X
D	Editorial modification	

Release:

Phase 2	
Release 96	
Release 97	
Release 98	
Release 99	X
Release 00	

Reason for change: The determination of QoS parameters for CS services for call setup and handover procedure needs to be enhanced.

Clauses affected: 5 and 9.1.1

Other specs affected:

Other 3G core specifications	[ ]	→ List of CRs:
Other GSM core specifications	[ ]	→ List of CRs:
MS test specifications	[ ]	→ List of CRs:
BSS test specifications	[ ]	→ List of CRs:
O&M specifications	[ ]	→ List of CRs:

Other comments:


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<--------- double-click here for help and instructions on how to create a CR.

5 CS QoS in release '99

For UMTS release '99 CS-CC, the QoS related bearer definitions of GSM (as defined in bearer capability information element, octet 6 and its extensions) are sufficient.

Based on the Bearer Capability information element the following services can be identified:

a) speech: from the Information Transfer Capability (ITC) parameter

b) data, non-transparent: from the ITC and Connection element (CE~~, HLC~~) parameters

- among the non-transparent data, **facsimile** is identified by the ITC

c) data, transparent: from the ITC and CE parameters

For each of the above services an appropriate UMTS Bearer service shall be defined. The definition shall include exact UMTS bearer attribute values or list of supported values.

Note: This service mapping is the task of TSG N3 and SA4.

The further mapping to Radio Access Bearer attributes is done according to the principles described in clause 8.

Note: The mapping from GSM CC to UMTS RAB parameters is in the responsibility of CN WG1 and CN WG3.

9 Interworking

The model for the UMTS QoS classes and parameters may not be any existing network or QoS protocol/mechanisms as such. The main goal of the specification is not to copy existing QoS mechanisms but rather to create a future proof concept that will provide means to transport different types of data with different QoS requirements. Thus the interworking of UMTS and existing network technologies has to be ensured. This chapter presents the most common technologies that UMTS shall be capable to interwork with.

9.1 UMTS-GSM CS/GPRS

9.1.1 UMTS-GSM CS

The mapping between UMTS-GSM CS is based on GSM CS mechanisms and CC parameters.

9.1.1.1 Handover from UMTS to GSM

In case a UMTS call is set up in the CN, the BC IE parameters are mapped into QoS RAB parameters at call setup.

If the CN has to perform a handover towards GSM, the non-anchor MSC needs to perform an assignment based on GSM specific traffic channel parameters.

As the BSSMAP protocol is used over the E-interface and as no appropriate procedure exists to map QoS parameters into BSSMAP parameters, the anchor MSC shall map BC IE parameters into GSM traffic channel parameters, according to existing GSM procedures for call setup.
This requires that the BC IE is coded according to GSM protocol requirements, i.e. all those parameters not applicable to UMTS should nevertheless be correctly specified by the UE in order to perform a handover to GSM according the above specified principles.

3GPP TSG SA WG2 Meeting #10
Abiko, Japan, 29.11-3.12. 1999

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e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

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23.107	CR	011	Current Version:	3.0.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	SA#6	for approval	X	strategic		(for SMG use only)
list expected approval meeting # here ↑		for information		non-strategic		

Form: CR cover sheet, version 2 for 3GPP and SMG *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects: *(at least one should be marked with an X)*	(U)SIM		ME	X	UTRAN / Radio	X	Core Network	X

Source: Nokia **Date:** 2.12.1999

Subject: Editorial changes to 23.107

Work item:

Category: *(only one category shall be marked with an X)*			**Release:**	
F	Correction		Phase 2	
A	Corresponds to a correction in an earlier release		Release 96	
B	Addition of feature		Release 97	
C	Functional modification of feature		Release 98	
D	Editorial modification	X	Release 99	X
			Release 00	

Reason for change: Several minor corrections to 23.107 are provided in this CR.

Clauses affected: 3, 4.3, 6.4.3, 6.4.4, 6.5.1, 6.5.2, Annex A, Annex B

Other specs affected:			
Other 3G core specifications		→ List of CRs:	
Other GSM core specifications		→ List of CRs:	
MS test specifications		→ List of CRs:	
BSS test specifications		→ List of CRs:	
O&M specifications		→ List of CRs:	

Other comments:

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<--------- double-click here for help and instructions on how to create a CR.

3 Abbreviations

For the purpose of this document the following abbreviations apply.

3G	3rd Generation
AMR	Adaptive Multirate speech codec
ATM	Asynchronous Transfer Mode
BER	Bit Error Rate
BS	Bearer Service
CC	Call Control
CN	Core Network
CRC	Cyclic Redundancy Check
CS	Circuit Switched
DTX	Discontinuous Transmission
FDD	Frequency Division Duplex
FER	Frame Erasure Ratio
FTP	File Transfer Protocol
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communication
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISDN	Integrated Services Digital Network
MO	Mobile Originating Call
MPEG	Moving Pictures Expert Group
~~MS~~	~~Mobile Station~~
MT	Mobile Terminal
MTC	Mobile Terminated Call
NS	Network Service
PDP	Packet Data Protocol
PDU	Protocol Data Unit
PS	Packet Switched
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RA	Routing Area
RAB	Radio Access Bearer
RAN	Radio Access Network
RLC	Radio Link Control
RSVP	Resource Reservation Protocol
RT	Real Time
RTP	Real Time Transport Protocol
SAP	Service Access Point
SDU	Service Data Unit
SGSN	Serving GPRS Support Node
SLA	Service Level Agreement
SMS	Short Message Service
SVC	Switched Virtual Circuit
UDP	User Datagram Protocol
TBC	Token Bucket Counter
TDD	Time Division Duplex
TE	Terminal Equipment
TSPEC	Traffic Specification
UE	User Equipment
UMTS	Universal Mobile Telecommunication System
UTRA	UMTS Terrestrial Radio Access
UTRAN	UMTS Terrestrial Radio Access Network

4.3 Technical Requirements for QoS

This chapter presents the general high-level technical requirements for the UMTS QoS. QoS will be defined with a set of parameters. These parameters should meet the following criteria:

- UMTS QoS control mechanisms shall provide QoS parameter control on a peer to peer basis between ~~MS~~UE and 3G gateway node.
- The UMTS QoS mechanisms shall provide a mapping between application requirements and UMTS services.
- The UMTS QoS control mechanisms shall be able to efficiently interwork with current QoS schemes. Further, the QoS concept should be capable of providing different levels of QoS by using UMTS specific control mechanisms (not related to QoS mechanisms in the external networks).
- A session based approach needs to be adopted for all packet mode communication within the 3G serving node with which UMTS QoS approach must be intimately linked, essential features are multiple QoS streams per address.
- The UMTS shall provide a finite set of QoS definitions.
- The overhead and additional complexity caused by the QoS scheme should be kept reasonably low, as well as the amount of state information transmitted and stored in the network.
- QoS shall support efficient resource utilisation.
- The QoS parameters are needed to support asymmetric bearers.
- Applications (or special software in ~~MS~~UE or 3G gateway node) should be able to indicate QoS values for their data transmissions.
- QoS behaviour should be dynamic , i.e., it shall be possible to modify QoS parameters during an active session.
- Number of parameters should be kept reasonably low (increasing number of parameters, increase system complexity).
- User QoS requirements shall be satisfied by the system, including when change of SGSN within the Core Network occurs.

6.4.3 UMTS Bearer Service Attributes

6.4.3.1 List of attributes

Note: The text within square brackets explaining the purpose of each attribute can be excluded later if that information is given elsewhere in the technical report.

Traffic class ([‘conversational’, ’streaming’, ’interactive’, ’background’])

Definition: type of application for which the UMTS bearer service is optimised

[(Purpose: By including the traffic class itself as an attribute, UMTS can make assumptions about the traffic source and optimise the transport for that traffic type.)]

Maximum bitrate ([kbps])

Definition: maximum number of bits delivered by UMTS at a SAP within a period of time, divided by the duration of the period. The traffic is conformant with Maximum bitrate as long as it follows a token bucket algorithm where token rate equals Maximum bitrate and bucket size equals Maximum SDU size.

The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.

[(Purpose: Maximum bitrate can be used to make code reservations in the downlink of the radio interface. Its purpose is to limit the delivered bitrate to applications or external networks with such limitations)]

Guaranteed bitrate ([kbps])

Definition: guaranteed number of bits delivered by UMTS at a SAP within a period of time (provided that there is data to deliver), divided by the duration of the period. The traffic is conformant with the guaranteed bitrate as long as it follows a token bucket algorithm where token rate equals Guaranteed bitrate and bucket size equals k*Maximum SDU size. For release 99, k=1. A value of k greater than one Maximum SDU size may be specified in future releases to capture burstiness of sources. Signalling to specify the value of k may be provided in future releases.

The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.

[(Purpose: Guaranteed bitrate may be used to facilitate admission control based on available resources, and for resource allocation within UMTS. Quality requirements expressed by e.g. delay and reliability attributes only apply to incoming traffic up to the guaranteed bitrate.)]

Delivery order ([y/n])

Definition: indicates whether the UMTS bearer shall provide in-sequence SDU delivery or not.

[(Purpose: the attribute is derived from the user protocol ([PDP type]) and specifies if out-of-sequence SDUs are acceptable or not. This information cannot be extracted from the traffic class. Whether out-of-sequence SDUs are dropped or re-ordered depends on the specified reliability)]

Maximum SDU size ([bits])

Definition: the maximum allowed SDU size

[(Purpose: The maximum SDU size is used for admission control and policing.)]

SDU format information ([bits])

Definition: list of possible exact sizes of SDUs

[(Purpose: UTRAN needs SDU size information to be able to operate in transparent RLC protocol mode, which is beneficial to spectral efficiency and delay when RLC re-transmission is not used. Thus, if the application can specify SDU sizes, the bearer is less expensive.)]

SDU error ratio

Definition: Indicates the fraction of SDUs lost or detected as erroneous. SDU error ratio is defined only for conforming traffic.

Note that by reserving resources, SDU error ratio performance is independent of the loading conditions, whereas without reserved resources, such as in Interactive and Background classes, SDU error ratio is used as target value.

~~[~~*(Purpose: Used to configure the retransmission protocol on layer 2 and the error detection coding on layer 1.*~~]~~*)*

Residual bit error ratio

Definition: Indicates the undetected bit error ratio in the delivered SDUs. If no error detection is requested, Residual bit error ratio indicates the bit error ratio in the delivered SDUs.

~~[~~*(Purpose: Used to configure channel coding and error detection coding on layer 1.*~~]~~*)*

Delivery of erroneous SDUs (y/n/-)

Definition: Indicates whether SDUs detected as erroneous shall be delivered or discarded.

Note: 'yes' implies that error detection is employed and that erroneous SDUs are delivered together with an error indication, 'no' implies that error detection is employed and that erroneous SDUs are discarded, and '-' implies that SDUs are delivered without considering error detection.

~~[~~*(Purpose: Used to decide whether frames with failed CRC on layer 1 shall be forwarded or not.*~~]~~*)*

Transfer delay (~~[~~s~~]~~)

Definition: time between request to transfer an SDU at one SAP to its delivery at the other SAP. Transfer delay is specified for one or more fixed SDU sizes. Exact statistical transfer delay definition and fixed SDU sizes are FFS.

~~[~~*(Purpose: used to specify the delay tolerated by the application. It allows UTRAN to set transport formats and ARQ parameters.*~~]~~*)*

Note: Transfer delay of an arbitrary SDU is not meaningful for a bursty source, since the last SDUs of a burst may have long delay due to queuing, whereas the meaningful response delay perceived by the user is the delay of the first SDU of the burst.

Traffic handling priority

Definition: specifies the relative importance for handling of all SDUs belonging to the UMTS bearer compared to the SDUs of other bearers.

~~[~~*(Purpose: Within the interactive class, there is a definite need to differentiate between bearer qualities. This is handled by using the traffic handling priority attribute, to allow UMTS to schedule traffic accordingly. By definition, priority is an alternative to absolute guarantees, and thus these two attribute types cannot be used together for a single bearer.*~~]~~*)*

Allocation/Retention Priority

Definition: specifies the relative importance compared to other UMTS bearers for allocation and retention of the UMTS bearer.

~~[~~*(Purpose: Priority is used for differentiating between bearers when performing allocation and retention of a bearer, and the value is typically related to the subscription.*

6.4.3.2 Attributes discussed per class

Conversational class

Although the bitrate of a conversational source codec may vary, conversational traffic is assumed to be relatively non-bursty. **Maximum bitrate** specifies the upper limit of the bitrate with which the UMTS bearer delivers SDUs at the SAPs. The UMTS bearer is not required to transfer traffic exceeding the **Guaranteed bitrate**. Maximum and guaranteed bitrate attributes are used for resource allocation within UMTS. Minimum resource requirement is

determined by guaranteed bitrate (When a conversational source generates less traffic than allocated for the bearer, the unused resources can of course be used by other bearers.)

Since the traffic is non-bursty, it is meaningful to guarantee a **transfer delay** of an arbitrary SDU.

Conversational bearers are likely to be realised in UTRAN without RLC re-transmissions. Hence, UTRAN transport is more efficient and thereby cheaper if RLC PDU size is adapted to UMTS bearer SDU size (RLC transparent mode). This motivates the use of **SDU format information**. The SDU periodicity knowledge needed to operate in RLC transparent mode is obtained through dividing the largest defined SDU format by Maximum bitrate. This must be considered when setting the attribute values in a service request.

The **Maximum SDU size** is only applicable if **SDU format information** is not specified and is used for admission control and policing. If **Maximum SDU size** is specified the SDU size is variable. If **SDU format information** is specified, with one or several possible sizes, each SDU must exactly conform to one of the specified sizes. By using the **SDU error ratio**, **Residual bit error ratio** and **Delivery of erroneous SDUs** attribute, the application requirement on error rate can be specified, as well as whether the application wants UMTS to detect and discard SDUs containing errors and an adequate forward error correction means can be selected.

Streaming class

As for conversational class, streaming traffic is assumed to be rather non-bursty. **Maximum bitrate** specifies the upper limit of the bitrate the UMTS bearer delivers SDUs at the SAPs. The UMTS bearer is not required to transfer traffic exceeding the Guaranteed bitrate. Maximum and guaranteed bitrate attributes are used for resource allocation within UMTS. Minimum resource requirement is determined by guaranteed bitrate. (When a streaming source generates less traffic than allocated for the bearer, the unused resources can of course be used by other bearers.)

Since the traffic is non-bursty, it is meaningful to guarantee a **transfer delay** of an arbitrary SDU.

The transfer delay requirements for streaming are typically in a range where at least in a part of this range RLC re-transmission may be used. It is assumed that the application's requirement on delay variation is expressed through the transfer delay attribute, which implies that there is no need for an explicit delay variation attribute.

It shall be possible for Streaming bearers to be realised in UTRAN without RLC re-transmissions. Hence, UTRAN transport is more efficient and thereby cheaper if RLC PDU size is adapted to UMTS bearer SDU size (RLC transparent mode). This motivates the use of **SDU format information**. The SDU periodicity knowledge needed to operate in RLC transparent mode is obtained through dividing the largest defined SDU format by Maximum bitrate. This must be considered when setting the attribute values in a service request.

The **Maximum SDU size** is only applicable if **SDU format information** is not specified and is used for admission control and policing. If **Maximum SDU size** is specified the SDU size is variable. If **SDU format information** is specified, with one or several possible sizes, each SDU must exactly conform to one of the specified sizes.

By using the **SDU error ratio**, **Residual bit error ratio** and **Delivery of erroneous SDUs** attribute, the application requirement on error rate can be specified, as well as whether the application wants UMTS to detect and discard SDUs containing errors.

Interactive class

This bearer class is optimised for transport of human or machine interaction with remote equipment, such as web browsing. The source characteristics are unknown but may be bursty.

To be able to limit the delivered data rate for applications and external networks by traffic conditioning, **maximum bitrate** is included.

There is a definite need to differentiate between quality for bearers within the interactive class. One alternative would be to set absolute guarantees on delay, bitrate etc, which however at present seems complex to implement within UTRAN/CN. Instead, **traffic handling priority** is used. SDUs of a UMTS bearer with higher traffic handling priority is given priority over SDUs of other bearers within the interactive class, through UMTS-internal scheduling.

It is principally impossible to combine this relative approach with attributes specifying delay, bitrate, packet loss etc, so an interactive bearer gives no quality guarantees, and the actual bearer quality will depend on the load of the system and the admission control policy of the network operator.

The only additional attribute that is reasonable to specify is the bit integrity of the delivered data, which is given by **SDU error ratio**, **Residual bit error ratio** and **Delivery of erroneous SDUs** . Because there are no reserved resources for interactive class, SDU error ratio should be used as a target value. SDU error ratio cannot be guaranteed under abnormal load conditions.

Background class

The background class is optimised for machine-to-machine communication that is not delay sensitive, such as messaging services. Background applications tolerate a higher delay than applications using the interactive class, which is the main difference between the background and interactive classes.

UMTS only transfers background class SDUs when there is definite spare capacity in the network. To be able to limit the delivered data rate for applications and external networks by traffic conditioning, **maximum bitrate** is included.

No other guarantee than bit integrity in the delivered data, given by **SDU error ratio**, **Residual bit error ratio** and **Delivery of erroneous SDUs** , is needed. Because there are no reserved resources for background class, SDU error ratio should be used as a target value. SDU error ratio cannot be guaranteed under abnormal load conditions.

6.4.3.3 UMTS bearer attributes: summary

In Table 2, the defined UMTS bearer attributes and their relevancy for each bearer class are summarised. Observe that traffic class is an attribute itself.

Table 2. UMTS bearer attributes defined for each bearer class.

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate	X	X	X	X
Delivery order	X	X	X	X
Maximum SDU size	X	X	X	X
SDU format information	X	X		
SDU error ratio	X	X	X	X
Residual bit error ratio	X	X	X	X
Delivery of erroneous SDUs	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/Retention priority	X	X	X	X

6.4.4 Radio Access Bearer Service Attributes

Radio Access Bearer Service Attributes shall be applied to both CS and PS domains.

6.4.4.1 List of attributes

Note: The text within square brackets explaining the purpose of each attribute can be excluded later if that information is given elsewhere in the technical report.

Traffic class ~~[~~‘conversational’, ’streaming’, ’interactive’, ’background’~~]~~

Definition: type of application for which the Radio Access Bearer service is optimised

~~[~~*(Purpose: By including the traffic class itself as an attribute, UTRAN can make assumptions about the traffic source and optimise the transport for that traffic type. In particular, buffer allocation may be based on traffic class.~~]~~*)*

Maximum bitrate ~~[~~kbps~~]~~

Definition: maximum number of bits delivered by UTRAN at a SAP within a period of time, divided by the duration of the period. The traffic is conformant with the Maximum bitrate as long as it follows a token bucket algorithm where token rate equals Maximum bitrate and bucket size equals Maximum SDU size.

The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.

[(Purpose: to limit the delivered bitrate to applications or external networks with such limitations])

Guaranteed bitrate ([kbps])

Definition: guaranteed number of bits delivered at a SAP within a period of time (provided that there is data to deliver), divided by the duration of the period. The traffic is conformant with the Guaranteed bitrate as long as it follows a token bucket algorithm where token rate equals Guaranteed bitrate and bucket size equals k Maximum SDU size. For Release 99, k = 1. A value of k greater than one Maximum SDU size may be specified in future releases to capture burstiness of sources. Signalling to specify the value of k may be provided in future releases.

The conformance definition should not be interpreted as a required implementation algorithm. The token bucket algorithm is described in Annex B.

[(Purpose: Guaranteed bitrate may be used to facilitate admission control based on available resources, and for resource allocation within UTRAN. Quality requirements expressed by e.g. delay and reliability attributes only apply to incoming traffic up to the guaranteed bitrate. The guaranteed bitrate at the RAB level may be different from that on UMTS bearer level, for example due to header compression.])

Delivery order ([y/n])

Definition: indicates whether the UMTS bearer shall provide in-sequence SDU delivery or not.

[(Purpose: specifies if out-of-sequence SDUs are acceptable or not. This information cannot be extracted from the traffic class. Whether out-of-sequence SDUs are dropped or re-ordered depends on the specified reliability])

Maximum SDU size ([bits])

Definition: the maximum allowed SDU size

[(Purpose: The maximum SDU size is used for admission control and policing.])

SDU format information ([bits])

Definition: list of possible exact sizes of SDUs. If unequal error protection shall be used by a Radio Access Bearer service, SDU format information defines the exact subflow format of the SDU payload.

Note: SDU format information is used by UTRAN to define which bits of the payload that belongs to each subflow. Exact syntax of SDU format information attribute is the task of RAN WG3

[(Purpose: UTRAN needs SDU format information to be able to operate in transparent RLC protocol mode, which is beneficial to spectral efficiency and delay when RLC re-transmission is not used. Thus, if the application can specify SDU sizes, the bearer is less expensive. Moreover, in case of unequal error protection, UTRAN needs to know the exact format of SDU payload to be able to demultiplex the SDU onto different radio bearer services.])

SDU error ratio

Definition: Indicates the fraction of SDUs lost or detected as erroneous. SDU error ratio is defined only for conforming traffic. In case of unequal error protection., SDU error ratio is set per subflow and represents the error ratio in each subflow. SDU error ratio is only set for subflows for which error detection is requested.

Note: By reserving resources, SDU error ratio performance is independent of the loading conditions, whereas without reserved resources, such as in Interactive and Background classes, SDU error ratio is used as target value.

[(Purpose: Used to configure the retransmission protocol on layer 2 and the error detection coding on layer 1])

Residual bit error ratio

Definition: Indicates the undetected bit error ratio for each subflow in the delivered SDUs. For equal error protection, only one value is needed. If no error detection is requested for a subflow, Residual bit error ratio indicates the bit error ratio in that subflow of the delivered SDUs.

~~[~~*(Purpose: Used to configure channel coding and error detection coding on layer 1. For services requiring unequal error protection, residual bit error ratio is given for each subflow.*~~]~~*)*

Delivery of erroneous SDUs (y/n/-)

Definition: Indicates whether SDUs with detected errors shall be delivered or not. In case of unequal error protection, the attribute is set per subflow.

Note: 'yes' implies that error detection is employed and that erroneous SDUs are delivered together with an error indication, 'no' implies that error detection is employed and that erroneous SDUs are discarded, and '-' implies that SDUs are delivered without considering error detection.

In case of unequal protection, different subflows may have different settings. Whenever there is a detected error in a subflow with 'no', the SDU is discarded, irrespective of settings in other subflows. For an SDU with multiple subflows with a 'yes' setting, there may be one error indication per subflow, or, if there is only one error indication per SDU, it indicates that an error was detected in at least one of these subflows. Exact definitions are the task of RAN3.

~~[~~*(Purpose: Used to decide whether frames with failed CRC on layer 1 shall be forwarded or discarded.*~~]~~*)*

Transfer delay ~~([~~s~~]~~)

Definition: time between request to transfer an SDU at one SAP to its delivery at the other SAP. Transfer delay is specified for one or more fixed SDU sizes. Exact statistical transfer delay definition and fixed SDU sizes are FFS.

~~[~~*(Purpose: specifies the UTRAN part of the total transfer delay for the UMTS bearer. It allows UTRAN to set transport formats and ARQ parameters.*~~]~~*)*

Traffic handling priority

Definition: specifies the relative importance for handling of all SDUs belonging to the radio access bearer compared to the SDUs of other bearers.

~~[~~*(Purpose: Within the interactive class, there is a definite need to differentiate between bearer qualities. This is handled by using the traffic handling priority attribute, to allow UTRAN to schedule traffic accordingly. By definition, priority is an alternative to absolute guarantees, and thus these two attribute types cannot be used together for a single bearer.*~~]~~*)*

Allocation/Retention Priority

Definition: specifies the relative importance compared to other Radio access bearers for allocation and retention of the Radio access bearer.

~~[~~*(Purpose: Priority is used for differentiating between bearers when performing allocation and retention of a bearer, and the value is typically related to the subscription.*

Source statistics descriptor ~~([~~'speech'/'unknown'~~]~~)

Definition: specifies characteristics of the source of submitted SDUs.

~~[~~*(Purpose: Conversational speech has a well-known statistical behaviour (or the discontinuous transmission (DTX) factor). By being informed that the SDUs for a RAB are generated by a speech source, UTRAN may, based on experience, calculate a statistical multiplex gain for use in admission control on the radio and Iu interfaces.* ~~]~~*)*

6.4.4.2 Attributes discussed per class

Conversational class

If the RAB carries a speech service, **Source statistics descriptor** can be set, which allows UTRAN to calculate a statistical multiplexing gain on radio and Iu interfaces and use that for admission control.

Unequal error protection can be supported in conversational class. In case unequal error protection is requested for a given RAB, the attributes Delivery of erroneous SDUs, Residual bit error ratio and SDU error ratio are specified per subflow**. Delivery of erroneous SDUs** determines whether error detection shall be used and, if so, whether SDUs with error in a certain subflow shall be delivered or not. **Residual bit error ratio** specifies the bit error ratio for undetected delivered bits. **SDU error ratio** specifies the fraction of SDUs with detected error in each subflow. It is only set for subflows for which error detection is requested.

In case of unequal error protection the payload of the user data SDU, transported by the Radio Access Bearer Service, must conform to a SDU format defined with possible exact sizes. The payload bits are statically structured into subflows. The **SDU format information** attribute defines the exact subflow format of SDU payload.

Streaming class

If the RAB carries streaming speech, **Source statistics descriptor** can be set, which allows UTRAN to calculate a statistical multiplexing gain on radio and Iu interfaces and use that for admission control.

Unequal error protection can be supported in streaming class. In case unequal error protection is requested for a given RAB, the attributes Delivery of erroneous SDUs, Residual bit error ratio and SDU error ratio are specified per subflow**. Delivery of erroneous SDUs** determines whether error detection shall be used and, if so, whether SDUs with error in a certain subflow shall be delivered or not. **Residual bit error ratio** specifies the bit error ratio for undetected delivered bits. **SDU error ratio** specifies the fraction of SDUs with detected error in each subflow. It is only set for subflows for which error detection is requested.

In case of unequal error protection the payload of the user data SDU, transported by the Radio Access Bearer Service, must conform to a SDU format defined with possible exact sizes. The payload bits are statically structured into subflows. The **SDU format information** attribute defines the exact subflow format of SDU payload.

Other classes

The RAB attribute sets and their use in, interactive and background classes are identical to those of UMTS bearer services (Section 6.4.2.2).

6.4.4.3 Radio Access Bearer attributes: summary

In Table 3, the defined Radio Access Bearer attributes and their relevancy for each bearer class are summarised. Observe that traffic class is an attribute itself.

Table 3. Radio Access Bearer attributes defined for each bearer class.

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate	X	X	X	X
Delivery order	X	X	X	X
Maximum SDU size	X	X	X	X
SDU format information	X	X		
SDU error ratio	X	X	X	X
Residual bit error ratio	X	X	X	X
Delivery of erroneous SDUs	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/ Retention priority	X	X	X	X
Source statistics descriptor	X	X		

6.5 Parameter Value Ranges

6.5.1 Ranges of UMTS Bearer Service Attributes

The following table lists the value ranges of the UMTS bearer service attributes. The value ranges reflect the capability of UMTS network.

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate ([kbps])	<2000 (1) (2)	<2000 (1) (2)	< 2000 - overhead (2) (3)	<2000 - overhead (2) (3)
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size ([octets])	<1500 (4)	<1500 (4)	<1500 (4)	<1500 (4)
SDU format information	(5)	(5)		
Delivery of erroneous SDUs	Yes/No/- (6)	Yes/No/- (6)	Yes/No/- (6)	Yes/No/- (6)
Residual BER	$5*10^{-2}$, 10^{-2}, 10^{-3}, 10^{-4} (7)	$5*10^{-2}$, 10^{-2}, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6} (7)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (8) (7)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (8) (7)
SDU error ratio	10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} (7)	10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} (7)	10^{-3}, 10^{-4}, 10^{-6} (7)	10^{-3}, 10^{-4}, 10^{-6} (7)
Transfer delay ([ms])	100 – maximum value(7)	500 – maximum value (7)		
Guaranteed bit rate ([kbps])	<2000 (1) (2)	<2000 (1) (2)		
Traffic handling priority			1,2,3 (9)	
Allocation/Retention priority	1,2,3 (9)	1,2,3 (9)	1,2,3 (9)	1,2,3 (9)

1) Bitrate of 2000 kbps requires that UTRAN operates in transparent RLC protocol mode, in this case the overhead from layer 2 protocols is negligible.

2) The granularity of the bit rate parameters must be studied. Although the UMTS network has capability to support a large number of different bitrate values, the number of possible values must be limited not to unnecessarily increase the complexity of for example terminals, charging and interworking functions. Exact list of supported values shall be defined together with S1, N1, N3 and R2.

3) Impact from layer 2 protocols on maximum bitrate in non-transparent RLC protocol mode shall be estimated.

4) Maximum SDU size shall at least allow UMTS network to support external PDUs having as high MTU as Internet/Ethernet (1500 octets). The need for higher values must be investigated by N1, N3, S1, R2, R3.

5) Definition of possible values of exact SDU sizes for which UTRAN can support transparent RLC protocol mode, is the task of RAN WG3.

6) If *Delivery of erroneous SDUs* is set to 'Yes' error indications can only be provided on the MT/TE side of the UMTS bearer. On the CN Gateway side error indications can not be signalled outside of UMTS network in release 1999.

7) Values are indicative. Exact values on Residual BER, SDU error ratio and transfer delay shall defined together with S1, N1, N3 and R2.

8) Values are derived from CRC lengths of 8, 16 and 24 bits on layer 1.

9) Number of priority levels shall be further analysed by S1, N1 and N3.

6.5.2 Ranges of Radio Access Bearer Service Attributes

The following table lists the value ranges of the radio access bearer service attributes. The value ranges reflect the capability of UTRAN.

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate ([kbps])	<2000 (1) (2)	<2000 (1) (2)	< 2000 - overhead (2) (3)	<2000 - overhead (2) (3)
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size ([octets])	<1500 (4)	<1500 (4)	<1500 (4)	<1500 (4)
SDU format information	(5)	(5)		
Delivery of erroneous SDUs	Yes/No/-	Yes/No/-	Yes/No/-	Yes/No/-
Residual BER	$5*10^{-2}$, 10^{-2}, 10^{-3}, 10^{-4} (6)	$5*10^{-2}$, 10^{-2}, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6} (6)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (6) (7)	$4*10^{-3}$, 10^{-5}, $6*10^{-8}$ (6) (7)
SDU error ratio	10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} (6)	10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} (6)	10^{-3}, 10^{-4}, 10^{-6} (6)	10^{-3}, 10^{-4}, 10^{-6} (6)
Transfer delay ([ms])	80 – maximum value(6)	500 – maximum value (6)		
Guaranteed bit rate ([kbps])	<2000 (1) (2)	<2000 (1) (2)		
Traffic handling priority			1,2,3 (8)	
Allocation/Retention priority	1,2,3 (8)	1,2,3 (8)	1,2,3 (8)	1,2,3 (8)
Source statistic descriptor	Speech/unknown	Speech/unknown	Speech/unknown	Speech/unknown

1) Bitrate of 2000 kbps requires that UTRAN operates in transparent RLC protocol mode, in this case the overhead from layer 2 protocols is negligible.

2) The granularity of the bit rate parameters must be studied. Although the UMTS network has capability to support a large number of different bitrate values, the number of possible values must be limited not to unnecessarily increase the complexity of for example terminals, charging and interworking functions. Exact list of supported values shall be defined together with S1, N1, N3 and R2.

3) Impact from layer 2 protocols on maximum bitrate in non-transparent RLC protocol mode shall be estimated.

4) Maximum SDU size shall at least allow UMTS network to support external PDUs having as high MTU as Internet/Ethernet (1500 octets). The need for higher values must be investigated by N1, N3, S1, R2, R3.

5) Definition of possible values of exact SDU sizes for which UTRAN can support transparent RLC protocol mode, is the task of RAN WG3.

6) Values are indicative. Exact values on Residual BER, SDU error ratio and transfer delay shall defined together with S1, N1, N3 and R2.

7) Values are derived from CRC lengths of 8, 16 and 24 bits on layer 1.

8) Number of priority levels shall be further analysed by S1, N1 and N3.

Annex A (informative): Error resilience in real-time packet multimedia payloads

A.1 Introduction

This annex provides some basic information with respect to the error resilience of different encoded media streams when considering the support of unequal error protection for real-time packet multimedia services. It provides some indicative figures for the residual bit error rates that could be tolerated by audio-visual H.323 payloads in a 3G environment.

H.323 employs the H.225.0 packetisation scheme, which in turn uses UDP/IP and RTP to transport each media stream. The structure of an H.323 packet is shown in Figure 4.

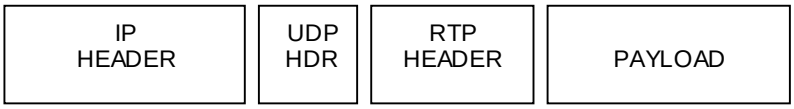

Figure 4: Structure of H.323 packet.

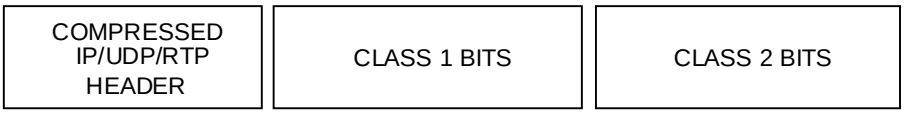

Figure 5: Structure of compressed H.323 packet.
Class 1 bits can tolerate medium BER; Class 2 bits can tolerate high BER.

It is assumed that some elements of the H.323 header information, which comprises the IP, UDP and RTP headers, can be compressed. It is also assumed that this information will require reliable transmission, such that any errors in the header will result in the loss of the complete H.323 packet. However, for real-time multimedia streams that cannot accommodate a large delay (and therefore packet retransmission), codecs can be used that are tolerant to residual bit errors.

This annex highlights the error resilience of audio and visual codecs, and provide some example tolerance figures for media streams of the type that are likely to comprise H.323 payloads.

A.1.1 Factors affecting error resilience

Specific error resilience figures will depend on a number of factors, including:

- the media type;
- the quality of service (QoS) required;
- the specific codec used;

Media streams may also be sub-divided into different classes on the basis of bit error sensitivity as shown in Figure 5. In some cases the most sensitive bits may be protected by in-band checksum information. It should also be noted that, in addition to the effect of residual bit errors in the media stream, the QoS will be further degraded by packet loss due to errors in the H.323 header.

A.2 Example figures

The following values are indicative of the QoS parameters required by audio and video media streams, including bit error rates (BER) and frame erasure rates (FER).

For the purposes of example, figures are provided for the AMR speech codec and the MPEG-4 video codec.

AMR speech codec payload

Bit rate: 4.75 - 12.2 kbit/s

Delay: end-to-end delay not to exceed 100ms (codec frame length is 20ms)

BER 10^{-4} for Class 1 bits

10^{-3} for Class 2 bits

for some applications, a higher BER class (~10^{-2}) might be feasible.

FER < 0.5% (with graceful degradation for higher erasure rates)

MPEG-4 video payload:

Bit rate: variable, average rate scalable from 24 to 128 kbit/s and higher

Delay: end-to-end delay between 150 and 400ms

video codec delay is typically less than 200 ms

BER 10^{-6} - no visible degradation

10^{-5} - little visible degradation

10^{-4} – some visible artefacts

> 10^{-3} - limited practical application

Packet loss rate FFS

Data and control:

Data (data refers to other types than audio and video e.g. file transfers, shared whiteboard) and control information must be transmitted reliably (i.e. residual bit errors should result in a lost packet).

Annex B (normative): Reference Algorithm for Conformance Definition of Bitrate

The annex shows a reference algorithm for the conformance definition of bitrate. This may be used for traffic contract between UMTS bearers and external network/user equipment. It should be noted that the reference algorithm will never imply a particular implementation for the traffic conditioner.

The algorithm is well known as “Token Bucket Algorithm” which has been described in IETF. Here, “tokens” represents the allowed data volume, for example in byte. “Tokens” are given at a constant “token rate” by a traffic contract, are stored temporarily in a “token bucket”, and are consumed by accepting the packet. This algorithm uses the following two parameters (r and b) for the traffic contract and one variable (TBC) for the internal usage.

- r: token rate, (corresponds to the monitored Maximum bitrate/Guaranteed bitrate)

- b: bucket size, (the upper bound of TBC, corresponds to bounded burst size)

- TBC(Token bucket counter): the number of given/remained tokens at any time

In words, conformance according to a token bucket can be defined as: “Data is conformant if the amount of data submitted during any arbitrarily chosen time period T does not exceed (b+rT).”

The algorithm is described in the following:

Token bucket counter (TBC) is usually increased by “r” in each small time unit. However, TBC has upper bound “b” and the value of TBC must never exceed “b”.

When a packet #i with length Li arrives, the receiver checks the current TBC. If the TBC value is equal to or larger than Li, the packet arrival is judged compliant, i.e., the traffic is conformant. At this moment tokens corresponding to the packet length is consumed, and TBC value decreases by Li.

When a packet #j with length Lj arrives, if TBC is less than Lj, the packet arrival is non-compliant, i.e., the traffic is not conformant. In this case, the value of TBC is not updated.

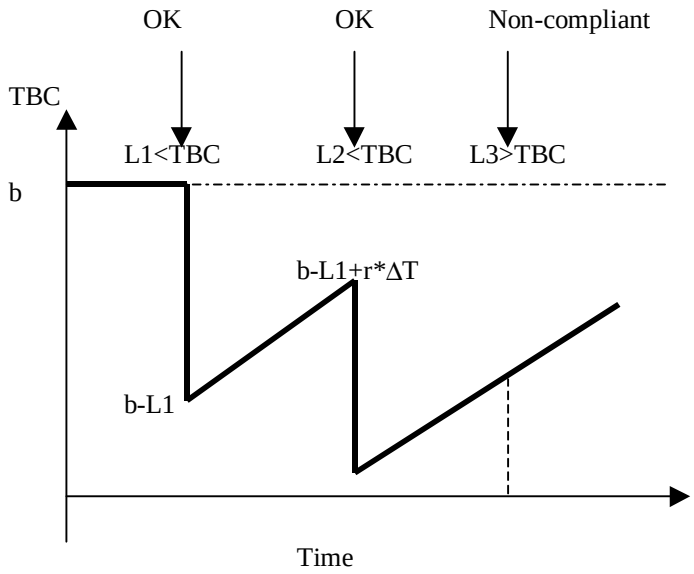

Figure 6. Operation example of the reference conformance algorithm.