RECOMMENDATION ITU-R BT.1720*

Quality of service ranking and measurement methods for digital video broadcasting services delivered over broadband Internet protocol networks

(Question ITU-R 100/6)

(2005)

Summary

This Recommendation specifies performance requirements and objective measuring methods of quality of service (QoS) for the delivery of digital video broadcasting services over broadband Internet protocol (IP) networks. The specified performance requirements are based on an IP QoS ranking at various levels, from "excellent" to "out-of-service". They rely on the objective end-to-end measurement of the values of a small number of parameters on the delivered IP streams, performed at the consumer premises equipment and relayed back to the head-end. The recommended objective measurement methods and parameters are known to influence the QoS delivered to the user.

The ITU Radiocommunication Assembly,

considering

a) that the development of broadcast and non-broadcast television systems is being widely undertaken, and, with their development, new levels of potential image quality are available;

b) that with the development of new image transmission technologies in broadcast and non-broadcast television, television system parameters can be chosen on the basis of compromises between image quality and the cost of image;

c) that for the definition of the requirements for television systems and for the various sections of the service delivery chain, the potential level of image quality to be provided by these systems is an important element;

d) that ISO/IEC MPEG-2 have standardized the encoding and transport mechanisms for audio, video and accompanying data which are adopted for digital video services,

further considering

a) that digital television services have begun to be distributed in IP broadband networks through IP multicast technologies and protocols (IP multicast distribution is analogous to broadcasting techniques in the radio transmission world);

b) that, in an IP network, interactive television services, such as video-on-demand (VoD), usually associated with a unicast content distribution method, are now available to end-users;

c) that, in an IP network, video receivers decode IP delivered television channels to the TV display,

* This Recommendation should be brought to the attention of Telecommunication Standardization Study Group 9.

noting

1 that packet loss ratio, latency and jitter are crucial IP transport requirements for end-to-end IP network performance assessment,

recommends

1 that methods for quality of service (QoS) measurements for digital television broadcasting services streamed in a broadband IP network should be tailored to the specific features of the transport services provided by an IP communications network;

2 that, for video services, the requirements in *noting* 1 should be measured and used for IP end-to-end network performance as described in Annex 1;

3 that end-to-end measurements should be performed on the video stream after its IP packetized structure is removed as described in Annex 2;

4 that QoS should be measured end-to-end in order to provide a close approximation to the quality offered to the end-user, taking into account the influence of the IP network on the video stream; Annex 3 shows a system measurement model of a chain for IP transmission of television services.

Annex 1

IP layer

1 IP transport requirements

IP networks are multi-hop, may be complex and different transmission technologies are usually employed along the network paths. The transmission control protocol (TCP)/IP protocol stack sees all these as "below layer 3" layers.

Measurements and quality parameters at the IP layer make it possible to define reference values for network requirements that are agnostic of the underlying transmission technologies and are suitable for use in end-to-end quality assessment.

The noise introduced in an IP packet network is described by the following parameters:

– *Packet loss ratio (PLR)*: The ratio between the number of the packets lost in the network and the total number of transmitted packets[1].

[1] According to the measurement scheme and the methodology proposed in this Recommendation, the total number of lost packets in the PLR parameter is the sum of IP packet loss ratio (IPLR) and IP packet error ratio (IPER) as defined in ITU-T Recommendation Y.1541. A more complete definition of this parameter is given in ITU-T Recommendation G.1020 where § 7.7.1 defines "Overall (frame/packet) loss ratio" for frames or packets. Being the measurement header on top of the transport layer, if, for an IP packet, the IP or user datagram protocol (UDP) checksum fails, this packet will not be presented to the measurement (or real time protocol (RTP)) layer.

– *Latency*: The time interval between initial transmission and final reception time of a packet.
– *Jitter*: The latency variation.

The quality of the video streams will impose a minimum value for the downstream throughput requirements; upstream end-to-end throughput requirements depend on application interactivity requirements.

This Annex 1 does not guarantee that the classification that it provides is sufficient for assessing the perceived quality on a TV broadcasting over IP system, since IP end-to-end network performance is measured before forward error correction (FEC) is applied.

2 Video streaming IP service class

Video services, such as VoD or TV services, are classified also as streaming services. In a high-quality television environment they have the following high-level requirements:

– good audio/video quality;
– high availability;
– medium interactivity.

These high-level requirements should be translated into values for transport requirements for an IP network.

As specified in Annex 3, it is up to the head-end to introduce good quality video content into the network according to the maximum end-to-end bandwidth and packet rate available for video services. Any packet loss will reduce the quality of the video.

To preserve good quality of the image, a low value of packet loss is required.

3 IP transport measurements

The IP network layer should be unaware if the video signal, or any upper layer, is employing forward-error correction (FEC) or any error-correction techniques, and it should only guarantee the performance needed before any error-correction scheme is applied at any of the above layers.

3.1 Parameters

Table 1 lists IP network measurement parameters. All measurements should be taken from points B to point C in the system measurement model described in Annex 3.

TABLE 1

Parameter	Equipment	Motivation	Monitoring method
PLR	Customer premises equipment (CPE) (set top box (STB))	Image quality, video information loss estimation	In service or through test streams with RTP/real-time control protocol (RTCP) or sequence numbers available on packet header Periodic PLR summary: Reports with one-minute resolution Measurements of PLR requires analysis of a number of packets at least ten times greater than the number related to the target PLR value This determines the rate at which the PLR is reported
Network latency	Test probe at user side, within CPE (STB) or as closest as possible to user access link	Smooth playout	Test stream
Jitter	CPE (STB)	Smooth playout	In service or through test streams with RTP/RTCP or time stamps available on packet header
Downstream throughput	CPE (STB)	Service qualification, monitoring	Test signal representative of worst-case encoding scenario, throughput test
Upstream throughput	CPE (STB)	Service qualification, monitoring	Throughput test

3.2 Values

Before giving reference values for transport requirements, it is important to note that, in video services delivery architecture, a receiver buffer is employed at the CPE (STB) end to eliminate (to some extent) the jitter introduced by the network and to have a continuous video frame reproduction.

Values that should be achieved in the network are outlined and motivated in the next paragraphs.

3.2.1 PLR value

It is preferable to a 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}.

This requirement on PLR is considerably more stringent than the IPLR objectives currently specified in ITU-T Recommendation Y.1541[2].

[2] There are plans to support digital video transport with some new QoS classes with values 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 bit error rate (BER) offered to the video decoder, typical error-correction schemes can be applied on the video streams.

3.2.2 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 playout 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.

4 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 Recommendation Y.1540, a video streaming services availability function can be defined using the same approach: If PLR > PLR_out, then service may be considered unavailable.

A value of 0.01 is proposed for PLR_out[3].

5 IP network service classification

In relation to video services, the performance of an IP network can be classified based on the value of PLR offered to the end-user. The PLR must be measured between points B and C of the system measurement model described in Annex 3.

NOTE 1 − In relation to the delivery of video services, the inclusion of the effect of latency and jitter for IP network classification purposes, as well as the evaluation of the impact of the definition of an FEC system needs further study.

[3] 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.

Appendix 1
to Annex 1

Example of an IP network service classification

This Appendix provides an example of an IP network service classification.

The classification used for digital television services is given below:

PLR $\leq 10^{-5}$	excellent service quality (ESQ)
PLR < $2 \times 10^{-4} - 10^{-5}$ >	intermediate service quality (ISQ)
PLR < PLR_out – 2×10^{-4} >	poor service quality (PSQ)
PLR < PLR_out – 1 >	IP end-to-end service not available.

Table 2 shows IP layer service classes that are related to the QoS service perceived by the end-user. The picture quality also depends on encoding conditions (bit rate, picture size, intra-refreshing method, etc.) and transmission parameters (packet size, FEC, etc.).

The evaluation interval for end-to-end service availability is from 1 to 5 min.

The network service classification is based on an evaluation interval of 30 min.

The end-to-end performance of an IP network can then be calculated adding up the time intervals in which the measured PLR was within the above thresholds during the reported time-slot. This is shown in the following example:

TABLE 2

Class	Time ESQ %	Time ISQ %	Time PSQ %	Note
A	≥ 99.8%	Between 0 and 0.2	Between 0 and 0.1	To be computed in service
B	≥ 99.8%	Between 0 and 0.1	Between 0.1 and 0.2	To be computed in service
D	< 99.8%	–	–	To be computed in service

The end-to-end unavailable service time is not included in the above example.

Annex 2

End-to-end measurements

An IP network allows each CPE (STB) to also behave as a measurement end-point. This offers the valuable opportunity to have a measurement probe at each installed video CPE. Measurements and monitoring taken at the CPE are the ones closest to the user’s real experience of the service.

Using a CPE as a measurement probe raises some point of attention since the CPE is not under the physical control of the network operator and measurements may be affected by the user’s equipment (cable not well plugged, vertical cabling issues, improper use of the home network). The STB should have the capability to give additional information about the quality of the video signal that is being decoded; receiver buffer fullness and frame rate are two important indicators of service availability and overall performance. CPEs measurements should be used to:

– measure the end-to-end IP network performance;
– measure the network performance at any hierarchical level or aggregation point through statistical analysis and data processing exploiting correlation among data;
– estimate the video quality offered to the end-user of the service;
– perform dedicated test sessions using test signals for qualification and troubleshooting.

As an example, some network operators currently perform end-to-end measurements at all the STBs available in their residential network, in order to evaluate end-to-end video service quality and network performance; STBs periodically send back frame rate and packet loss reports to provide a continuous quality feedback about the service in progress.

1 Video receiver measurements

Table 3 shows the parameters that should be measured at video receivers to estimate video quality, as described in the system measurement model.

These measurements can be used for all the assessments outlined above.

TABLE 3

Parameter	Value	Equipment	Purpose	Monitoring method	Measurement path[(1)]
Video frame rate	As required by the video standards	STB	Image quality	In service through codec specific methods. Sampling	From A to D
Buffer underflows	Not applicable	STB	Image quality, smooth playout	In service, while playing video. Sampling. Measure underflows events and percentage of service time spent by the STB in an "underflow" state	D
Buffer overflows	Not applicable	STB	Image quality, smooth playout	In service while playing video. Sampling. Measure underflows events and percentage of service time spent by the STB in an "overflow" state	D
Coding specific parameters	Not applicable	STB	Image/service quality	In service, while playing video. Sampling	Not applicable

(1) See Fig. 2.

NOTE 1 − Further studies should address video quality parameters which can be returned by the STB decoder and that may help in better evaluating the video reproduction process that takes place at the decoder.

2 Frame rate analysis

Television standards may use different frame rates.

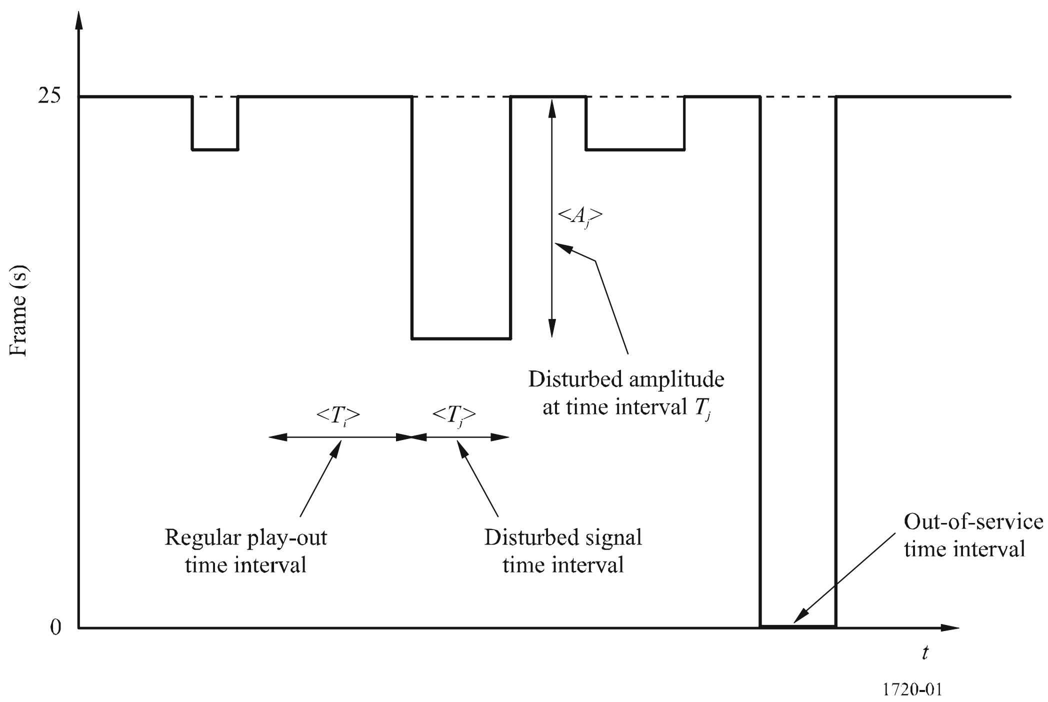
The output of the decoder will produce exactly the original frame rate, except in the presence of video information loss.

Measure of the frame rate at the output of the decoder, gives a rough estimate of the continuity of the service.

Figure 1 shows, as an example for a 25 frame/s video stream, possible information that can be retrieved through frame rate analysis:

FIGURE 1

Frame rate analysis



Annex 3

System measurement model

In its simplest form, the television services distribution model, in an IP network, consists of three parts:

– *Head-end*: This includes all the devices and applications needed to produce the video signals that are sent into the network.

– *Transport network*: This transports the video signal to the end-user CPEs.

– *The CPE*: This is an IP end-point (usually an STB) that decodes the video signal and displays it on a television set normally connected to it.

Explicit service level agreements (SLAs) need to be established for the transport of the video streams between the head-end and the transport network (in particular between the service provider and the network operator if they are not the same).

Audio, video, data and interactive services can be delivered in the IP transport network if the head-end and the STBs provide the necessary compliance. All the services and standards are compatible with the TCP/IP stack; the IP network should guarantee the required performance level and it should provide some test point where it can be measured. This Recommendation assumes that the quality of the input video signal that is delivered to the IP network is under the responsibility and control of the head-end.

The head-end should inject the video streams in the network according to transport rules appropriate to the IP network. These rules should define:

– maximum packet rate per stream;

– maximum number of sustainable streams;

– maximum bandwidth per stream (or packet rate for a given packet size);

– transport protocol to be used;

– frame size (transport layer);

– packet size;

– allowed inter-packet gap profile;

– maximum burst size.

On its side, the IP network should guarantee the agreed service level for the delivery of video streams to end-users.

In an IP network,VoD services are usually associated with unicast content distribution methods while television services are distributed by using IP multicast-based protocols.

IP transport protocol used for unicast distribution may be UDP or TCP while multicast distribution is transported on top of UDP.

The determination of the service level should be based on end-to-end measurements, which should provide information on:

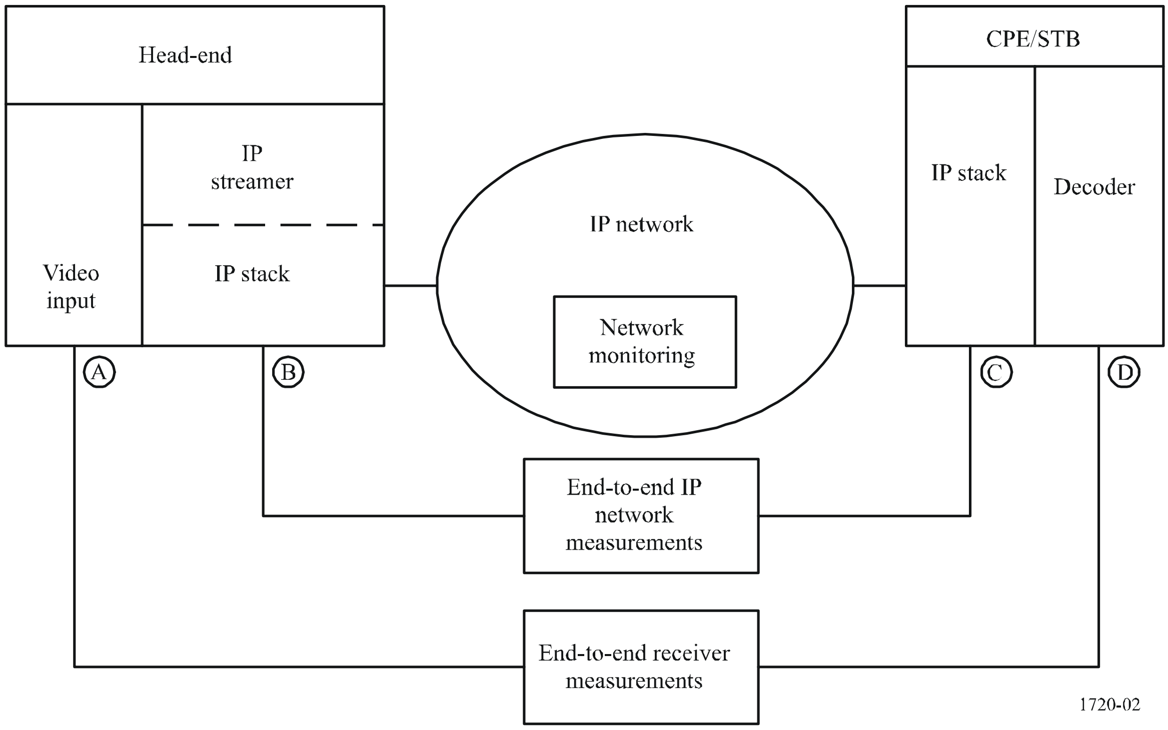
– the quality offered to the user;

– the influence of the IP network on the video signal.

Figure 2 shows the system measurement model that summarizes this approach.

FIGURE 2

System measurement model



The reference points A, B, C and D are described as follows:

Reference point	Description
A	Video encoder
B	IP layer at head-end (raw IP data)
C	IP layer at CPE (raw IP data)
D	Video decoder

Glossary

BER: bit error rate

CPE: customer premises equipment

FEC: forward-error correction

IPER: IP packet error ratio

IPLR: IP packet loss ratio

MPEG: Moving Picture Experts Group

PLR: packet loss ratio

QoS: quality of service

RTP: real-time protocol

RTCP: real-time control protocol

SLA: service level agreement

STB: set top box

TCP/IP: transmission control protocol/Internet protocol

UDP: user datagram protocol

VoD: video-on-demand

References

ITU-T Recommendation G.1020 (informative reference)

ITU-T Recommendation Y.1540 (normative reference)

ITU-T Recommendation Y.1541 (normative reference)