



ATIS-0100032

RELATION BETWEEN ITU-T (Y.1541/Y.1221) AND
3GPP UMTS/LTE QoS CLASSES

TECHNICAL REPORT



ATIS is the leading technical planning and standards development organization committed to the rapid development of global, market-driven standards for the information, entertainment and communications industry. More than 200 companies actively formulate standards in ATIS' Committees, covering issues including: IPTV, Cloud Services, Energy Efficiency, IP-Based and Wireless Technologies, Quality of Service, Billing and Operational Support, Emergency Services, Architectural Platforms and Emerging Networks. In addition, numerous Incubators, Focus and Exploratory Groups address evolving industry priorities including Smart Grid, Machine-to-Machine, Networked Car, IP Downloadable Security, Policy Management and Network Optimization.

ATIS is the North American Organizational Partner for the 3rd Generation Partnership Project (3GPP), a member and major U.S. contributor to the International Telecommunication Union (ITU) Radio and Telecommunications' Sectors, and a member of the Inter-American Telecommunication Commission (CITEL). ATIS is accredited by the American National Standards Institute (ANSI). For more information, please visit < <http://www.atis.org> >.

Notice of Disclaimer & Limitation of Liability

The information provided in this document is directed solely to professionals who have the appropriate degree of experience to understand and interpret its contents in accordance with generally accepted engineering or other professional standards and applicable regulations. No recommendation as to products or vendors is made or should be implied.

NO REPRESENTATION OR WARRANTY IS MADE THAT THE INFORMATION IS TECHNICALLY ACCURATE OR SUFFICIENT OR CONFORMS TO ANY STATUTE, GOVERNMENTAL RULE OR REGULATION, AND FURTHER, NO REPRESENTATION OR WARRANTY IS MADE OF MERCHANTABILITY OR FITNESS FOR ANY PARTICULAR PURPOSE OR AGAINST INFRINGEMENT OF INTELLECTUAL PROPERTY RIGHTS. ATIS SHALL NOT BE LIABLE, BEYOND THE AMOUNT OF ANY SUM RECEIVED IN PAYMENT BY ATIS FOR THIS DOCUMENT, WITH RESPECT TO ANY CLAIM, AND IN NO EVENT SHALL ATIS BE LIABLE FOR LOST PROFITS OR OTHER INCIDENTAL OR CONSEQUENTIAL DAMAGES. ATIS EXPRESSLY ADVISES ANY AND ALL USE OF OR RELIANCE UPON THIS INFORMATION PROVIDED IN THIS DOCUMENT IS AT THE RISK OF THE USER.

NOTE - The user's attention is called to the possibility that compliance with this standard may require use of an invention covered by patent rights. By publication of this standard, no position is taken with respect to whether use of an invention covered by patent rights will be required, and if any such use is required no position is taken regarding the validity of this claim or any patent rights in connection therewith.
--

ATIS-0100032, *Relation Between ITU-T (Y.1541/Y.1221) and 3GPP UMTS/LTE QoS Classes*

Is an ATIS Standard developed by the **ATIS Network Performance, Reliability, and Quality of Service Committee (PRQC)**.

Published by

**Alliance for Telecommunications Industry Solutions
1200 G Street, NW, Suite 500
Washington, DC 20005**

Copyright © 2011 by Alliance for Telecommunications Industry Solutions
All rights reserved.

No part of this publication may be reproduced in any form, in an electronic retrieval system or otherwise, without the prior written permission of the publisher. For information contact ATIS at 202.628.6380. ATIS is online at < <http://www.atis.org> >.

Printed in the United States of America.

Technical Report on

Relation Between ITU-T (Y.1541/Y.1221) and 3GPP UMTS/LTE QoS Classes

Alliance for Telecommunications Industry Solutions

Approved March 2011

Abstract

This Technical Report addresses the relation between QoS classes defined in the ITU-T (Y.1541 performance classes) and the 3GPP (UMTS and LTE).

FOREWORD

The Alliance for Telecommunication Industry Solutions (ATIS) serves the public through improved understanding between providers, customers, and manufacturers. The Network Performance, Reliability, and Quality of Service Committee (PRQC) develops and recommends standards, requirements, and technical reports related to the performance, reliability, and associated security aspects of communications networks, as well as the processing of voice, audio, data, image, and video signals, and their multimedia integration. PRQC also develops and recommends positions on, and foster consistency with, standards and related subjects under consideration in other North American and international standards bodies.

The mandatory requirements are designated by the word *shall* and recommendations by the word *should*. Where both a mandatory requirement and a recommendation are specified for the same criterion, the recommendation represents a goal currently identifiable as having distinct compatibility or performance advantages. The word *may* denotes a optional capability that could augment the standard. The standard is fully functional without the incorporation of this optional capability.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, PRQC, 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, PRQC, which was responsible for its development, had the following roster:

- P. Tarapore, PRQC Chair (AT&T)
- M. Neibert, PRQC Vice-Chair(Telcordia)
- P. Tarapre, Technical Editor (AT&T)
- C. Dvorak, Technical Editor (AT&Y)
- C. Underkoffler, ATIS Chief Editor

Active Participants:

K. Biholar	M. Niebert
C. Dvorak	A. Nguyen
E. Geelen	H. Pant
Y. Kogan	E. Rojek
O. Lima	J. Schiavone
M. Linnell	P. Tarapore
S. Makris	A. Webster
A. Morton	

Acknowledgment: Almost all of the progress on the mapping between the ITU and 3GPP QoS classes (UMTS at that time) was accomplished by Neal Seitz (NTIA). Neal provided many technical contributions in former ATIS Technical Subcommittee T1A1, including documents that became USA contributions to the ITU-T in 2004. Much of the content of Sections 6-9 is taken directly from Neal's earlier ATIS contributions.

TABLE OF CONTENTS

1	Scope and Purpose.....	1
2	References.....	1
2.1	Normative References.....	1
2.2.1	3GPP References.....	1
2.2.2	ITU-T References.....	2
2.2	Informative References.....	2
3	Definitions.....	2
4	Acronyms & Abbreviations.....	2
5	Relationship Between 3GPP & ITU-T QoS Classes – General Discussion.....	3
6	Comparison of Y.1541 and UMTS QoS Classes.....	4
7	Mapping Between Y.1541 Classes and 3GPP UMTS Service Attributes.....	7
8	Y.1221 Traffic Classes and 3GPP UMTS Service Attributes.....	11
9	A Hypothetical Y.1541-to-UMTS Mapping Example.....	12
9.1	Y.1541 to UMTS.....	13
9.2	UMTS to Y.1541.....	14
9.3	Summary Observations from the Mapping Example.....	14
10	3GPP LTE Classes and Their Relationship to UMTS Classes.....	15
11	Problems of Supporting End-to-End QoS (Y.1541) with LTE QCI.....	18
12	Conclusions and Recommendations.....	19

TABLE OF FIGURES

FIGURE 1 - TAXONOMY OF PACKET (OR SDU) TRANSFER OUTCOMES.....	9
FIGURE 2 - CORRESPONDENCE BETWEEN UE AND GGSN OF UMTS AND TE, UNI, AND NNI OF ITU-T.....	13
FIGURE 3 - SCOPE OF QCI CHARACTERISTICS FOR CLIENT/SERVER (UPPER FIGURE) AND PEER/PEER (LOWER FIGURE) COMMUNICATION.....	16

TABLE OF TABLES

TABLE 1– IP QoS CLASS DEFINITIONS AND NETWORK PERFORMANCE OBJECTIVES (FOOTNOTES OMITTED).....	4
TABLE 2– GUIDANCE FOR IP QoS CLASSES.....	5
TABLE 3– 3GPP UMTS QoS CLASSES.....	6
TABLE 4– UMTS BEARER ATTRIBUTES DEFINED FOR EACH BEARER TRAFFIC CLASS.....	8
TABLE 5– VALUE RANGES FOR UMTS BEARER SERVICE ATTRIBUTES (FOOTNOTES OMITTED).....	10
TABLE 6– RELATIONSHIPS AMONG ITU-T (Y.1541) AND 3GPP (TS 23-107) UMTS QoS CLASSES, PARAMETERS, AND BEARER ATTRIBUTES.....	11
TABLE 7 – STANDARDIZED QCI CHARACTERISTICS.....	17
TABLE 8 – MAPPING BETWEEN LTE QCI CLASSES AND UMTS TRAFFIC CLASSES.....	18

Technical Report on –

Relation Between ITU-T (Y.1541/Y.1221) and 3GPP UMTS/LTE QoS Classes

1 SCOPE AND PURPOSE

This Technical Report addresses the incompatibilities that exist in the way that the 3GPP and the ITU have addressed QoS classes for IP-based packet flows.

This Technical Report has several purposes. It formally documents important work done earlier in the former Committee T1's T1A1 Subcommittee, specifically the mapping of QoS classes between the ITU-T's Y.1541/Y.1221 and the 3GPP's TS 23-107 (for UMTS). (This work was captured in various T1A1 contributions, and in external liaisons to the ITU and the 3GPP, but it was never documented in an ATIS publication, so it is done so here.)

With the formal documentation of the QoS class mapping between the ITU and UMTS presented, the next purpose of this document is to examine the same question of QoS class mapping between the ITU and LTE classes. It is shown that, as was the case for UMTS, a rigorous mapping that fulfills the numerical performance objectives of both domains is neither meaningful nor possible.

As the ultimate goal in the industry is fixed/mobile network convergence, some form of interworking will be necessary, and it is the purpose of this document to establish the foundation for quantitative methods capable of supporting this critically needed interworking.

2 REFERENCES

2.1 Normative References

The following standards contain provisions which, through reference in this text, constitute provisions of this ATIS Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this ATIS Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

2.2.1 3GPP References¹

[TS23-107] – 3GPP Technical Specification TS 23-107 (2008), *Quality of Service (QoS) Concept and Architecture*.

[TS23-203] - 3GPP Technical Specification TS 23-203 (2009), *Policy and Charging Control Architecture*.

[TS23-401] - 3GPP Technical Specification TS 23-401 (2009), *General Packet Radio Service (GPRS) Enhancements for Evolved Universal Terrestrial Radio Access Network (E-UTRAN) Access*.

[TS23-207] – 3GPP Technical Specification TS 23-207 (2008), *End-to-End Quality of Service Concept and Architecture*.

¹ These documents are available from the Third Generation Partnership Project (3GPP) at < <http://www.3gpp.org/specs/specs.htm> >.

2.2.2 ITU-T References²

[Y.1221] – ITU-T Recommendation Y.1221 (2010), *Traffic Control and Congestion Control in IP Networks*.

[Y.1540] – ITU-T Recommendation Y.1540 (2002), *Internet Protocol Data Communication Service – IP Packet Transfer and Availability Performance Parameters*.

[Y.1541] – ITU-T Recommendation Y.1541 (2006), *Network Performance Objectives for IP-Based Services*.

2.2 Informative References³

[b-3A175] – ATIS Committee T1 Contribution 3A100750 (2004), *Mapping Between ITU-T (Y.1541/Y.1221) and 3GPP (TS 23-107) QoS Classes and Traffic Descriptors*, G. Bain & N. Seitz.

3 DEFINITIONS

None identified in this document.

4 ACRONYMS & ABBREVIATIONS

3GPP	Third Generation Partnership Project
BER	Bit Error Rate
FTP	File Transfer Protocol
GBR	Guaranteed Bit Rate
IP	Internet Protocol
IPDV	IP Delay Variation
IPER	IP Error Ratio
IPTD	IP Transfer Delay
ISP	Internet Service Provider
ITU-T	International Telecommunications Union – Standardization Sector
LTE	Long Term Evolution
NNI	Network-Network Interconnect
PCEF	Policy & Charging Enforcement Function
PDB	Packet Delay Budget
QCI	QoS Class Identifier
QoS	Quality of Service
SDF	Service Data Flow
SDU	Service Data Unit
SLA	Service Level Agreement
TS	Technical Specification
UE	User Element
UMTS	Universal Mobile Telecommunications System
UNI	User-Network Interface
VoIP	Voice over IP

² These documents are available from the International Telecommunications Union. < <http://www.itu.int/ITU-T/> >.

³ This reference is a committee contribution. PRQC committee participants can access this document at < <http://contributions.atis.org> >. Copies of this contribution will be made available to all other interested parties upon request. Such request should be made to the ATIS Document Center Administrator at < doccenter@atis.org >.

5 RELATIONSHIP BETWEEN 3GPP & ITU-T QoS CLASSES – GENERAL DISCUSSION

The delivery of assured end-to-end IP Quality of Service (QoS) over end-to-end paths involving both wireless access networks and wireline access and/or core networks is of significant interest to service providers. There is a need to support highly interactive applications such as business quality voice, video teleconferencing, transactional data, and the signaling associated with each. However, as documented in the report, incompatibilities between the QoS classes standardized in ITU-T and those being standardized in 3GPP present serious obstacles to directly getting known end-to-end QoS for paths involving wireless access.⁴

ITU-T Recommendation Y.1541 [Y.1541] specifies six IP network QoS classes, each characterized by numerical performance objectives for four IP network performance parameters defined in companion Recommendation Y.1540 [Y.1540]. The Y.1541 performance objectives apply to the end-to-end IP path between user-network interfaces (whereas the 3GPP objectives apply only to the wireless access network).

In wireless space, 3GPP Technical Specification 23-107 [TS23-107] specifies four universal mobile telecommunications system (UMTS) QoS classes (also called “traffic classes”), distinguished primarily by their delay sensitivity, but excluding any consideration for IP packet delay variation (IPDV). More recently, 3GPP Technical Specification 23-203 [TS23-203] specifies nine QoS classes as part of their Long Term Evolution (LTE) architecture. The relationships between the four UMTS classes and the nine LTE classes are noted in 3GPP Technical Specification 23-401 [Table E.3 in TS23-401].

The two sets of QoS classes (ITU-T and 3GPP) have similarities but no complete mapping between them is possible for several reasons. First, the 3GPP classes do not have all the parameters as the ITU-T set of parameters; second, some parameters like delay are defined and measured completely differently; and third, the ITU-T QoS classes are end-to-end but the 3GPP classes are not. Recognizing these incompatibilities, Committee T1 previously considered [b-3A175] one possible mapping between the Y.1541 QoS classes (and associated traffic descriptors) and a corresponding set of values for 3GPP “bearer service attributes” defined for the four UMTS classes .

This Technical Report provides a formal documentation of the earlier Committee T1 effort [b-3A175] (sections 6 to 9). It then provides descriptions of the newer 3GPP LTE QoS classes (section 10) and demonstrates inconsistencies between the LTE and Y.1541 classes – similar to those observed between the UMTS and Y.1541 classes (section 11). Conclusions and recommendations are provided in section 12 all in the context of addressing a good interworking solution—a critical precursor to achieving truly harmonized end-to-end QoS as wireless and wireline networks converge.

For convenience, a summary of conclusions is presented here as follows. There are two major incompatibilities in deriving a meaningful and consistent mapping between the Y.1541 classes and the 3GPP UMTS/LTE:

- *Transfer Delay* - The 3GPP Transfer Delays are defined as *maxima* whereas the Y.1541 Transfer Delays are expressed as *mean* values. Also, the 3GPP Transfer Delays are defined for the 3GPP domain only whereas the Y.1541 Transfer Delays are expressed as end-to-end, but the ITU experts were never able to agree on an allocation to segments like wireless access.
- *Delay Variation* – Delay Variation in the packet stream is specified for the Y.1541 classes (a distribution statistic) but is not addressed in any way for the 3GPP classes.

⁴ As an example of the concerns expressed, [b-3P149] states in part: “The QoS Parameters, Parameter Values and QoS Classes defined in 3GPP specifications are different from, and may be incompatible with those in ITU-T specifications. This may result in interoperability problems between 3GPP-based wireless networks and ITU-T based wireline networks. Adversely impacted services may include Voice over IP, Video Streaming, and multimedia services such as Telecommunications for Disaster Relief. Either alignment of specifications or the definition of standardized interworking that will not adversely impact service delivery between networks based on 3GPP and ITU-T specifications is therefore required.”

Going forward, it is recommended to approach the end-to-end QoS problem by developing a detailed specification of interworking between different network segments (e.g., between LTE and optical IP backbones). This would necessitate, for example, developing an industry-wide default markings between things such as LTE QCI-related markings and Diffserv Code Points, or alternatively, development of interworking guidelines that can be used to derive appropriate Service Level Agreements (SLA) between service providers.

6 COMPARISON OF Y.1541 AND UMTS QoS CLASSES

The content of sections 6 through 9 are based on earlier ATIS Committee T1 work by Neal Seitz. The content in these sections is taken from Neal’s earlier ATIS contributions [b-3A175].

Table 1 illustrates the Y.1541 QoS classes and associated network performance objectives. These specifications apply between user-network interfaces that delimit end-to-end IP flows. The objectives are designed to be achievable on common IP network implementations. Classes 0 and 1 place upper bounds on packet transfer delay and packet loss. They also limit packet delay variation. Classes 2 and 3 place upper bounds on packet transfer delay and packet loss, but do not limit packet delay variation. Classes 0 and 2 differ from Classes 1 and 3 in their packet transfer delay objectives. Class 4 limits packet loss and provides a very loose upper bound on delay. A single packet error ratio objective is specified for classes 0-4; this value is chosen to ensure that packet loss is the dominant cause of defects presented to upper layers. Y.1541 also defines a “best effort” QoS class (Class 5) with no specific performance guarantees.

Y.1541 assumes that the user and network provider have agreed on a traffic profile that applies to one or more packet flows in a QoS class. At present, the agreeing parties may use whatever capacity specifications they consider appropriate so long as they allow both enforcement and verification. For example, peak bit rate (including lower layer overhead) may be sufficient. When protocols and systems supporting dynamic requests are available, users may negotiate a *traffic contract* that specifies one or several traffic parameters. ITU-T Recommendation Y.1221 [Y.1221] defines the traffic parameters in the context of three fundamental types of flows IP-based networks can support (dedicated bandwidth, statistical bandwidth, and best effort). The Y.1221 traffic parameters and corresponding UMTS service attributes are discussed below.

Table 1– IP QoS class definitions and network performance objectives (footnotes omitted)

		QoS Classes					
Network Performance Parameter	Nature of Network Performance Objective	Class 0	Class 1	Class 2	Class 3	Class 4	Class 5 (Un-specified)
IPTD	Upper bound on the mean IPTD	100 ms	400 ms	100 ms	400 ms	1 s	U
IPDV	Upper bound on the 1-10 ⁻³ quantile of IPTD minus the minimum IPTD	50 ms	50 ms	U	U	U	U
IPLR	Upper bound on the packet loss probability	1*10 ⁻³	U				
IPER	Upper bound	1*10 ⁻⁴					U

Table 2 identifies some typical applications for each Y.1541 QoS class, and some typical node mechanisms and network techniques that could be used to implement them. For example, the node mechanisms could involve separate queues with preferential servicing and different drop priorities, or traffic grooming; the network techniques could involve constrains on routing and distance.⁵ Y.1541 emphasizes that these guidelines are discretionary; network providers using the standard may employ whatever node mechanisms, routing constraints, provisioning strategies, or other QoS control techniques they choose.

Table 2– Guidance for IP QoS classes

QoS Class	Applications (Examples)	Node Mechanisms	Network Techniques
0	Real-Time, Jitter sensitive, high interaction(VoIP, VTC)	Separate Queue with preferential servicing, Traffic grooming	Constrained Routing and Distance
1	Real-Time, Jitter sensitive, interactive (VoIP, VTC).		Less constrained Routing and Distances
2	Transaction Data, Highly Interactive, (Signaling)	Separate Queue, Drop priority	Constrained Routing and Distance
3	Transaction Data, Interactive		Less constrained Routing and Distances
4	Low Loss Only (Short Transactions, Bulk Data, Video Streaming)	Long Queue, Drop priority	Any route/path
5	Traditional Applications of Default IP Networks	Separate Queue (lowest priority)	Any route/path

Table 3 illustrates the QoS (also called traffic) classes defined for the 3GPP defined universal mobile telecommunications system (UMTS). Four QoS classes are defined in 3GPP Technical Specification 23-107 [TS23-107]: conversational, streaming, interactive, and background. The main distinguishing factor among these classes is the delay sensitivity of the traffic. The conversational and streaming classes are intended to be used primarily in carrying real-time traffic flows. The conversational class supports real-time services like video telephony that are particularly delay sensitive. The streaming class supports one-way flows, and therefore is somewhat less delay sensitive. The interactive and background classes are primarily meant to be used by traditional Internet applications like WWW, e-mail, telnet, FTP, and news. Because they have looser delay requirements than the conversational and streaming classes, they can provide better error rates using channel coding and retransmission. The main difference between the interactive and background classes is that the interactive class is mainly used by interactive applications (e.g. interactive e-mail or interactive Web browsing), while the background class is meant for background traffic (e.g. background download of e-mail or other files). Responsiveness of the interactive applications is ensured by separating the interactive and background applications. Interactive traffic is intended to have a higher priority in scheduling than background traffic, so that background applications use transmission resources only when interactive applications

⁵ Recommendation Y.1541 notes that there will be very long paths that cannot support Classes 0 and 2; nevertheless, it was considered important to specify (and offer) low delay services where feasible.

do not need them. TS 23-107 notes that such prioritization is particularly important in a wireless environment, where the bandwidth is low compared to fixed networks.

Table 3– 3GPP UMTS QoS classes

Traffic class	Conversational class conversational RT	Streaming class streaming RT	Interactive class Interactive best effort	Background Background best effort
Fundamental characteristics	- Preserve time relation (variation) between information entities of the stream - Conversational pattern (stringent and low delay)	- Preserve time relation (variation) between information entities of the stream	- Request response pattern - Preserve payload content	- Destination is not expecting the data within a certain time - Preserve payload content
Example of the application	- Voice	- Streaming video	- Web browsing	- Background download of emails

Comparing Tables 1-3, it appears that Y.1541 classes 0 and 1 correspond generally with the 3GPP conversational and streaming classes, respectively. In each specification regime, the two classes are intended to support real time services and the first class has a more stringent delay requirement than the second. In both regimes delay variation is intended to be limited.⁶

Similarly, it appears that Y.1541 classes 2-4 correspond generally with the 3GPP interactive class. In both specification regimes, a key application of interest is interactive data. Y.1541 supports this application category with three classes, distinguished by different quantitative delay limits. TS 23-107 states (para. 6.4.3.2):

“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 ... Instead, **traffic handling priority** is used. SDUs of a UMTS bearer with higher traffic handling priority [are] given priority over SDUs of other bearers within the interactive class, through UMTS-internal scheduling.”

Thus TS 23-107 envisions a QoS distinction for interactive traffic similar to that defined in Y.1541, but specifies a *relative* QoS mechanism for implementing it.⁷

Y.1541 class 5 corresponds closely with the 3GPP background class.

The most fundamental difference between the Y.1541 and TS 23-107 classes is that the former classes specify quantitative performance limits while the latter classes, in themselves, do not. Clearly, such limits would need to be specified in wireless/wireline interworking situations to assure particular QoS levels end-to-end.

⁶ The limit is implicit in the requirement to “preserve time relation (variation) between info entities of the stream” in the 3GPP case.

⁷ Another 3GPP specification (TS 29-207) defines six QoS classes by expanding the interactive class into three classes, distinguished by three traffic handling priorities. This specification would align the UMTS and Y.1541 classes more closely, since a UMTS class would be associated with each of Y.1541 classes 2-4 (see Table 6).

7 MAPPING BETWEEN Y.1541 CLASSES AND 3GPP UMTS SERVICE ATTRIBUTES

Although the 3GPP QoS classes do not in themselves provide a basis for QoS interworking with external IP networks, TS 23-107 specifies a related set of “bearer service attributes” that may. TS 23-107 in fact states that QoS will be defined by specifying such attributes. Any particular set of attributes that can be requested by the user is defined as a “QoS profile.” The QoS profile is communicated among UMTS entities to activate QoS mechanisms ensuring provision of the negotiated UMTS service QoS. TS 23-107 further states:

“The end-to-end service is provided by translation/mapping with UMTS external services. A Translation Function converts between the internal service primitives for UMTS bearer service control and the various protocols for service control of interfacing external networks. *The translation includes converting between UMTS bearer service attributes and QoS attributes of the external network’s service control protocol.*⁸

Thus, the QoS mapping envisioned in TS 23-107 is a translation of *bearer service attributes*. The attributes specify values for particular performance (and traffic) parameters. If the set of bearer service attributes translated between 3GPP and external systems were comprehensive enough (including, for example, attributes related to error, delay, etc.), it could be possible to map between the 3GPP bearer service attributes and the Y.1541 QoS classes.

TS 23-107 discusses the attributes that can be specified (and mapped between 3GPP and external systems) for each of the four 3GPP QoS (or traffic) classes.⁹ Table 4 summarizes the defined UMTS bearer attributes and their relevance for each traffic class. The bearer service attributes are listed in the column on the left. Three of these attributes describe bearer service *performance* (as opposed to traffic or functionality): transfer delay, SDU error ratio, and residual bit error ratio. The latter parameter cannot be easily related to the Y.1541 parameters because it is bit-based. Transfer delay and SDU error ratio are relatable to the Y.1541 parameters IPTD and IPLR/IPER, respectively, if the UMTS SDU corresponds to an IP packet. In that case, the delay parameter definitions are similar enough that their values could be related. However, the specifications for the two delay parameters represent different distribution statistics: the IPTD specifications are means, while the SDU transfer delay specifications are maxima.

TS 23-107 defines SDU error ratio as “the fraction of SDUs lost or detected as erroneous.” The Y.1540/Y.1541 lost packet and errored packet outcomes are thus, in effect, combined in the SDU error outcome, as illustrated in Figure 1.¹⁰

Although the definitions for the 3GPP conversational and streaming classes imply that IPDV must be limited to “preserve time relation (variation) between information entities of the stream,” TS 23-107 does not define or specify a value for delay variation. It will not be possible to assure an end-to-end limit on IPDV in the absence of a quantitative limit on delay variation for the 3GPP portion. A QoS mapping between 3GPP and external IP networks could take IPDV into account qualitatively by mapping Y.1541 classes 0 and 1 to the conversational or streaming classes.

⁸ Emphasis added. TS 23-107 also notes that the QoS (or traffic) classes are “attributes” in themselves.

⁹ Para. 6.4.3.2, “Attributes discussed per traffic class.”

¹⁰ Delivery order is not addressed in the taxonomy of Figure 1.

Table 4– UMTS bearer attributes defined for each bearer traffic 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		
Signaling indication			X	

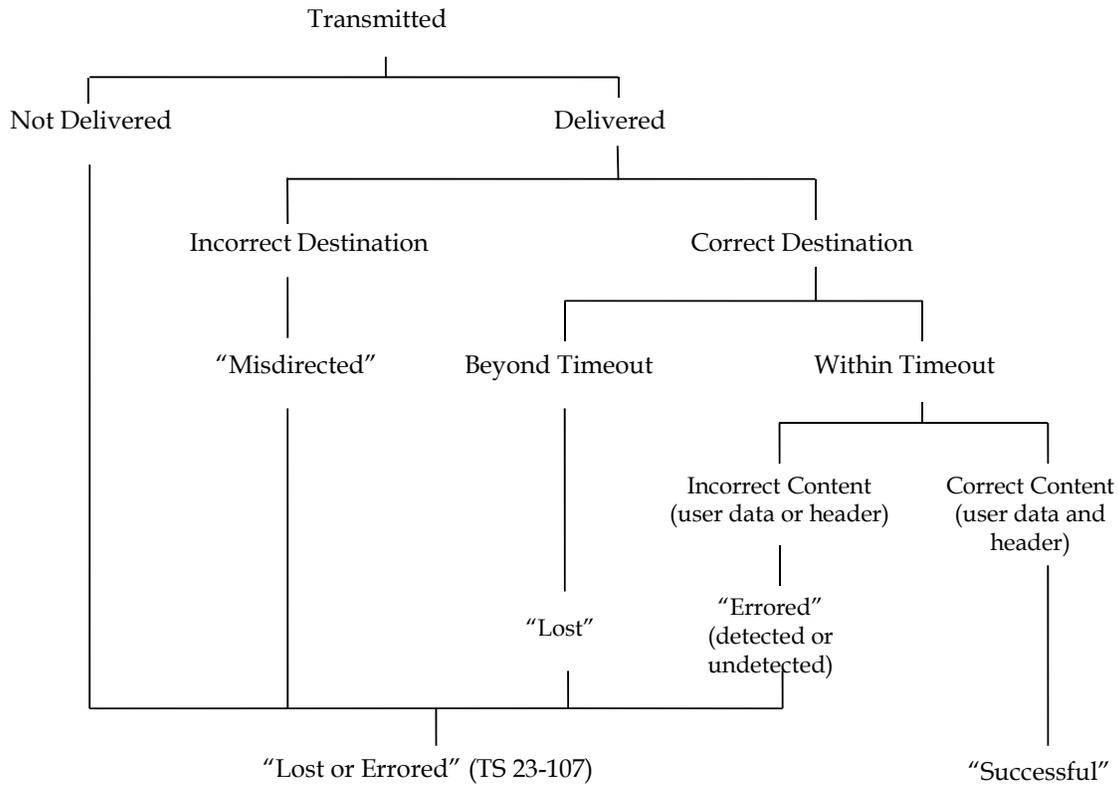


Figure 1 - Taxonomy of packet (or SDU) transfer outcomes.

Table 5 lists a set of distinct attribute values or identifies the allowed range of values for each attribute as they apply to 3GPP UMTS networks. As noted earlier, the attributes that describe bearer service *performance* (of particular interest here) are transfer delay and SDU error ratio. The value list/value range defines the values that are *possible* to be used for an attribute, considering *every possible* service condition. 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.

Table 5– Value ranges for UMTS Bearer Service Attributes (footnotes omitted)

Traffic class	Conversational class	Streaming class	Interactive class	Background class
Maximum bitrate (kbps)	<= 16 000	<= 16 000	<= 16 000 - overhead	<= 16 000 - overhead
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size (octets)	<=1 500 or 1 502	<=1 500 or 1 502	<=1 500 or 1 502	<=1 500 or 1 502
SDU format information	(RAN WG3)	(RAN WG3)		
Delivery of erroneous SDUs	Yes/No	Yes/No	Yes/No	Yes/No
Residual BER	$5 \cdot 10^{-2}, 10^{-2}, 5 \cdot 10^{-3}, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6}$	$5 \cdot 10^{-2}, 10^{-2}, 5 \cdot 10^{-3}, 10^{-3}, 10^{-4}, 10^{-5}, 10^{-6}$	$4 \cdot 10^{-3}, 10^{-5}, 6 \cdot 10^{-8}$	$4 \cdot 10^{-3}, 10^{-5}, 6 \cdot 10^{-8}$
SDU error ratio	$10^{-2}, 7 \cdot 10^{-3}, 10^{-3}, 10^{-4}, 10^{-5}$	$10^{-1}, 10^{-2}, 7 \cdot 10^{-3}, 10^{-3}, 10^{-4}, 10^{-5}$	$10^{-3}, 10^{-4}, 10^{-6}$	$10^{-3}, 10^{-4}, 10^{-6}$
Transfer delay (ms)	100 – maximum value	280 – maximum value		
Guaranteed bit rate (kbps)	<= 16 000	<= 16 000		
Traffic handling priority			1,2,3	
Allocation/Retention priority	1,2,3	1,2,3	1,2,3	1,2,3
Source statistic descriptor	Speech/unknown	Speech/unknown		
Signalling Indication			Yes/No	

Table 6 lists the Y.1541 QoS classes and the 3GPP UMTS QoS classes and associated bearer service attributes and values in the rows and columns of a matrix and indicates, at selected intersections, the most closely related classes and how they differ. This matrix provides a basis for assessing the opportunities and difficulties in mapping performance values between the Y.1541 QoS classes and the 3GPP service attributes as they are currently defined.

Table 6– Relationships among ITU-T (Y.1541) and 3GPP (TS 23-107) UMTS QoS classes, parameters, and bearer attributes

3GPP UMTS QoS Class (and Relevant Attribute Values)		Real Time		Best Effort	
		Conversational	Streaming	Interactive	Background
Y.1541 QoS Class (and Relevant Parameter Values)		<ul style="list-style-type: none"> Preserve time relation (variation) between info entities of the stream Conversational pattern (stringent and low delay) 	<ul style="list-style-type: none"> Preserve time relation (variation) between info entities of the stream 	<ul style="list-style-type: none"> Request/response pattern Preserve payload content 	<ul style="list-style-type: none"> Destination is not expecting data within a certain time Preserve payload content
		<ul style="list-style-type: none"> Transfer delay: 100 ms (maximum value) SDU error ratio (ER): 10^{-2}, 7×10^{-3}, 10^{-3}, 10^{-4}, 10^{-5} 	<ul style="list-style-type: none"> Transfer delay: 280 ms (maximum value) SDU error ratio (ER): 10^{-1}, 10^{-2}, 7×10^{-3}, 10^{-3}, 10^{-4}, 10^{-5} 	<ul style="list-style-type: none"> Transfer delay: 'traffic handling priority' SDU error ratio (ER): 10^{-3}, 10^{-4}, 10^{-6} 	<ul style="list-style-type: none"> SDU error ratio (ER): 10^{-3}, 10^{-4}, 10^{-6}
Class 0	IPTD \leq 100 ms IPDV \leq 50 ms IPLR $\leq 10^{-3}$ IPER $\leq 10^{-4}$	IPTD is a mean value; transfer delay is a maximum Y.1541 specifies IPDV limit Y.1541 specifies IPLR/IPER; TS 23-107 specifies SDU ER			
Class 1	IPTD < 400 ms IPDV < 50 ms IPLR < 10^{-3} IPER < 10^{-4}		IPTD is a mean value; transfer delay is a maximum Y.1541 specifies IPDV limit Y.1541 specifies IPLR/IPER; TS 23-107 specifies SDU ER		
Class 2	IPTD < 100 ms IPLR < 10^{-3} IPER < 10^{-4}				
Class 3	IPTD \leq 400 ms IPLR $\leq 10^{-3}$ IPER < 10^{-4}			Y.1541 specifies IPTD limits; TS 23-107 specifies 'traffic handling priority' Y.1541 specifies IPLR/IPER;	
Class 4	IPTD \leq 1 second IPLR < 10^{-3} IPER < 10^{-4}			TS 23-107 specifies SDU ER 'target'	
Class 5	Best Effort				TS 23-107 specifies SDU ER 'target'

8 Y.1221 TRAFFIC CLASSES AND 3GPP UMTS SERVICE ATTRIBUTES

The other key requirement for QoS interworking between 3GPP wireless and non-3GPP wireline networks is compatibility in the capacity made available (and the traffic control applied) to particular end-to-end IP flows. In general, such compatibility will require a mapping of traffic descriptors among the two specification domains, analogous to the QoS mapping posited in the previous section. As noted earlier, ITU-T has defined a set of IP network traffic parameters in Recommendation Y.1221 [Y.1221]. The corresponding UMTS traffic descriptors are service attributes defined in TS 23-107 (see Tables 4 and 5 above).

Although the units of measure differ, there appears to be good general correspondence between the Y.1221 traffic parameters and the traffic related subset of the TS 23-107 service attributes. Y.1221 defines five traffic parameters for which quantitative values may be specified: peak rate, peak token bucket size, sustainable rate, sustainable token bucket size, and maximum packet size. The related TS 23-107 service attributes listed in Tables 4 and 5 are maximum bit rate, guaranteed bit rate, and

maximum SDU size. TS-23-107 also defines maximum and guaranteed bit rate token bucket sizes, both of which are equated to the maximum SDU size.¹¹

9 A HYPOTHETICAL Y.1541-TO-UMTS MAPPING EXAMPLE

This section describes a hypothetical QoS mapping (in each direction) between two concatenated networks: a 3GPP network providing UMTS service in accordance with the TS 23-107 QoS classes and bearer service attributes, and an “external” (non-3GPP) IP network supporting assured-quality IP flows in accordance with Recommendation Y.1541 - Figure 2. For simplicity, the UMTS SDU is assumed to correspond to an IP packet. The end-to-end (NI-NI) IP packet transfer service provided by the concatenated networks is intended to meet the end-to-end QoS objectives of Y.1541.¹² The objective in mapping QoS classes (and bearer attribute values) between the UMTS network and the IP network is to divide the end-to-end “impairment budget” for each Y.1541 performance parameter (delay, delay variation, packet loss, packet error) appropriately between them. For illustration an equal division is assumed in this example -- e.g., each network would get 50 ms of a 100 ms end-to-end IPTD objective.¹³ A QoS translator in the interworking function between the UMTS network and the IP network would map QoS classes and attribute values between the two networks so as to ensure, where possible, that the end-to-end QoS objectives are met. The purpose of this example is not to suggest a specific mapping for implementation, but to explore the utility and limitations of the mapping approach to interworking, given the ITU-T and 3GPP UMTS QoS specifications as they exist today.

¹¹ Both Y.1221 and TS 23-107 frame the traffic conformance definitions in terms of a token bucket reference algorithm. TS 23-107 states (Annex B) that the token bucket algorithm “may be used for traffic contract between UMTS bearers and external network/user equipment.”

¹² TS 23-107 notes that the UMTS attribute values and ranges apply to the UMTS network, not the end-to-end service.

¹³ Y.1541 limits IPDV by specifying the numerical distance between the upper quantile and the minimum value of the IP packet delay distribution. Because IPDV is a distribution statistic, its value cannot be apportioned in a simple additive fashion. If independence is assumed, the distribution of the end-to-end delay introduced by two (or more) concatenated networks can be calculated by convolving the individual network delay distributions. However, closed-form solutions for the reverse (“de-convolution”) problem exist only in special cases.

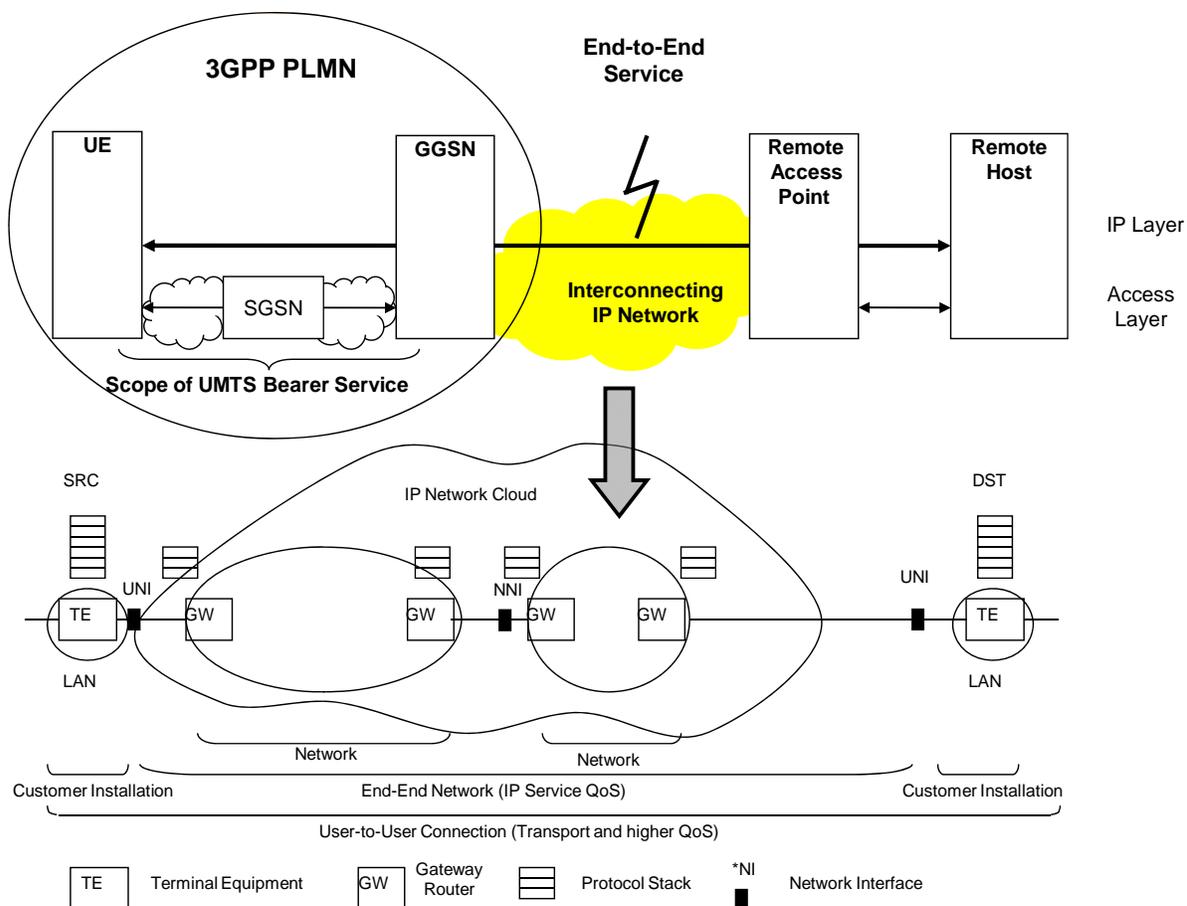


Figure 2 - Correspondence between UE and GGSN of UMTS and TE, UNI, and NNI of ITU-T.

9.1 Y.1541 to UMTS

For the example and assumptions outlined above, the QoS translator would map Y.1541 class 0 to the UMTS conversational class, selecting the 10^{-4} value for the SDU error ratio attribute.¹⁴ The UMTS SDU transfer delay value (100 ms maximum) might or might not meet the example objective for the UMTS network portion (50 ms mean), depending on the SDU transfer delay distribution. The UMTS SDU error ratio value (10^{-4}) would meet the Y.1541 IPLR and IPER objectives assumed for the UMTS network portion (5×10^{-4} , 5×10^{-5}), since the former parameter definition combines the Y.1541 packet loss and packet error outcomes. The UMTS conversational class requirement to “preserve time relation (variation) between information entities of the stream” would relate qualitatively to the Y.1541 IPDV objective, but the end-to-end objective would not be assured since the UMTS specification does not currently limit IPDV.

Y.1541 class 1 would be mapped to the UMTS streaming class, again selecting the 10^{-4} SDU error ratio value. The UMTS SDU transfer delay value (280 ms maximum) might or might not meet the example objective for the UMTS network portion (200 ms mean), depending on the delay distribution. The

¹⁴ See Table 7 for additional information on these and other mappings.

UMTS SDU error ratio value would meet the example Y.1541 IPLR and IPER objectives as described for class 0 above. The Y.1541 IPDV objective would be addressed qualitatively but without end-to-end assurance as noted above.

Y.1541 classes 2-4 could be mapped to the UMTS interactive class with a 10^{-4} SDU error ratio. The three Y.1541 classes could be mapped to different UMTS interactive class priority levels to reflect their different IPTD objectives; but as noted in TS 23-107, these relative priorities would not provide assured quality levels. If more assured IPTD values were required, Y.1541 classes 2-4 could be mapped to the UMTS conversational or streaming class. The SDU transfer delay limit of the UMTS conversational class (100 ms maximum) might or might not meet the example IPTD objective of class 2 (50 ms mean); it would definitely meet the assumed IPTD objectives of classes 3 and 4 (200 ms and 500 ms mean, respectively). Similarly, the SDU transfer delay limit of the UMTS streaming class (280 ms maximum) might or might not meet the assumed IPTD objectives of classes 2 and 3 (50 ms and 200 ms mean respectively), but would definitely meet the assumed IPTD objective of class 4 (500 ms mean).

Y.1541 class 5 would be mapped to the UMTS background class.

The mappings suggested above are probably the most reasonable ones for the stated example, and they could meet the postulated IPLR and IPER requirements for all of the Y.1541 classes. The suggested mappings would not meet the end-to-end delay requirements for some classes, and as noted, would place no quantitative bounds on end-to-end IPDV.

9.2 UMTS to Y.1541

Assuming the hypothetical conditions and mapping principles outlined above, the mapping from UMTS QoS classes to Y.1541 QoS classes would essentially reverse that described in 5.1 above. The UMTS conversational class would be mapped to Y.1541 class 0. The UMTS streaming class would be mapped to Y.1541 class 1.¹⁵ The UMTS interactive class could be mapped to Y.1541 class 2, 3, or 4 depending on the specified traffic handling priority; the Y.1541 classes would provide quantitative limits supporting up to three priority levels. The UMTS background class would be mapped to Y.1541 class 5.

These mappings are (again) probably the most reasonable ones for the stated example, but as noted they would not meet the end-to-end delay requirements for some classes and would place no quantitative bounds on end-to-end IPDV.

Although they do not themselves apply to end-to-end services, the more stringent SDU error ratio specifications presented in Tables 5 and 6 (10^{-5} , 10^{-6}) suggest that there may be a need for end-to-end IPLR and IPER objectives lower than those currently specified in Recommendation Y.1541 (10^{-3} , 10^{-4}). That possible need should be further investigated, and should be addressed in a future revision to Recommendation Y.1541 if it is validated for important user applications.

9.3 Summary Observations from the Mapping Example

A mapping between the currently defined ITU-T and 3GPP UMTS QoS specifications would improve interworking between wireless and IP-based wireline networks to some extent, but the result would be far from ideal. On the positive side, such a mapping could enable concatenated UMTS and IP networks to support quantitative end-to-end IP packet loss and packet error ratios (and associated values for UMTS SDU error ratio); to support assured end-to-end delay limits in some cases; and to relate end-to-end IPDV requirements with UMTS transfer capabilities in a qualitative way. Relatively good existing correspondence between the Y.1221 traffic parameters and TS 23-107 traffic attributes would make it possible to coordinate capacity assignment and traffic control between the two specification domains.

¹⁵ When UMTS and wireline IP networks are concatenated, the end-to-end delay performance of the conversational class may fall in Class 1.

On the negative side, a mapping between the existing QoS classes and attribute values would not support end-to-end QoS limits that will be required to provide real-time telephony and other important applications in wireline/wireless interworking situations. Transfer delay specifications are not relatable between the two domains because Y.1541 specifies IPTD as a mean value, while TS 23-107 specifies SDU transfer delay as a maximum. IPDV cannot currently be limited end to end because the UMTS specification does not define or quantitatively limit delay variation.¹⁶ TS 23-107's relative priorities would not support assured end-to-end QoS levels for interactive data applications. Combining the SDU loss and error outcomes in the TS 23-107 parameter SDU error ratio sacrifices some specificity that could be useful in UMTS network design and operation, although interoperation with IP-based networks is still possible as noted above.

All of the translations described in this paper will be more complicated in situations where the SDU and IP packet sizes substantially differ. IP QoS translation between 3GPP wireless and non-3GPP wireline networks would be greatly simplified if the SDU were defined to correspond to an IP packet in the relevant 3GPP specifications. It would be useful to make the observation intervals and payload sizes used in defining parameters and objectives in the two specification regimes equivalent as well.

10 3GPP LTE CLASSES AND THEIR RELATIONSHIP TO UMTS CLASSES

3GPP Technical Specification 23-203 [TS23-203] specifies nine QoS classes for the Long Term Evolution (LTE) architecture. Each Service Data Flow (SDF) is associated with one and only one QoS Class Identifier (QCI). Standardized characteristics are associated with each QCI. These characteristics are:

- Resource Type: Guaranteed Bit Rate (GBR) or Non-GBR
- Priority
- Packet Delay Budget
- Packet Error Loss Rate

The scope of the standardized QCI characteristics is depicted in [TS23-203 - Figure 6.1.7-1] as follows:

¹⁶ TS 23-107 states: "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." In general, a maximum transfer delay specification only limits delay variation to be less than the maximum. A different 3GPP specification indicates that delay variation may be defined and limited in Release 6.

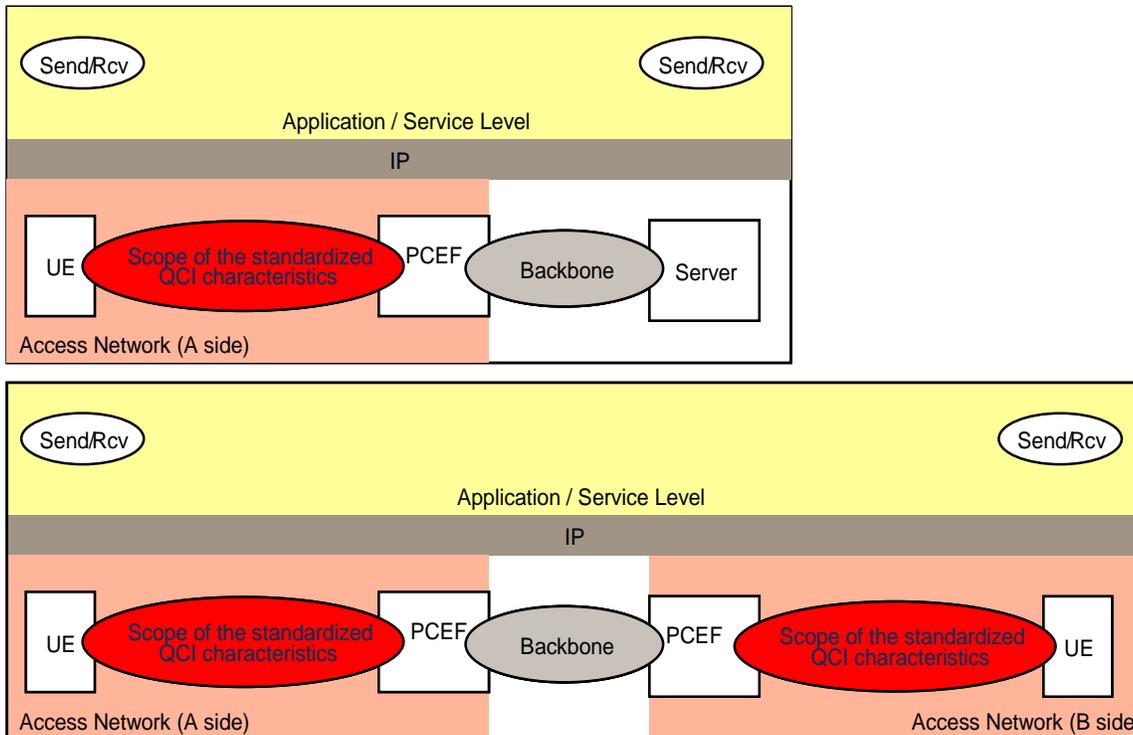


Figure 3 - Scope of QCI Characteristics for Client/Server (Upper Figure) and Peer/Peer (Lower Figure) Communication

The interpretation of the standardized QCI characteristics in [TS23-203] is stated as follows:

“The standardized characteristics are not signalled on any interface. They should be understood as guidelines for the pre-configuration of node specific parameters for each QCI. The goal of standardizing a QCI with corresponding characteristics is to ensure that applications/services mapped to that QCI receive the same minimum level of QoS in multi-vendor network deployments and in case of roaming. A standardized QCI and corresponding characteristics is independent of the UE's current access (3GPP or Non-3GPP).”

Standardized values for each QCI are specified in [TS23-203] – see Table 7 below.

Table 7 – Standardized QCI Characteristics

QCI	Resource Type	Priority	Packet Delay Budget (NOTE 1)	Packet Error Loss Rate (NOTE 2)	Example Services	
1 (NOTE 3)	GBR	2	100 ms	10 ⁻²	Conversational Voice	
2 (NOTE 3)		4	150 ms	10 ⁻³	Conversational Video (Live Streaming)	
3 (NOTE 3)		3	50 ms	10 ⁻³	Real Time Gaming	
4 (NOTE 3)		5	300 ms	10 ⁻⁶	Non-Conversational Video (Buffered Streaming)	
5 (NOTE 3)	Non-GBR	1	100 ms	10 ⁻⁶	IMS Signalling	
6 (NOTE 4)		6	300 ms	10 ⁻⁶	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)	
7 (NOTE 3)		7	100 ms	10 ⁻³	Voice, Video (Live Streaming) Interactive Gaming	
8 (NOTE 5)		8	9	300 ms	10 ⁻⁶	Video (Buffered Streaming) TCP-based (e.g., www, e-mail, chat, ftp, p2p file sharing, progressive video, etc.)
9 (NOTE 6)						

NOTE 1: A delay of 20 ms for the delay between a PCEF and a radio base station should be subtracted from a given PDB to derive the packet delay budget that applies to the radio interface. This delay is the average between the case where the PCEF is located "close" to the radio base station (roughly 10 ms) and the case where the PCEF is located "far" from the radio base station, e.g. in case of roaming with home routed traffic (the one-way packet delay between Europe and the US west coast is roughly 50 ms). The average takes into account that roaming is a less typical scenario. It is expected that subtracting this average delay of 20 ms from a given PDB will lead to desired end-to-end performance in most typical cases. Also, note that the PDB defines an upper bound. Actual packet delays - in particular for GBR traffic - should typically be lower than the PDB specified for a QCI as long as the UE has sufficient radio channel quality.

NOTE 2: The rate of non congestion related packet losses that may occur between a radio base station and a PCEF should be regarded to be negligible. A PELR value specified for a standardized QCI therefore applies completely to the radio interface between a UE and radio base station.

NOTE 3: This QCI is typically associated with an operator controlled service, i.e., a service where the SDF aggregate's uplink / downlink packet filters are known at the point in time when the SDF aggregate is authorized. In case of E-UTRAN this is the point in time when a corresponding dedicated EPS bearer is established / modified.

NOTE 4: This QCI could be used for prioritization of specific services according to operator configuration.

NOTE 5: This QCI could be used for a dedicated "premium bearer" (e.g. associated with premium content) for any subscriber / subscriber group. Also in this case, the SDF aggregate's uplink / downlink packet filters are known at the point in time when the SDF aggregate is authorized. Alternatively, this QCI could be used for the default bearer of a UE/PDN for "premium subscribers".

NOTE 6: This QCI is typically used for the default bearer of a UE/PDN for non privileged subscribers. Note that AMBR can be used as a "tool" to provide subscriber differentiation between subscriber groups connected to the same PDN with the same QCI on the default bearer.

The Packet Delay Budget (PDB) is the maximum delay encountered only in the 3GPP wireless domain as the time spent between the User Element (UE) and the Policy Charging and Enforcement Function (PCEF) [TS23-203]:

“The Packet Delay Budget (PDB) defines an upper bound for the time that a packet may be delayed between the UE and the PCEF. For a certain QCI the value of the PDB is the same in uplink and downlink. The purpose of the PDB is to support the configuration of scheduling and link layer functions. The PDB shall be interpreted as a maximum delay with a confidence level of 98 percent.”

Note the difference between the LTE PDB and the Y.1541 IPTD: the LTE PDB is a maximum value that is defined only for the 3GPP wireless domain whereas the Y.1541 IPTD is end-to-end and is a mean value.

Note further that no delay variation is specified for the LTE QCI classes.

The QCI Packet Error Loss Rate is defined as “an upper bound for a rate of non congestion related packet losses”.

The mapping between the nine LTE QCI classes and the four UMTS classes is specified in 3GPP Technical Specification TS 23-401 [TS23-401] as shown in Table 8.

Table 8 – Mapping Between LTE QCI Classes and UMTS Traffic Classes

LTE QCI	UMTS Traffic Class	Traffic Handling Priority	Signalling Indication	Source Statistics Descriptor
1	Conversational	N/A	N/A	Speech
2	Conversational	N/A	N/A	Unknown (NOTE 1)
3	Conversational	N/A	N/A	Unknown (NOTE 2)
4	Streaming	N/A	N/A	Unknown (NOTE 3)
5	Interactive	1	Yes	N/A
6	Interactive	1	No	N/A
7	Interactive	2	No	N/A
8	Interactive	3	No	N/A
9	Background	N/A	N/A	N/A

NOTE 1: When QCI 2 is mapped to pre-Rel-8 QoS parameter values, the Transfer Delay parameter is set to 150 ms. When pre-Rel-8 QoS parameter values are mapped to a QCI, QCI 2 is used for conversational/unknown if the Transfer Delay parameter is greater or equal to 150 ms.

NOTE 2: When QCI 3 is mapped to pre-Rel-8 QoS parameter values, the Transfer Delay parameter is set to 80 ms as the lowest possible value, according to [TS 23.107]. When pre-Rel-8 QoS parameter values are mapped to a QCI, QCI 3 is used for conversational/unknown if the Transfer Delay parameter is lower than 150 ms.

NOTE 3: When QCI 4 is mapped to pre-Rel-8 QoS parameter values, it is mapped to Streaming/Unknown. When pre-Rel-8 QoS parameter values are mapped to a QCI, Streaming/Unknown and Streaming/Speech are both mapped to QCI 4.

11 PROBLEMS OF SUPPORTING END-TO-END QoS (Y.1541) WITH LTE QCIs

The significant inconsistencies between UMTS and Y.1541 classes that were elucidated in section 9 will be at least as bad for LTE and Y.1541 classes. Accordingly a detailed elaboration of these incompatibilities will not be done, as the goal for a robust “mapping” as the method of interworking is not possible. Additionally, it is important not to propagate the concept of allocating end-to-end objectives (Y.1541) to LTE path segments in a static way (“budgeting”) as was done earlier, because this method has been strongly rejected by the ITU-T (and by service providers in particular).

So the issue is then, simply, how can the end-to-end QoS classes specified in Y.1541 be supported by the LTE QCIs defined for wireless network segments? Given the many, fundamental incompatibilities between the two, it is re-stating the obvious to say that a direct, meaningful and robust mapping is not possible. The “mapping” issue thus evolves into a question of how LTE domains might interwork with other domains (e.g., DiffServ-based) so that marked packets are handled in a way that they get the treatment needed so that any accumulated impairments over the end-to-end path do not result in unsatisfactory performance of the application supported by that packet flow.

An important aspect of this issue is that, while the 3GPP has chosen not to adopt or define the QoS parameters defined by others, it has fortunately specified [TS23-207] that 3GPP domains need to interwork with other IP domains implementing QoS mechanisms—specifically DiffServ.

While there is still work to be done to determine the exact details of how these different domains will interwork, at least having such interworking recognized as being a requirement is an important step forward. Thus, the focus now needs to shift to exactly how marked packets from IP backbones will be treated by LTE domains, and vice-versa.

12 CONCLUSIONS AND RECOMMENDATIONS

Section 9 illustrated the difficulties in deriving a meaningful and consistent mapping between the Y.1541 classes and the 3GPP UMTS classes. Examples of these incompatibilities, which also exist for LTE, can be summarized as follows:

- *Transfer Delay* - There are two potential difficulties as follows:
 - The 3GPP Transfer Delays are defined as *maxima* whereas the Y.1541 Transfer Delays are expressed as *mean* values. Where packet streams are involved, there can be a significant difference.
 - The 3GPP Transfer Delays are defined for the 3GPP domain only. The Y.1541 Transfer Delays are expressed as end-to-end, but the ITU experts were never able to agree on an allocation to segments like wireless access.
- *Delay Variation* – Delay Variation in the packet stream is specified for the Y.1541 classes (a distribution statistic) but is not addressed in any way for the 3GPP classes. For packet streams terminating in UMTS or LTE networks, one of two methods could be used to preserve the temporal characteristics of the signal: Either the delay variation is corrected as a form of interworking at the first point of entry to the 3GPP network; or else the packet stream has to be sent unprocessed directly to the terminal system—if and only if it has adequate buffers for the delay variation.

Given these incompatibilities regarding how the performance classes are defined, it is not currently feasible to derive a meaningful and well-defined mapping between the specific parameters used by the 3GPP for LTE domains and by the ITU for end-to-end IP paths.

Going forward, it is instead recommended to approach the end-to-end QoS problem by developing a detailed specification of interworking between different network segments (e.g., between LTE and optical IP backbones). This would necessitate, for example, developing an industry-wide set of default markings between things such as LTE QCI-related markings and Diffserv Code Points, or alternatively, development of interworking guidelines that can be used to derive appropriate SLA agreements between service providers. Methods like this might serve as the basis of the desirable interworking, or perhaps some other techniques can be explored—a matter for further study.