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International Telecommunication Union
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Recommendation

# ITU-T Y.1567 (10/2023)

SERIES Y: Global information infrastructure, Internet protocol aspects, next-generation networks, Internet of Things and smart cities

Internet protocol aspects – Quality of service and network performance

# Latency under load metrics and methods of measurement

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# Recommendation ITU-T Y.1567

## Latency under load metrics and methods of measurement


### Summary

There is considerable industry interest in conducting tests that capture latency of the user's path while in-use. The traffic load level associated with "while in-use" is widely proportional to the additional latency measured with respect to the minimum latency, and latency increases rapidly near the maximum load level. A user using application traffic will experience latency within measurable bounds: the lower limit of the minimum latency and the upper limit of latency at maximum capacity.

Although many measurement applications available at the time of writing assess different metrics of latency, they mostly employ the TCP protocol. Therefore, user datagram protocol (UDP) based latency under load/responsiveness is a clear gap. This gap is significant, because:

1. User applications/traffic with strict response time requirements will use UDP.
2. The percentage of UDP traffic is significant and continues to grow in proportion to TCP traffic.

Recommendation ITU-T Y.1567 specifies metrics of latency under simultaneous traffic load and defines methods of measurement to increase the specificity and repeatability of metric assessment.



### History [*]

| Edition | Recommendation | Approval | Study Group | Unique ID |
|---|---|---|---|---|
| 1.0 | ITU-T Y.1567 | 2023-10-29 | 12 | 11.1002/1000/15696 |



### Keywords

Internet protocol, measurements, performance, responsiveness.


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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.

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# Recommendation ITU-T Y.1567

## Latency under load metrics and methods of measurement

### 1 Scope

This Recommendation specifies metrics of latency under simultaneous traffic load and defines methods of measurement to increase the specificity and repeatability of metric assessment.

Key points defining this work are:

- Multiple metrics are required to assist with interpreting the results, such as the minimum delay, and one or more metrics of delay when the path is operating at maximum capacity.
- It is important to distinguish the network latency from the contribution to latency in end-hosts, since the latter varies with hosts and falls under different administrations. This Recommendation focuses on measuring the network latency.
- All metrics need to be carefully defined. Latency is a common term, and round-trip time or round-trip packet delay are more exact. Statistical measures will be used to summarize the many individual measurements of packet delay collected during a measurement interval.
- The methods are primarily intended for access paths to the Internet, possibly extending to one or more Internet gateways. Multiple transport protocols will be considered.
- Mobile networks are included in the scope, and with them the possibility of variable capacity over time.
- Bimodal capacity is characterized by the presence of two-time ranges where capacity, observed by repeated measurements, is distinguishably different. Typically, but not exclusively so, the capacity during the initial seconds of transmission is high, followed by a stable, lower capacity for the remainder of the transmission. It is intended to characterize the latency under load for each mode when present. Multimodal paths may also be present, requiring the characterization of each mode. There are at least two audiences for latency under load metrics: technical experts will be able to understand network performance metrics, but typical consumers do not (yet) have an appreciation of their needs on the latency continuum in terms of milliseconds. The results may need to be expressed in categories for consumers (such as red, yellow, green).

Note that technically the latency experienced by a packet queued in a buffer of constant depth depends on the capacity to forward (dequeue) the temporary buffered packets. If the dequeuing capacity is variable, so is the latency experienced by repeated measurement flows. The latency is proportional to the inverse of the dequeue capacity.

### 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 G.109] Recommendation ITU-T G.109 (2007), *Definition of categories of speech transmission quality.*

[ITU-T G.1070] Recommendation ITU-T G.1070 (2018), *Opinion model for video-telephony applications*.

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| [ITU-T G.1071] | Recommendation ITU-T G.1071 (2016), *Opinion model for network planning of video and audio streaming applications*. |
| [ITU-T G.1072] | Recommendation ITU-T G.1072 (2020), *Opinion model predicting gaming quality of experience for cloud gaming services*. |
| [ITU-T Y.1540] | Recommendation ITU-T Y.1540 (2023), *Internet protocol data communication service - IP packet transfer and availability performance parameters*. |
| [ITU-T Y.1565] | Recommendation ITU-T Y.1565 (2011), *Home network performance parameters*. |
| [IETF RFC 7799] | IETF RFC 7799 (2016), *Active and passive metrics and methods (with hybrid types in-between).* |

# 3 Definitions

## 3.1 Terms defined elsewhere

This Recommendation uses the following terms defined elsewhere:

**3.1.1 successful round-trip packet transfer outcome** [ITU-T Y.1565].

**3.1.2 mean round-trip packet transfer delay** [ITU-T Y.1565].

**3.1.3 population of interest** [ITU-T Y.1540].

**3.1.4 duration of the test** [ITU-T Y.1540].

NOTE – Called "duration of the load generation test" in this Recommendation to distinguish the duration of the load generation test interval from the duration of the latency under load measurement interval.

## 3.2 Terms defined in this Recommendation

This Recommendation defines the following term:

**3.2.1 latency under load**: Latency metric or measurement resulting from maximum IP-layer capacity load conditions in the case of UDP transport, and maximum throughput load conditions in the case of TCP transport.

# 4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

| | |
|---|---|
| ACK | Acknowledgement packet |
| AQM | Active Queue Management |
| CCA | Congestion Control Algorithm |
| HTTP | Hypertext Transfer Protocol |
| ICMP | Internet Control Message Protocol |
| MSS | Maximum Segment Size |
| MTU | Maximum Transmission Unit |
| QoE | Quality of Service |
| QoS | Quality of Experience |
| QUIC | Quick User datagram protocol Internet Connections |
| TCP | Transmission Control Protocol |

UDP User Datagram Protocol

## 5 Conventions

None.

## 6 Metric definitions

This clause defines the packet transfer parameters used to characterize latency using the specified measurement stream along with other specified operating conditions. This and the following clauses rely heavily on Figure 3 of [ITU-T Y.1565], included as Figure 1.

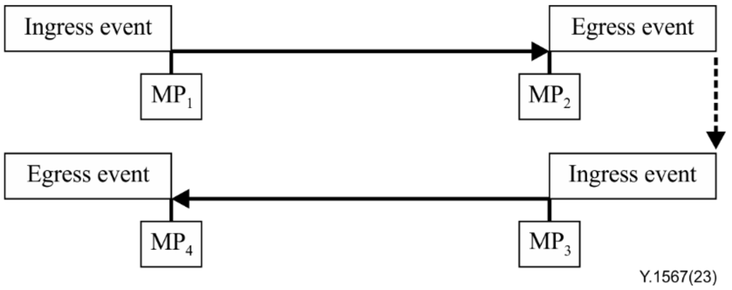


**Figure 1 – Round-trip reference events and measurement points (MPs) (Figure 3 of [ITU-T Y.1565])**

### 6.1 Singleton round-trip packet transfer delay

**Round-trip packet transfer delay**: The time to complete a successful round-trip packet transfer outcome, from $MP_1$ to $MP_4$ in Figure 3 of [ITU-T Y.1565]. Ideally, the time from $MP_1$ to $MP_4$ excludes the processing time from $MP_2$ to $MP_3$, but this may not be possible in practice or with some test protocols. The protocol layer at which this metric is assessed shall be reported with the results.

### 6.2 Statistics for round-trip packet transfer delay

The measurement system shall generate a stream of packets. Packets resulting in successful round-trip packet transfer outcomes comprise the population of interest for statistical summary.

**Minimum round-trip packet transfer delay**: Within the population of interest, the shortest time to complete a successful round-trip packet outcome, from $MP_1$ to $MP_4$ in Figure 3 of [ITU-T Y.1565].

**Maximum round-trip packet transfer delay**: Within the population of interest, the longest time to complete a successful round-trip packet outcome, from $MP_1$ to $MP_4$ in Figure 3 of [ITU-T Y.1565].

**Maximum round-trip packet transfer delay variation**: Within the population of interest, the longest time minus the shortest time to complete a successful round-trip packet outcome, from $MP_1$ to $MP_4$ in Figure 3 of [ITU-T Y.1565].

**Inter-quartile mean of round-trip packet transfer delay**: Within the population of interest reduced to the measurements above the first quartile and below the fourth quartile, the arithmetic mean of the time to complete a successful round-trip packet outcome, from $MP_1$ to $MP_4$ in Figure 3 of [ITU-T Y.1565].

## 7 Methods of measurement for metrics

The class of measurement methods for this Recommendation shall be the Active class, as defined in [IETF RFC 7799].

Referring to Figure 3 of [ITU-T Y.1565], the measurement system shall launch a test packet from $MP_1$ to $MP_2$ and when the packet is received at $MP_2$, as immediately as possible send a test packet from $MP_3$ to $MP_4$ in Figure 3 of [ITU-T Y.1565], recording the times at $MP_1$ and $MP_4$ (as a minimum).

Timestamp resolution shall be at least 1 millisecond, and relative timestamp accuracy of plus-or-minus 1 millisecond should be easily achievable for round-trip measurements (the times at $MP_1$ and $MP_4$ are read from the same clock).

The test packets should:

1. Employ the same transport protocol as the load-generating packets (described in clause 8). However, it is recognized that a specific test protocol may be more convenient to use for measurements (such as internet control message protocol (ICMP) when transition control protocol (TCP) transport is used for load generation).
2. Sample the round-trip characteristics of the measurement path sufficiently frequently to assess the extent of the delay distribution in a 10- to 15-second measurement. Sample measurements at 10 to 20 test packets per second have been used successfully. Measurement durations should be < 30 seconds total for sequential tests of upstream and downstream directions. At least 100 test packet transfer attempts should comprise the population of interest.
3. When loading both directions of the round-trip path simultaneously, it may be possible to conduct measurements on the near-complete set of load-generating packets, and to conduct the reciprocal round-trip measurements as well. This operation requires the measurement system to establish correspondence between each packet received at $MP_2$ and the resulting test packet sent from $MP_3$.
4. Typical application packet streams exhibit bursty characteristics, and the load generation should follow suit. Evenly spaced load packet streams are not recommended. Likewise, emulating (one or more) specific user application streams as the load is not recommended (due to complexity, adaptivity and design changes over time that mean that the stream characteristics also change). Since the burst characteristics are likely to change over the range of load rates, no defaults are supplied, except that the maximum transmission unit (MTU) or maximum segment size (MSS) shall be selected to avoid fragmentation.

Many of the measurement considerations applicable to maximum IP-layer capacity (UDP) measurement are directly applicable to the UDP measurements of latency under load. See [b-ITU-T Y.Suppl. 60] for additional specific considerations.

If a specific system can support measurement of one-way delay or delay variation, then these measurements may be helpful when interpreting the results. Although one-way delay measurements require sufficiently accurate time synchronization to support the timestamp resolution supplied, delay variation is a relative measurement and the time error cancels-out.

The methods make no assumption about the bottleneck buffer design, whether it be tail-drop or some form of active queue management (AQM).

## 8 Load generation

This Recommendation considers two methods to create the conditions necessary to measure latency under load: UDP-based methods and TCP-based methods. Note that HTTP-based methods (which may also use TCP) require application-layer support in the testing hosts, and therefore conflate the host/stack performance with the underlying network performance (the latter is the goal in this Recommendation), and that the use of strong encryption with QUIC [b-IETF RFC 9000] can increase the host processing time contribution to round-trip packet transfer delay.

A desirable feature for all methods is the ability to mark packets with differentiated services or explicit congestion codes. If the tested path exhibits different behaviour with packet markings, then it becomes important to exercise the relevant markings (the markings the network responds to) and report the results in this context.

## 8.1 UDP-based methods

The methods for UDP load generation are as specified in [ITU-T Y.1540] for maximum IP-layer capacity testing, supported by Annex B of [ITU-T Y.1540]. (Specifically, either the Type B and C load adjustment/search algorithm may be used. A fixed rate stream sufficient to reach maximum capacity may be applied too. Note, however, that a fixed rate stream may cause persistent congestion of the access path under measurement. To avoid that, a fixed rate measurement stream should be terminated after experiencing packet loss for a single digit second interval).

The population of interest to obtain the latency under load metrics should be measured during test interval of the load generation test, where the maximum load (IP-layer capacity) is attained. If the duration of the load generation test is identical with the latency under load measurement interval, capturing the minimum round-trip packet transfer delay of the access at low load conditions requires collecting the first round-trip packet transfer delay sample while load generation starts. This requires more than 5 seconds test duration using the Type B algorithm with 1 Gbit/s capacity, and more than 1 second when using the Type C algorithm. If the duration of the load generation test is larger than the latency under load measurement interval, capturing the minimum round-trip packet transfer delay of the access at low load conditions requires collecting a separate round-trip packet transfer delay sample during the initial load generation test sub-interval.

NOTE – Measurement of maximum capacity in channels with highly varying capacity as, e.g., in mobile networks may need dedicated parameter settings of the ITU-T Y.1540 Annex B Type B or C algorithm to reflect the actual maximum capacity in the observation period.

A typical test set-up requires transmitting the load generation stream(s) from $MP_1$ to $MP_2$ in Figure 3 of [ITU-T Y.1565], and arranging for a corresponding return stream from $MP_3$ to $MP_4$ in Figure 3 of [ITU-T Y.1565] to complete the path for round-trip packet transfer delay measurements.

Simultaneous measurement of maximum load in each direction would require another load generation transmitter from $MP_3$ to $MP_4$ in Figure 3 of [ITU-T Y.1565], and the corresponding return stream from $MP_1$ to $MP_2$.

Sequential measurement of the reciprocal direction with load generation transmitted from $MP_3$ to $MP_4$ in Figure 3 of [ITU-T Y.1565] is also recommended (with the corresponding return stream from $MP_1$ to $MP_2$).

## 8.2 TCP-based methods

The measurement system shall generate one or more TCP connections with sufficient byte streams to keep each connection maximally utilized. The TCP congestion control algorithm governing each connection will determine the transmission rates (and load) of the aggregate stream. The population of interest to obtain the latency under load metrics should be measured during test interval of the load generation test, where the maximum load (IP-layer capacity) is attained. If the duration of the load generation test is identical to the latency under load measurement interval, capturing the minimum round-trip packet transfer delay of the access at low load conditions requires collecting the first round-trip packet transfer delay sample while load generation starts. If the duration of the load generation test is larger than the latency under load measurement interval, capturing the minimum round-trip packet transfer delay of the access at low load conditions requires collecting a separate round-trip packet transfer delay sample during the initial load generation test sub-interval.

A typical test set-up requires transmitting the load on TCP connection(s) from $MP_1$ to $MP_2$ in Figure 3 of [ITU-T Y.1565], and the ACK (acknowledgement packet) stream from $MP_3$ to $MP_4$ in Figure 3 of [ITU-T Y.1565] to complete the path for round-trip packet transfer delay measurements. A testing protocol may be more convenient to use for round-trip measurements (such as ICMP).

TCP methods should include the possibility for sequential and simultaneous measurements with load in the reciprocal direction or both directions (as described for UDP-based methods above).

## 9 Results reporting

The context of each test should be reported as completely as known. This includes time of day, contracted access properties, access technology, IP-layer measurement path and other reliably known data pertaining to the test.

The population of interest used in the test shall be described. The description shall include the IP address family, the transport layer, the maximum packet size, the frequency of sampled round-trip packet transfer delay measurements, the direction of the load steams(s) when loading upstream or downstream or the simultaneous load case.

The layer of the measurement shall be reported. This is typically the IP-layer (including IP header bits) or the TCP payload layer (without headers).

The report shall include the version of the TCP congestion control algorithm (CCA), or the ITU-T Y.1540 Annex B load adjustment algorithm, and any parameter modifications.

The report shall include the maximum IP-layer capacity or TCP throughput attained during the load generation test interval.

The complete report shall include the complete set of latency metrics (in clause 3.1.2 and clause 6), expressed using the statistics provided. These results shall be expressed in milliseconds.

If a specific system can support measurement of one-way delay or delay variation, then these measurements may be helpful when interpreting the results.

## 10 Interpreting the results for multiple audiences

A key challenge when presenting numerical results is to provide a frame of reference to aid qualitative interpretation by users with a wide range of experience and expertise in performance measurement. Some examples:

– A 500 ms inter-quartile mean of round-trip packet transfer delay will result in poor interactivity for conversational applications such as voice and video conferencing.

– A 10 ms or less inter-quartile mean of round-trip packet transfer delay will result in poor cloud gaming quality if the maximum IP-layer capacity is below 1 Mbit/s.

A practitioner in quality of service (QoS) and quality of experience (QoE) assessment will immediately know that, but the casual user needs guidance to interpret the result. The Recommendation provides a method to interpret the latency results that will be useful to all users.

The method is based on the availability of QoS and/or QoE requirements for various interactive applications, and agreement on categorization of requirements into good, fair or poor categories with additional colourization of the results to enforce the categories, "green, yellow, red". ITU-T Study Group 12 has extensive literature to support the development of the objectives for many applications and has already conducted a categorization for speech quality results obtained using the E-model [ITU-T G.109].

Compared to a fix 'fail/pass' criterion, a categorization into, e.g., three categories is better suited to reflect a user's perception.

Wherever possible, the boundaries of maximum latency values for the categories should be taken from ITU-T or other normative references with, for example, planning models, such as [ITUT G.1070] [ITU-T G.1071] [ITU-T G.1072].

A well-defined example for a normative reference-based categorization of a video conferencing application by latency under load metrics is given in Appendix I.

# Appendix I

## Example of categorization of video conferencing applications for latency under load

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

This appendix provides a well-defined example for the categorization of a video conferencing application. Table I.1 provides the categorization of latency values based on [ITU-T G.1070].

**Table I.1 – Example of categorization of video conferencing applications for latency under load**

| Category | Good | Fair | Poor |
|---|---|---|---|
| Inter-quartile mean of round-trip packet transfer delay/ms | ≤ 240 | ≤ 1000 | > 1000 |

Each categorization requires a description of the application whose latency under load is to be categorized. The application description for the video conferencing example given here is based on the normative ITU-T G.1070 model. The assumptions of the latter were used to set the latency objectives applied above.

The application description below only provides background information to use for the network planning models referenced by [ITU-T G.1070], nothing more.

NOTE – The application objectives are derived from the presence of latency alone, given that sufficient capacity is present. When multiple sources of impairments are present on the measured path, it is expected that application tolerance to latency would be potentially more demanding than the objectives provided by sources focusing purely on latency.

**Table I.2 – Video conferencing application description (with selections)**

| Parameter | Selection |
|---|---|
| Video codec | ITU-T H.264 |
| Video resolution | VGA (720p) |
| Video framerate | 30 fps |
| Video bitrate (down/up) | 3 Mbit/s |
| Audio codec | ITU-T G.722 (opus not available) |
| Audio bitrate (down/up) | 100 kbit/s |
| Control bitrate (down/up) | 100 kbit/s (3.2 Mbit/s total) |
| Delay in terminals (to determine network delay threshold) | 200 ms |
| NOTES:<br>The most popular audio codec, Opus, is not available in the model, so a wideband alternative was chosen (ITU-T G.722).<br>Speech and Video components have MOS > 4.0 with delay = 243 ms. The model only applies for delay < 1000 ms. | |

The above-recommended method of categorization can be applied to other types of applications by using latency thresholds as recommended for those applications by the relevant standards.

Some applications tend to adapt to delay conditions with the goal to buffer as many long-delayed packets for subsequent play-out as possible. For applications of that kind, the statistic related to the upper bound of round-trip packet transfer delay measurements shall be used with Table I.1 to interpret the results; this statistic is the maximum round-trip packet transfer delay variation for latency under load.

It should be considered that the definition of an application into 'cloud gaming' or 'extended reality' might be too rough a classification. For actual applications of the latter kind, subdivisions based on required responsiveness of the targeted application and possibly for individual scenes or sections of an application should be made.

# Bibliography


[b-ITU-T G.722] Recommendation ITU-T G.722 (2012), *7 kHz audio-coding within 64 kbit/s*.

[b-ITU-T Y.Suppl. 60] ITU-T Y-series Suppl. 60 (2022), *Interpreting ITU-T Y.1540 maximum IP-layer capacity measurements*.

IETF RFC 9000] IETF RFC 9000 (2021), *QUIC: A UDP-Based Multiplexed and Secure Transport*.

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