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Internet protocol aspects – Quality of service and network performance

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Network performance objectives for IP-based services

Recommendation ITU-T Y.1541

ITU-T

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Recommendation ITU-T Y.1541

Network performance objectives for IP-based services

Summary

This Recommendation defines classes of network quality of service (QoS) with objectives for Internet Protocol network performance parameters. Two of the classes contain provisional performance objectives. These classes are intended to be the basis for agreements among network providers, and between end users and their network providers.

Appendix I provides information about how asynchronous transfer mode (ATM) might support IP layer performance. Appendix II discusses alternatives for defining IP delay variation. Appendix III presents the hypothetical reference paths (HRP) against which the ITU-T Y.1541 QoS objectives were tested for feasibility. Appendix IV gives example computations of packet delay variation. Appendix V discusses issues that must be considered whenever IP measurements are made. Appendix VI describes the relationship between this Recommendation and the IETF-defined mechanisms for managing QoS. Appendix VII gives estimates of speech transmission quality for the hypothetical reference paths of Appendix III. Appendix VIII discusses digital television transport on IP networks. Appendix IX estimates transmission control protocol (TCP) file transfer performance on paths conforming to ITU-T Y.1541 objectives. Appendix X gives example calculations for combining delay variation measurements from multiple sections to estimate user network interface to user network interface (UNI-UNI) performance, and Appendix XI estimates the packet loss requirement for digital circuit emulation.

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FOREWORD

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The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

Compliance with this Recommendation is voluntary. However, the Recommendation may contain certain mandatory provisions (to ensure, e.g., interoperability or applicability) and compliance with the Recommendation is achieved when all of these mandatory provisions are met. The words "shall" or some other obligatory language such as "must" and the negative equivalents are used to express requirements. The use of such words does not suggest that compliance with the Recommendation is required of any party.

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As of the date of approval of this Recommendation, ITU had received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementers are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database at http://www.itu.int/ITU-T/ipr/.

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Network performance objectives for IP-based services

1 Introduction and Scope

1.1 Introduction

Customers require network performance levels that, when combined with their hosts, terminals, and other devices, satisfactorily support their applications. The adoption of IP-based network services [IETF RFC 791] has not changed this fact, except that networks must be constrained in terms of packet transfer performance parameters (as defined in [ITU-T Y.1540]).

Traditional application performance requirements are well-understood, but several key contributors are often beyond the network service provider's control (e.g., home networks, LAN, application gateways, terminals, hosts, and other customer devices). We note that objectives on the performance of customer equipment are available, such as [ITU-T P.1010] for VoIP terminals and gateways, and combining these objectives with specific network performance levels (as appendices of this Recommendation illustrate), a view of application performance can be directly related to network performance.

In response, service providers have agreed on network performance levels that they will work together to meet, and have codified the numerical objectives in this Recommendation. Agreement on levels of network performance is highly beneficial, because it constrains a critical and often dominating factor in application performance [ITU-T I.350].

The objectives are organized in sets called network quality of service (QoS) classes (in Table 1) that can be matched with well-designed customer equipment to satisfactorily support various applications (as indicated in Table 2). Classes with provisional objectives are found in Table 3. The number of classes has been deliberately kept small to simplify the engineering of paths traversing multiple operators' networks, so the objectives in each class must satisfy the needs of multiple applications. Readers of this Recommendation should plan for at least eight classes when considering protocol fields and values, since future expansion of the classes is possible.

The objective values result from analysis of key applications such as conversational telephony, multimedia conferencing, reliable data exchange using TCP, and digital television, in concert with network feasibility analysis. The appendices provide significant, detailed testimony as to how the objectives in the network QoS classes can be used to determine the end-to-end (application) quality provided. Another factor in the development of objective values has been network feasibility. When paths span wide geographical distances, very long propagation times will prevent low delay objectives from being met, thus additional classes are required to address these cases.

It is important to clarify how designers of new applications should make use of the ITU-T Y.1541 classes. Designers should consider the packet performance objectives as representative of well-managed IP-based networks and include mitigations for these impairment levels in their designs. Only after application requirements have been carefully rationalized and a range of impairment mitigations have been examined, should new QoS classes be considered to address unmet requirements.

The network QoS classes form an important link in the chain of developments required to assure end-to-end performance. They are part of the lexicon for QoS negotiation among users and networks, especially when signalling protocols communicate QoS requests on a dynamic basis.

Verification that the service meets network objectives is another key area of customer interest. This has been addressed here through recommended evaluation intervals, packet payload sizes, and other aspects useful to measurement designers. In addition, the UNI-UNI objectives are directly verifiable by users, in contrast with objectives that apply to non-user interfaces or utilize information unknown to customers, such as route distance.

1.2 Scope

This Recommendation specifies network (UNI-UNI) IP performance values for each of the performance parameters defined in [ITU-T Y.1540]. The specific performance values vary, depending on the network QoS class. This Recommendation defines eight network QoS classes, two of which are provisional. This Recommendation applies to international IP network paths (UNI-UNI). The network QoS classes defined here are intended to be the basis of agreements between end-users and network service providers, and between service providers. The classes should continue to be used when static agreements give way to dynamic requests supported by QoS specification protocols.

The QoS classes defined here support an extremely wide range of applications, including the following: conversational telephony, multimedia conferencing, digital video, and interactive data transfer. Designers of new user applications should first consider using the existing QoS classes, and possibly include technologies to mitigate packet transfer impairments in their design. If one or more packet transfer requirements is not satisfied, then a new class may be considered rather than modifying the current/stable classes. However, any desire for new classes must be balanced with the requirement of feasible implementation, and the number of classes must be small for implementations to scale in global networks. Thus, the extent of user application coverage may expand over time, and readers of this Recommendation are urged to consult the latest version, including the appendices.

Since the QoS classes have been developed to support user applications, their numerical objectives are likely to support the same applications on networks using alternate technologies or combinations of technologies, providing that the fundamental transfer unit has a one-to-one correspondence with IP packets as used here (no fragmentation), and that the overhead of the alternate technology is a non-substantial addition to the IP header (e.g., multi-protocol label switching (MPLS) label and Ethernet frame overhead).

The QoS objectives are primarily applicable when access link speeds are at the T1 or E1 rate and higher. This limitation recognizes that IP packet serialization time is included in the definition of IP packet transfer delay (IPTD), and that sub-T1 access rates can produce serialization times of over 100 ms for packets with 1500 octet payloads. Also, this Recommendation effectively requires the deployment of network QoS mechanisms on access devices in order to achieve the IP packet delay variation (IPDV) objective, especially when the access rate is low (e.g., T1 rate). Network designs may include lower access rates if:

1) Network planners understand the effect of additional serialization time on the user network interface (UNI) to UNI objective for IPTD.
2) QoS mechanisms limit the access contribution to IPDV, and the UNI to UNI objective for IPDV is met. The current IPDV objective is necessary to achieve high quality application performance, as Appendices III and VII clearly show.

This Recommendation provides the network QoS classes needed to support user-oriented QoS categories. Accordingly, this Recommendation is consistent with the general framework for defining quality of communication services in [ITU-T G.1000], and with the end-user multimedia QoS categories needed to support user applications given in [ITU-T G.1010].

NOTE – This Recommendation utilizes parameters defined in [ITU-T Y.1540] that can be used to characterize IP services that are provided using IPv4 and IPv6. [ITU-T Y.1540] was used as the foundation of MPLS performance parameters in [ITU-T Y.1561] and Ethernet service performance parameters in [ITU-T Y.1563]. Applicability or extension to other protocols is for further study.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T E.651] Recommendation ITU-T E.651 (2000), *Reference connections for traffic engineering of IP access networks*.

[ITU-T G.1000] Recommendation ITU-T G.1000 (2001), *Communications Quality of Service: A framework and definitions*.

[ITU-T G.1010] Recommendation ITU-T G.1010 (2001), *End-user multimedia QoS categories*.

[ITU-T I.350] Recommendation ITU-T I.350 (1993), *General aspects of quality of service and network performance in digital networks, including ISDNs*.

[ITU-T P.1010] Recommendation ITU-T P.1010 (2004), *Fundamental voice transmission objectives for VoIP terminals and gateways*.

[ITU-T Y.1221] Recommendation ITU-T Y.1221 (2010), *Traffic control and congestion control in IP-based networks*.

[ITU-T Y.1231] Recommendation ITU-T Y.1231 (2000), *IP Access Network Architecture*.

[ITU-T Y.1540] Recommendation ITU-T Y.1540 (2011), *Internet protocol data communication service – IP packet transfer and availability performance parameters*.

[ITU-T Y.1561] Recommendation ITU-T Y.1561 (2004), *Performance and availability parameters for MPLS networks*.

[ITU-T Y.1563] Recommendation ITU-T Y.1563 (2009), *Ethernet frame transfer and availability performance*.

[IETF RFC 791] IETF RFC 791 (1981), *Internet Protocol, DARPA Internet Program Protocol Specification*.

3 Abbreviations, acronyms and conventions

3.1 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

AF Assured Forwarding

ATM Asynchronous Transfer Mode

BBER Background Block Error Ratio

BE Best-Effort

CBR Constant Bit Rate

CDV Cell Delay Variation

CER	Cell Error Ratio
CLR	Cell Loss Ratio
CMR	Cell Misinsertion Ratio
CS	Circuit Section
DBW	Dedicated Bandwidth
DS	Differentiated Services
DST	Destination host
EF	Expedited Forwarding
ESR	Errored Second Ratio
FEC/I	Forward Error Correction and Interleaving
FIFO	First-In, First-Out
FTP	File Transfer Protocol
GW	Gateway
HRE	Hypothetical Reference Endpoint
HRP	Hypothetical Reference Path
IP	Internet Protocol
IPDV	IP packet Delay Variation
IPER	IP packet Error Ratio
IPLR	IP packet Loss Ratio
IPOT	Octet based IP packet Throughput
IPPT	IP Packet Throughput
IPRE	IP packet transfer Reference Event
IPRR	IP packet Reordering Ratio
IPTD	IP packet Transfer Delay
ISP	Internet Service Provider
LL	Lower Layers, protocols and technology supporting the IP layer
LP	Loss Period
LAN	Local Area Network
MP	Measurement Point
MPLS	Multi-Protocol Label Switching
MPLS-FRR	MPLS-Fast Re-Route
MTBA	Mean Time Between visible Artefacts
MTBISO	Mean Time between IP Service Outages
MTTISR	Mean Time to IP Service Restoral
NS	Network Section
NSE	Network Section Ensemble
NSP	Network Service Provider

OSPF	Open Shortest Path First
PDB	Per Domain Behaviour
PDH	Plesiochronous Digital Hierarchy
PHB	Per Hop Behaviour
PIA	Percent IP service Availability
PIU	Percent IP service Unavailability
PLC	Packet Loss Concealment
QoS	Quality of Service
R	Router
RSVP	Resource Reservation Protocol
RTP	Real-Time Transport Protocol
RTT	Round Trip Times
SACK	Selective Acknowledgements
SBW	Statistical Bandwidth
SDH	Synchronous Digital Hierarchy
SESR	Severely Errored Second Ratio
SPR	Spurious Packet Ratio
SRC	Source host
TC	Transfer Capability
TCP	Transmission Control Protocol
TDMA	Time Division Multiple Access
TE	Terminal Equipment
ToS	Type of Service
TS	Transport Stream
TTL	Time To Live
UDP	User Datagram Protocol
UNI	User Network Interface
VoIP	Voice over Internet Protocol
VTC	Video Teleconference

3.2 Conventions

E1	Digital hierarchy transmission at 2.048 Mbit/s
E3	Digital hierarchy transmission at 34 Mbit/s
M_{av}	The minimum number of packets recommended for assessing the availability state
N	The number of packets in a throughput probe of size N
pkt	IP datagram (IP packet)
T1	Digital hierarchy transmission at 1.544 Mbit/s

T3 Digital hierarchy transmission at 45 Mbit/s

T_{av} Minimum length of time of IP availability; minimum length of time of IP unavailability

T_{max} Maximum IP packet delay beyond which the packet is declared to be lost

4 Transfer capacity, capacity agreements, and the applicability of QoS classes

This clause addresses the topic of network transfer capacity (the effective bit rate delivered to a flow over a time interval), and its relationship to the packet transfer quality of service (QoS) parameters defined in [ITU-T Y.1540], and the objectives specified here.

Transfer capacity is a fundamental QoS parameter having primary influence on the performance perceived by end users. Many user applications have minimum capacity requirements; these requirements should be considered when entering into service agreements.

It is assumed that the user and network provider have agreed on the maximum access capacity that will be available to one or more packet flows in a specific QoS class (except the Unspecified class). A packet flow is the traffic associated with a given connection or connectionless stream having the same source host (SRC), destination host (DST), class of service, and session identification. Other documents may use the terms microflow or subflow when referring to traffic streams with this degree of classification. Initially, the agreeing parties may use whatever capacity specifications they consider appropriate, so long as they allow both network provider enforcement and user verification. For example, specifying the peak bit rate on an access link (including lower layer overhead) may be sufficient. The network provider agrees to transfer packets at the specified capacity in accordance with the agreed QoS class.

When the protocols and systems that support dynamic requests are available, the user will negotiate a traffic contract. Such a contract specifies one or several traffic parameters (such as those defined in [ITU-T Y.1221], or RSVP) and the QoS class, and applies to a specific flow.

The network performance objectives may no longer be applicable when there are packets submitted in excess of the capacity agreement or the negotiated traffic contract. If excess packets are observed, the network is allowed to discard a number of packets equal to the number of excess packets. Such discarded packets must not be included in the population of interest, which is the set of packets evaluated using the network performance parameters. In particular, discarded packets must not be counted as lost packets in assessing the network's IP packet loss ratio (IPLR performance). A discarded packet might be retransmitted, but then it must be considered as a new packet in assessing network performance.

It is a network privilege to define its response to flows with excess packets, possibly based on the number of excess packets observed. When a flow includes excess packets, no network performance commitments need be honoured. However, the network may offer modified network performance commitments.

5 Network performance objectives

This clause discusses objectives for the user information transfer performance of public IP services. These objectives are stated in terms of the IP layer performance parameters defined in [ITU-T Y.1540]. A summary of the objectives can be found in Table 1 together with its associated general notes. All values in Table 1 are stable.

NOTE – From a users' perspective, network QoS objectives contribute to only part of the transmission performance (e.g., mouth-to-ear quality in voice over IP). Appendix VII provides pointers to the appropriate Recommendations in this area.

5.1 General discussion of QoS

The QoS class definitions in Table 1 present bounds on the network performance between user network interfaces (UNI). As long as the users (and individual networks) do not exceed the agreed capacity specification or traffic contract, and a path is available (as defined in [ITU-T Y.1540]), network providers should collaboratively support these UNI-to-UNI bounds for the lifetime of the flow.

The actual network QoS offered to a given flow will depend on the distance and complexity of the path traversed. It will often be better than the bounds included with the QoS class definitions in Table 1.

Static QoS class agreements can be implemented by associating packet markings (e.g., Type of Service precedence bits or Diff-Serv Code Point) with a specific class.

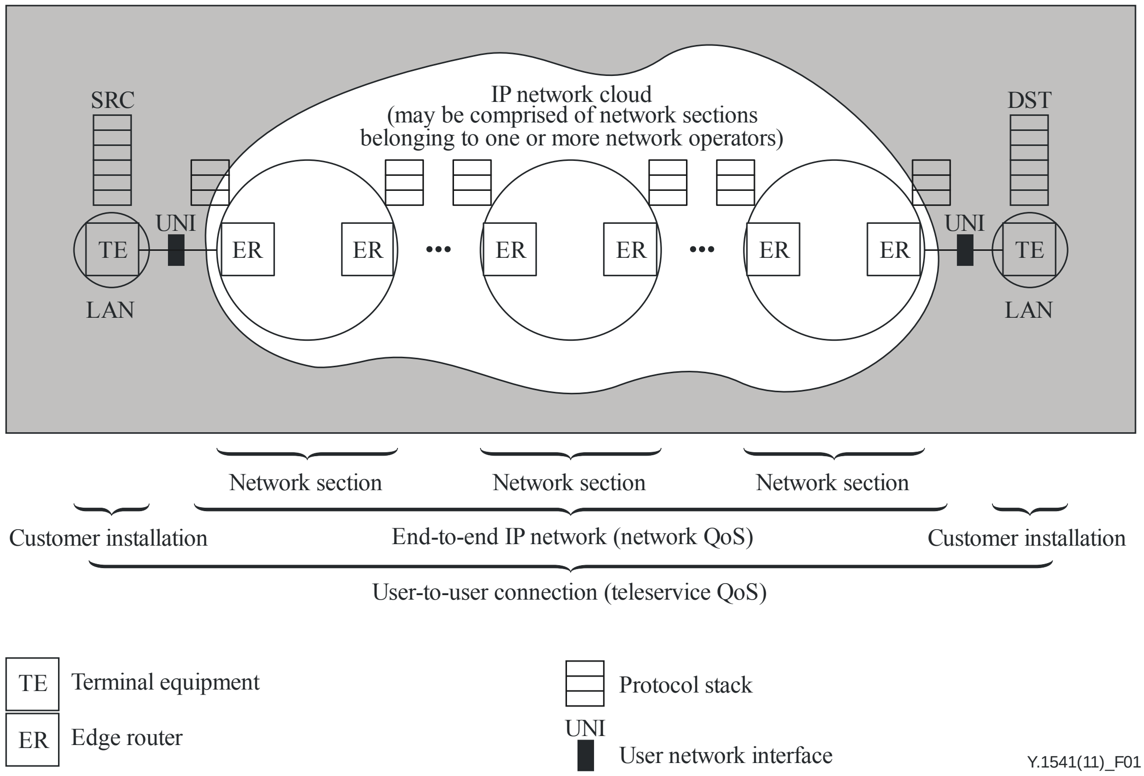
Protocols to support dynamic QoS requests between users and network providers, and between network providers, are under study. When these protocols and supporting systems are implemented, users or networks may request and receive different QoS classes on a flow-by-flow basis. In this fashion, the distinct performance needs of different services and applications can be communicated, evaluated, and acknowledged (or rejected, or modified).

5.2 Reference path for UNI to UNI QoS

Flows contain one or more packets and each packet in a flow follows a specific path from UNI to UNI.

NOTE – The phrase "End-to-End" has a different meaning in Recommendations concerning user QoS classes, where end-to-end means, for example, from mouth to ear in voice quality Recommendations. Within the context of this Recommendation, end-to-end is to be understood as from UNI-to-UNI.

The UNI-to-UNI performance objectives are defined for the IP performance parameters corresponding to the IP packet transfer reference events (IPRE). The UNI-to-UNI IP performance objectives apply from user network interface-to-user network interface in Figure 1. The UNI-to-UNI IP network path includes the set of network sections (NS) and inter-network links that provide the transport of IP packets transmitted from the UNI at the SRC side to the UNI at the DST side; the protocols below and including the IP layer (layer 1 to layer 3) may also be considered part of an IP network. Network sections (defined in [ITU-T Y.1540]) are synonymous with operator domains, and may include IP access network architectures as described in [ITU-T E.651] and [ITU-T Y.1231]. The reference path in Figure 1 is an adaptation of the ITU-T Y.1540 performance model.



NOTE – Customer Installation equipment (shaded area) is for illustrative purposes only.

Figure 1 – UNI-to-UNI reference path for network QoS objectives

The customer installation includes all terminal equipment (TE), such as a host and any router or LAN if present. There will be only one human user in some applications. It is important to note that specifications for TE and the user-to-user connection are beyond the scope of this Recommendation. The edge routers that connect with the terminal equipment may also be called access gateways.

Reference paths have the following attributes:

1) IP clouds may support user-to-user connections, user-to-host connections, and other endpoint variations.
2) Network sections may be represented as clouds with edge routers on their borders, and some number of interior routers with various roles.
3) The number of network sections in a given path may depend upon the class of service offered, along with the complexity and geographic span of each network section.
4) The scope of this Recommendation allows one or more network sections in a path.
5) The network sections supporting the packets in a flow may change during its life.
6) IP connectivity spans international boundaries, but does not follow circuit switched conventions (e.g., there may not be identifiable gateways at an international boundary if the same network section is used on both sides of the boundary).

5.3 Network QoS classes

This clause describes the currently defined network QoS classes. Each network QoS class creates a specific combination of bounds on the performance values. Any flow that satisfies all the performance objectives of a QoS class can be considered fully compliant with the normative recommendations of this Recommendation for that class. This clause includes guidance as to when each network QoS class might be used, but it does not mandate the use of any particular network QoS class in any particular context.

Table 1 – IP network QoS class definitions and network performance objectives

Network performance parameter	Nature of network performance objective	QoS Classes					
		Class 0	Class 1	Class 2	Class 3	Class 4	Class 5 Unspecified
IPTD	Upper bound on the mean IPTD (Note 1)	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 (Note 2)	50 ms (Note 3)	50 ms (Note 3)	U	U	U	U
IPLR	Upper bound on the packet loss probability	1×10^{-3} (Note 4)	1×10^{-3} (Note 4)	1×10^{-3}	1×10^{-3}	1×10^{-3}	U
IPER	Upper bound	1×10^{-4} (Note 5)					U

General notes:

The objectives apply to public IP networks. The objectives are believed to be achievable on common IP network implementations. The network providers' commitment to the user is to attempt to deliver packets in a way that achieves each of the applicable objectives. The vast majority of IP paths advertising conformity with [ITU-T Y.1541] should meet those objectives. For some parameters, performance on shorter and/or less complex paths may be significantly better.

An evaluation interval of 1 minute is suggested for IPTD, IPDV, and IPLR and, in all cases, the interval must be recorded with the observed value. Any minute observed should meet these objectives.

Individual network providers may choose to offer performance commitments better than these objectives.

"U" means "unspecified" or "unbounded". When the performance relative to a particular parameter is identified as being "U", ITU-T establishes no objective for this parameter and any default ITU-T Y.1541 objective can be ignored. When the objective for a parameter is set to "U", performance with respect to that parameter may, at times, be arbitrarily poor.

NOTE 1 – Very long propagation times will prevent low end-to-end delay objectives from being met. In these and some other circumstances, the IPTD objectives in classes 0 and 2 will not always be achievable. Every network provider will encounter these circumstances and the range of IPTD objectives in Table 1 provides achievable QoS classes as alternatives. The delay objectives of a class do not preclude a network provider from offering services with shorter delay commitments. According to the definition of IPTD in [ITU-T Y.1540], packet insertion time is included in the IPTD objective. This Recommendation suggests a maximum packet information field of 1500 bytes for evaluating these objectives.

NOTE 2 – The definition of the IPDV objective (specified in [ITU-T Y.1540]) is the 2-point IP packet delay variation. See [ITU-T Y.1540] and Appendix II for more details on the nature of this objective. For planning purposes, the bound on the mean IPTD may be taken as an upper bound on the minimum IPTD and, therefore, the bound on the $1 - 10^{-3}$ quantile may be obtained by adding the mean IPTD and the IPDV value (e.g., 150 ms in class 0).

Table 1 – IP network QoS class definitions and network performance objectives

NOTE 3 –This value is dependent on the capacity of inter-network links. Smaller variations are possible when all capacities are higher than the primary rate (T1 or E1), or when competing packet information fields are smaller than 1500 bytes (see Appendix IV).
NOTE 4 – The class 0 and 1 objectives for IPLR are partly based on studies showing that high quality voice applications and voice codecs will be essentially unaffected by a 10^{-3} IPLR.
NOTE 5 – This value ensures that packet loss is the dominant source of defects presented to upper layers, and is feasible with IP transport on ATM.

5.3.1 Nature of the network performance objectives

The objectives in Table 1 apply to public IP networks, between MPs that delimit the end-to-end IP network. The objectives are believed to be achievable on common implementations of IP networks.

The left-hand part of Table 1 indicates the statistical nature of the performance objectives that appear in the subsequent rows.

The performance objectives for IP packet transfer delay are upper bounds on the underlying mean IPTD for the flow. Although many individual packets may have transfer delays that exceed this bound, the average IPTD for the lifetime of the flow (a statistical estimator of the mean) should normally be less than the applicable bound from Table 1.

The performance objectives for 2-point IP packet delay variation (defined in [ITU-T Y.1540]) are based on an upper bound on the $1 - 10^{-3}$ quantile of the underlying IPTD distribution for the flow. The $1 - 10^{-3}$ quantile allows short evaluation intervals (e.g., a sample with 1000 packets is the minimum necessary to evaluate this bound). Also, this allows more flexibility in network designs where engineering of delay buildout buffers and router queue lengths must achieve an overall IPLR objective on the order of 10^{-3}. Use of lower quantile values will result in under-estimates of de-jitter buffer size, and the effective packet loss would exceed the overall IPLR objective (e.g., an upper quantile of $1 - 10^{-2}$ may have an overall packet loss of 1.1%, with IPLR = 10^{-3}). Other statistical techniques and definitions for IPDV are being studied as described in Appendix II, and Appendix IV discusses IPDV performance estimation.

The performance objectives for the IP packet loss ratios are upper bounds on the IP packet loss for the flow. Although individual packets will be lost, the underlying probability that any individual packet is lost during the flow should be less than the applicable bound from Table 1.

Objectives for less-prevalent packet transfer outcomes and their associated parameters are for further study, such as the spurious packet ratio (SPR) defined in [ITU-T Y.1540].

5.3.2 Evaluation intervals

The objectives in Table 1 cannot be assessed instantaneously. Evaluation intervals produce subsets of the packet population of interest (as defined in [ITU-T Y.1540]). Ideally, these intervals are:

- Sufficiently long to include enough packets of the desired flow, with respect to the ratios and quantiles specified.
- Sufficiently long to reflect a period of typical usage (flow lifetime), or user evaluation.
- Sufficiently short to ensure a balance of acceptable performance throughout each interval (intervals of poor performance should be identified, not obscured within a very long evaluation interval).
- Sufficiently short to address the practical aspects of measurement.

For evaluations associated with telephony, a minimum interval of the order of 10 to 20 seconds is needed with typical packet rates (50 to 100 packets per second), and intervals should have an upper limit on the order of minutes. A value of 1 minute is suggested and, in any case, the value used must be recorded with the observed value, along with any assumptions and confidence intervals. Any minute observed should meet the IPTD, IPDV, and IPLR objectives of Table 1. Minimally acceptable estimation methodologies are intended for future revisions of this Recommendation.

Methods to verify achievement of the objectives are for further study. Either continuous or non-continuous evaluation may be used. One possible method of measurement is given in [b-IETF RFC 3432], where the requirement for random measurement start times and evaluation intervals of finite length result in a non-continuous evaluation.

5.3.3 Packet size for evaluation

Packet size influences the results for most performance parameters. A range of packet sizes may be appropriate since many flows have considerable size variation. However, evaluation is simplified with a single packet size when evaluating IPDV, or when the assessment is targeting flows that support constant bit rate sources. Therefore, a fixed information field size is recommended. Information fields of either 160 octets or 1500 octets are suggested, and the field size used must be recorded. Also, an information field of 1500 octets is recommended for performance estimation of IP parameters when using lower layer tests, such as bit error measurements.

5.3.4 Unspecified (unbounded) performance

For some network QoS classes, the value for some performance parameters is designated "U". In these cases, ITU-T sets no objectives regarding these parameters. Network operators may unilaterally elect to assure some minimum quality level for the unspecified parameters, but ITU-T does not recommend any such minimum.

Users of these QoS classes should be aware that the performance of unspecified parameters can, at times, be arbitrarily poor. However, the general expectation is that mean IPTD will be no greater than 1 second.

NOTE – The word "unspecified" may have a different meaning in Recommendations concerning B-ISDN signalling.

5.3.5 Discussion of the IPTD objectives

Very long propagation times will prevent low UNI-to-UNI delay objectives from being met, e.g., in cases of very long geographical distances, or in cases where geostationary satellites are employed. In these and some other circumstances, the IPTD objectives in classes 0 and 2 will not always be achievable. It should be noted that the delay objectives of a class do not preclude a network provider from offering services with shorter delay commitments. Any such commitment should be explicitly stated. See Appendix III for an example calculation of IPTD on a global route. Every network provider will encounter these circumstances (either as a single network, or when working in cooperation with other networks to provide the UNI-to-UNI path), and the range of IPTD objectives in Table 1 provides achievable network QoS classes as alternatives. Despite different routing and distance considerations, related classes (e.g., classes 0 and 1) would typically be implemented using the same node mechanisms.

According to the definition of IPTD in [ITU-T Y.1540], packet insertion time is included in the IPTD objectives. This Recommendation suggests a maximum packet information field of 1500 bytes for evaluating the objectives.

5.3.6 Guidance on class usage

Table 2 gives some guidance for the applicability and engineering of the network QoS classes.

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 (Signalling)	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
NOTE – Any example application listed in Table 2 could also be used in class 5 with unspecified performance objectives, as long as the users are willing to accept the level of performance prevalent during their session.			

Traffic policing and/or shaping may also be applied in network nodes.

Table 2 conveys one of the principles of QoS class development, that the requirements of multiple applications are addressed by a single set of network performance objectives. This approach keeps the number of QoS classes small and manageable. Figure 2 below illustrates the approach of satisfying applications with common performance requirements in a single QoS class (for example, applications 2, 3, and 4 are all satisfied by QoS classes Y and Y*, where Y* modifies one or more performance requirements based on network feasibility).

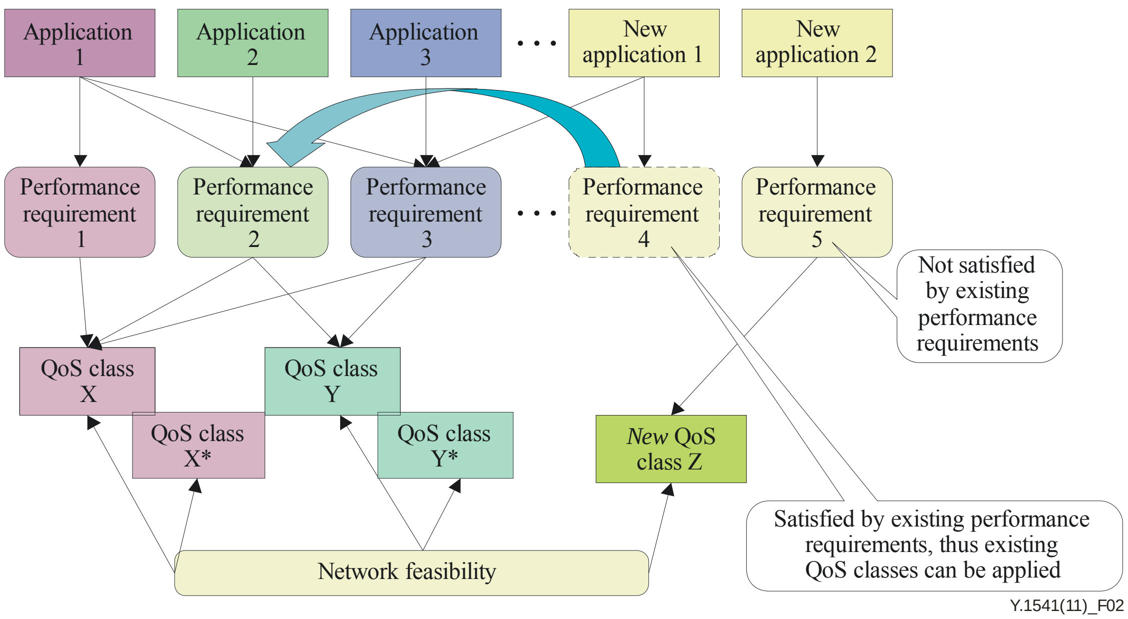


Figure 2 – Principle of multiple applications supported by a lesser number of QoS classes

With the growing number of user applications, it is important that new applications consider first how their requirements are similar to the applications already supported and seek to satisfy their requirements using one of the existing QoS classes.

5.3.7 Provisional network QoS classes

This clause presents a set of provisional network QoS classes. The distinction between these classes (see Table 3) and those in Table 1, is that the values of all objectives are provisional and they need not be met by networks until they are revised (up or down), based on real operational experience.

In this revision, there is agreement that the applications which provided the original rationale for the IPLR in Table 3 have evolved. In at least one case (IPTV) the performance objectives of Table 1 may now be sufficient (see Appendix VIII). New applications with strict performance needs are emerging, and their requirements are for further study. However, the provisional status of the classes and numerical objectives in Table 3 remains unchanged in this revision, pending further study and agreement.

Table 3 – Provisional IP network QoS class definitions and network performance objectives

<table>
<tr><th rowspan="2">Network performance parameter</th><th rowspan="2">Nature of network performance objective</th><th colspan="2">QoS Classes</th></tr>
<tr><th>Class 6</th><th>Class 7</th></tr>
<tr><td>IPTD</td><td>Upper bound on the mean IPTD</td><td>100 ms</td><td>400 ms</td></tr>
<tr><td>IPDV</td><td>Upper bound on the $1 - 10^{-5}$ quantile of IPTD minus the minimum IPTD (Note 1)</td><td colspan="2">50 ms</td></tr>
<tr><td>IPLR</td><td>Upper bound on the packet loss ratio</td><td colspan="2">1×10^{-5}</td></tr>
<tr><td>IPER</td><td>Upper bound</td><td colspan="2">1×10^{-6}</td></tr>
<tr><td>IPRR</td><td>Upper bound</td><td colspan="2">1×10^{-6}</td></tr>
<tr><td colspan="4">General notes:
Evaluation intervals for these classes should be 1 minute or longer. Evaluations should use 1500 byte payloads. An evaluation interval of 1 minute is suggested for IPTD, IPDV, and IPLR, and any minute observed should meet these objectives.
One rationale for the IP packet loss ratio (IPLR) objective was to minimize the effect of loss on TCP capacity, even when TCP parameters and the operating system have been tuned, and the large windows option has been utilized. Appendix IX provides background information on this and other support rationales. TCP selective acknowledgements (SACK), multi-path connections, and revised congestion control may reduce the loss objective, and they are subjects of further study.
The value for IPLR specified above, is not sufficient to support all the quality levels envisioned by the community of digital video users, and forward error correction and interleaving (FEC/I) or other forms of packet loss mitigation are required. Appendix VIII supplies background on the quality expectations of video transport users, and multiple forms of packet loss mitigation needed to produce low loss ratios.
The objective for IP packet error ratio (IPER) was set so as to contribute insignificantly to the overall packet loss.
The IP packet reordering ratio (IPRR) has been defined in [ITU-T Y.1540]. Reordered packets may appear as lost to a TCP sender, depending on the distance from their original positions. Therefore, the IPRR was set so as to contribute insignificantly to the overall packet loss.
New performance parameters for stream repair have been agreed and included in [ITU-T Y.1540]; they are applicable to the user applications considered in this Table. The role of these new metrics in network QoS performance objectives is for further study.</td></tr>
</table>

Table 3 – Provisional IP network QoS class definitions and network performance objectives

Network performance parameter	Nature of network performance objective	QoS Classes	
		Class 6	Class 7
The value for IPDV is under study, and contributions are invited to examine the rationale and feasibility of other (lower) values. In particular, if this table's scope was restricted to a higher category of access speeds than Table 1, then considerably lower IPDV objectives are possible (as Appendix IV shows). NOTE 1 – The definition of the IPDV objective (specified in [ITU-T Y.1540] is the 2-point IP packet delay variation. See [ITU-T Y.1540] and Appendix II for more details on the nature of this objective. For planning purposes, the bound on the mean IPTD may be taken as an upper bound on the minimum IPTD, and therefore the bound on the $1 - 10^{-5}$ quantile may be obtained by adding the mean IPTD and the IPDV value (e.g., 150 ms in class 6).			

These classes are intended to support the performance requirements of high bit rate user applications that have more stringent loss/error requirements than those supported by classes 0 through 4 in Table 1.

Discussions of broadcast quality television transport on IP may be found in Appendix VIII. Appendix IX estimates TCP file transfer performance on paths conforming to [ITU-T Y.1541] objectives. Appendix XI estimates the packet loss requirement for digital circuit emulation. Some of these appendices are expected to be revised following further study, and new appendices may be added to describe models of new user applications and the estimated performance requirements based on those models.

6 Availability objectives

This clause will include information about availability objectives based on the availability parameter defined in [ITU-T Y.1540]. The objectives require more study, since fundamental network design options are rapidly changing.

7 Achievement of the performance objectives

Further study is required to determine how to achieve these performance objectives when multiple network providers are involved. There are promising standards development activities that are intended to complete other aspects needed for UNI-UNI QoS assurance.

Clause 8 gives the relationships for concatenating the performance levels of two or more network sections to determine whether the UNI-UNI objectives are met.

8 Concatenating network sections and their QoS values

8.1 Introduction

This clause addresses the estimation of the UNI-UNI performance of a path, knowing the performance of sub-sections. The purpose is to provide standard relationships to compose these UNI-UNI estimates.

These relationships produce reasonably accurate estimates of the UNI-UNI performance. Errors in the estimation process are believed to be in balance with potential errors of the individual values themselves. When the values come from recent measurements or modelling activities, they can be subject to considerable error if conditions are not stationary, or the principal assumption of independence between network sections does not hold.

These relationships are intended to support accumulation of impairments facilitated by QoS signalling protocol(s). They must not be used to support allocation of UNI-UNI values.

8.2 Composing UNI-UNI values

8.2.1 Mean transfer delay

For the mean IP packet transfer delay (IPTD) performance parameter, the UNI-UNI performance is the sum of the means contributed by network sections.

The units of IPTD values are seconds, with resolution of at least 1 microsecond. If lesser resolution is available in a value, the unused digits shall be set to zero.

8.2.2 Loss ratio

For the IP packet loss ratio (IPLR) performance parameter, the UNI-UNI performance may be estimated by inverting the probability of successful packet transfer across *n* network sections, as follows:

$$IPLR_{UNI\text{-}UNI} = 1 - \{ (1 - IPLR_{NS1}) \times (1 - IPLR_{NS2}) \times (1 - IPLR_{NS3}) \times ... \times (1 - IPLR_{NSn}) \}$$

This relationship does not have limits on the parameter values, so it is preferred over other approximations, such as the simple sum of loss ratios. All measurements will use the same value of T_{max} (the waiting time to declare a packet lost).

The units of IPLR values are lost packets per total packets sent, with a resolution of at least 10^{-9}. If a lesser resolution is available in a value, the unused digits shall be set to zero.

8.2.3 Error packet ratio

For the IP packet error ratio (IPER) performance parameter, the UNI-UNI performance may be estimated by inverting the probability of error-free packet transfer across *n* network sections, as follows:

$$IPER_{UNI\text{-}UNI} = 1 - \{ (1 - IPER_{NS1}) \times (1 - IPER_{NS2}) \times (1 - IPER_{NS3}) \times ... \times (1 - IPER_{NSn}) \}$$

This relationship does not have limits on the parameter values, so it is preferred over other approximations, such as the simple sum of packet error ratios.

The units of IPER values are errored packets per total packets sent, with a resolution of at least 10^{-9}. If lesser resolution is available in a value, the unused digits shall be set to zero.

8.2.4 Relationship for delay variation

The relationship for estimating the UNI-UNI delay variation (IPDV) performance from the network section values, must recognize their sub-additive nature and it is difficult to estimate accurately without considerable information about the individual delay distributions. If, for example, characterizations of independent delay distributions are known or measured, they may be convolved to estimate the combined distribution. This detailed information will seldom be shared among operators, and may not be available in the form of a continuous distribution. As a result, the UNI-UNI IPDV estimation may have accuracy limitations. Since study continues in this area, the estimation relationship given below has been specified on a provisional basis, and this clause may change in the future, based on new findings or real operational experience.

The relationship for combining IPDV values is given below.

The problem under consideration can be stated as follows: estimate the quantile *t* of the UNI-UNI delay *T* as defined by the condition:

$$\Pr(T < t) = p$$

Step 1

Measure the mean and variance for the delay for each of *n* network sections. Estimate the mean and variance of the UNI-UNI delay by summing the means and variances of the component distributions.

$$\mu = \sum_{k=1}^{n} \mu_k$$

$$\sigma^2 = \sum_{k=1}^{n} \sigma_k^2$$

Step 2

Measure the quantiles for each delay component at the probability of interest, $p = 0.999$. Estimate the corresponding skewness and third moment using the formula shown below, where $x_{0.999} = 3.090$ is the value satisfying $\Phi(x_{0.999}) = 0.999$, where Φ denotes the standard normal (mean 0, variance 1) distribution function.

$$\gamma_k = 6 \cdot \frac{x_p - \frac{t_k - \mu_k}{\sigma_k}}{1 - x_p^2}$$

$$\omega_k = \gamma_k \cdot \sigma_k^3$$

Assuming independence of the delay distributions, the third moment of the UNI-UNI delay is just the sum of the network section third moments.

$$\omega = \omega_1 + \omega_2 + \omega_3 + \ldots = \sum_{k=1}^{n} \omega_k$$

The UNI-UNI skewness is computed by dividing by σ^3 as shown below.

$$\gamma = \omega / \sigma^3$$

Step 3

The estimate of the 99.9-th percentile ($p = 0.999$) of UNI-UNI delay *t* is as follows.

$$t = \mu + \sigma \cdot \left\{ x_p - \frac{\gamma}{6}\left(1 - x_p{}^2\right) \right\}$$

where $x_p = x_{0.999} = 3.090$.

As stated earlier, the nature of the IPDV objective is the upper bound on the $1 - 10^{-3}$ quantile of IPTD minus the minimum IPTD (i.e., the distribution of IPDV is normalized to the minimum IPTD). The units of IPDV values are seconds, with a resolution of at least 1 microsecond. If a lesser resolution is available in a value, the unused digits shall be set to zero.

8.3 Impairment accumulation procedures

There are two principal ways in which the relationships above may be applied to estimate the UNI-UNI performance levels. Both are acceptable.

When the values from all network sections in the path are available in one place for computation, then they should be used in the relationships above as individual values. In a signalling protocol, the individual values would be collected from the source to the destination and communicated to the entity responsible for computation and action on the result.

The values may also be accumulated each time a new value is available. In this case, the relationships above are used to combine the cumulative estimate with the value from the current network (or router, if that is the basis of combination). The calculated estimate becomes the new cumulative value, and would be communicated further along the path to the destination.

9 Security

This Recommendation does not specify a protocol, and there are limited areas where security issues may arise. All are associated with verification of the performance objectives with measurement system implementations.

Measurement systems that assess the performance of networks to determine compliance with numerical objectives defined in this Recommendation must limit the measurement traffic to appropriate levels to avoid abuse (e.g., denial of service attack). Parties participating in measurement activities, including administrations or operators of networks that carry the traffic, should agree in advance on acceptable traffic levels.

Systems that monitor user traffic for the purpose of measurement must maintain the confidentiality of user information.

Systems that attempt to make measurements may employ techniques (e.g., cryptographic hash) to determine if additional traffic has been inserted by an attacker appearing to be part of the population of interest.

Appendix I

ATM network QoS support of IP QoS

(This appendix does not form an integral part of this Recommendation.)

This appendix presents an analysis of mapping IP performance parameters on top of the ATM QoS class objectives as specified in [b-ITU-T I.356]. The purpose of this analysis is to estimate IP level performance obtained when ATM is used as the underlying transport. Because there are no routers considered in this analysis, the IP performance numbers shown here are the best that can be expected. In scenarios where intermediate routers exist, the IP performance will be worse.

Table I.1 – IP packet loss ratio (IPLR) values corresponding to ATM QoS service classes 1 and 2 (IP packet size 40 bytes; all errored packets are assumed lost)

ATM QoS class	Delivered ATM CER	Delivered ATM CLR	Resulting IPLR
1	4.00 E-06	3.00 E-07	4.30 E-06
2		1.00 E-05	1.40 E-05

Table I.2 – IP packet transfer delay (IPTD) values for a flow over a national portion and an end-to-end flow

Network portion	IPTD resulting from ATM QoS class 1 (no delay from IP routers)
National portion	~27.4 ms
End-to-end	400 ms

Note that class 0 and class 2 mean IPTD cannot be met on the 27 500 km reference connection of [b-ITU-T I.356].

The value of the cell error ratio (CER) in the ATM classes is 4×10^{-6}. If IP packets are long (1500 bytes) and errored cells cause errored IP packets, the value of the IP packet error ratio will be about 10^{-4}.

Cell misinsertion ratio (CMR) is currently specified as 1/day. The implications of CMR on SPR requires more study.

Appendix II

IP delay variation parameter definition considerations

(This appendix does not form an integral part of this Recommendation.)

This appendix discusses considerations for the definition of IPDV and the use of alternate statistical methods for the IPDV objective.

In order to provide guidance to designers of jitter buffers in edge equipment, the parameter(s) need to capture the effects of the following on IPDV:

- routine congestion in the network (high frequency IPTD variations);
- TCP windowing behaviour (low frequency IPTD variations);
- periodic and aperiodic variations in average network loading (low frequency IPTD variations);
- routing update effects on IPTD (instantaneous (and possibly large) changes in IPTD).

The current definition of IP delay variation is:

$$IPDV = IPTD_{upper} - IPTD_{min}$$

where:

$IPTD_{upper}$ is the $1 - 10^{-3}$ quantile of IPTD in the evaluation interval;

$IPTD_{min}$ is the minimum IPTD in the evaluation interval.

The definition of IPDV is based on the reference events given in clause 6.2.2 of [ITU-T Y.1540]. Here, the nominal delay is based on the packet with the minimum one-way delay (as an alternative to the first packet, or the average of the population as the nominal delay).

The specification of the $1 - 10^{-3}$ quantile (equivalent to the 99.9th percentile) is influenced by the size of the packet sample in a 1 minute measurement interval and the IPLR objective $\leq 10^{-3}$, resulting in an overall loss ratio objective of about 10^{-3}. Smaller quantiles would add more losses, as shown below.

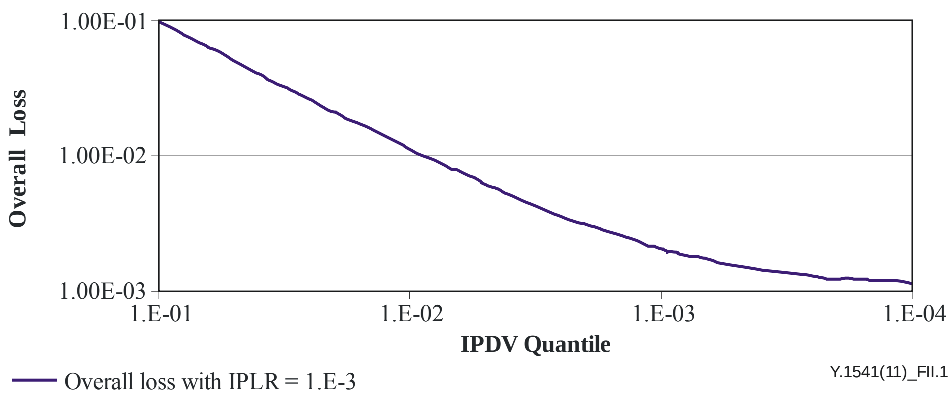


Figure II.1 – Effect of different IPDV quantiles on overall loss when IPLR = 0.001

An example alternate definition of IP delay variation is given here. IP delay variation may be defined as the maximum IPTD minus the minimum IPTD during a given short measurement interval.

$$IPDV = IPTD_{max} - IPTD_{min}$$

where:

$IPTD_{max}$ is the maximum IPTD recorded during a measurement interval;

$IPTD_{min}$ is the minimum IPTD recorded during a measurement interval.

Several values of IPDV are measured over a large time interval, comprising several short measurement intervals. The 95th percentile of these IPDV values is expected to meet a desired objective. This is a simple and fairly accurate method for calculating IPDV in real-time. The actual value of the measurement interval is for further study. The measurement interval influences the ability of the metric to capture low and high frequency variations in the IP packet delay behaviour.

Appendix III

Example hypothetical reference paths for validating the IP performance objectives

(This appendix does not form an integral part of this Recommendation.)

This appendix presents the hypothetical reference paths considered in validating the feasibility of the end-to-end performance objectives presented in clause 5. These hypothetical reference paths (HRP) are examples only. The material in this appendix is not normative and does not recommend or advocate any particular path architectures.

Each packet in a flow follows a specific path. Any flow (with one or more packets on a path) that satisfies the performance objectives of clause 5 can be considered fully compliant with the normative recommendations in the main body of the Recommendation.

The end-to-end performance objectives are defined for the IP performance parameters corresponding to the IP packet transfer reference events (IPREs). The end-to-end IP network includes the set of network sections (NS) and inter-network links that provide the transport of IP packets transmitted from SRC to DST; the protocols below and including the IP layer (layer 1 to layer 3) within the SRC and DST may also be considered part of an IP network.

NOTE – For information concerning the effects on end-to-end quality as perceived by the user of the delay figures given by the presented hypothetical reference paths refer to Appendix VII.

III.1 Number IP nodes in the HRP

HRPs have similar attributes to the reference path of clause 5.

Network sections are defined (in [ITU-T Y.1540]) as sets of hosts together with all of their interconnecting links that together provide a part of the IP service between an SRC and a DST, and are under a single (or collaborative) jurisdictional responsibility. Network sections are synonymous with operator domains. Network sections may be represented as clouds with edge routers on their borders, and some number of interior routers with various roles. In this case, HRPs are equivalent to the "path digest" of [b-IETF RFC 2330].

Each NS may be composed of IP nodes performing access, distribution, and core roles, as illustrated in Figure III.1.

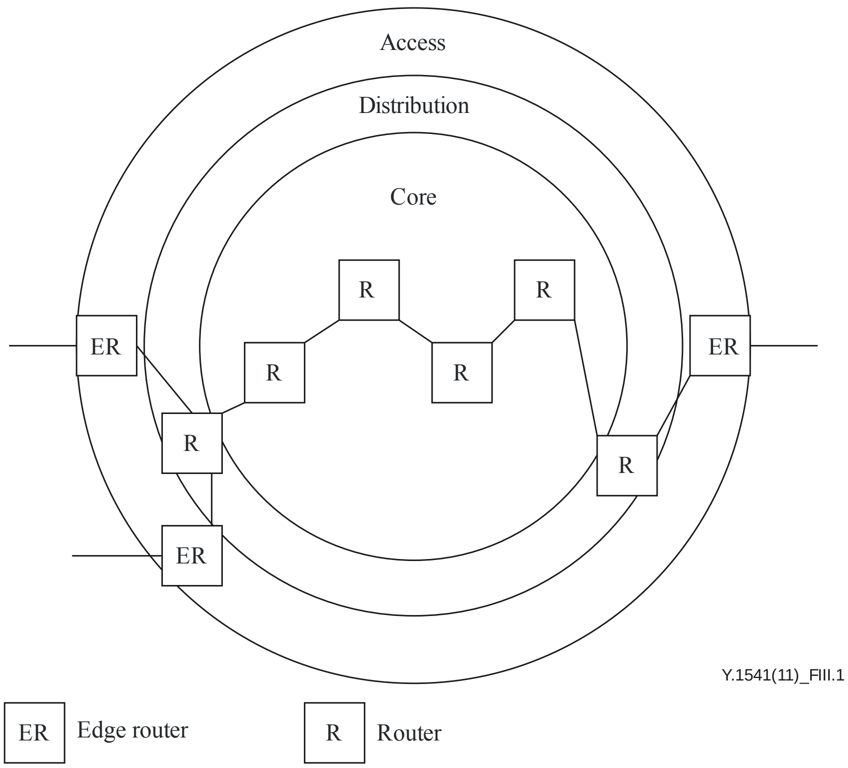


Figure III.1 – Role of IP nodes in a network section

Note that one or more routers are needed to complete each role, and the core path illustrated has four routers in tandem. A path through an NS could encounter as few as three routers, or as many as eight in this example.

Router contribution to various parameters may vary according to their role. Edge routers generally perform one of two roles, as access gateway routers or internetworking gateway routers.

Table III.1 – Examples of typical delay contribution by router role

Role	Average total delay (sum of queueing and processing)	Delay variation
Access gateway	10 ms	16 ms
Internetworking gateway	3 ms	3 ms
Distribution	3 ms	3 ms
Core	2 ms	3 ms

NOTE – Internetworking gateways typically have performance characteristics different from access gateways.

Route length calculation

If the distance-based component is proportional to the actual terrestrial distance, plus a proportional allowance, for a typical physical-route-to-actual-distance ratio. The route length calculation used here is based on [b-ITU-T G.826], and only for the long distances considered here. If D_{km} is the air-route distance between the two MPs that bound the portion, then the route length calculation is:

- if $D_{km} > 1200$, $R_{km} = 1.25 \times D_{km}$

The above does not apply when the portion contains a satellite hop.

III.2 Example computations to support end-end class 0 and class 1 delay

Class X network delay computation (X = 0 through 4)

This clause calculates the IPTD for any path portion supporting a QoS class X flow. When a flow portion does not contain a satellite hop, its computed IPTD is (using the delay for optical transport given in [b-ITU-T G.114]):

$$\text{IPTD (in microseconds)} \leq (R_{km} \times 5) + (N_A \times D_A) + (N_D \times D_D) + (N_C \times D_C) + (N_I \times D_I)$$

In this formula:

- R_{km} represents the route length assumption computed above.
- ($R_{km} \times 5$) is an allowance for "distance" within the portion.
- N_A, N_D, N_C, and N_I represent the number of IP access gateway, distribution, core and internetwork gateway nodes respectively; consistent with the network section example in Figure III.1.
- D_A, D_D, D_C, and D_I represent the delay of IP access gateway, distribution, core and internetwork gateway nodes respectively; consistent with the values for class X (e.g., Table III.1).

Maximum IPDV may be calculated similarly.

As an example of this calculation, consider the following HRP. This path contains two IP networks, and an internetworking point.

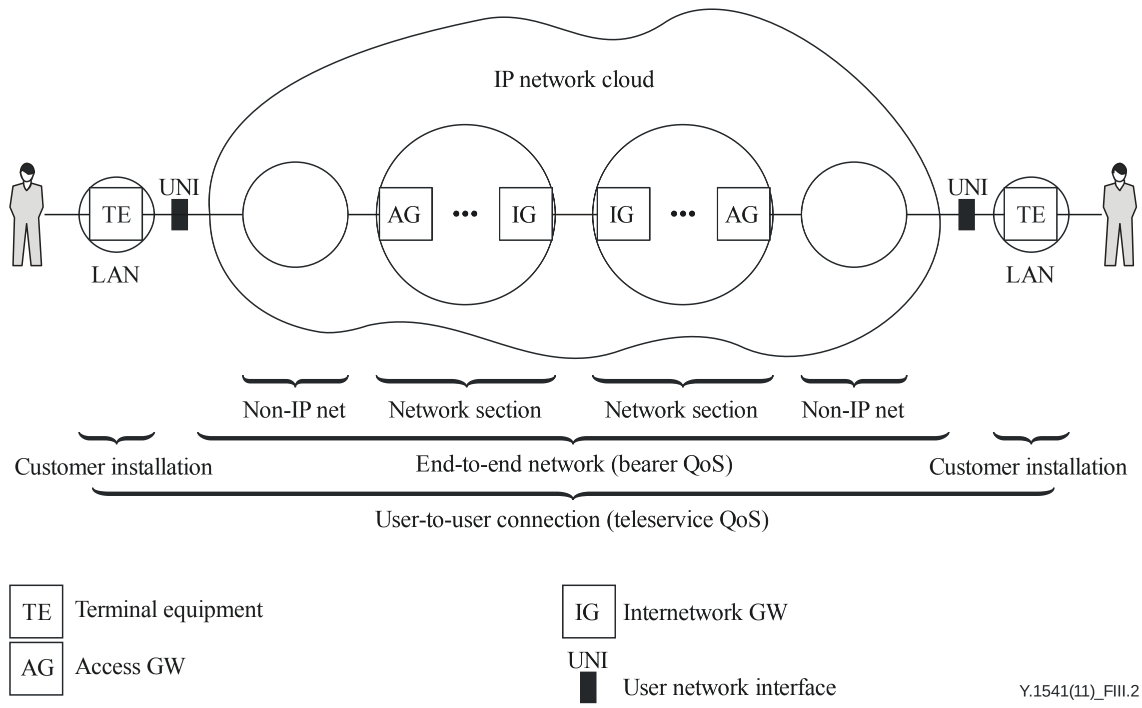


Figure III.2 – Hypothetical reference path for QoS class 0

Interior router configurations are not shown in the hypothetical reference path (HRP) of Figure III.2. The number of core and distribution routers can be found in Table III.2.

Assumptions:

1) Distance used is approximately the span between Daytona Beach and Seattle (US Diagonal, longer than Lisbon to Moscow).

2) Access links are T1 capacity, others are larger than T1 (e.g., OC-3).
3) Largest packet size is 1500 bytes, and VoIP packet size is 200 bytes.
4) Non-IP networks are needed between the NI and Access GW.

Table III.2 – Analysis of example class 0 path

Element	Unit	IPTD/ Unit	Ave IPTD	IPDV/ Unit	Max IPDV
Distance	4070 km				
Route	5087.5 km		25		
Insertion Time	200 bytes (1500 bytes)		1 (8)		
Non IP Net 1			15		0
IP Net 1					
Access, N_A	1	10	10	16	16
Distribution, N_D	1	3	3	3	3
Core, N_C	2	2	4	3	6
Internetwork GW, N_I	1	3	3	3	3
IP Net 2					
Access, N_A	1	10	10	16	16
Distribution, N_D	1	3	3	3	3
Core, N_C	4	2	8	3	12
Internetwork GW, N_I	1	3	3	3	3
Non IP Net 2			15		0
Total, ms			**100**		**62**

Table III.2 gives the HRP configuration in terms of number and type of routers, distance, and contribution of all HRP components to delay (IPTD) and delay variation (IPDV). Note that the calculation of maximum IPDV here is very pessimistic (assuming the worst case addition of each node), and is therefore greater than the specification of IPDV in the body of this Recommendation.

III.3 Example end-end class 1 delay computation

Class 1 is available to support longer path lengths and more complex network paths. Using the same assumptions as described in Table III.2, but with a 12 000 km distance, the mean IPTD will be 150 ms, and an R-value of approximately 83 is possible.

In a second example, we add a transit IP network section, for a total of 3 NS.

Table III.3 – Example calculation for class 1 path

Element	Unit	IPTD/ Unit	Ave IPDT	IPDV/ Unit	Max IPDV
Distance	km				
Route	27 500 km		138		
Insertion Time	200 bytes (1500 bytes)		1 (8)		
Non IP Net 1			15		0
IP Net 1					
Access, N_A	1	10	10	16	16
Distribution, N_D	1	3	3	3	3
Core, N_C	2	2	4	3	6
Internetwork GW, N_I	1	3	3	3	3
IP Net 2					
Distribution, N_D	2	3	6	3	6
Core, N_C	4	2	8	3	12
Internetwork GW, N_I	2	3	6	3	6
IP Net 3					
Access, N_A	1	10	10	16	16
Distribution, N_D	1	3	3	3	3
Core, N_C	4	2	8	3	12
Internetwork GW, N_I	1	3	3	3	3
Non IP Net 2			15		0
Total, ms			**233**		**86**

Table III.3 gives the HRP configuration in terms of number and type of routers, distance, and contribution of all HRP components to delay (IPTD) and delay variation (IPDV).

III.4 Example computations to support end-end class 4 delay

Following the form of the calculation above, we can expand the number of NS having delay contributions given in Table III.1, or we can expand the contributions as follows:

Table III.4 – Class 4 delay contribution by router role

Role	Average total delay (sum of queueing and processing)
Access Gateway	200 ms
Internetworking Gateway	64 ms
Distribution	64 ms
Core	3 ms

Here, with a route length of 27 500 km, the average 1-way delay would be 884 ms (using the HRP with node configuration as described in Table III.2).

III.5 Loading within the HRP

The fraction of each transmission link occupied by active packets is one of the factors to be considered in the HRPs. The load levels at which the network will continuously operate is another factor.

III.6 Geostationary satellites within the HRP

The use of geostationary satellites was considered during the study of the HRPs. A single geostationary satellite can be used within the HRPs and still achieve end-to-end objectives on the assumption that it replaces significant terrestrial distance, multiple IP nodes, and/or transit network sections.

The use of low and medium-Earth orbit satellites was not considered in connection with these HRPs.

When a path contains a satellite hop, this portion will require an IPTD of 320 ms, to account for a low earth station viewing angle, low rate TDMA systems, or both. In the case of a satellite possessing on-board processing capabilities, 330 ms of IPTD is needed to account for on-board processing and packet queueing delays.

It is expected that most HRPs which include a geostationary satellite will achieve IPTD below 400 ms. However, in some cases, the value of 400 ms may be exceeded. For very long paths to remote areas, network providers may need to make additional bilateral agreements to improve the probability of achieving the 400 ms objective.

Appendix IV

Example calculations of IP packet delay variation

(This appendix does not form an integral part of this Recommendation.)

This appendix provides material to facilitate the calculation of the IP packet delay variation (IPDV) for those IP QoS classes where a rather strict value for the IPDV is specified, i.e., IP QoS class 0 and class 1.

For the calculations here it is assumed that a network operator provides a choice of different IP QoS classes also including QoS classes for which no IPDV objectives are specified. This mix of properties motivates the notion of "delay variation-sensitive" flows (e.g., QoS class 0 and class 1) and "delay variation-insensitive" flows (e.g., QoS classes 2, 3, 4, and 5). It is further assumed that an operator providing such a mix of QoS classes, makes a reasonable effort to separate the variation-sensitive from the variation-insensitive flows. Key elements in such an effort consist of a packet scheduling strategy and additional traffic control measures. For the calculations in this appendix, it is assumed that packets of variation-sensitive flows are scheduled with non-pre-emptive priority over packets from variation-insensitive flows, and that the scheduling within each of these two categories is FIFO.

NOTE – This simple assumption only serves the purpose to arrive at a 'calculable' model. Other packet scheduling strategies (such as weighted fair queueing) or traffic control measures, are not excluded. It is further assumed that the performance of other approaches is either better, or not much worse than, the performance of the approach used for these calculations.

IV.1 Contributors to IP packet delay variation

The following factors are taken into account as the most significant contributors to IP packet delay variation (IPDV) for the variation-sensitive flows:

- Variable delay because the processing delay for the packet's forwarding decision (routing look-up) is not a single fixed value but may vary from packet-to-packet.
- Variable delay because the packet has to wait behind other variation-sensitive packets which arrived earlier.
- Variable delay because the packet has to wait for the service completion of a variation-insensitive packet which arrived earlier and is already in service.

IV.2 Models and calculation procedures to establish an upper bound to the IPDV

IV.2.1 Delay variation due to routing look-up

For an arriving packet, the router needs to establish the outgoing port to which the packet is to be forwarded, based on the IP address. The time required for this forwarding decision may vary from packet-to-packet.

High performance routers may cache recently used IP addresses to speed-up this process for subsequent packets. Then, all packets of a flow, except the first one, are expected to experience a short look-up delay and very small variation between them. Though, strictly, the longer delay of the first packet contributes to the IPDV, the exceptional delay of the first packet is disregarded in these calculations because it is a 'one off' event and its effect will vanish in flows with a relative long duration (e.g., a VoIP flow).

It is expected that the packet-to-packet variation in the routing look-up delay is not more than a few tens of microseconds in each router. For the calculations, the variability is assumed to be less than 30 µs per router.

Because there is little information available about the distribution of this delay component, the aggregated variability over several routers in tandem is set to the sum of the individual variabilities, i.e., statistical effects are not taken into account for this IPDV component.

IV.2.2 Delay variation due to variation-sensitive packets

A variation-sensitive packet will have to wait for other variation-sensitive packets to be serviced which have arrived earlier (FIFO discipline). Each variation-sensitive flow is modelled as a continuous flow of packets with negligible 1-point IP packet delay variation, comparable to the concept of 'negligible CDV' used for a CBR stream of ATM cells (see [b-ITU-T E.736]).

For the calculations, it is further assumed that all variation-sensitive packets have a fixed size of 1500 bytes. This allows the well-known M/D/1 queueing model (see [b-ITU-T E.736]) to be applied for the calculation of this component in the packet delay variation. The fixed service time is determined by the assumed fixed packet size (1500 bytes) and the router's output link rate, e.g., 80.13 µs on an STM-1 link.

For the aggregation of this delay component over several routers in tandem, the convolution of the relevant delay distributions is to be used, taking into account different output link rates when applicable. The lower quantile is assumed to be zero, the higher $(1 - 10^{-3})$ quantile can be approximated accurately using large deviations theory, in particular the Bahadur-Rao estimate as worked out in [b-Mandjes].

Figure IV.1 illustrates the result of such calculations. It shows the $(1 - 10^{-3})$ delay variation quantile for the aggregated delay component due to interference from variation-sensitive traffic, for different load levels of variation-sensitive traffic and for different numbers of router hops in tandem.

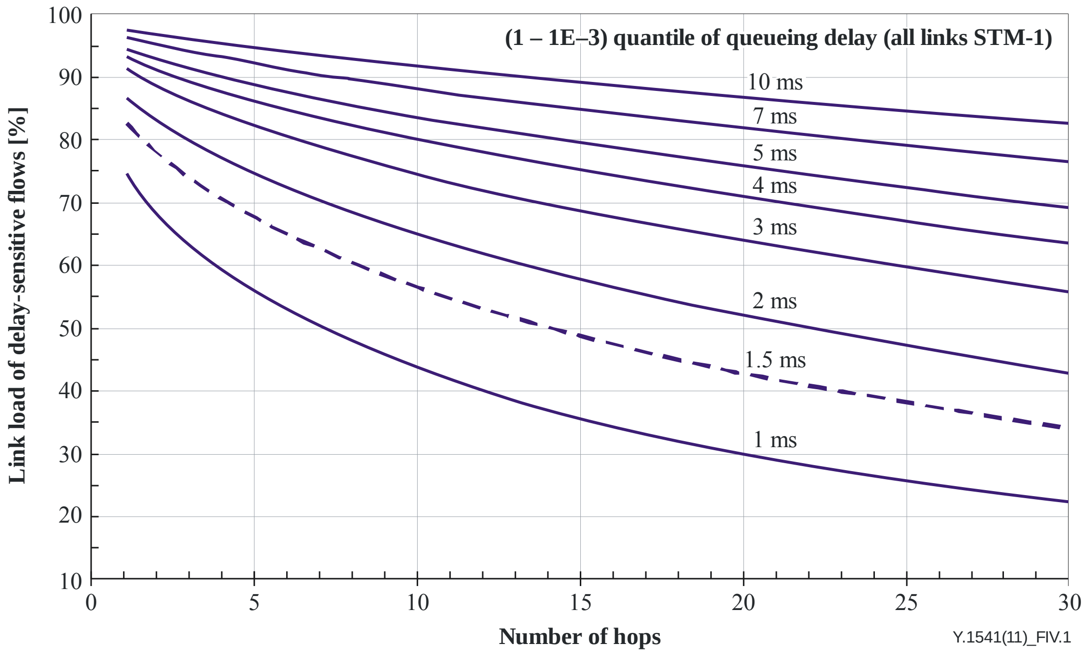


Figure IV.1 – The $(1 - 10^{-3})$ quantile of the aggregated queueing delay component due to variation-sensitive traffic for different levels of the variation-sensitive traffic and for different numbers of router hops in tandem

Figure IV.1 assumes that all links in the network are STM-1 and all links show the same load level for variation-sensitive traffic. If one or more links have a higher capacity than STM-1, the resulting end-to-end delay will be lower; if some links have a lower capacity, the resulting end-to-end delay will be higher. These effects can be calculated (see clause IV.2.4) but cannot easily be reflected in Figure IV.1.

Finally, it is assumed that in a network which supports both variation-sensitive and variation-insensitive traffic, the load of variation-sensitive traffic on a link is not more than 50% of the link to reflect the observed trend towards 'more data than voice'. Then, from Figure IV.1 it can be derived that this delay component contributes no more than about 2.48 ms to the IPDV on the path, even if the patch crosses a very high number of 25 STM-1 router hops.

IV.2.3 Delay variation due to a variation-insensitive packet

An arriving variation-sensitive packet does not pre-empt the servicing of a variation-insensitive packet which arrived earlier. Consequently, the variation-sensitive packet may experience a queueing component in each router bounded by the time it takes to serve a variation-insensitive packet.

For the calculation, it is assumed that each variation-sensitive packet experiences a random delay due to a variation-insensitive packet which is uniformly distributed between zero and the service time of maximum sized (1500 byte) variation-insensitive packets on the relevant output link rate. On an STM-1 output link this corresponds to a uniformly distributed delay between 0 and 80.13 µs in each router.

For the aggregation of this delay component over several routers in tandem, the convolution of the relevant delay distributions may be used, taking into account different output link rates when applicable. The lower quantile is assumed to be zero, the higher $(1 - 10^{-3})$ quantile can be calculated exactly. In most cases a good approximation is achieved by using an approximation by a normal (Gaussian) distribution or the worst case, whichever yields the smallest value. The $(1 - 10^{-3})$ quantile is found at $(\mu + 3.72 \cdot \sigma)$.

IV.2.4 Aggregated delay variation for variation-sensitive packets

An upper bound to the IPDV on a HRP is found by adding the values calculated for each of the three components in clauses IV.2.1 to IV.2.3.

NOTE – The resultant calculated value is expected to be higher than the value experienced in a real network. The following factors are noted:

- The addition of three quantile values yields a higher value than the actual delay quantile.
- The actual size of variation-sensitive packets (such as VoIP packets) is expected to be much smaller than the assumed size of 1500 bytes. In addition, the load with variation-sensitive traffic on most links is expected to be smaller than the assumed value of 50%. Therefore, the actual queueing delay due to interference with variation-sensitive traffic is expected to be smaller than calculated.
- The actual distribution of variation-insensitive packets (e.g., TCP acknowledgements) also contains packets which are (much) smaller than the assumed size of 1500 bytes. In addition, the total load (variation-sensitive plus variation-insensitive traffic) on most links is expected to be usually smaller than the assumed value of 100%. Therefore, the actual queueing delay due to interference with variation-insensitive traffic is expected to be smaller than calculated.

IV.3 Calculation examples

The following shows three examples for the calculation of the IPDV induced on a user-to-user HRP (see Figure II.1).

- An example where all links are relatively high speed (STM-1 or higher).
- An example where the links between customer and network and the links between network sections have a lower speed (E3 or T3).

- An example where the links between customer and network are low speed (e.g., 1.544 Mbit/s, T1).

IV.3.1 Example with STM-1 links

In this example, all links are assumed to be STM-1. The HRP between the network interfaces of the IP network cloud (see Figure III.2) consists of 12 router hops. Thus, the contributing factors to the IPDV on this path can be calculated as follows.

- Router look-up delay variation (see clause IV.2.1): 12×30 μs = 0.36 ms.
- Queueing delay variation due to variation-sensitive traffic (see Figure IV.1 for 50% load and 12 hops STM-1): ≈ 1.36 ms.
- Queueing delay variation due to variation-insensitive traffic (see clause IV.2.3): ≈ 9.01×80.13 μs = 0.72 ms.

Thus, the IPDV on this high link rate path can be expected to be smaller than 2.44 ms.

IV.3.2 Example with E3 interconnecting links

In this example, all links are assumed to be STM-1 except the user-network links and the link between network sections which are assumed to be E3 (34 Mbit/s). The HRP between the network interfaces of the IP network cloud (see Figure III.2) consists of 12 router hops, of which 2 hops have the lower E3 bit rate. Thus, the contributing factors to the IPDV on this path can be calculated as follows.

- Router look-up delay variation (see clause IV.2.1): 12×30 μs = 0.36 ms.
- Queueing delay variation due to variation-sensitive traffic (for 50% load and 10 hops STM-1 plus 2 hops E3): ≈ 2.92 ms.
- Queueing delay variation due to variation-insensitive traffic (for 10 hops STM-1 plus 2 hops E3): ≈ 1.19 ms.

Thus, the IPDV on this mixed link rate path can be expected to be smaller than 4.47 ms.

IV.3.3 Example with low rate access links

In this example, all links are assumed to be STM-1 except the user-network links which are assumed to be about 1.5 Mbit/s T1. The HRP between the network interfaces of the IP network cloud (see Figure III.2) consists of 12 router hops, of which 1 hop has the lower bit rate. In this case, the access link contribution is treated separately. The contributing factors to the IPDV on the high rate part of this path can be calculated as follows.

- Router look-up delay variation (see clause IV.2.1): 12×30 μs = 0.36 ms.
- Queueing delay variation due to variation-sensitive traffic (for 50% load and 11 hops STM-1): ≈ 1.29 ms.
- Queueing delay variation due to variation-insensitive traffic (for 11 hops STM-1): ≈ 8.364×80.13 μs = 0.67 ms.

Thus, the IPDV on this high link core path can be expected to be smaller than 2.32 ms.

On the access links, the delay contribution due to interference with a variation-insensitive packet may be as much as 15.6 ms when two 1500 byte packets are served ahead of a variation-sensitive packet (one of these packets may be part of the delay sensitive flow). The contribution to the IPDV due to interference with other variation-sensitive flows highly depends on the number of these flows and on the actual packet sizes used.

Note that the number of variation-sensitive flows, and the related packet size on the low rate access link, is determined by applications selected by the end-users. Without some influence, the network operator will find himself in a difficult position to commit to a stringent value for the IPDV network performance objective in the presence of a low rate access link.

If the delay-sensitive traffic has constant packet size (each containing 20 ms of G.711 coded voice, consistent with Appendix III), and occupies no more than 50% of the access link, then delay can be estimated as follows. There may be up to 9 voice flows of 50 packet/s, each 160 byte payload plus 40 byte RTP, UDP and IP headers (each total 80 kbit/s).

- Queueing delay variation due to variation-sensitive traffic (for 46.9% load and 1 hop T1), using the M/D/1 queueing model shows that the delay contribution, due to those relatively small variation-sensitive packets on the access link, is 5.12 ms.
- Queueing delay variation due to variation-insensitive traffic (for 1 hop T1): 7.81 ms.

The contribution to the delay variation on the access link thus aggregates to 12.93 ms thus totalling to 15.25 ms. The access link contribution thus dominates the IPDV in this case.

IV.3.4 Example summary and conclusions

The calculation examples show that a network operator who makes a modest effort to support both variation-sensitive and variation-insensitive traffic can commit to rather stringent values for the IPDV on a long HRP where all links have a reasonably high rate (e.g., a mix of STM-1 and E3/T3 or higher). Committing to an IPDV value in the order of 10 ms leaves ample room for additional lower rate (E3/T3) links or for an additional network section.

If a low rate link (1.5 Mbit/s T1, or E1) is present, committing to any low IPDV value becomes difficult. The network operator has little or no control over the actual number of variation-sensitive flows and the actual packet size of the variation-sensitive packets. Therefore, the IPDV commitments made by the network in this case will be dominated by the access link, and will need to be considerably larger than 10 ms, as shown in Table 1. On the access link, the end-user has control over the number and type of flows designated for a delay sensitive class, and therefore over the resulting IPDV. Under the assumption that the access link is only modestly loaded (<50%) with variation-sensitive traffic and that the dominant size of those packets will be small compared to the 1500 byte maximum size, an additional allowance of 20 ms for one low rate access link may be sufficient.

Appendix V

Material relevant to IP performance measurement methods

(This appendix does not form an integral part of this Recommendation.)

This appendix, which is for further study, will describe important issues to be considered as IP performance measurement methods are developed. It will describe the effects of conditions external to the sections under test, including traffic considerations, on measured performance.

The following conditions should be specified and controlled during IP performance measurements:

1) The exact sections being measured:
   - SRC and DST for end-to-end measurements;
   - MP bounding an NSE being measured;

NOTE – It is not necessary to measure between all MP pairs or all SRC and DST pairs in order to characterize performance.

2) Measurement time:
   - how long samples were collected;
   - when the measurement occurred.
3) Exact traffic characteristics:
   - rate at which the SRC is offering traffic;
   - SRC traffic pattern;
   - competing traffic at the SRC and DST;
   - IP packet size.
4) Type of measurement:
   - in-service or out-of-service;
   - active or passive.
5) Summaries of the measured data:
   - means, worst-case, empirical quantiles;
   - summarizing period:
     - short period (e.g., one minute);
     - long period (e.g., one hour, one day, one week, one month).

Appendix VI

Applicability of the ITU-T Y.1221 transfer capabilities and IETF differentiated services to IP QoS classes

(This appendix does not form an integral part of this Recommendation.)

This appendix addresses the applicability of the transfer capabilities defined in [ITU-T Y.1221] in support of the ITU-T Y.1541 IP QoS classes. It also specifies the relationship between ITU-T Y.1221 transfer capabilities and IETF differentiated services per hop behaviours consistent with what is specified in [ITU-T Y.1221].

[ITU-T Y.1221] defines three transfer capabilities (TC) called dedicated bandwidth (DBW), statistical bandwidth (SBW), and best-effort (BE). Each of the service models specified as part of the definitions of the ITU-T Y.1221 transfer capabilities currently specify a set of network performance parameters consistent with those specified in Table 1. Transfer capabilities defined in [ITU-T Y.1221] can be used to meet the performance objectives of the six QoS classes defined in this Recommendation.

QoS classes 0 and 1 in Table 1 define bounds on both IP packet delay and delay variation, and on IP packet loss ratio. The transfer capability of [ITU-T Y.1221] that allows a traffic contract to specify bounds on IP packet delay/delay variation and IP packet loss is the dedicated bandwidth transfer capability. QoS classes 2, 3 and 4 in Table 1 define bounds on IP packet loss ratio but not on IP packet delay variation. The transfer capability of [ITU-T Y.1221] that allows a traffic contract to specify bounds on both IP packet loss and delay is under study. QoS class 5 in Table 1 does not define bounds on IP packet loss ratio or IP packet delay/delay variation. The transfer capability of [ITU-T Y.1221] that does not offer any QoS commitment is the best-effort transfer capability. Table VI.1 specifies the mapping between ITU-T Y.1541 QoS classes and ITU-T Y.1221 transfer capabilities.

[ITU-T Y.1221] provides a mapping between the three transfer capabilities it defines and the IETF differentiated services per hop behaviours that should be used in networks that use the DiffServ architecture. Table VI.1 specifies the mapping between ITU-T Y.1221 transfer capabilities and IETF DiffServ per hop behaviours.

Table VI.1 – Association of ITU-T Y.1541 QoS classes with ITU-T Y.1221 transfer capabilities and differentiated services PHBs

Y.1221 transfer capabilities	Associated DiffServ PHBs	IP QoS class	Remarks
Best-effort (BE)	Default	Unspecified QoS class 5	A legacy IP service, when operated on a lightly loaded network may achieve a good level of IP QoS.
Delay-sensitive Statistical Bandwidth (DSBW)	AF	QoS classes 2, 3, 4	The IPLR objective only applies to the IP packets in the higher priority levels of each AF class. The IPTD applies to all packets.
Dedicated Bandwidth (DBW)	EF	QoS classes 0 and 1	

Appendix VII

Effects of network QoS on end-to-end speech transmission performance as perceived by the user

(This appendix does not form an integral part of this Recommendation.)

This appendix gives calculations of end-to-end speech quality using the objectives of ITU-T Y.1541 network QoS class 0 and class 1 as a starting point. These objectives constrain key contributors to application performance that are often dominant in the calculations. When combined with the performance of well-designed customer equipment, it is believed that the objectives provided by this Recommendation do allow for the achievement of a high end-to-end speech transmission performance as perceived by the users. However, the material provided by the G.100-series of ITU-T Recommendations should also be taken into account.

[b-ITU G.107], [b-ITU-T G.108], [b-ITU-T G.109], [b-ITU-T G.113], [b-ITU-T G.114] are the key documents required to derive an estimation of the mouth-to-ear speech quality which can be achieved with the values of the relevant network QoS class.

[b-ITU-T G.114] provides end-to-end limits and allocations for mean one-way delay, independent of other transmission impairments. The need to consider the combined effects of all impairments on overall transmission quality is addressed by [b-ITU-T G.107], the so-called E-model as the common ITU-T transmission rating model, which is the recommended ITU-T method for end-to-end speech transmission planning. [b-ITU-T G.108] gives detailed examples on how to use the model to assess the transmission performance of connections involving various impairments, including delay; and [b-ITU-T G.109] maps transmission rating predictions of the model into categories of speech transmission quality. Thus, while [b-ITU-T G.114] provides useful information regarding mean one-way delay as a parameter by itself, [b-ITU-T G.107] (and [b-ITU-T G.108] and [b-ITU-T G.109]) should be used to assess the effects of delay in conjunction with other impairments (e.g., distortions due to speech processing).

Furthermore, [b-ITU-T G.101] (the transmission plan) and related Recommendations are currently undergoing a basic revision.

VII.1 Example VoIP calculations with ITU-T Y.1541 class 0 network performance

As an example, a telephony hypothetical reference endpoint (HRE) for speech media may be as shown below. Information flows from the talker down through the protocol stack on the left, across the HRP, and up the protocol stack on the right to the listener (only one sending direction is shown).

Talker		Listener
ITU-T G.711 coder		ITU-T G.711 decoder, Appendix I of ITU-T G.711 packet loss concealment
RTP 20 ms payload size		60 ms jitter buffer
UDP		UDP
IP		IP
	(lower layers)	

Figure VII.1 – Example VoIP hypothetical reference endpoint

Using the hypothetical reference endpoint in Figure VII.1, endpoint delay is as below. These calculations follow from the formulas given in [b-ITU-T G.1020] for overall delay.

Table VII.1 – Endpoint delay analysis

	Delay, ms	Notes
Packet Formation	40	Two times frame size plus 0 look-ahead
Jitter Buffer, ave.	30	Centre of 60 ms buffer
Packet Loss Conceal.	10	One PLC "frame"
Total, ms	**80**	

The endpoint delay calculated in Table VII.1 is consistent with the objective for an ITU-T P.1010 category B terminal. If we combine this mean endpoint delay with the class 0 network delay, the total average delay for the user-to-user path is 100 + 80 = 180 ms. The example class 0 reference path in Appendix III indicates that this delay may be achieved over a distance of 4070 km.

A 50 ms customer installation (1-way send and receive) is possible with a packet formation time of 10 ms and a 50 ms de-jitter buffer.

Table VII.2 – Low delay endpoint delay analysis

	Delay, ms	Notes
Packet Formation	20	Two times frame size plus 0 look-ahead
De-Jitter Buffer, ave.	25	Centre of 50 ms buffer
Packet Loss Conceal.	0	"Repeat Previous" requires no additional delay
Other Equipment	5	
Total, ms	**50**	

The endpoint delay calculated in Table VII.2 is consistent with the objective for an ITU-T P.1010 category A terminal. The class 0 path IPTD and customer installation delays sum to a 1-way mouth-to-ear transmission time of 150 ms, satisfying the needs of most applications (as per [b-ITU-T G.114]).

It must be noted that a de-jitter buffer's contribution to mouth-ear delay is based on the average time packets spend in the buffer, not the peak buffer size. Packets that encounter the minimum transfer delay will wait the maximum time in the de-jitter buffer before being played out as a synchronous stream, while the reverse is true for packets with the maximum accommodated transfer delay (these packets spend the minimum time in the de-jitter buffer). In this way, the de-jitter buffer compensates for transfer delay variations and ensures that packets can be removed according to a synchronous play-out clock. [b-ITU-T G.1020] gives a more detailed description of the de-jitter buffer and its contribution to overall delay.

VII.2 Example VoIP calculations with ITU-T Y.1541 class 1 network performance

Using the same assumptions and the hypothetical reference path endpoint delays of Table VII.1, and the class 1 example path from Appendix III, the total average delay for a 27 500 km user-to-user path is 233 + 80 = 313 ms.

VII.3 Speech quality calculations for ITU-T Y.1541 hypothetical reference paths

It is possible to estimate the speech quality of IP networks using the ITU-T G.107 transmission planning tool, also known as the E-model.

Appendix III gives assumptions and configuration details of calculations for network (UNI-UNI). The example endpoint assumptions and delay calculations above include ITU-T G.711 codec, packet size, packet loss concealment, de-jitter buffer size, etc. Alternate speech codecs with lower bit rates, alternate packet sizes, and other variations are possible.

Figure VII.2 gives the reference connection for this analysis.

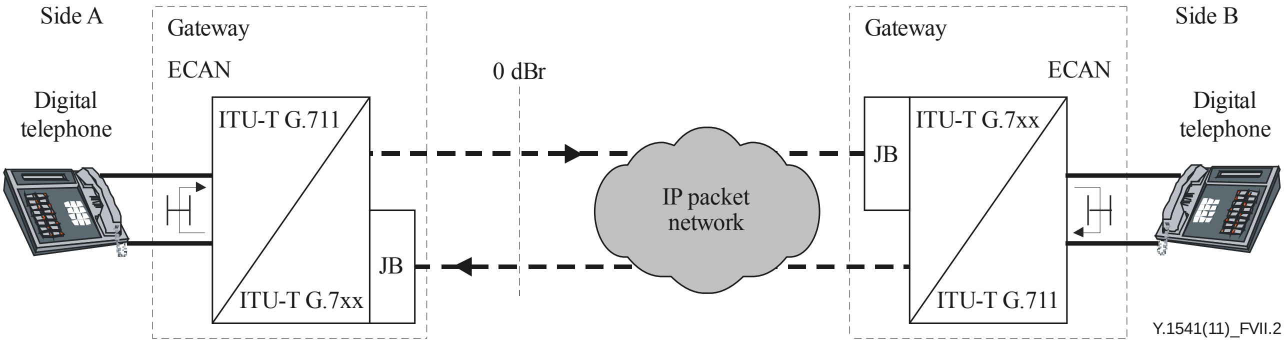


Figure VII.2 – Reference connection

Table VII.3 gives the E-model parameters used in the analysis.

Table VII.3 – E-model parameters

Parameters		Model input values		
Symbol	**Definition**	**ITU-T G.107 default**	**Input values**	**Unit**
Nc	Electric Circuit Noise Referred to at the 0 dBr point	(–70)	–70.0	dBm0p
Pos	Room Noise (Send)	(35)	35.0	dB(A)
Por	Room Noise (Receive)	(35)	35.0	dB(A)
SLR	Send Loudness Rating	(8)	8.0	dB
RLR	Receive Loudness Rating	(2)	2.0	dB
Ds	D-factor (Send)	(3)	3.0	
LSTR	Listener's Sidetone Rating	(equ.)	18.0	dB
Nfor	Noise Floor	(–64)	–64.0	dBmp
STMR	Sidetone Masking Rating	(15)	15.0	dB
qdu	Quantizing Distortion Units	(1)	1.0	units
T	Mean One-Way Delay	(0)	**150.0**	ms
TELR	Talker Echo Loudness Rating	(65)	65.0	dB
WEPL	Weighted Echo Path Loss	(110)	110.0	dB
Ta	Absolute Delay from (S) to (R)	(0)	**150.0**	ms
Tr	Round-Trip Delay	(0)	**300.0**	ms
Ie	Equipment Impairment Factor	(0)	0.0	
Bpl	Packet Loss Robustness Factor	(1)	**4.8**	
Ppl	Random Packet Loss Probability	(0)	0.0	%
A	Expectation Factor	(0)	0.0	
Dr	D-factor (Receive)	(3)	3.0	

We have assumed the default values for all parameters, except T, Ta, and Tr. The mean absolute 1-way delay was calculated using 100 ms for network delay (UNI-UNI, conforming to the QoS class 0 objective) and 50 ms for the end-terminal, including ITU-T G.711 packetization and de-jitter buffer (100 + 50 = 150 ms = T = Ta = Tr/2). Here, R = 89.5.

Packet loss also influences speech quality. We include a column below where approximately 0.1% loss is combined with a packet loss robustness factor, Bpl = 4.8 when the packet loss concealment used with ITU-T G.711 is Repeat 1, followed by silence. When using the PLC in [b-ITU-T G.711 APP I], we take the packet loss robustness factor, Bpl = 25.1.

Appendix III also provides calculations showing longer mean network delays, and larger terminal delays. Table VII.4 summarizes the findings.

Table VII.4 – E-model results with ITU-T Y.1541 hypothetical reference paths and end-terminals

Network, mean 1-way delay, ms	Terminal mean 1-way delay, ms	Total, mean 1-way delay, ms	Packet size, ms	Packet loss conceal.	R, no loss	R, with ~0.1% packet loss	Y.1541 QoS class
100	50	150	10	Rpt.1/Sil	89.5	87.6	0
100	80	180	20	G.711ApI	87.8	87.5	0
150	80	230	20	G.711ApI	81.9	81.5	1
233	80	313	20	G.711ApI	71.1	70.7	1

Appendix VIII

Effects of IP network performance on digital television transmission QoS

(This appendix does not form an integral part of this Recommendation.)

VIII.1 Introduction

This appendix details a part of the analysis behind the specification of provisional network QoS classes 6 and 7 in Table 3. The objective values were selected in order to support digital television transmission. The IP packet loss ratio (IPLR) objective in classes 0 through 4 was insufficient to support this application, as stated in the previous version of this appendix.

VIII.2 Hypothetical reference endpoint (HRE) for high-bandwidth video signals

It is important to first establish a reference endpoint for video transport. The proposed endpoint is based on work done previously by the ATIS T1A1 sub-committee, as well as analysis of typical video transport endpoint models spanning both compressed and uncompressed video by the Video Services Forum (VSF). There may ultimately be a need to establish more than one HRE to allow point-to-point and point-to-multipoint transmission, but this analysis is restricted to the simpler case of the point-to-point HRE.

Sender		Receiver
Video (uncompressed SDI, multi- or single-compressed-MPEG-2 stream DVB-ASI, etc.), multiple audio streams, ancillary data		Video (uncompressed SDI, multi- or single-compressed-MPEG-2 stream DVB-ASI, etc.), multiple audio streams, ancillary data
Embedder		De-embedder
Packetizer/Interleaver/FEC		FEC-1/De-inteleaver/De-packetizer
RTP		RTP, Sufficient De-Jitter Buffer
UDP		UDP
IP		IP
	(Physical Layer)	

Figure VIII.1 – Hypothetical reference endpoint for digital television

The digital television transport uses an IP network where uncompressed video packets or MPEG-compressed video packets are encapsulated into either UDP/IP or RTP/UDP/IP. We assume that RTP/UDP/IP is the protocol used and that the following protocol overhead applies:

IP packet length = (7 × 188-Byte MPEG packets) + RTP/UDP/IP packet overhead

The following clauses describe three profiles of video services and give a rationale for the deployment of error correction mechanisms in IP networks to guarantee the appropriate level of quality and reliability.

VIII.3 Service profiles and end-to-end packet performance requirements

The technical requirements for this appendix will be limited to three service profiles: contribution services profile, primary distribution service profile and access distribution service profile. These three profiles encompass the vast majority of the video industry's applications and needs. We also present the performance requirements for these profiles in terms of packet loss at three different viewer quality levels, or hit rates.

VIII.3.1 Contribution video services profile

Contribution services typically have the highest performance and can vary from uncompressed to mildly compressed video and audio signals. Contribution connections allow exchange of content by a network or its affiliates for further use, e.g., for bringing signals back from fixed, temporary, or remote locations to the studio for editing or immediate rebroadcast. In those scenarios, for long-haul applications, terrestrial fibre, microwave or satellite infrastructure endpoint connections can be utilized.

Contribution can also mean the outbound delivery of signals from the main network studio to network affiliates for rebroadcasting and typically employs satellite or long-haul terrestrial network services. Today, these outbound connections are provided by way of fixed or on-demand private leased lines (fibre), or in certain, less-extensive applications, ATM services offering DS-3, OC-3, or OC-12 bandwidth.

In addition to those real-time applications, sometimes IP services are used for non-real-time file exchange between video and audio servers and for monitoring and control of remote systems. As the same user may use their IP service for contribution video and file transfer, the contribution service profile also easily accommodates file transfer and remote control.

VIII.3.2 Primary distribution video service profile

Distribution means delivery of video and audio content either directly to the consumer or to cable head-ends for transmission through a cable television plant. In these applications, typically a lower signal quality (lower data rates) is needed, as little additional signal processing will be applied. Traditionally for these applications, terrestrial or satellite services are used. There are two types of distribution signals, primary and access. Primary distribution connections are feeds from the local affiliate to the cable head-end or to the television transmission tower, and ordinarily, these connections are comparable to, or slightly lower in quality than, contribution connections. Primary distribution may be provided by satellite, short-haul terrestrial microwave, or fibre optic connection. Access distribution involves the delivery of the content from the cable head-end to the final consumer over the cable television plant or through the air in the form of a broadcast emission from the television transmitter tower antenna. The VSF recommends that 40 Mbit/s represent the bit rate of this type of service.

VIII.3.3 Access distribution service profile

Access distribution service profile is defined as TV services currently being delivered by cable and satellite networks. Since the quality achieved by these networks is somewhat subjective, this contribution will characterize quality as an upper bound on video data errors (due to network) in a specific window of time.

VIII.3.4 Performance requirements for the service profiles

Quality of service for this application will be given in terms of actual number of errors (performance hits) in a specific time period. Table VIII.1 was constructed based on recommendations from active members of the Video Services Forum and represents expected error rates that service providers (e.g., DirecTV), as well as users (e.g., Fox Sports Network), would demand.

Table VIII.1 – Digital television loss/error ratio recommendations

Profile (Typical bit rate)	One performance hit per 10 days	One performance hit per day	10 performance hits per day
Contribution (270 Mbit/s)	4×10^{-11}	4×10^{-10}	4×10^{-9}
Primary Distribution (40 Mbit/s)	3×10^{-10}	3×10^{-9}	3×10^{-8}
Access Distribution (3 Mbit/s)	4×10^{-9}	4×10^{-8}	4×10^{-7}

This table assumes all lost packets cause a performance hit (possibly visible or audible impairment), and seven MPEG transport stream (TS) packets are encapsulated in a single IP packet. The required packet loss ratio is given at the intersection of a hit rate and profile. For example, access distribution allowing a quality level of 1 performance hit per day requires a packet loss ratio of 4×10^{-8}.

VIII.4 Forward error correction (FEC)/Interleaving to improve UNI-UNI performance

Even an IP network conforming to QoS classes 6 or 7 is not capable of providing the packet loss rates required for the profiles above, and edge equipment is needed to correct packet errors, packet losses and reordered packets. We assume the service uses FEC/Interleaving as defined by the Pro-MPEG Forum COP-3 recommendation (Code of Practice) and as reflected in Table VIII.2. Note that this 2-dimensional FEC/Interleaving specification is slightly more powerful than the base layer of digital video broadcast application-layer-FEC (DVB AL-FEC) of Annex E [b-ETSI TS 102 034]. The DVB AL-FEC base layer is consistent with the 1-dimensional Pro-MPEG FEC.

Table VIII.2 – FEC/Interleaving to achieve desired end-to-end hit rates

	Minimal correction	Moderate correction	High correction
Minimum Network Performance			
Loss Distance (Packets)	100	50	50
Loss Period (Packets)	5	5	10
Applied FEC			
FEC L, D	5, 20	5, 10	10, 5
FEC Overhead (%)	5	10	20
Resulting Video Performance Quality	High	High	High

Note that the specification of network performance above utilizes two new terms. Loss distance (LD) and loss period (LP), defined in [b-IETF RFC 3357], are packet loss pattern parameters. LP defines the maximum number of consecutive packets that can be lost, while LD defines the minimum number of good packets that must arrive between lost packets for the algorithm to properly correct for losses. The LD and LP values describe the minimum network performance correctable by the corresponding FEC in the same column. The FEC is defined by length (L) and depth (D) algorithm parameters that define the robustness of the method.

Correction of network impairments is not free, as it consumes additional bandwidth. The overhead values in the table represent three levels of robustness, where 5% represents minimal correction, 10% represents moderate correction and 20% represents the highest amount of correction. Note that the more robust the algorithm we choose, the higher the overhead. It is the VSF's position that these three values encompass the majority of the needs in the industry.

As an example, a 2 Mbit/s video service requiring minimal correction would be configured with (L, D) settings of (5, 20). This would generate an extra 100 kbit/s (5% of 2 Mbit/s) of network traffic for the FEC packets, resulting in a total data rate of 2.1 Mbit/s. Similarly, a 270 Mbit/s service requiring high correction would be configured with (L, D) values of (10, 5) which would generate an additional 54 Mbit/s of network traffic, resulting in an aggregate rate of 324 Mbit/s.

VIII.5 Laboratory assessment of forward error correction (FEC)/Interleaving effectiveness

Laboratory test results with the Pro-MPEG Forum COP-3 recommendation 2-dimensional FEC/Interleave (5, 50) indicate that:

– UNI-UNI loss ratio of 10^{-4} improves to 1.5×10^{-8} (covers most of the access profile);

– UNI-UNI loss ratio of 10^{-5} improves to 2×10^{-10} (covers most profiles).

It was concluded that an IP network with UNI-UNI IPLR and IPER conforming to Table 3, class 6 or 7 will support the digital television application described above, providing that the appropriate FEC/Interleaving is applied.

VIII.6 Additional performance parameters

The Video Services Forum concluded that the values for IPTD and IPDV specified in Table 3, classes 6 and 7 are sufficient for digital television transport.

VIII.7 Further analysis with advanced FEC schemes

The IPTV focus group (see http://www.itu.int/ITU-T/IPTV/index.phtml) prepared an analysis of application layer error recovery mechanisms. Their numerical results utilize the enhancement layer of the DVB-IP AL-FEC mechanism. This is a decoder enhanced according to clause E.5.1.2 of [b-ETSI TS 102 034], which describes the digital fountain raptor code (and is apparently more powerful than the Pro-MPEG Forum COP-3 recommendation 2-dimensional FEC/Interleave code).

The IPTV FG analysis used the following assumptions and inputs:

1) Mean time between visible artefacts (MTBA) or 4 hours (slightly more demanding than the 10 hits per day level used in the VSF study).

2) Two video stream rates: 2.1 Mbit/s for standard definition and 9.4 Mbit/s for high definition.

3) Seven MPEG-2 TS packets per RTP payload.

4) A set of FEC protection periods, ranging from 100 ms to 1000 ms.

5) A fixed average packet loss ratio of 10^{-3}.

6) Two network loss models, one with independent random packet loss, and another with fixed length bursts of loss corresponding to 8 ms of time (less than 2 packets for SD and 8 packets for HD). The fixed length burst loss model is intended to simulate a DSL access line subjected to electrical impulse noise, and each impulse causes an outage equal in length to the DSL interleaving depth, which is taken to be 8 ms.

The IPTV focus group results are shown in Table VIII.3 below, over a range of protection periods.

Table VIII.3 – Required overhead for DVB-IP AL-FEC for different bit rates, different channel models at IPLR of 10e-3, and different protection periods

Protection period	Random, 2.1 Mbit/s	Random, 9.4 Mbit/s	Burst, 2.1 Mbit/s	Burst, 9.4 Mbit/s
100 ms	16%	5%	20%	12%
200 ms	8%	3.5%	10%	6%
400 ms	5%	3%	7%	4%
600 ms	4%	2%	4%	2.5%
800 ms	3.5%	2%	4%	2.5%
1000 ms	3%	2%	4%	2%

The FEC overhead is reasonable and within the same range used with the Pro-MPEG Forum COP-3 recommendation 2-dimensional FEC/Interleave in Table VIII.2.

When the network characteristics are similar to the two cases examined (8 ms loss bursts or random independent loss), the analysis using clause E.5.1.2 FEC [b-ETSI TS 102 034], shows that the ITU-T Y.1541 class 0 or 1 objectives are sufficient.

Note that the degree to which these two network models represent the actual conditions experienced in digital video transmission over packet networks is not known at this time and requires further study.

VIII.8 Analysis of retransmission schemes

This clause currently identifies an area for further study. There may be existing analyses that can be summarized in this clause, following review.

VIII.9 Recovery from errors and losses due to protection switching schemes

This clause currently identifies another area for further study.

Most protection switching schemes, such as SONET rings and MPLS-Fast Re-Route (MPLS-FRR) require at least 50 ms to replace a failed primary path with a backup path. In practice, restoration times on the order of 100-200 ms are possible. None of the example correction schemes considered above can compensate for such long outages. However, if a design goal is correcting outages of this long duration, it may be possible to devise a scheme that can correct the burst losses with additional penalties of longer delay and more overhead.

Appendix IX

Effects of network QoS on end-to-end data transmission performance using TCP

(This appendix does not form an integral part of this Recommendation.)

IX.1 Introduction

This appendix details a part of the analysis behind the specification of provisional network QoS classes 6 and 7 in Table 3. The objective values were selected in order to support applications using the reliable byte stream transfer services of the transmission control protocol (TCP) [b-IETF RFC 793] at the largest possible data rate. The existing IP packet loss ratio (IPLR) objective (in classes 0 through 4) supports TCP with the limitations of widely deployed legacy settings, or assumes that some bottleneck will be encountered beyond the UNI-UNI path.

There are two key factors that limit TCP transfer capacity:

1) The congestion-aware flow-control mechanisms infer that congestion has been encountered on the path when packet loss occurs. In response to loss, the flow-control cuts the sending window in half, and allows linear increase when a full window of packets has been transferred successfully. Thus, packet loss can limit capacity.

2) The maximum window size may be limited by the sender or receiver TCP settings, or by the operating system itself (limiting the amount of memory available to a specific application for buffering network data). This is the classic delay bandwidth product, where the transmission rate is given as one window of octets per round-trip time (for acknowledgement).

Given that packet transfer time is usually dominated by propagation time, the goal of the analysis was to determine an objective for IPLR that provides very high TCP transfer capacity when other factors, such as window size or bottleneck bandwidth, do not encumber the process. A packet loss ratio of 10^{-5} was selected for classes 6 and 7, and the analysis below shows what capacities can be achieved.

IX.2 Model of TCP performance

The basis for this study is the model of TCP Reno [b-IETF RFC 2001] developed and verified by [b-Padhye1]. Their model can be approximated as follows:

$$B(p) \approx \min\left(\frac{W_{\max}}{RTT}, \frac{1}{RTT\sqrt{\frac{2bp}{3}} + T_0 \min\left(1, 3\sqrt{\frac{3bp}{8}}\right) p\left(1+32p^2\right)} \right)$$

where:

$B(p)$: approximate model of TCP throughput [packet/s]

W_{max} : maximum window buffer size of receiver [packets]

RTT : Round Trip Time [sec]

b : number of packets that are acknowledged by a received ACK

p : probability that a packet is lost

T_0 : time-out for re-transmitting an unacknowledged (lost) packet [sec]

There are many combinations of TCP features, and the different combinations are sometimes named according to the meeting place where they were agreed (Vegas, Tahoe, and Reno). A discussion of TCP features is available in many other references. For an even simpler TCP model with a single fitting parameter that is useful across versions, see [b-Mathis].

IX.3 TCP hypothetical reference endpoint (HRE)

Various appendices of this Recommendation specify hypothetical reference endpoints (HRE) and pair them with hypothetical reference paths to assess the user application quality levels that the network performance objectives can support. We define a TCP hypothetical reference endpoint below.

Sending application		**Receiving application**
TCP Reno Max window = 16 kbyte, 64 kbyte, or 256 kbyte T0 timeout = 1 s Large windows option		TCP Reno Max window = 16 kbyte, 64 kbyte, or 256 kbyte b = 1 ACK/2 packets Large windows option
IP		IP
	(lower layers)	

Figure IX.1 – TCP hypothetical reference endpoint

We assume that the sending application supplies a continuous byte stream with no idle intervals, and that the receiving host contribution to RTT is insignificant. Note that the sending and receiving max window sizes will vary in the analysis that follows.

IX.4 Observations

Figure IX.2 shows the estimate of "Legacy" TCP Reno capacity vs. round-trip time (including host processing) and packet loss. The 3-dimensional surface is scribed with lines that correspond to round trip times (RTT) of 20, 40, 100, 200, 400, 1000, 2000, and 4000 ms, intersecting with lines at Loss Ratios of 10^{-1}, 10^{-2}, 10^{-3}, 10^{-4}, 10^{-5}, and 10^{-6}. The height of the surface indicates the TCP capacity in bits/ second, and the surface colour changes when it crosses a labelled capacity level.

We note that none of the long-delay mitigations have been applied here, such as IETF RFC 1323 large windows or IETF RFC 2018 selective acknowledgements (SACK).

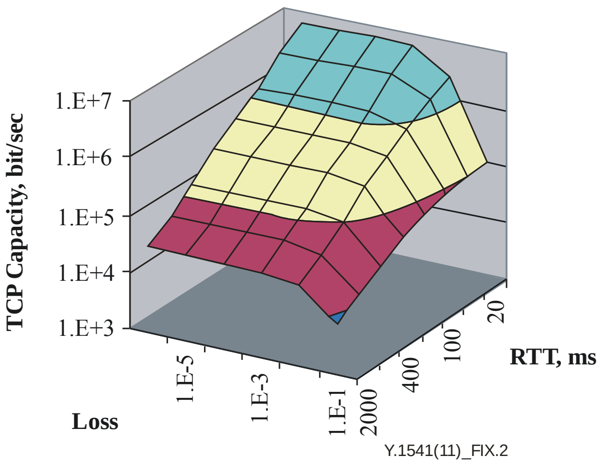


Figure IX.2 – TCP capacity with 16 kbyte window ("Legacy")

An 8 kbyte or 16 kbyte window is the default setting for many legacy TCP implementations. Figure IX.2 shows that packet loss > 10^{-3} has an effect on capacity, but the window size limitation dominates the capacity vs. loss performance over a wide range of round trip times (RTT). Therefore, the IPLR objective <10^{-3} is sufficient under these circumstances, and network QoS classes 2, 3, and 4 will produce satisfactory capacity.

Although transfer capacities in the order of 10 Mbit/s are possible at very low RTT, packet transfer time also influences capacity for the "legacy" TCP sender-receiver pair.

Figure IX.3 shows the TCP Reno capacity when the maximum window size is set to 64 kbytes. This is usually possible with simple tuning procedures, but the overwhelming majority of IP network users do not attempt tuning, or have no need. Users who want to realize the full potential of broadband access while reducing the transfer time for extremely large files (e.g., Linux distribution ISO-files contain 700 Mbyte CD-ROM images) may seek the benefits of tuning.

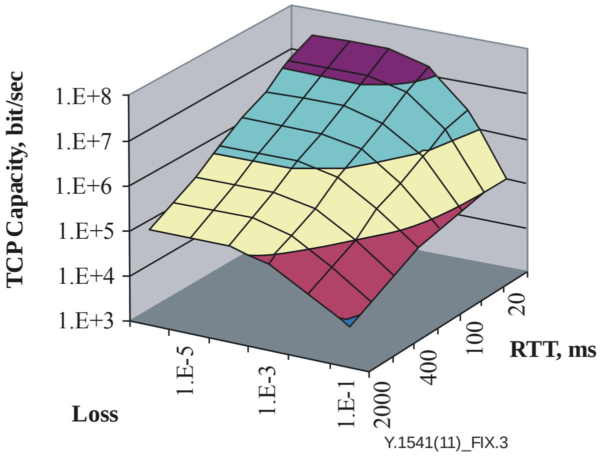


Figure IX.3 – TCP capacity with 64 kbyte window

A 64 kbyte window is the maximum setting for standard TCP implementations that do not enable IETF RFC 1323 large windows. Figure IX.3 shows that packet loss > 10^{-4} has an effect on capacity, but the window size limitation dominates the capacity from there on.

Figure IX.4 shows the TCP Reno capacity when the maximum window size is set to 256 kbytes. This is possible with many operating systems, and the TCP large windows option must be available.

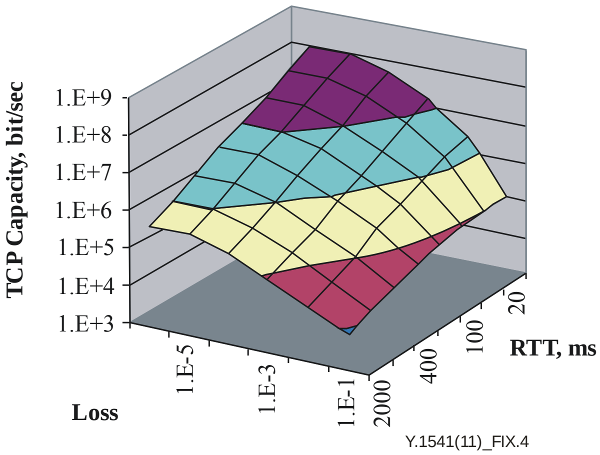


Figure IX.4 – TCP capacity with 256 kbyte window (and [b-IETF RFC 1323])

Figure IX.4 shows that packet loss > 10^{-5} has an effect on capacity, but the window size limitation dominates the capacity vs. loss performance characteristic beyond that point. Therefore, these are circumstances where the new provisional classes (with IPLR objective < 10^{-5}) are needed to support maximum capacity.

Transfer capacities on the order of 100 Mbit/s are possible at very low RTT, and the large window option [b-IETF RFC 1323] reduces the negative affect of RTT on capacity.

IX.5 Summary of TCP capacity estimates

Table IX.1 provides a numerical summary of Figures IX.2 through IX.4 at the values of delay and loss ratio appearing in the objectives.

Table IX.1 – Summary of TCP capacity estimates, bits/s

Window size	Packet Loss, p	IPTD = RTT/2 = 100 ms	IPTD = RTT/2 = 400 ms
16 kbytes	10^{-3}	640 000	160 000
	10^{-5}	640 000	160 000
64 kbytes	10^{-3}	**1 624 887**	**409 640**
	10^{-5}	2 560 000	640 000
256 kbytes	10^{-3}	**1 624 887**	**409 640**
	10^{-5}	10 240 000	2 560 000

Note that **Bold** values are limited by packet loss ratio, otherwise window size limits capacity. Packet loss ratio of 10^{-5} does not limit capacity at any window size examined, clearly showing the benefits of the new network QoS classes.

Appendix X

An example showing how to calculate IPDV across multiple sections

(This appendix does not form an integral part of this Recommendation.)

This appendix provides an example on how to calculate IPDV when a number of network sections are involved. It builds on the information contained in clause 8.2.4 of this Recommendation and also provides some background information on the method.

The definition of IP delay variation used here (see discussion in Appendix II) is:

$$IPDV = IPTD_{upper} - IPTD_{min}$$

where:

$IPTD_{upper}$ is the $1 - 10^{-3}$ quantile (99.9th percentile) of IPTD in the evaluation interval

$IPTD_{min}$ is the minimum IPTD in the evaluation interval

It assumes that there are a number of network sections S_1, S_2, ... , S_n for which estimates of $IPDV_1$, $IPDV_2$, ... , $IPDV_n$ are available. The individual estimates must have been made under comparable network conditions for any end-to-end combination to be meaningful. For example, they might have all been measured during the busiest hour of the month in each of their individual sections. In this case, the resulting combinations will generally not correspond to any real end-to-end measurement that could be made as all of the component sections could not be expected to experience their busiest hours simultaneously. Nevertheless, the result would produce an upper bound that could be used for planning and network monitoring purposes.

The relationship for estimating the UNI-UNI IP delay variation (IPDV) performance from the network section values must recognize their sub-additive nature and is difficult to estimate accurately without considerable information about the individual delay distributions. If, for example, characterizations of independent delay distributions are known or measured, they may be convolved to estimate the combined distribution. This detailed information will seldom be shared among operators, and may not be available in the form of a continuous distribution. As a result, the UNI-UNI IPDV estimation may have accuracy limitations. Since study continues in this area, the estimation relationship given below has been specified on a provisional basis, and this clause may change in the future based on new findings or real operational experience.

X.1 Calculation of delay variation

The relationship for combining IPDV values is given below.

The problem under consideration can be stated as follows: estimate the quantile t of the UNI-UNI delay T as defined by the condition:

$$\Pr(T < t) = p$$

We will assume that $p = 0.999$ (99.9th percentile) and for simplicity that all delay measurements have been normalized by removing the measured minimum delay. For the numerical examples below, it is assumed that there are three network sections ($n = 3$) and that all delays are expressed in ms.

Step 1

Measure the mean and variance of the delay in each of the n network sections. For a set of measurements, $D_1, D_2, \ldots, D_n$ for the kth section the mean, μ_k, and variance, σ_k^2, are computed as:

$$\mu_k = \frac{1}{n}\sum_{i=1}^{n} D_i$$

$$\sigma_k^2 = \frac{1}{n}\sum_{i=1}^{n}(D_i - \mu_k)^2$$

For our example we suppose that we have found that:

$$\mu_1 = 1.0 \qquad \mu_2 = 2.0 \qquad \mu_3 = 3.0$$

$$\sigma_1^2 = 0.5 \qquad \sigma_2^2 = 1.0 \qquad \sigma_3^2 = 1.5$$

Estimate the mean and variance of the UNI-UNI delay by summing the means and variances of the component distributions.

$$\mu = \sum_{k=1}^{n}\mu_k = 1.0 + 2.0 + 3.0 = 5.0$$

$$\sigma^2 = \sum_{k=1}^{n}\sigma_k^2 = 0.5 + 1.0 + 1.5 = 3.0$$

Step 2

Measure the quantiles, t_k, for each delay section at the probability of interest, $p = 0.999$. These can be determined simply by sorting the measurements, D_i, so that without loss of generality:

$$D_1 \le D_2 \le \ldots \le D_n$$

and then selecting as the pth quantile the mth measurement D_m (that is $t_k = D_m$) where m is the smallest integer satisfying $p \le m/n$. If $n = 1000$, then $m = 999$ for $p = 0.999$. Suppose for our example we find that:

$$t_1 = 4.32 \qquad t_2 = 6.02 \qquad t_3 = 7.55$$

Estimate the skewness, γ_k, and third moment, ω_k, for the kth section using the formulas shown below, where $x_{0.999} = 3.090$ is the value satisfying $\Phi(x_{0.999}) = 0.999$ where Φ denotes the standard normal (mean 0, variance 1) distribution function.

$$\gamma_k = 6 \cdot \frac{x_p - \dfrac{t_k - \mu_k}{\sigma_k}}{1 - x_p^2} \qquad \omega_k = \gamma_k \cdot \sigma_k^3$$

$$\gamma_1 = 6 \cdot \frac{3.09 - \dfrac{4.32 - 1}{\sqrt{0.5}}}{1 - 3.090^2} = 1.126 \qquad \omega_1 = 1.126 \cdot \left(\sqrt{0.5}\right)^3 = 0.398$$

$$\gamma_2 = 6 \cdot \frac{3.09 - \dfrac{6.02 - 2}{\sqrt{1.0}}}{1 - 3.090^2} = 0.653 \qquad \omega_2 = 0.653 \cdot \left(\sqrt{1.0}\right)^3 = 0.653$$

$$\gamma_3 = 6 \cdot \frac{3.09 - \frac{7.55-3}{\sqrt{1.5}}}{1-3.090^2} = 0.439 \qquad \omega_3 = 0.439 \cdot \left(\sqrt{1.5}\right)^3 = 0.806$$

Assuming independence of the delay distributions, the third moment of the UNI-UNI delay is just the sum of the network section third moments.

$$\omega = \sum_{k=1}^{n} \omega_k = 0.398 + 0.653 + 0.806 = 1.856$$

The UNI-UNI skewness is computed by dividing by $\sigma^{3/2}$ as shown below.

$$\gamma = \frac{\omega}{\sigma^3} = \frac{1.856}{\left(\sqrt{3}\right)^3} = 0.357$$

Step 3

The estimate of the 99.9th percentile ($p = 0.999$) of UNI-UNI delay t (in ms) is as follows.

$$t = \mu + \sigma \cdot \left\{ x_p - \frac{\gamma}{6}\left(1 - x_p{}^2\right) \right\} = 6 + \sqrt{3} \cdot \left\{ 3.09 - \frac{0.357}{6}\left(1 - 3.09^2\right) \right\} = 12.23$$

As stated earlier, the nature of the IPDV objective is the upper bound on the $1–10^{-3}$ quantile of IPTD minus the minimum IPTD (i.e., the distribution of IPDV is normalized to the minimum IPTD). In general, units of IPDV values are seconds, with resolution of at least 1 microsecond. If lesser resolution is available in a value, the unused digits shall be set to zero.

X.2 Mathematical background

If the distributions of each of the components T_k were known in detail, the distribution of the end-to-end delay T could be computed using convolutions. Convolutions are challenging in practice: most implementations will rely on Laplace transform techniques including methods to invert transforms numerically to recover the underlying probability distributions. To use this method, assumptions would have to be made about the exact nature of the component distributions.

Instead, an alternative method is employed that uses the available information without requiring additional assumptions or complex methods.

The basic idea is to transform a random variable T with known mean μ, variance σ^2, and skewness γ into a symmetric random variable Z which is standard normal (mean 0, variance 1) or nearly so. One such method, called the normal power approximation (see [b-Ramsay]) works as follows:

- Define the standardized variable $X = \frac{T - \mu}{\sigma}$
- The normal power approximation states that $X \approx Z + \frac{\gamma}{6}(Z^2 - 1)$ where Z is a standard normal (mean 0, variance 1) random variable.

Once the details are worked through, the following approximation is obtained:

$$\Pr(T < t) \approx \Phi\left(\frac{1}{\gamma} \sqrt{9 + 6\gamma\left(\frac{t-\mu}{\sigma}\right) + \gamma^2} - \frac{3}{\gamma} \right)$$

where Φ is the cumulative standard normal distribution function:

$$\Phi(x)=\int_{-\infty}^{x}\frac{e^{\frac{-x^2}{2}}}{\sqrt{2\pi}}dx$$

Although the values of this function are readily available, a more transparent relationship can be derived that eliminates all reference to Φ and allows the quantile t to be directly computed from the component quantiles t_k.

In fact, since the probabilities in all the quantile definitions $\Pr(T_k < t_k) = p$, $\Pr(T < t) = p$ have the common value p, if we define x_p to be the unique value satisfying $\Phi(x_p) = p$, then we have:

$$\frac{1}{\gamma_k}\sqrt{9+6\gamma_k\left(\frac{t_k-\mu}{\sigma_k}\right)+\gamma_k{}^2}-\frac{3}{\gamma_k}=x_p \text{ and}$$

$$\frac{1}{\gamma}\sqrt{9+6\gamma\left(\frac{t-\mu}{\sigma}\right)+\gamma^2}-\frac{3}{\gamma}=x_p$$

If we multiply the above by σ_k^2 and σ^2 respectively and add over all the components we deduce from the additivity of variances of independent distributions that:

$$\sigma^2\cdot\frac{\sqrt{1+2\delta\cdot\left(\frac{t-\mu}{\sigma}\right)+\delta^2}-1}{\delta}=\sum_{k=1}^{n}\sigma_k^2\cdot\frac{\sqrt{1+2\delta_k\cdot\left(\frac{t_k-\mu_k}{\sigma_k}\right)+\delta_k^2}-1}{\delta_k}$$

where we have set $\delta=\frac{\gamma}{3}$ and $\delta_k=\frac{\gamma_k}{3}$. Although this looks complex, it requires only simple algebra to compute the end-to-end quantile t from the components t_k and the available measured quantities.

X.3 Special cases

In the approximation

$$\Pr(T<t)\approx\Phi\left(\frac{1}{\gamma}\sqrt{9+6\gamma\left(\frac{t-\mu}{\sigma}\right)+\gamma^2}-\frac{3}{\gamma}\right)$$

if we let $\gamma\to 0$ we produce the result

$$\Pr(T<t)\approx\Phi\left(\frac{t-\mu}{\sigma}\right)$$

corresponding to the case where T has a normal distribution with mean μ, variance σ^2. If we let all the skewness terms $\gamma\to 0$, $\gamma_k\to 0$ the algebraic expression of the previous section reduces to:

$$\sigma\cdot(t-\mu)=\sum_{k=1}^{n}\sigma_k\cdot(t_k-\mu_k)$$

Some further manipulation removes the variances to produce:

$$(t-\mu)^2 = \sum_{k=1}^{n} (t_k - \mu_k)^2$$

This shows that when the component delays T_k are normally distributed with mean μ_k, and variance σ_k^2, then the corresponding quantiles follow a composition law similar to that for variances.

This composition law for normal variates can also be derived directly. The algebraic expression of the previous section can be viewed as a generalization of this particular composition law.

X.4 Estimating skewness from quantiles

Consider a random variable T whose mean μ and variance σ^2 are known and where the quantile t in $\Pr(T < t) = p$ is known but where the skewness γ is not known. Using the normal power approximation we have:

$$\Pr(T < t) \approx \Phi\left(\frac{1}{\gamma}\sqrt{9 + 6\gamma\left(\frac{t-\mu}{\sigma}\right) + \gamma^2} - \frac{3}{\gamma}\right) = p$$

From tabulated values of the standard normal distribution function Φ we can find the unique value x_p satisfying $\Phi(x_p) = p$. Therefore:

$$\frac{1}{\gamma}\sqrt{9 + 6\gamma\left(\frac{t-\mu}{\sigma}\right) + \gamma^2} - \frac{3}{\gamma} = x_p$$

This can be solved for γ producing:

$$\gamma = 6 \cdot \frac{x_p - \frac{t-\mu}{\sigma}}{1 - x_p^2}$$

Appendix XI

Digital circuit (ISDN) emulation requirements on IP-based networks

(This appendix does not form an integral part of this Recommendation.)

XI.1 Introduction

The purpose of this appendix is to derive a packet loss requirement for the support of the circuit-mode unrestricted 64 kbit/s bearer service defined in [b-ITU-T I.231.1] over an IP network, based on the requirements of [b-ITU-T G.826] for error rates on transmission systems carrying ISDN connections. Next, the possible mitigation of the packet loss requirement through the use of forward error correction (FEC) techniques is examined.

We consider a multiplex of RTP packet streams as an emulation of a TDM digital transport connection, which should meet the ITU-T G.826 requirements for digital connections.

XI.2 Packetization and transport assumptions

Consider transmission of a single ISDN B-channel as a 64 kbit/s stream in 10 ms packets, 100 packets/s per stream. The resulting stream of RTP packets has a 9-octet POS overhead, 40-octet RTP/UDP/IP headers and 80-octet user data payload.

Suppose then that the packetized B-channel is routed with other packetized 64 kbit/s streams (voice and ISDN) across a core network using STM-1 transmission systems. This system transports 92.9 Mbit/s user data payload using an overall bit rate of 149.76 Mbit/s. Block size at STM-1 is 18'792 bits and there are 8'000 blocks/s. Thus a given 64 kbit/s stream contributes a packet to every 80th block and the multiplex can carry about 149'760'000/(129 * 8 * 100) = 1'451 64 kbit/s streams. A block contains about 18 packets, all from different 64 kbit/s streams.

XI.3 Range of packet loss requirements

This clause derives the UNI-UNI packet loss ratio necessary to meet various transport accuracy requirements.

The first approach to derive a packet loss requirement refers to the ITU-T G.826 specification for a background block error ratio (BBER) of 2×10^{-4}, and loss of a single packet will result in a background block error. Hence packet loss ratio must be better than $2 \times 10^{-4}/18 = 1.1 \times 10^{-5}$ to meet the BBER specification.

The second approach examines the ITU-T G.826 errored second ratio (ESR) of 0.16. Since loss of a single packet will result in an ES, and there are 145'100 packets/s, the packet loss ratio must be better than $0.16/145'100 = 1.1 \times 10^{-6}$ assuming random packet loss. Because 0.16 is not very much less than 1, there is a small correction arising from the finite probability of two errors in the same second, which we have neglected for this approximate calculation. The packet loss ratio derived from ESR is about 10 times more demanding than from the BBER derivation, so the more stringent requirement of these two will be adopted.

The third approach considers the ITU-T G.826 specifications for ESR and severely errored second ratio (SESR) for sub-primary rate connections. With 10 ms packetization, loss of a packet results in loss of 640 payload bits which must be replaced by dummy data. On average, 320 bits will be in error, and an SES is a second in which the error ratio is 1.0×10^{-3}, so an SES will result for connections at rates less than or equal to 320 kbit/s (5 × 64 kbit/s, requiring 500 packets/s). The SESR for sub-primary-rate connections is 2×10^{-3}. Hence the packet loss ratio must be less than $2 \times 10^{-3}/500 = 4 \times 10^{-6}$ assuming random packet loss.

Thus, the different requirements within [b-ITU-T G.826] lead to somewhat different values of packet loss for an international IP network. The derived requirements range between 1.1×10^{-6} and 4×10^{-6} depending on the assumptions above and the specification from [b-ITU-T G.826]. However, both these figures are very much more stringent than the IPLR of 1×10^{-3} for QoS classes 0 through 4 of Y.1541.

XI.4 Effect of forward error correction

An alternative to achieving the very low packet loss ratio needed for the ITU-T I.231.1 service is to use forward error correction, trading bandwidth and extra delay for a less-demanding packet loss requirement. [b-IETF RFC 2733] describes a scheme for protected transmission of RTP streams through networks with packet loss. The scheme permits the use of a range of block FEC methods.

For example, (n, k) block codes may be used, generating n-k redundant packets from each k data packets and transmitting all n packets. All k data packets can be recovered provided any k packets of the n are received without loss or error. The probability of a residual error (one not corrected by the scheme) affecting a block, P_b, is equal to the probability of losing more than n-k packets from the block, hence approximately equal to the probability of losing $n - k + 1$ packets from the block. If packet loss is random, this is given by:

$$P_b \approx \Pr(n-k+1) = \frac{n!}{(n-k+1)!(k-1)!} p^{n-k+1}(1-p)^{k-1}$$

where p is the probability of loss of a single packet. As $p << 1$, the term involving $(1 - p)$ will always be close to 1 for the expected small values of k.

Consider a stream of payload packets of rate R packets/s. The rate of generation of blocks is R/k. Hence, the rate at which such blocks suffer from loss of more than n-k packets is RP_b/k. If a block suffers from loss of more than n-k packets, the worst case is that no payload packet is recoverable from the block, so the worst-case rate of loss of payload packets after FEC is $kRP_b/k=RP_b$. This is to be compared with a rate of loss of payload packets of $(R \times p)$ in the absence of FEC. Thus, P_b is an effective packet loss probability after FEC. P_b may be an overestimate of the effective packet loss probability, in cases where the FEC code allows recovery of some payload packets even after loss of more than n-k packets from the transmitted block.

To achieve the packet loss requirements derived above, we wish to make the post-FEC packet loss probability around 1×10^{-6} when operating on an IPLR of $p = 1 \times 10^{-3}$, which is assured on paths that are compliant with ITU-T Y.1541 classes 0 through 4.

A (k + 1, k) scheme was chosen for further analysis, because of its simplicity. Any (k + 1, k) block code which adds a single parity packet leads to a probability of a residual block error equal to a numerical factor (greater than 1) multiplied by p^2. The two simplest such codes are (2,1) (simple repetition, requiring double bandwidth) and (3,2) (needing only the exclusive-OR operation, increasing bandwidth by 3/2).

The probability of an error in the block for the (2,1) code is just p^2. For the (3,2) code, it is $3p^2$. For a loss requirement of 1.1×10^{-6} after correction, the requirement before correction is 1.05×10^{-3} for the (2,1) repetition "code". For the (3,2) code, the requirement before correction is 6.0×10^{-4}. Note that these are requirements on total packet loss, including both IPLR and packets arriving too late to be played out.

The 1.05×10^{-3} requirement is numerically close to the IPLR values for QoS classes 0 and 1, however this does not account for additional packets that arrive too late for play-out as permitted by the $1 - 10^{-3}$ quantile used in the IPDV specification.

Provisional QoS classes 5 and 6 offer much more stringent IPLR objectives and IPDV quantiles (1×10^{-5} and $1 - 10^{-5}$, respectively). Using the overall loss based on these values, it is clear that an FEC code can be designed to meet the loss requirement of 1.1×10^{-6} after correction using much less overhead. For example, a (14,13) code can correct a $p = 10^{-4}$ loss ratio to $91p^2$, or 9.1×10^{-7}.

The desire for low overhead must be tempered by the delay consumed in the FEC processing. Delay increases by at least $(k - 1)$ times the packetization time.

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