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Part 2: Reference benchmarking
and KPIs for High speed internet

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650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ).

The present document is part 2 of a multi-part deliverable covering Reference benchmarking, background traffic profiles and KPIs as identified below:

Part 1: "Reference benchmarking, background traffic profiles and KPIs for VoIP and FoIP in fixed networks";

Part 2: "Reference benchmarking and KPIs for High speed internet";

Part 3: "Reference benchmarking, background traffic profiles and KPIs for UMTS and VoLTE";

Part 4: "Reference benchmarking for IPTV, Web TV and RCS-e Video Share".

Modal verbs terminology

In the present document "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**will**", "**will not**", "**can**" and "**cannot**" are to be interpreted as described in clause 3.2 of the ETSI Drafting Rules (Verbal forms for the expression of provisions).

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Introduction

The present document describes the reference benchmarking, background traffic profiles and key performance indicators for high-speed internet.

1 Scope

The offer of new NGN services requires new KPIs, QoS measurement and benchmarking methods which are needed to ensure the quality of new services. To ensure the comparability of test results, reference benchmarking methods and background traffic load profiles are needed. The present document describes key performance indicators and benchmarking methods for the spectrum of potential applications. All access technologies offered by the operator under test are considered.

The present document is the second part of the multi-part deliverable which consists of four parts. The present document contains the Reference benchmarking and KPIs for High speed internet.

2 References

2.1 Normative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at https://docbox.etsi.org/Reference.

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

The following referenced documents are necessary for the application of the present document.

[1] Recommendation ITU-T E.800 (2008): "Definitions of terms related to quality of service".

[2] ETSI TS 102 250-2: "Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 2: Definition of Quality of Service parameters and their computation".

[3] Recommendation ITU-T Y.1540: "Internet protocol data communication service - IP packet transfer and availability performance parameters".

2.2 Informative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] ETSI TR 101 578: "Speech and multimedia Transmission Quality (STQ); QoS aspects of TCP-based video services like YouTubeTM".

3 Definition of terms, symbols and abbreviations

3.1 Terms

For the purposes of the present document, the following terms apply:

benchmark: evaluation of performance value/s of a parameter or set of parameters for the purpose of establishing value/s as the norm against which future performance achievements may be compared or assessed

NOTE: The definition is taken from Recommendation ITU-T E.800 [1].

3.2 Symbols

Void.

3.3 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACK	Acknowledgement message
DL	DownLoad
DNS	Domain Name System
E2E	End-to-End
FFS	For Further Study
FIN	FIN message
FTP	File Transfer Protocol
GET	HTTP method GET
HTTP	Hypertext Transfer Protocol
IAD	Integrated Access Device
ICMP	Internet Control Message Protocol
IP	Internet Protocol
IPDV	IP Delay Variation
IPER	IP Packet Error Ratio
IPLR	IP Packet Loss Ratio
IPSBR	IP Packet Sending Bit Rate
IPTD	IP Packet Transfer Delay
IPTV	Internet Protocol Television
KPI	Key Performance Indicator
NGN	Next Generation Networks
PING	Packet INternet Groper (Send a packet to a computer and wait for its return)
POST	HTTP method POST
QoE	Quality of Experience
QoS	Quality of Service
RTP	Real-Time Transport Protocol
RTT	Round-Trip Time
SYN	SYNchronize Message
TCP	Transmission Control Protocol
UDP	User Datagram Protocol
UE	User Equipment
UL	UpLoad
UMTS	Universal Mobile Telecommunications System
URI	Uniform Resource Identifier
VoIP	Voice over IP

4 Quality of service parameter values for High-speed internet

4.0 KPIs overview

To determine the quality of high-speed internet connections, the following measurement and KPI values are determined.

Table 4.0-1: Overview of quality benchmarks for DNS

1.	DNS Lookup Time [ms]; see clause 5.1
2.	DNS Lookup Failure Ratio [%]; see clause 5.10 of [2]
3.	DNS Host Name Resolution Time; see clause 5.11 of [2]

Table 4.0-2: Overview of quality benchmarks for PING

1.	PING Average Time [ms]; is defined in clause 6.3.1 of [2]
2.	PING packets missing [%]; see clause 4.1.1
3.	PING Packets Errors [%]; see clause 4.1.2
4.	PING Failure Ratio; see clause 4.1.3

Table 4.0-3: Overview of quality benchmarks for HTTP

1.	HTTP Service Non-Accessibility [%]; see clause 6.8.1 of [2]
2.	HTTP Setup Time [s]; see clause 6.8.2 of [2]
3.	HTTP IP Service Access Failure Ratio [%]; see clause 6.8.3 of [2]
4.	HTTP IP Service Setup Time [s]; see clause 6.8.4 of [2]
5.	HTTP Session Failure Ratio [%]; see clause 6.8.5 of [2]
6.	HTTP Session Time [s]; see clause 6.8.6 of [2]
7.	HTTP Mean Data Rate [kbit/s]; see clause 6.8.7 of [2]
8.	HTTP Data Transfer Cut-off Ratio [%]; see clause 6.8.8 of [2]
9.	HTTP Content Compression Ratio [%]; see clause 6.8.9 of [2]
10.	HTTP peak download throughput [kbit/s]; see clause 4.2.11
11.	HTTP Response Time; see clause 4.2.1
12.	HTTP Download Throughput; see clause 4.2.2
13.	HTTP Download Throughput < x % of Bandwidth; see clause 4.2.3
14.	HTTP Percentage of invoiced Bandwidth; see clause 4.2.4
15.	HTTP Upload Response Time; see clause 4.5
16.	HTTP Upload Throughput; see clause 4.2.6
17.	HTTP Upload Throughput; see clause 4.2.7
18.	HTTP; Number of HTTP Sessions; see clause 4.2.8
19.	HTTP Dropped Sessions; see clause 4.2.9
22.	Number of HTTP Sessions (Daily number of HTTP sessions); see clause 4.2.9
23.	Traffic Volume DL (Daily total volume in Downlink)
24.	Traffic Volume UL (Daily total volume in Uplink)
25.	Number of active users
26.	Average duration of HTTP sessions [min]

Table 4.0-4: Overview of quality of service parameters for FTP

1.	FTP {Download|Upload} Setup Time [s]; see clause 6.1.2 of [2]
2.	FTP {Download|Upload} IP-Service Access Failure Ratio [%]; see clause 6.1.3 of [2]
3.	FTP {Download|Upload} IP-Service Setup Time [s]; see clause 6.1.4 of [2]
4.	FTP {Download|Upload} Session Failure Ratio [%]; see clause 6.1.5 of [2]
5.	FTP {Download|Upload} Session Time [s]; see clause 6.1.6 of [2]
6.	FTP {Download|Upload} Mean Data Rate [kbit/s]]; see clause 6.1.7 of [2]
7.	FTP {Download|Upload} Data Transfer Cut-off Ratio [%]; see clause 4.3.7
8.	FTP {Download|Upload} Service non - Accessibility [%]; see clause 4.3.8
9.	FTP {Download|Upload} throughput [kbit/s]; defined in clause 4.3.1
10.	FTP Upload/Download Throughput < x % of Bandwidth; defined in clause 4.3.2
11.	FTP Percentage of invoiced Bandwidth; defined in clause 4.3.3
12.	FTP Download Throughput < x % of Bandwidth; defined in clause 4.3.4
13.	File Download/Upload Data Transfer Cut-off; defined in clause 4.3.5
14.	FTP {Download|Upload} Service non - Accessibility [%]; defined in clause 4.3.6
15.	FTP Retransmission Ratio; defined in clause 4.3.9
16.	FTP {Download|Upload} packet loss rate [%]; defined in clause 4.3.8
17.	FTP {Download|Upload} Traffic Volume (Daily total volume in Downlink)
18.	Active Users (Daily number of unique users)
19.	FTP upload with parallel HTTP Download
20.	FTP upload Number of FTP Sessions with parallel HTTP Download
21.	FTP upload Service non - Accessibility with parallel HTTP Download [%]
22.	FTP upload Setup Time with parallel HTTP Download
23.	FTP upload IP service Access Failure Ratio with parallel HTTP Download [%]
24.	FTP upload IP-Service Access Setup Time with parallel HTTP Download [%]
25.	FTP upload Session Time [s] with parallel HTTP Download
26.	FTP upload Mean Data Rate with parallel HTTP Download [kbit/s]
27.	FTP upload Data Transfer Cut-off Ratio with parallel HTTP Download [%]

Table 4.0-5: Overview of TCP quality benchmarks

1.	TCP Round Trip Time (Server side); see clause 4.4.1
2.	TCP Round Trip Time (Client side); see clause 4.4.2
3.	TCP Retransmission Ratio; see clause 4.4.3
4.	TCP Data Call Access Failure Ratio; see clause 4.4.4
5.	TCP Data Call Access Time; see clause 4.4.5
6.	TCP Server Response Time; see clause 4.4.6

Table 4.0-6: Overview of IP/UDP quality benchmarks

1.	IP Sending Rate (Sender side); see clause 4.5.2
2.	Maximum IP Capacity (Receiver side); see clause 4.5.3

Table 4.0-7: Overview of quality benchmarks E-mail

1.	E-Mail {Download|Upload} Service Non-Accessibility [%]; see clause 7.2 of [2]
2.	E-Mail {Download|Upload} Setup Time [s]; see clause 7.2 of [2]
3.	E-Mail {Download|Upload} IP-Service Access Failure Ratio [%]; see clause 7.2 of [2]
4.	E-Mail {Download|Upload} IP-Service Setup Time [s]; see clause 7.2 of [2]
5.	E-mail {Upload|Download} Session Failure Ratio [%]; see clause 7.2 of [2]
6.	E-mail {Upload|Header Download|Download} Session Time [s]; see clause 7.2 of [2]
7.	E-mail {Upload|Header Download|Download} Mean Data Rate [kbit/s]; see clause 7.2 of [2]
8.	E-mail {Upload|Header Download|Download} Data Transfer Cut-off Ratio [%]; see clause 7.2 of [2]
9.	E-mail {Upload|Header Download|Download} Data Transfer Time [s]; see clause 7.2 of [2]
10.	E-mail Login Non-Accessibility [%]; see clause 7.2 of [2]
11.	E-mail Login Access Time [s]; see clause 7.2 of [2]
12.	E-mail Notification Push Failure Ratio [%]; see clause 7.2 of [2]
13.	E-mail Notification Push Transfer Time [s]; see clause 7.2 of [2]
14.	E-mail End-to-End Failure Ratio [%]; see clause 7.2 of [2]

Table 4.0-8: Network Diagnostic parameters

1.	Number of open parallel UDP ports (Default value 11)
2.	Number of open parallel TCP ports (Default value 16)
3.	Reachability of DNS entries (Default value 45)
4.	Reachability of Transparent connections (Default value 2) - for further study
5.	Reachability of Reference Webpage (Default value 1) - for further study
6.	Unchanged content transport (Default value 2) - for further study
7.	Radio level - for further study
8.	Stability of Radio level - for further study

Table 4.0-9: Overview of quality benchmarks for Network performance objectives for IP-based services

1.	Maximal IPDV
2.	Maximal IPTD
3.	Maximal IPLR
4.	Maximal IPER

Table 4.0-10: Overview of quality benchmarks for voice Media RTP

1.	RTP DL Throughput
2.	RTP UL Throughput
3.	RTP Latency
4.	RTP Jitter
5.	RTP Out of sequence
6.	RTP Dropped sequence
7.	RTP duplicate packets

Table 4.0-11: Overview of quality benchmarks for real-time video

1.	DL Throughput
2.	UL Throughput
3.	Latency
4.	Jitter
5.	Out of sequence
6.	Dropped sequence
7.	Duplicate packets

Table 4.0-12: Web Browsing QoS Parameters

1.	Website Response Time; see clause 5.2.3
2.	Website Load Duration Time; see clause 5.2.4
3.	Website Session Duration Time; see clause 5.2.5
4.	Website Download Failure Ratio; see clause 5.2.6
5.	HTTP Service Non-Accessibility; defined in clause 6.8.1 of [2]
6.	Number HTTP Sessions; defined in clause 4.2.8
7.	TCP Round Trip Time (Client side); defined in clause 4.4.1
8.	TCP Round Trip Time (Server side); defined in clause 4.4.2
9.	TCP Retransmission Ratio; defined in clause 4.4.3
10.	HTTP Service Setup Time; defined in clause 6.8.4 of [2]
11.	HTTP IP-Service Setup Time [s]; see clause 6.8.4 of [2]
12.	HTTP Setup Time [s]; see clause 6.8.2 of [2]

Table 4.0-13: Streaming QoS Parameters

1.	Streaming Service Non-Accessibility [%]; see clause 6.5.4 of [2]
2.	Streaming Service Access Time [s]; see clause 6.5.5 of [2]
3.	Streaming Reproduction Cut-off Ratio [%];see clause 6.5.6 of [2]
4.	Streaming Audio Quality; see clause 6.5.7 of [2]
5.	Streaming Video Quality; see clause 6.5.8 of [2]
6.	Streaming Audio/Video De-Synchronization; see clause 6.5.9 of [2]
7.	Streaming Reproduction Start Failure Ratio [%]; see clause 6.5.10 of [2]
8.	Streaming Reproduction Start Delay [s]; see clause 6.5.11 of [2]
9.	Streaming Teardown Failure Ratio [%]; see clause 6.5.12 of [2]
10.	Streaming Teardown Time [s]; see clause 6.5.13 of [2]
11.	Streaming Rebuffering Failure Ratio [%]; see clause 6.5.14 of [2]
12.	Streaming Rebuffering Time [s]; see clause 6.5.15 of [2]
13.	Number Streaming Sessions; see clause 6.12
14.	Streaming Mean Data Rate; see clause 6.9
15.	Streaming Peak Data Rate; see clause 6.10
16.	Effective Reproduction Time; see clause 6.11

Table 4.0-14: File Download/Upload QoS Parameter

1.	TCP Round Trip Time (Server side); see clause 4.4.1
2.	TCP Round Trip Time (Client side); see clause 4.4.2
3.	TCP Retransmission Ratio; see clause 4.4.3
4.	TCP Data Call Access Failure Ratio; see clause 4.4.4
5.	TCP Data Call Access Time; see clause 4.4.5
6.	TCP Server Response Time; see clause 4.4.6
7.	FTP {Download|Upload} Setup Time [s]; see clause 6.1.2 of [2]
8.	FTP {Download|Upload} IP-Service Access Failure Ratio [%]; see clause 6.1.3 of [2]
9.	FTP {Download|Upload} IP-Service Setup Time [s]; see clause 6.1.4 of [2]
10.	FTP {Download|Upload} Session Failure Ratio [%]; see clause 6.1.5 of [2]
11.	FTP {Download|Upload} Session Time [s]; see clause 6.1.6 of [2]
12.	FTP {Download|Upload} Mean Data Rate [kbit/s]]; see clause 6.1.7 of [2]
13.	FTP {Download|Upload} Data Transfer Cut-off Ratio [%]; see clause 4.3.7
14.	FTP {Download|Upload} Service non - Accessibility [%]; see clause 4.3.8
15.	FTP {Download|Upload} throughput [kbit/s]; defined in clause 4.3.1
16.	FTP Upload/Download Throughput < x % of Bandwidth; defined in clause 4.3.2
17.	FTP Percentage of invoiced Bandwidth; defined in clause 4.3.3
18.	FTP Download Throughput < x % of Bandwidth; defined in clause 4.3.4
19.	File Download/Upload Data Transfer Cut-off; defined in clause 4.3.5
20.	FTP {Download|Upload} Service non - Accessibility [%]; defined in clause 4.3.6
21.	FTP {Download|Upload} Retransmission Ratio; defined in clause 4.3.7
22.	FTP {Download|Upload} packet loss rate [%]; defined in clause 4.3.8
23.	FTP {Download|Upload} number of FTP Sessions
24.	FTP {Download|Upload} Traffic Volume (Daily total volume in Downlink)
25.	Active Users (Daily number of unique users)
26.	HTTP Service Non-Accessibility [%]; see clause 6.8.1 of [2]
27.	HTTP Setup Time [s]; see clause 6.8.2 of [2]
28.	HTTP IP-Service Access Failure Ratio [%]; see clause 6.8.3 of [2]
29.	HTTP IP-Service Setup Time [s]; see clause 6.8.4 of [2]
30.	HTTP Session Failure Ratio [%]; see clause 6.8.5 of [2]
31.	HTTP Session Time [s], see clause 6.8.6 of [2]
32.	HTTP Mean Data Rate [kbit/s]; see clause 6.8.7 of [2]
33.	HTTP Data Transfer Cut-off Ratio [%]; see clause 6.8.8 of [2]
34.	HTTP Content Compression Ratio [%]; see clause 6.8.9 of [2]
35.	HTTP peak download throughput [kbit/s]; see clause 4.2.11
36.	HTTP Response Time; see clause 4.2.1
37.	HTTP Download Throughput; see clause 4.2.2
38.	HTTP Download Throughput < x % of Bandwidth; see clause 4.2.3
39.	HTTP Percentage of invoiced Bandwidth; see clause 4.2.4
40.	HTTP Upload Response Time; see clause 4.5
41.	HTTP Upload Throughput; see clause 4.2.6
42.	HTTP Upload Throughput; see clause 4.2.7
43.	HTTP; Number of HTTP Sessions; see clause 4.2.8

44.	HTTP Dropped Sessions; see clause 4.2.9
45.	Number of HTTP Sessions (Daily number of HTTP sessions); see clause 4.2.9
46.	Traffic Volume DL (Daily total volume in Downlink)
47.	Traffic Volume UL (Daily total volume in Uplink)
48.	HTTP Dropped Connections [%]
49.	HTTP IP-Service Access Failure Ratio [%]
50.	HTTP Confirmation Time [ms]; see clause 6.1 (optional Website Benchmarking)
51.	Number of HTTP Sessions (Daily number of HTTP sessions); see clause 4.2.9
52.	Traffic Volume DL (Daily total volume in Downlink)
53.	Traffic Volume UL (Daily total volume in Uplink)
54.	Average duration of HTTP sessions [min]

Table 4.0-15: QoS aspects of TCP-based video services like YouTube™

1.	Player IP Service Access Failure Ratio [%]; see clause 4.3.1 of [i.1]
2.	Player IP Service Access Time [s]; see clause 4.3.2 of [i.1]
3.	Player Download Cut-off Ratio [%]; see clause 4.3.3 of [i.1]
4.	Player Download Time [s]; see clause 4.3.4 of [i.1]
5.	Player Session Failure Ratio [%]; see clause 4.3.5 of [i.1]
6.	Player Session Time [s]; [%]; see clause 4.3.6 of [i.1]
7.	Player Session Time [s]; see clause 4.3.7 of [i.1]
8.	Video IP Service Access Failure Ratio [%]; see clause 4.3.8 of [i.1]
9.	Video IP Service Access Time [s]; see clause 4.3.9 of [i.1]
10.	Video Reproduction Start Failure Ratio [%]; see clause 4.3.10 of [i.1]
11.	Video Reproduction Start Delay [s]; see clause 4.3.11 of [i.1]
12.	Video Play Start Failure Ratio [%]; see clause 4.3.12 of [i.1]
13.	Video Play Start Time [s]; see clause 4.3.13 of [i.1]
14.	Video Play Start Time [s]; see clause 4.3.14 of [i.1]
15.	IP Service Access Failure Ratio [%]; see clause 4.3.15 of [i.1]
16.	IP Service Access Time [s]; see clause 4.3.16 of [i.1]
17.	Video Session Cut-off Ratio [%];see clause 4.3.17 of [i.1]
18.	Video Session Time [s]; see clause 4.3.18 of [i.1]
19.	Impairment Free Video Session Ratio [%]; see clause 4.3.19 of [i.1]
20.	Video Expected Size [kbit]; see clause 4.3.20 of [i.1]
21.	Video Downloaded Size [kbit]; see clause 4.3.21 of [i.1]
22.	Video Compression Ratio [%]; see clause 4.3.22 of [i.1]
23.	Video Transfer Cut-off Ratio [%]; see clause 4.3.23 of [i.1]
24.	Video Transfer Time [s]; see clause 4.3.24 of [i.1]
25.	Video Mean User Data Rate [kbit/s]; see clause 4.3.25 of [i.1]
26.	Video Playout Cut-off Ratio [%]; see clause 4.3.26 of [i.1]
27.	Video Playout Cut-off Time [s]; see clause 4.3.27 of [i.1]
28.	Video Expected Duration [s]; see clause 4.3.28 of [i.1]
29.	Video Freeze Occurrences; see clause 4.3.29 of [i.1]
30.	Accumulated Video Freezing Duration [s]; see clause 4.3.30 of [i.1]
31.	Video Skip Occurrences; see clause 4.3.31 of [i.1]
32.	Accumulated Video Skips Duration [s]; see clause 4.3.32 of [i.1]
33.	Video Maximum Freezing Duration [s]; see clause 4.3.33 of [i.1]
34.	Video Freezing Impairment Ratio [%]; see clause 4.3.34 of [i.1]
35.	Video Freezing Time Proportion; see clause 4.3.35 of [i.1]
36.	End-to-End Session Failure Ratio [%]; see clause 4.3.36 of [i.1]

4.1 PING

4.1.1 PING Packets Missing [%]

The parameter value "PING Packets Missing" indicates the ratio in percentage of ICMP echo replies not received to the total number of initiated ICMP echo requests. An ICMP echo reply is deemed not received if it does not arrive within the timeout of 10 seconds from an ICMP echo request.

$$\text{PING Packets misssing \%} = \frac{\text{number of ICMP echo replies not received}}{\text{total number of ICMP echo replies received}} \times 100$$

4.1.2 PING Packet Errors [%]

The parameter value PING Packet Errors indicates the percentage ratio of ICMP echo replies with errors to the total number of ICMP echo replies received.

$$\text{PINGPacketsErrors \%} = \frac{\text{numberof ICMPechoerrorsreceived}}{\text{totalnumberof ICMPechorepliesreceived}} \times 100$$

4.1.3 PING Failure Ratio [%]

The parameter value PING Failure Ration represents the PING Failure Ratio as a percentage.

A PING measurement is considered unsuccessful if ICMP echo replies are missing, have errors (e.g. the target does not respond with the same value in the "Sequence Number"), or the mean response time (PING Average Time) exceeds 1 second.

$$\text{PING FailureRatio \%} = \frac{\text{unsuccessfull IMCPrequests}}{\text{successful IMCPrequests}} \times 100$$

4.2 HTTP

4.2.1 HTTP Response Time

The HTTP Response Time is defined as the time elapsed between the sending of the initial Sending of First TCP SYN and the receipt of the first HTTP GET command. To determine the various HTTP download parameters of the products, a standardized HTTP download to data reference systems is performed. In addition, four HTTP streams are set up in parallel. These HTTP downloads access sufficiently large files, the download of which is simultaneously interrupted after a defined period of time.

In this way, four HTTP Response Times are recorded per standardized HTTP download. The minimum value of the HTTP Response Times recorded is deemed the HTTP Response Time of the standardized HTTP download.

4.2.2 HTTP Download Throughput

NOTE: Following the testing specified and results presented in Recommendation ITU-T Y.1540 [3], this metric should not be used to measure access performance (using TCP transport). TCP throughput measurements tend to under-estimate the network operator's access performance.

A standardized HTTP download is conducted to determine the available downstream bandwidth. The average throughput is measured from opening the data connection to the end of the successful transfer of the content.

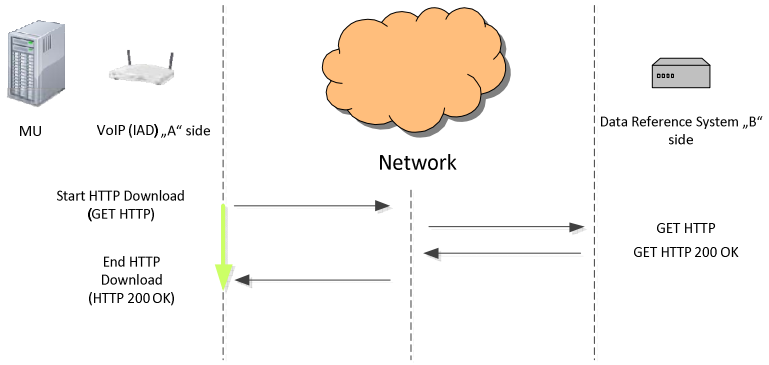

Figure 4.2.2-1: Measurement of the HTTP Download Throughput

Table 4.2.2-1: Description of the HTTP Download Throughput measurement

Event from abstract equation	Trigger point from user's point of view	Technical description/protocol part
$t_{data\ transfer\ start}$: Time of successfully started data transfer	Start: Web page download starts.	Start Method A: Reception of the first data packet containing content. Start Method B: Sending of the first GET command.
$T_{data\ transfer\ complete}$: Time when data transfer is complete	Stop: Web page download successfully is completed.	Stop: Reception of the last data packet containing content.
NOTE: There are web pages that do not have a well-defined last data packet, e.g. due to dynamic content. If such pages are used for measurement, then suitable end of page indications, e.g. provided by the browser, may be used as an alternative.		

4.2.3 HTTP Download Throughput [%] of Bandwidth

The value HTTP Download Throughput [%] of Bandwidth is the percentage ratio of the reduced HTTP downloads to the HTTP downloads undertaken in total.

$$HTTP\,Download\,Throughput\,\% = \frac{ratio\,of\,the\,reduced\,HTTP\,downloads}{HTTP\,downloads\,undertaken\,in\,total} \times 100$$

4.2.4 HTTP Percentage of invoiced Bandwidth [%]

The HTTP Percentage of invoiced Bandwidth indicates the ratio in percent of the actual measured HTTP download throughput against the nominally invoiced bandwidth.

$$HTTP\ invoiced\ bandwith\ \% = \frac{actual\ measured\ \ HTTP\ download}{nominal\ invoiced\ bandwith} \times 100$$

4.2.5 HTTP Upload Response Time

The HTTP Upload Response Time is defined as the time elapsed between the sending of an initial HTTP request (GET HTTP) to the receipt of the first HTTP Response Packet (TCP Packet).

4.2.6 HTTP Upload Throughput

The HTTP upload throughput is defined as the available upstream bandwidth in kbit/s.

NOTE: Following the testing specified and results presented in Recommendation ITU-T Y.1540 [3], this metric should not be used to measure access performance (using TCP transport). TCP throughput measurements tend to under-estimate the network operator's access performance.

4.2.7 HTTP Upload Throughput [%] of Bandwidth

The value HTTP Upload Throughput [%] of Bandwidth is the percentage ratio of the reduced HTTP uploads to the Nominal invoiced bandwidth.

$$\text{HTTP Upload Throughput \%} = \frac{\text{reduced HTTP upload}}{\text{HTTP nominal invoiced bandwith}} \times 100$$

4.2.8 Number of HTTP Sessions

Number HTTP Sessions is defined as a number of HTTP flows grouped together as per the heuristics below in order to comprise the view of a unique web page.

4.2.9 HTTP Dropped Sessions [%]

The number percentage of HTTP Dropped Sessions indicates the ratio between the dropped file DL/UL connections and the total file DL/UL Sessions

$$\text{HTTP dropped sessions Upload Throughput \%} = \frac{\text{HTTP dropped sessions}}{\text{total HTTP number of sessions}} \times 100$$

4.2.10 HTTP Download Failure Ratio [%]

The HTTP Download Failure Ratio is defined as the ratio of failed HTTP downloads to the total number of initiated HTTP downloads as a percentage.

$$\text{HTTP download Faillure Ratio} = \frac{\text{failed HTTP ddownloads}}{\text{total number of initiated HTTP downloads}} \times 100$$

4.2.11 HTTP peak download throughput

The definition of HTTP peak download throughput is for FFS.

4.3 FTP

4.3.1 FTP Upload/Download Throughput

The FTP Upload/Download Throughput determines the available upstream/downstream bandwidth.

NOTE: Following the testing specified and results presented in Recommendation ITU-T Y.1540 [3], this metric should not be used to measure access performance (using TCP transport). TCP throughput measurements tend to under-estimate the network operator's access performance.

4.3.2 FTP Upload/Download Throughput [%] of Bandwidth

The value FTP Upload Throughput [%] of Bandwidth is the percentage ratio of the reduced FTP Uploads/Download to the FTP Uploads undertaken in total.

$$\text{FTP throughpu t \%} = \frac{\text{reduced UL/DL FTP throughpu t}}{\text{FTP UL/DL taken in total}} \times 100$$

4.3.3 FTP Percentage of invoiced Bandwidth [%]

The FTP Percentage of invoiced Bandwidth indicates the ratio in percent of the FTP Upload/Download Throughput measured as against the bandwidth invoiced.

$$\text{FTP percentage of invoiced bandwith\%} = \frac{\text{reduced UL/DL FTP throughput}}{\text{nominal invoiced bandwith}} \times 100$$

4.3.4 FTP Download Throughput [%] of Bandwidth

The value FTP Upload Throughput [%] of Bandwidth is the percentage ratio of the reduced FTP Download to the FTP Download undertaken in total.

4.3.5 File Download/Upload Data Transfer Cut-off [%]

The parameter "File Download/Upload Data Transfer Cut-off" describes the probability that a successfully started FTP stream reproduction is ended by a cause other than the intentional termination by the user.

$$\textit{Data Cut-off Ratio [\%]} = \frac{\textit{unintenionally terminated FTP stream}}{\textit{all succesfully started FTP streams}} \times 100$$

4.3.6 FTP {Download|Upload} Service non-accessibility [%]

In networks where dedicated bearers are mandatory for access to a special service, an activation FTP failure results in service non accessibility for the user of the IP network requesting the service.

The dedicated FTP activation failure ratio measures the probability that a dedicated FTP transfer cannot be activated. It is the proportion of unsuccessful FTP activation attempts and the total number of dedicated FTP activation attempts.

4.3.7 FTP {Download|Upload} Retransmission Ratio

FTP Retransmission Ratio defines the number of counted retransmitted FTP/TCP packets divided by total number of counted packets within a certain direction (uplink/downlink) per user.

The formula used to compute this indicator is:

$$TCP\ Retransmission\ Ratio = 100 \frac{\sum_{TCP\ flows} Number\ of\ Resend\ Packets_i + Number\ of\ Wrong\ resent\ Packets_i}{\sum_{TCP\ flow\ interims} \text{total number of counted packets within a certain direction}}$$

Where:

- The number of Resent Packets for a given TCP flow is the sum of packets received with a TCP sequence number smaller than the next one expected in the same direction (that is last TCP sequence number plus last packet TCP payload size in the same direction).
- The number of wrong resent packets is the number of packets with a sequence number smaller than the last acknowledged one from the other peer.
- The total number of packets for a given TCP flow is the count of all the packets processed for a single TCP flow between both peers.

4.3.8 FTP {Download|Upload} packet loss rate [%]

The FTP {Download|Upload} packet loss rate [%] indicates the percentage ratio of lost FTP packets replies with the total number of FTP Packets received.

$$\text{FTP Packet lost rate \%} = \frac{\text{number of loss FTP } packets \text{ received}}{\text{total number of FTP packets received}} \times 100$$

4.3.9 FTP Retransmission Ratio [%]

The FTP Retransmission Ratio [%] indicates the ratio of retransmitted FTP packets replied with the total number of FTP Packets received.

$$\text{FTP retransmision ratio\%} = \frac{\text{number of retransmitted FTP } packets \text{ received}}{\text{total number of FTP packets received}} \times 100$$

4.4 TCP

4.4.1 TCP Round Trip Time (Server side)

Time from the SYN to the SYN/ACK packet in the TCP handshake (SYN - SYN/ACK - ACK) (see figure 4.4.2-1).

4.4.2 TCP Round Trip Time (Client side)

Time from the SYN/ACK to the ACK packet in the TCP handshake (SYN - SYN/ACK - ACK) (see figure 4.4.2-1).

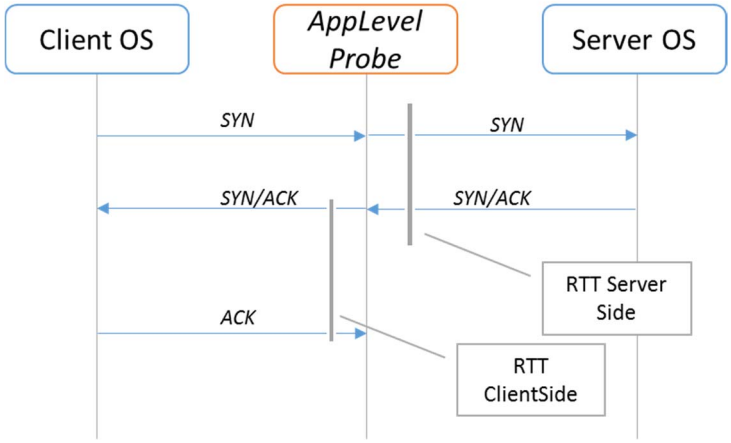

Figure 4.4.2-1: TCP RTT (Round-Trip Time): Client & Server Side

4.4.3 TCP Retransmission Ratio

TCP Retransmission Ratio defines the number of counted retransmitted TCP packets divided by total number of counted packets within a certain direction (uplink/downlink) per user.

The formula used to compute this indicator is:

$$TCP\ Retransmission\ Ratio = 100\frac{\sum_{TCP\ flows} Resend\ Packets_i + Wrong\ resent\ Packets_i}{\sum_{TCP\ flow\ interims} \text{total number of counted packets within a certain direction}}$$

Where:

- The number of Resent Packets for a given TCP flow is the sum of packets received with a TCP sequence number smaller than the next one expected in the same direction (that is last TCP sequence number plus last packet TCP payload size in the same direction).
- The number of wrong resent packets is the number of packets with a sequence number smaller than the last acknowledged one from the other peer.

- The total number of packets for a given TCP flow is the count of all the packets processed for a single TCP flow between both peers.

4.4.4 TCP Data Call Access Failure Ratio

The Data Call Access Failure Ratio defines the failure ratio in the period from initiating the data call to alerting or to the busy signal.

The formula used to compute this indicator is:

$$Data\ Call\ Failure\ Ratio\ [\%] = 100\ \frac{\sum_{Web\ browsing,file-sharing,streaming,email} number\ of\ flows\ with\ no\ downlink\ payload}{\sum_{Web\ browsing,file-sharing,streaming,email} number\ of\ initiating\ data\ calls}$$

4.4.5 TCP Data Call Access Time

Time from the TCP connection creation request (SYN) to the first packet of response (e.g. HTTP GET response) (Equivalent to the "IP Service Setup Time" for the services above: browsing).

4.4.6 TCP Server Response Time

The TCP Server Response Time is defined as the time from the first request packet (e.g. HTTP GET request) to the first received packet (e.g. first packet of HTTP GET response).

4.5 IP/UDP Capacity

4.5.1 Conventions and Definitions for IP/UDP Metrics

Recommendation ITU-T Y.1540 [3] defines a set of conventions for IP-layer metrics, which are re-used here. A key definition composed using these conventions determines when a packet has been successfully transferred between a Source host and Destination host (the two ends in end-to-end case referred to below). A transmitted packet is successfully received at the Destination host if it has error-free header(s) and payload identical to the corresponding packet transmitted at the Source.

Another key definition is the population of interest: for the *end-to-end case*, the population of interest is usually the total set of packets being sent from Source host to Destination host. The measurement points in the end-to-end case are at the Source and Destination hosts, and the locations of the Source and Destination hosts shall be determined by the direction of transmission intended to be measured. For example, a Download measurement would position the Destination host at the customer's location.

Using these definitions above, it is possible to define the IP-layer bits transferred: For a given population of interest, the IP-layer bits transferred are defined as eight (8) times the number of octets in all IP packets generating successful IP packet transfer outcomes at the Destination host, from the first octet of the IP header to the last octet of the IP packet payload, inclusive.

4.5.2 IP packet sending bit rate

For a given population of interest, the IP packet sending bit rate (IPSBR) generated by the Source host is the number of bits in the IP packet headers and payloads resulting in IP packet reference events divided by the time interval duration.

NOTE: This definition is consistent with Recommendation ITU-T Y.1540 [3].

4.5.3 Maximum IP-layer Capacity

For a given population of interest, the maximum IP-layer capacity during time interval [t, $t + \Delta t$] is:

$$Maximum\_C(t,\ \Delta t) = \frac{\max_{[t,\Delta t]}\left(n_0(dt_n, dt_{n+1})\right)}{dt}$$

where:

time interval [t, $t + \Delta t$] is composed of x equal sub-intervals, dt in length;

n_0 is the total number of IP-layer header and payload bits that can be transferred over a basic section generating successful IP packet transfer outcomes at the egress measurement point during a specified time interval, from [dt_1, dt_2] or other intervals dt in length, and the maximum_$C(t, \Delta t)$ corresponds to the maximum value of n_0 measured in any sub-interval [dt_n, dt_{n+1}] within time interval [t, $t + \Delta t$], divided by the duration of the sub-interval.

Note that UDP transport shall be used when assessing the Maximum IP Capacity Metric.

NOTE: This definition is consistent with Recommendation ITU-T Y.1540 [3].

4.5.4 Method of Measurement for Maximum IP-layer Capacity

The method(s) of measurement for the maximum IP-layer capacity shall meet the requirements in annex A of Recommendation ITU-T Y.1540 [3].

NOTE: The method(s) of measurement in Recommendation ITU-T Y.1540 [3] may be refined over time, resulting in new annexes.

5 Cloud Services

5.0 Introduction

With the cloud service tests the response time measurements are performed accessing favourite websites.

5.1 DNS Lookup Time

The DNS Lookup Time is defined as the time that elapses from the sending of a DNS request (DNS query) to the IAD until receipt of the resolved IP address (DNS query response).

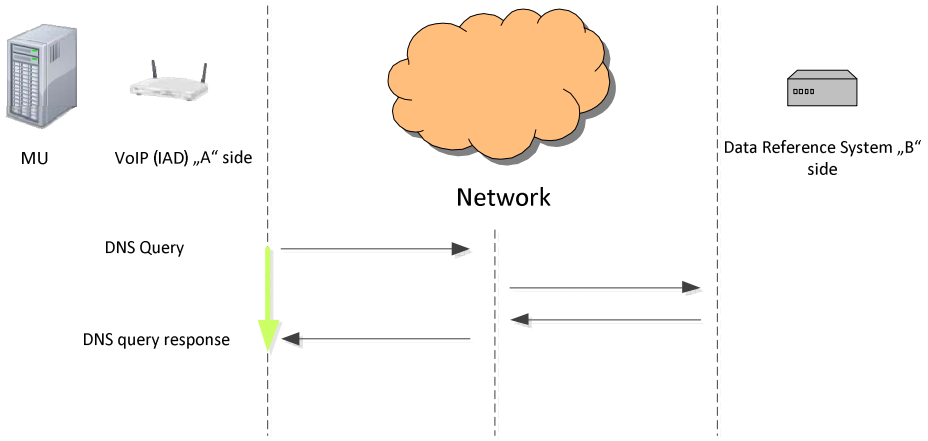

Figure 5.1-1: DNS Lookup Time

5.2 Web browsing indicators

5.2.1 Introduction

Web browsing indicators are some of the most important ones in order to measure the QoE on a per-subscriber basis. Figure 5.2.1-1 shows a sequence diagram with the measurements of the different indicators.

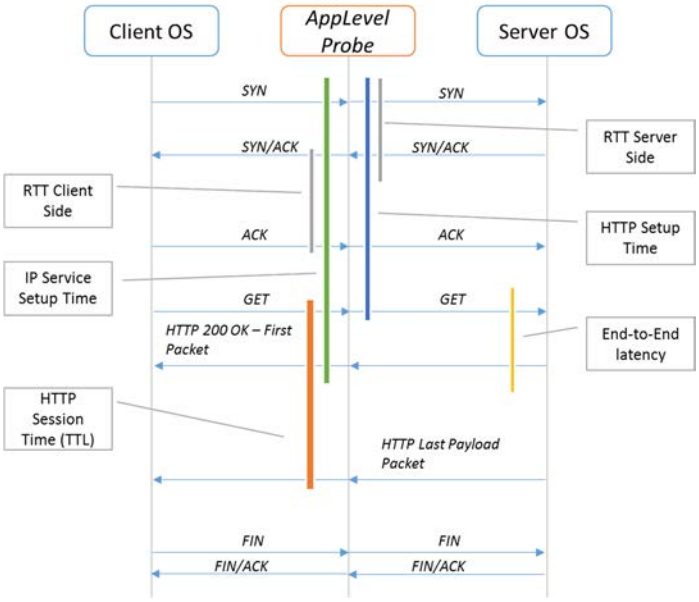

Figure 5.2.1-1: Web browsing

5.2.2 HTTP Concepts

The following terms are used to define the HTTP KPIs formulae:

- HTTP Flow: An HTTP flow is any TCP connection established between a Source IP Address with the Source IP Port to a Destination IP Address with a Destination IP Port where the destination IP port is :80 o :8080.

In the scope of this clause, Web Browsing indicators, the HTTP Flow is one that has been classified by the probe as a web browsing one, and it is distinguished from other HTTP flows that have been classified as streaming or file sharing.

As shown in figure 5.2.2-1 any TCP connection begins with a three-way handshake.

User A	Message	User B
SYN	⇨	SYN
SYN ACK	⇦	SYN ACK
ACK	⇨	ACK

Figure 5.2.2-1: Establishing TCP connection with a three-way handshake

- A GET/POST Method is a request for a resource to the destination server that includes the resource identification (URI) and a specific HTTP protocol version (e.g. HTTP/1.1).

Note that it is common that the web browsers open parallel flows when loading a specific web page, and that those flows can be kept opened and reused to load new pages on the same site.

- HTTP Session: HTTP Session is a heuristic term representing the duration of a web page download requested by the user.

Note that an HTTP Session can include multiple HTTP flows, for instance, when the browser opens flows in parallel to speed up the page download, or when the web page requires information from different hosts (for instance advertisement frames).

- HTTP Transactions at SITE level: The HTTP Transactions measured at Site levels are HTTP flows, independently of its classification by the probe. Hence, the HTTP transactions include web browsing, file sharing and streaming flows. The HTTP transactions KPIs are not part of the subscriber quality of experience KPIs and, if provided, are only provided at HTTP host level (SITE).

In order to understand the meaning of the web browsing QoE indicators, it is important to explain two different concepts:

- HTTP Flow: It refers to the TCP connection to a web server. It is called "flow" in the sense of an IP flow (defined as the quintuplet of Protocol - Source IP Address - Source Port - Destination IP Address - Destination Port, where protocol is the L4 protocol, like TCP or UDP).
- HTTP Session: It refers to the loading of a web page requested from a web browser, from the moment of the initial HTTP GET for the desired web page, until the reception of the last packet of the last object contained in that web page (like javascripts, pictures, advertisements, etc., which are automatically requested by the browser, by sending additional HTTP GETs).

Note that the relationship between HTTP Flows and HTTP Sessions is complex, because:

- An HTTP Session can include multiple HTTP Flows, for instance:
  - when the browser opens multiple TCP connections in parallel to the same server, to speed up the page download; or
  - when the web page includes objects from different hosts (for instance, advertisements).
- An HTTP Flow can include multiple HTTP Sessions, since most modern web browsers keep open the TCP connections, to be reused to load new pages from the same server.

5.2.3 Website Response Time

The Website Response Time is defined as the time that elapses between the sending of the initial HTTP request (GET HTTP) and the arrival of the complete HTTP response (see green arrow in figure 5.2.3-1).

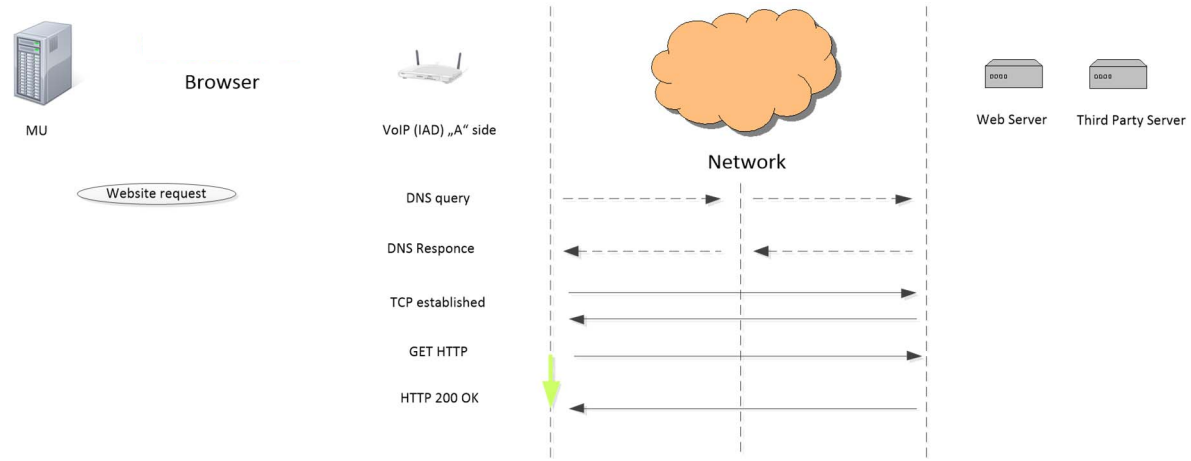

Figure 5.2.3-1: Website Response Time

5.2.4 Website Load Duration Time

The Website Load Duration Time (see figure 5.2.4-1) is defined as the time that elapses between the sending of the website request in the browser to the receipt and processing of all elements.

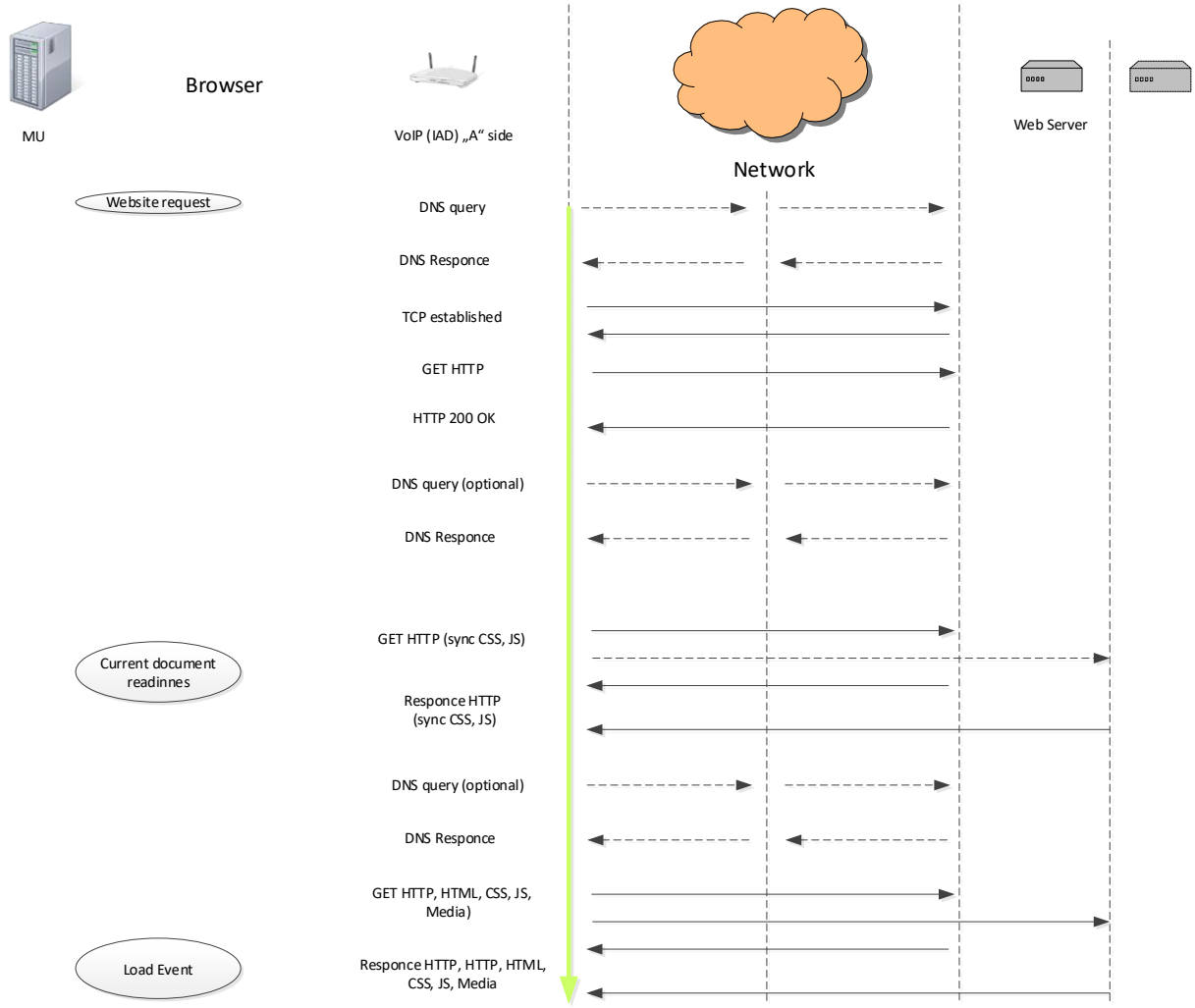

Figure 5.2.4-1: Website Load Duration Time

5.2.5 Website Session Duration Time

The Website Session Duration Time (see figure 5.2.5-1) is defined as the time that elapses between the sending of the website request in the browser to the receipt of all HTTP responses.

Figure 5.2.5-1: Website Session Duration Time

5.2.6 Website Download Failure Ratio

The HTTP Download Failure Ratio is the ratio of failed HTTP downloads to the total initiated HTTP downloads as a percentage.

If the HTTP download of a website is incorrect or exceeds the Website Response Time of 1 second, the HTTP download is rated as failed.

6 Streaming

6.0 QoE Streaming Indicators

Figure 6.0-1 shows what is measured by each of the key QoE indicators for the streaming service.

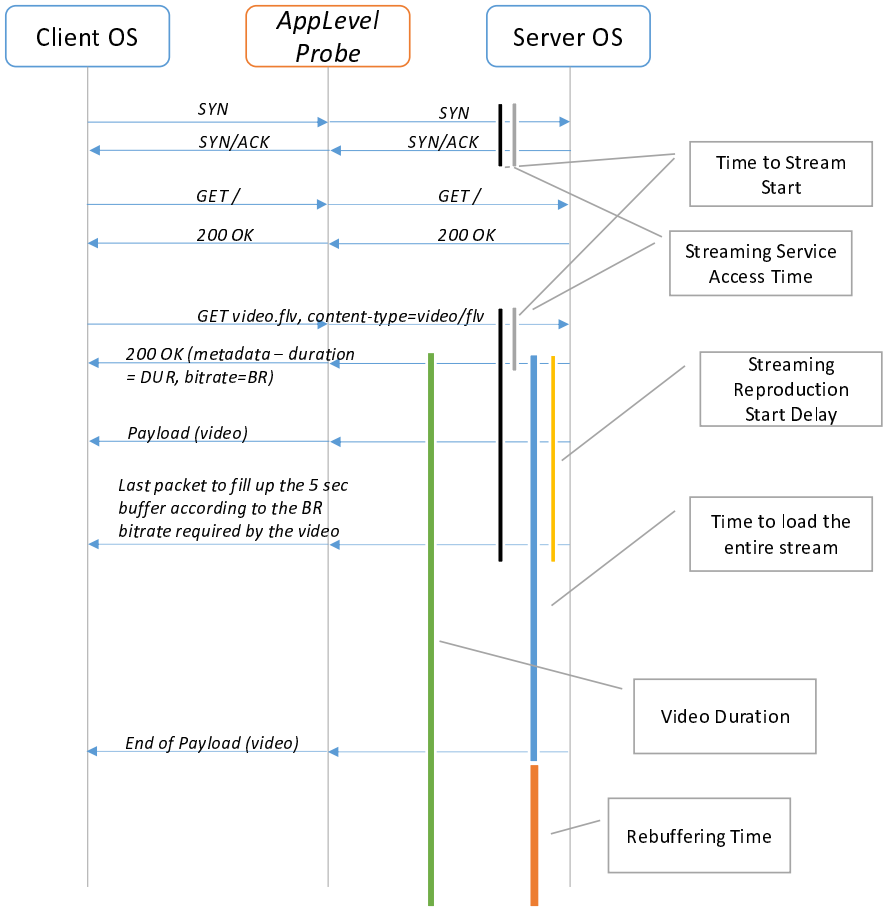

Figure 6.0-1: Streaming QoE indicators

6.1 Streaming indicators

This clause provides a thorough description of the QoE indicators related to the streaming category.

Figure 6.1-1 displays a summary of the different indicators in a sequence diagram.

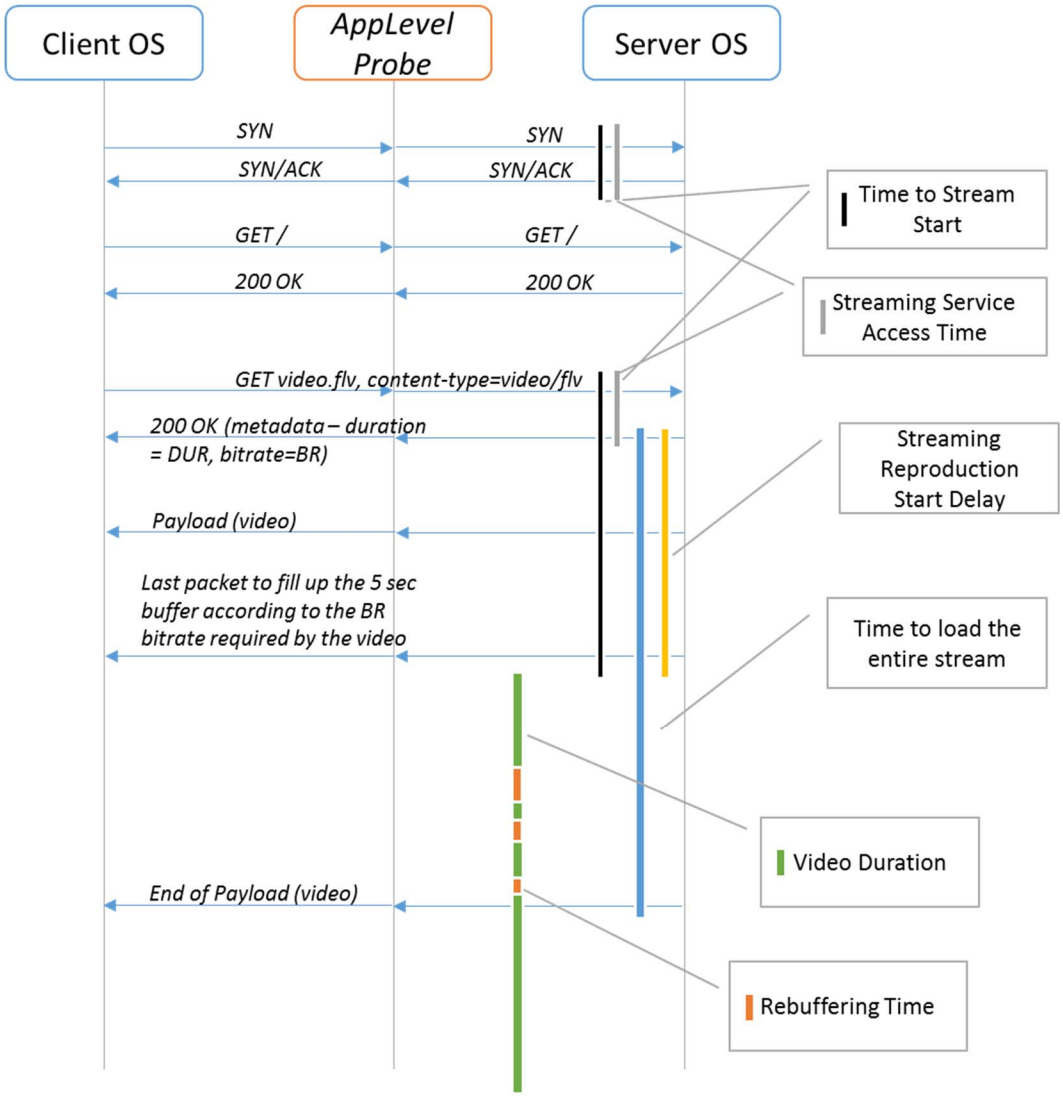

Figure 6.1-1: Streaming Indicators

6.2 Streaming Reproduction Cut-Off Ratio

Streaming Reproduction Cut-Off Ratio is described in clause 6.5.6 of [2].

6.3 Time to Stream Start

The Time to Stream Start is defined as the time between the User request for the Stream until the Stream starts playing. Streaming Service Access Time + Streaming Reproduction Start Delay.

This indicator is computed as:

$$Stream\ Start = \frac{\sum_{Streaming\ flows} Streaming\ Service\ Access\ Time_i + Streaming\ Reproduction\ Start\ Delay_i}{\#Reproduction\ Started\ Streaming\ Flows}$$

Where:

- The Streaming Service Access Time is the duration of a service access from requesting the stream at the portal until the reception of the first stream data packet at the UE.
- The Streaming Reproduction Start Delay is the duration between the reception at the UE of the first stream data packet and the start of the reproduction of the stream on the UE.

- The Number of Reproduction Started Streaming Flows is the total number of videos that started reproduction on the UE (flow with streaming payload).

6.4 Streaming Reproduction Start Failure Ratio [%]

Streaming Reproduction Start Failure Ratio describes the probability of unsuccessful stream reproduction. This includes streaming access failure before the service has been accessed.

This indicator is computed in the following way:

$$Streaming\ Reproduction\ Start\ Failure\ Ratio = 100 \frac{Number\ of\ Streaming\ Error\ Flows}{Number\ of\ Streaming\ Flows}$$

Where:

- Streaming Error Flows is the total number of flows requesting a video content (an HTTP transaction to a streaming site) ending with a 4XX or 5XX response or a TCP reset from the server or client.
- Streaming Flows is the total number of TCP flows identified from streaming sites by the application level probe.

6.5 Total Number of Videos

The "Total Number of Videos" is the total number of videos that started reproduction on the user equipment and have finished during the aggregation period.

6.6 Total Reproduction Time

Total Reproduction Time is the total duration of the video stream which has been downloaded. Note it is not the total duration of the video as stored on the server.

6.7 Effective Reproduction Time

Effective Reproduction Time is the average duration of the video stream which has been downloaded.

This indicator is computed in the following way:

$$Effective\ Reproduction\ Time = \frac{\sum Reproduction\ Time}{Number\ of\ Reproduction\ Started\ Videos}$$

Where:

- Reproduction Time is the total duration the video stream has been downloaded.
- Reproduction Started Videos is the number of videos that successfully started reproduction in the user equipment and finished during the aggregation period.

6.8 Rebuffering Time Percentage

Rebuffering Time Percentage is the percentage of the Streaming Rebuffering Time over Total Streaming Reproduction Time.

This indicator is computed as follows:

$$Rebuffering\ Time\ Percentage = 100 \frac{\sum_{streaming\ flows} Time(Rebuffering_i)}{\sum_{streaming\ flows} Streaming\ Effective\ Reproduction\ Time_i}$$

Where:

- *The "Rebuffering Time"* is the time for a given in-the flow spent downloading data from the point the video has been stopped playing by the player (stalled) due to exhaustion of its buffer to the point it resumes playing. Time to stream start is not taken into account in the time measurement. Once the player buffer is filled in again to the threshold required, the video resumes playing and the time measurement stops.

6.9 Streaming Mean Data Rate

The Streaming Mean Data Rate is defined as Streaming Session Average Throughput on streaming traffic.

6.10 Streaming Peak Data Rate

Streaming Peak Data Rate is defined as the Streaming Session Peak Throughput.

6.11 Effective Reproduction Time

The Effective Reproduction Time is defined as the time that the video has been actually played (not the nominal duration from metadata).

6.12 Number Streaming Sessions

Number Streaming Sessions is defined as the number of connections established against streaming services.

7 Gaming

7.0 Introduction

To determine the quality of high-speed internet gaming connections, the following measurement and KPI values are determined.

Table 7.0-1: Overview of quality benchmarks for gaming

Gaming	
1.	Number of Gaming Sessions
2.	Gaming Sessions Average Duration
3.	Gaming Sessions Average Duration [min]
4.	Gaming Average Throughput Uplink [kbit/s]
5.	Gaming Average Throughput Downlink [kbit/s]
6.	Gaming Overall response time [ms]
7.	Gaming, TCP Retransmission Rate Uplink
8.	Gaming, TCP Retransmission Rate Downlink
9.	Gaming, Packet loss rate Downlink
10.	Gaming, Packet loss rate Uplink
11.	Gaming, Latency Average E2E RTT
12.	Traffic Volume DL (Daily total volume in Downlink)
13.	Traffic Volume UL (Daily total volume in Uplink)
14.	Active Users (Daily number of unique users)

7.1 Number of Gaming Sessions

Number of Gaming Sessions is the number of flows identified as gaming sessions.

This metric is computed by summing up all the flows identified as gaming in the given time interval:

$$\#Gaming\ Flows = \sum\nolimits_{-} Gaming flows$$

7.2 Gaming Session Duration

The Gaming Session Duration is the average duration of gaming session flows.

This metric is computed by dividing the total duration of all gaming flows by the number of all flows in a given time interval:

$$Avg\ \text{Gaming Session Duration} = \frac{\sum_{\text{Gaming flows}} \text{Duration}_{\text{flow}}}{\text{Number of Gaming Flows}}$$

7.3 Gaming Average Throughput Uplink

Gaming Average Throughput Uplink is considered to be the average gaming throughput, so it is measured in the following way:

$$\text{Gaming Average Throughput Uplink} = \frac{\sum_{\text{Gaming Flows}} \text{Session Volume Uplink}}{\sum_{\text{Gaming Flows}} \text{Effective Session Time Uplink}}$$

Where:

- $Flow\ Volume$ is the total effective volume in the aggregation period.
- $Effective\ Flow\ Time$ is the total effective flow time for all flows altogether.

7.4 Gaming Average Throughput Downlink

The Gaming Average Throughput Downlink is considered to be the average gaming throughput, so it is measured in the following way:

$$Gaming\ Average\ Throughput\ Downlink = \frac{\sum_{Gaming\ Flows} Session\ Volume\ Downlink}{\sum_{Gaming\ Flows} Effective\ Session\ Time\ Downlink}$$

Where:

- $Flow\ Volume$ is the total effective volume in the aggregation period.
- $Effective\ Flow\ Time$ is the total effective flow time for all flows altogether.

7.5 Gaming Packet Loss Ratio (Uplink/Downlink)

Gaming Packet Loss Ratio (Uplink/Downlink) is the Number of counted loss TCP packets divided by total number of counted packets within a certain direction (uplink/downlink) per user.

The formula used to compute this indicator is:

$$TCP\ Loss\ Packets\ Ratio = 100 \frac{\sum_{TCP\ Gaming\ flows} Number\ of\ Loss\ Packets_i}{\sum_{TCP\ Gaming\ flow\ interims} Total\ number\ of\ flow\ Packets_i}$$

Where:

- The number of Loss Packets for a given TCP gaming flow is the sum of packets received with a TCP sequence number greater than the next one expected in the same direction.
- The total number of packets for a given TCP gaming flow is the count of all the packets processed for a single TCP flow between both peers.

7.6 Gaming TCP Retransmission Ratio (Uplink/Downlink)

Gaming TCP Retransmission Ratio (Uplink/Downlink) is the Number of counted retransmitted TCP packets divided by total number of counted packets within a certain direction (uplink/downlink) per user.

The formula used to compute this indicator is:

$$TCP\ Retr\ Ratio = 100\frac{\sum_{TCP\ Gaming\ flows} Number\ of\ Resend\ Packets_i + Number\ of\ Wrong\ resent\ Packets_i}{\sum_{TCP\ Gaming\ flow\ interims} Number\ of\ Total\ flow\ Packets_i}$$

Where:

- The number of Resend Packets for a given TCP gaming flow is the sum of packets received with a TCP sequence number smaller than the next one expected in the same direction (that is last TCP sequence number plus last packet TCP payload size in the same direction).
- The number of wrong resent packets is the number of packets with a sequence number smaller than the last acknowledged one from the other peer.
- The total number of packets for a given TCP gaming flow is the count of all the packets processed for a single TCP flow between both peers.

7.7 Gaming End-to-end Latency

Gaming End-to-end Latency is the measured round-trip time covering the complete end to end perspective (both within Operator network (only Operator network segment) and including external network elements (for example end to end latency for accessing public Internet servers).

The formula used to compute this indicator is:

$$E2E\ Latency = \frac{\sum_{Gaming} time(First\ DL\ packet\ with\ payload_i) - time(Last\ UL\ packet\ before\ first\ DL\ packet_i)}{\sum_{Gaming} Number\ of\ Flows\ with\ Payload}$$

Where:

- The Time of the "First DL packet with payload" is the time of the first packet seen in the downlink direction containing a payload.
- The Time of the "Last UL packet before first DL Packet" is the time of the last packet seen in the uplink direction before seeing any packet in the Downlink direction with payload.
- The number of "Flows with Payload" is the total number of TCP flows with payload seen by the probe in the aggregation interval.

From figure 7.7-1:

- It is the time of the first IP downlink packet with payload following the HTTP GET - time of the GET packet.
- Or the time of the first IP downlink packet with payload following the HTTP POST - time of the POST packet.

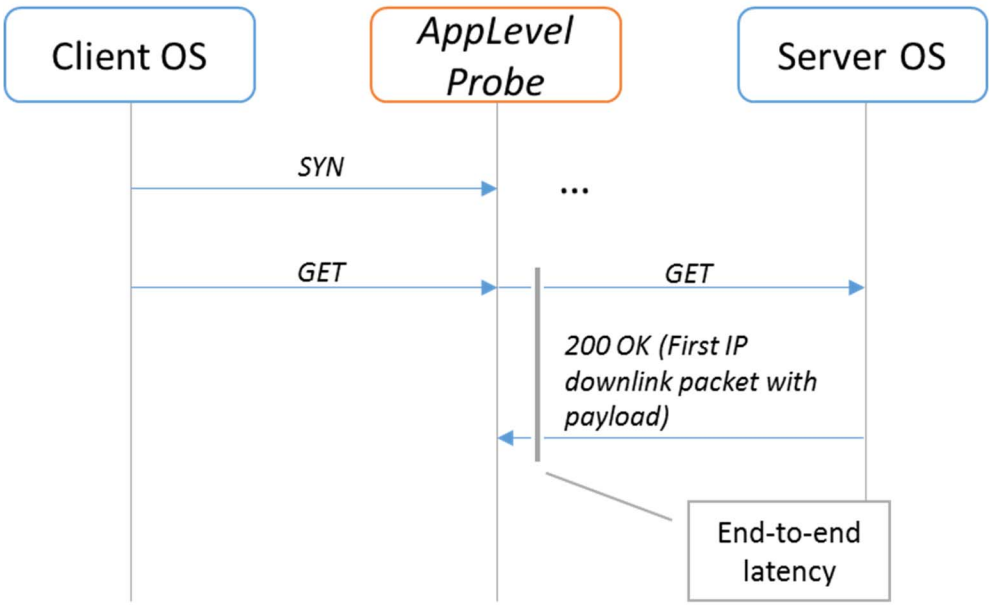

Figure 7.7-1: End-to-end latency

Notice, how in this case, the measurement is taken from the time of the GET to the time of the first packet of the 200 OK response (taken from the measurement point).

7.8 Gaming Traffic Volume (Uplink/Downlink)

Gaming Traffic Volume is the total number of bytes corresponding to gaming flows.

This metric is computed by adding up all the volume corresponding to flows identified as gaming in the given time interval:

$$\#Gaming\ Flows = \sum_{Gaming flows} Volume_{flow}$$

7.9 Daily Number of Unique Users

Daily Number of Unique Users is the daily number of unique users making use of gaming flows.

$$Time\ to\ Stream\ Start = \frac{\sum_{Streaming\ flows} Streaming\ Service\ Access\ Time_i + Streaming\ Reproduction\ Start\ Delay_i}{\#Reproduction\ Started\ Streaming\ Flows}$$

Where:

- The "Streaming Service Access Time" is the duration of a service access from requesting the stream at the portal until the reception of the first stream data packet at the UE.
- The "Streaming Reproduction Start Delay" is the duration between the reception at the UE of the first stream data packet and the start of the reproduction of the stream on the UE.
- The Number of "Reproduction Started Streaming Flows" is the total number of videos that started reproduction on the UE (flow with streaming payload).

History

Document history		
V1.1.1	December 2017	Publication
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