IPネットワークの伝送品質に関するパラメータ①

資料IP作1－6

(ITU-T Rec J.241 Quality of service ranking and measurement methods for digital video services delivered over broadband IP networks)

- IP – transport requirements

Packet Loss Ratio ･･･The ratio between the number of the packets lost in the network and the total number of transmitted packets1.

Latency ･･･The time interval between initial transmission and final reception of a packet.

Jitter ･･･The latency variation.

- IP network measurement parameters

Parameter	Equipment	Motivation	Monitoring method
Packet Loss Ratio	CPE (STB)	Image quality, video information loss estimation	• In service or through test streams with RTP/RTCP or sequence numbers available on packet header. • Periodic PLR summary. • Reports with one-minute resolution. • Measurement of PLR requires analysis of a number of packets at least 10 times greater than the number related to the target PLR value. • This determines the rate at which the PLR is reported.
Latency	Test probe at user side, within CPE (STB) or as closest as possible to user access link.	Smooth play-out	• Test stream
Jitter	CPE (STB)	Smooth play-out	• In service or through test streams with RTP/RTCP or timestamps available on packet header.

IPネットワークの伝送品質に関するパラメータ②

- IP transport measurements andIP end-to-end service availability

Packet Loss Ratio value

・・・It is preferable to specify PLR value that is "codec independent" and dimensioned on a worst-case scenario.
The PLR value needed to guarantee that an IP network seamlessly delivers video services is 10^{-5}.
The requirement for PLR $< 10^{-5}$ is considerably more stringent than the IPLR objectives currently specified in ITU-T Rec. Y.1541.
However, there are plans to support digital video transport with some new QoS Classes with the value of IPLR $< 10^{-5}$.
A PLR of 10^{-5} may appear a stringent requirement for the PLR. A rough estimation is done considering that potentially any video information loss will be noticed by the user.
The actual result of a packet loss is not predictable since it depends on the type of frame that is corrupted or on the part of the frame that is missing at the decoder (foreground, background, spatial, temporal, etc.). The degree of signal recovery in the presence of a certain loss depends on the power of the codec itself. Finally, the kind of scene that is being reproduced (steady, moving, etc.) greatly influences the chance that the user perceives video signal degradation.
To further reduce the BER offered to the video decoder, typical error-correction schemes can be applied on the video streams..

Latency and Jitter

・・・Latency and jitter values may vary according to specific multimedia service characteristics, such as interactivity, and according to the size of the de-jitter buffer and of the play-out delay employed at the CPE (STB) side.
For example, for high quality video streaming services, latency in the order of hundreds of milliseconds and jitter in the order of tenths of milliseconds may be tolerated.
It is recognized that the definition of objective values for jitter and latency needs further study, even taking into account the different application interactivity evolution, such as videoconferencing, which will impact the traditionally mainly unidirectional television service.

IP end-to-end service availability

・・The video service availability depends on the availability of all the elements that are controlled by the operator and that are significant for video service distribution, from the network device closest to the video source, to the access device closest to the user.
A classification of IP service availability is found in ITU-T Rec. Y.1540, a video streaming services availability function can be defined using the same approach: If PLR > PLR_out, then the service may be considered unavailable.
A value of 0.01 is proposed for PLR_out.
This value refers to a system where no FEC is employed; further study defining the FEC scheme, may, in the future, result in defining a different value for PLR_out. This evolution will be reflected in this Recommendation..

ＩＰネットワークのQoSクラス①
(ITU-T Rec Y.1541 Network performance objectives for IP-based services)

- 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

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).

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.

ＩＰネットワークのQoSクラス②

- <u>Guidance for IP QoS classes</u>

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.

（参考）略語

CPE	・・・	Customer Premises Equipment
RTP	・・・	Real Time Protocol
RTCP	・・・	Real Time Control Protocol
PLR	・・・	Packet Loss Ratio
IPLR	・・・	IP packet Loss Ratio
QoS	・・・	Quality of Service
BER	・・・	Bit Error Ratio
IPTD	・・・	IP packet Transfer Delay
IPDV	・・・	IP packet Delay Variation
IPER	・・・	IP packet Error Ratio
ATM	・・・	Asynchronous Transfer Mode
VTC	・・・	Video Teleconference
T1	・・・	Digital hierarchy transmission at 1.544 Mbit/s
E1	・・・	Digital hierarchy transmission at 2.048 Mbit/s