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Speech and multimedia Transmission Quality (STQ);
Throughput Measurement Guidelines

Reference

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Contents

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Foreword

This final draft ETSI Guide (EG) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ), and is now submitted for the ETSI standards Membership Approval Procedure.

Introduction

The main purpose of the present document is to help the reader understand and differentiate between various throughput definitions and calculation methods described in the TS 102 250 series and technical reports produced by STQ.

This guide describes the different aspects (e.g. protocol-specific, measurement-environmental, statistical) which should be considered during planning, execution and evaluation of throughput measurements in order to avoid the major problems which can occur.

TS 102 250-2 [i.2] standardizes throughput QoS parameters for popular IP based services used in mobile networks from the user's point of view. Based on these definitions TS 102 250-7 [i.6] defines a model for network quality. Finally, TR 102 678 [i.8] introduces a new method of throughput calculation based on fixed data transfer times.

1 Scope

The present document focuses on aspects of throughput measurements and their evaluation, providing different approaches. It contains factors, guidelines and background information that should be considered during selection, planning, execution and evaluation of throughput measurements.

2 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 http://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.

2.1 Normative references

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

Not applicable.

2.2 Informative references

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 TS 102 250-1: "Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 1: Assessment of Quality of Service".

[i.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".

[i.3] ETSI TS 102 250-3: "Speech and Multimedia Transmission and Quality (STQ); QoS aspects for popular services in mobile networks; Part 3: Typical procedures for Quality of Service measurement equipment".

[i.4] ETSI TS 102 250-4: "Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in mobile networks; Part 4: Requirements for Quality of Service measurement equipment".

[i.5] ETSI TS 102 250-6: "Speech Processing, Transmission and Quality Aspects (STQ); QoS aspects for popular services in GSM and 3G networks; Part 6: Post processing and statistical methods".

[i.6] ETSI TS 102 250-7: "Speech and multimedia Transmission Quality (STQ); QoS aspects for popular services in GSM and 3G networks; Part 7: Network based Quality of Service measurements".

[i.7] ETSI TR 102 607: "Speech Processing, Transmission and Quality Aspects (STQ); TCP IP Stack Parameter Settings for Microsoft Windows XP and Microsoft Windows Vista; Comparison and Recommendations".

[i.8] ETSI TR 102 678: "Speech and multimedia Transmission Quality (STQ); QoS Parameter Measurements based on fixed Data Transfer Times".

[i.9] ETSI TR 102 807: "Speech and multimedia Transmission Quality (STQ); Process description for the transaction view model".

[i.10] ITU-T Recommendation X.290: "OSI conformance testing methodology and framework for protocol Recommendations for ITU-T applications - General concepts".

[i.11] IETF RFC 793: "Transmission Control Protocol".

[i.12] IETF RFC1323: "TCP Extensions for High Performance".

[i.13] IETF RFC 3481: "TCP over Second (2.5G) and Third (3G) Generation Wireless Networks".

[i.14] M. Mathis, J. Semske, J. Mahdavi, and T. Ott: "The macroscopic behavior of the TCP congestion avoidance algorithm." Computer Communication Review, 27(3), July 1997.

3 Definitions, symbols and abbreviations

3.1 Definitions

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

application: software, using a particular service for providing related functionality to the user

Point of Control and Observation (PCO): point within a testing environment where the occurrence of test events is to be controlled and observed

NOTE: A PCO is identified by a) a reference point or interface and b) a service access point (SAP) at the specified reference point or interface, indicating unambiguously where (usually in a protocol stack) events are observed (or other measurements are made). See ITU-T Recommendation X.290 [i.10].

service: capability of a specific layer and the layers beneath to provide a set of functions to higher layers

NOTE: One example of the higher layers is the application layer.

transmission capacity: maximum achievable throughput, which is determined by the physical characteristics of the transmission media

NOTE: Examples of the physical characteristics of the transmission media are transmission capacity, applied modulation and coding scheme.

3.2 Symbols

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

Δt_d Predefined measurement time period

3.3 Abbreviations

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

CDF	Cumulative Distribution Function
CPU	Central Processing Unit
FDTT	Fixed Data Transfer Time
FTP	File Transfer Protocol
FTTX	Fiber To The X (of any type)
GGSN	Gateway GPRS Support Node
HDD	Hard Disk Drive
HSPA	High Speed Packet Access
HTTP	HyperText Transfer Protocol
IP	Internet Protocol

ISO	International Organisation for Standardization
MSS	Maximum Segment Size
NAT	Network Address Translation
NP	Network Performance
OSI	Open System Interconnection
PCO	Point of Control and Observation
PEP	Performance Enhancement Proxy
QoS	Quality of Service
RAM	Random Access Memory
RTP	Real Time Protocol
RTT	Round-Trip Time
SDP	Session Description Protocol
SIM	Subscriber Identity Module
SYN	TCP synchronise flag
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol / Internet Protocol
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications System
USB	Universal Serial Bus

4 General throughput measurement aspects

When measuring throughput, certain protocol-specific aspects always have an impact on measurement results. Thus, the purpose of the measurement and the respective measurement methodology has to be taken into account when evaluating and comparing measurement results. The specific protocol used to perform a measurement may have an impact on measurement results. In particular, throughput measurement results obtained using one protocol cannot generally be assumed to be transferrable to other protocols. For example, performing application layer measurements in two different mobile networks using both FTP and HTTP can result in a situation where one network produces better results for FTP measurements while the other one produces better HTTP results.

4.1 Purpose of the measurement

The three main parameters to characterise the actual performance of an IP network are delay, packet loss and transmission capacity.

The main cause why throughput is measured is to determine the achievable portion of the transmission capacity, being influenced by delay and packet loss, for different services. From a user's point of view the throughput is a key performance measure for a dedicated service perceived by the user on application level.

There is a notable difference between network throughput and application layer throughput which allows statements about Network Performance (NP) and Quality of Service (QoS), respectively. A proper disambiguation of the different perspectives can be found in clauses 5.1 to 5.3 of TS 102 250-1 [i.1].

4.2 Throughput equation

Throughput can be calculated using the following equation.

$$Throughput = \frac{Amount\ Of\ Transferred\ Data}{Duration}$$

The achievable IP transmission capacity is a highly variable stochastic parameter that varies in both small and large timescales. Every aspect of the throughput value has to be carefully listed; otherwise it can lead to misunderstandings.

4.3 Throughput on different layers

During packet transmission many protocols are involved. On every protocol stack layer the throughput metric can be interpreted. While from a user's point of view only the highest protocol stack layer is important, from a network operator's point of view the throughput on every protocol stack layer has its own meaning.

Depending on the used layer different PCOs apply and thus, the respective measurement results need to be treated accordingly. The concept of PCOs is explained in clause 7.1 of TS 102 250-1 [i.1] and in clause 5.1 of TR 102 807 [i.9].

For network throughput measurements, different measurement environments and also different tools are needed to be compare to application layer measurements, since different protocol layers are involved accessible at their own reference points. Thus, the protocol layer has to be chosen and protocol-specific aspects like e.g. stationary state for TCP have to be taken into account. For performance assessments of different services with respect to these network characteristics, corresponding application traffic has to be generated and measured on a network with "live" overall traffic load. Network throughput measurements and application layer throughput measurements cannot be mixed, even though both use service protocols such as FTP or HTTP.

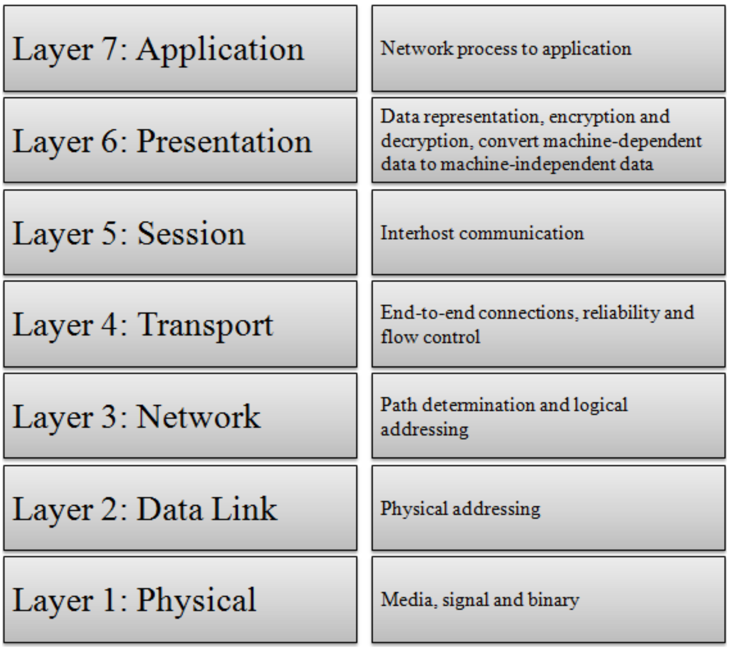


Figure 1: Idealistic view: ISO OSI Reference Model

The ISO OSI Reference Model is shown in Figure 1. The relation between the different protocol layer throughputs is typically the following:

$$Lower\ Layer\ Throughput \geq Higher\ Layer\ Throughput$$

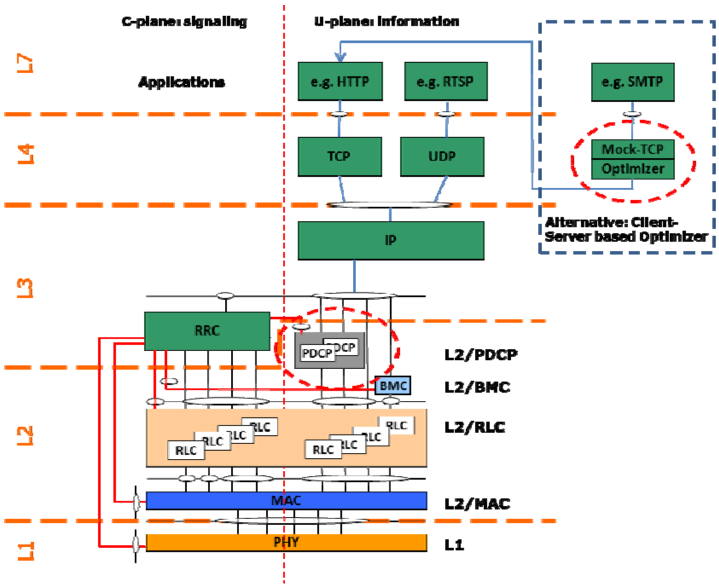
NOTE 1: Due to headers on higher layers the lower layer throughput can normally be assumed to be greater than higher layer throughput, i.e. ">" and not "≥".

Since every newly introduced layer increases the overhead, the amount of transferred data needed just for the communication increases, thus decreasing protocol efficiency:

$$Protocol\ Efficiency = \frac{Usable\ Traffic}{All\ Traffic}$$

NOTE 2: As stated above, higher layers include headers which add traffic and additional data to the transferred data which is not directly available or usable for the user.

NOTE 3: In the real world there are exceptions from the guidelines stated above as described in Figure 2 (real world example). Certain protocols, e.g. the Packet Data Convergence Protocol, are designed to increase protocol efficiency as compared to higher layers by performing a header compression with the goal of efficiently using scarce radio resources. Another example is the use of optimizer/compression software that can modify user data in order to increase the perceived throughput. In this case one can also say that lower layer throughput is bigger than higher layer throughput.



NOTE: Red circles indicate examples of header/payload compression.

Figure 2: Real world example: UMTS Protocol Stack (simplified) with internet applications and possible optimization

Therefore, the throughput performance of an end-to-end communication is usually different from a user's point of view compared to a network's point of view. In IP environments, upper layer protocols are executed in the end points (e.g. terminals) while the network only performs a transport function involving only lower layer protocols. The end user who is affected by the whole end-to-end path – including terminals - has to deal with the full protocol stack while the network only deals with a subsection of the path involving only a few protocol layers.

Reporting just the measurement results without providing information about the layer involved and the protocol used and the purpose of the measurement, e.g. network or application measurement, can lead to misinterpretation. Furthermore, different service protocols use the underlying transport protocols differently and should thus be reported, as well. For further details, please refer to clause 5.5.2.

In certain testing environments a service will not or cannot be evaluated by trigger points defined on just one layer. Start and end trigger points can for example be on different layers or need to be mapped to other layers. For example, when using a Performance Enhancement Proxy (PEP) the mapping of trigger points may refer to a point between the network and the application layer, most commonly the transport layer. A detailed look at the influences of PEPs can be found in clause 4.2.1 of TS 102 250-2 [i.2].

4.4 Throughput at different reference points

From a user's perspective the achievable throughput on the end-user side is important. This metric can be quite different if it is applied at any other point of the network.

In case of TCP for instance, a connection terminates in two dedicated points from the user's point of view, e.g. the PC running the web browser that is accessing a web server on a remote machine. From the network point of view, many intermediate connections and therefore many more endpoints can be found, e.g. routers or firewalls. The different properties of these TCP connections (traffic, retransmission rate) will result in different throughputs.

Performing network throughput measurements against these endpoints (destination hosts) can help finding bottlenecks in the network or to compare different IP aggregation points along the service line. As an example a feasible measurement scenario for testing different GGSN performance can be to measure the achievable TCP transmission capacity against a FTP server located at the GGSN's Gi interface. Another measurement scenario would be to test the performance of firewalls or NAT or L3-L7 deep packet inspection tools along the service line by placing an FTP server before and after these equipments and comparing the measurement results. Measurement results for different destination hosts on different logical places along the network should not be aggregated.

4.5 Active and passive measurement

An active measurement generates specific traffic on the network under test and is commonly used on access level. A passive measurement captures "live" traffic at dedicated points in the network. This type of measurement is usually used in the core network.

It is difficult to calculate the achievable throughput per user from the captured "live" traffic in the core network. For such passive measurements it is e.g. not known if the user did not want to or could not generate more traffic on the access side due to technical limitations of the client applications. Furthermore, the user behaviour is not predictable, but has direct impact on the calculated throughput.

When using an active measurement to determine the core link capacity in terms of the maximum achievable throughput, it should be considered that there usually is additional user and/or signalling traffic on the core link. In such a scenario, the active measurement will measure the throughput that is achievable with the remaining (free) network capacities not already allocated to other resources/traffic.

Commonly, active throughput measurements are performed on access network side generating traffic in order to saturate the available transmission capacity. Passive throughput measurements, on the other hand, are more common in the core network.

4.6 Load generation for active measurement of IP throughput

There are several ways to evaluate the IP network throughput from an end-user's perspective. Most commonly the IP network throughput is measured based on UDP or TCP since it cannot be measured directly on layer 3. In IP networks the respective traffic generation is usually done using TCP, because the majority of applications nowadays use this protocol to transfer data. UDP can also be used for throughput measurements, since the measured UDP throughput represents the upper bounds for TCP throughput in case the IP packets are in the same QoS class.

TCP was designed to adapt to the actual IP transmission capacity and to avoid congestion. To this end TCP provides mechanisms such as e.g. slow start, fast retransmit and fast recovery. In addition TCP constantly monitors the actual RTT and packet loss rates and uses those values to optimize the transmission capacity utilisation. For the setup and the evaluation of the measurement many protocol specific aspects should be considered with respect to TCP, such as e.g.:

- TCP/IP stack parameters;
- stationary state of TCP;
- RTT;
- packet loss.

The aspects listed above influence the achievable throughput. For further information concerning the TCP/IP stack parameters, please refer to TR 102 607 [i.7].

When speaking about throughput a stationary state is assumed, i.e. the TCP congestion control mechanisms had enough time to adapt to the actual capacity of the network.

Usually the stationary state throughput itself reflects the throughput TCP can reach in the long run, with the time it takes to reach it also being important. To assess how fast users can access network resources, measurements with small files are recommended due to the fact that the TCP slow-start has a stronger effect compared to large files.

A common way to generate TCP traffic is by using FTP or HTTP. In case of the evaluation of the achievable throughput the focus is not on the service evaluation, i.e. the service is used in order to generate traffic. However, it has to be taken into account that other network specific aspects can influence the achievable throughput for different services and thus the achievable IP throughput, such as e.g.:

- protocol-dependent routing;
- intermediate nodes, e.g.:
  - PEP;
  - NAT;
  - firewalls.

When the traffic generator, e.g. some FTP server, is able to send out data packets with a higher rate than the network can transmit the difference between the different application layer protocols is negligible assuming that any Layer 7 analyser running in the network analyses the packets at the same speed and TCP has achieved a stationary state. E.g. all IP services using TCP as a transport service should provide nearly the same results when TCP has achieved a stationary state. This might not necessarily be true when QoS classes are used on the Internet and in access networks (mobile, xDSL, FTTX) assigning services to different service classes, for which specific service profiles might be defined. With respect to this, the protocol used to measure the achievable IP throughput for a specific service class has to be chosen carefully, e.g. FTP for background traffic, HTTP for interactive traffic and RTP for real-time traffic.

In case of UDP the measurement is more straightforward as UDP does not use any throughput control mechanisms like TCP does: packets are sent by the sender without any regard of the capabilities of the intermediate network or the receiver, i.e. the sender will not adapt to the actual transmission capacity and the measurement only counts the received bytes.

From a test methodology's point of view the packet send rate should be chosen carefully and has to be higher than the expected achievable transmission capacity of the network under test. Also, the expected packet loss should be considered.

Although a UDP measurement provides a very good estimation for IP throughput if the measurement is well designed, it does not provide any usable estimation for TCP throughput. The reason behind this is that no RTTs and packet loss rates are measured and no traffic and congestion control functions will be taken into account. In addition network elements such as traffic shaper or profiler usually only influence TCP traffic by dropping packets, but not UDP traffic.

4.7 "Best effort" versus "windowed" approach

In the following clauses the "best effort" and "windowed" approaches are introduced and compared using two examples.

4.7.1 "Best effort" approach

The "best effort" approach is a best-practice method to achieve a maximum of measurement samples for a fixed test file size within a given time. Measurement locations/areas with higher achievable throughput contribute more measurement samples than those with lower achievable throughput. The mean value of all measured speed samples would be biased by this effect.

Using the "best effort" approach as many data transfers as possible are measured within a given time. The number of transfers depends on data transfer speed whereas the transferred amount of data per transfer is constant (not considering retransmissions).

Figure 3: "Best effort" approach

Figure 3 illustrates the best effort approach. A number of subsequent transfers is measured but not all transfers last for the same duration.

4.7.2 "Windowed" approach

Using the "windowed" approach, data transfers are executed within measurement windows of predetermined length. The number of transfers is constant and the transferred amount of data per transfer depends on the data transmission ratio.

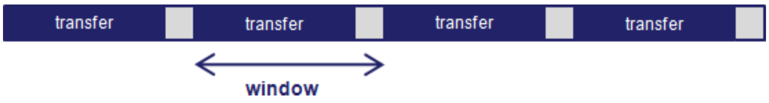


Figure 4: "Windowed" approach

Figure 4 illustrates the windowed approach. Here it can be seen that all transfers last for the same duration, i.e. the configured window size.

The "windowed" approach is best suited to achieve a more regular kind of sampling for network performance. As the sampling is regular in the time domain one could approximately achieve a regular geographical sampling by doing tests, e.g. as drive test with low but constant velocity. The mean value of all measured transfer speeds would better reflect the average experienced speed under the assumption that users use the network at random locations.

As in reality network usage is not random or evenly distributed onto the area a geographic traffic distribution would have to be considered to achieve a mean speed value even better reflecting user experience.

With respect to the findings above and especially in order to fulfil the common request for many QoS measurements to have a limited and regular runtime for individual measurement tasks, TR 102 678 [i.8] describes the concept of Fixed Data Transfer Time QoS (FDTT-QoS) parameters for data measurements, especially for FTP and HTTP data transfers.

NOTE: The "windowed" approach cannot be used to determine valid session failure ratios for the service used since the data transfer is interrupted after a given time.

4.7.3 Bias effect for "best effort" and "windowed" approaches

With the "best effort" approach measurement locations/areas with higher achievable throughput contribute more measurement samples than those with lower achievable throughput. "Slow" samples have just a slight effect on the mean user data rate calculated from all measurement samples.

With the "windowed" approach measurement locations/areas with higher achievable throughput contribute in the same way as those with lower achievable throughput.

4.7.4 First example

Figure 5 shows an example comparing "best effort" and "windowed" approach for one network.

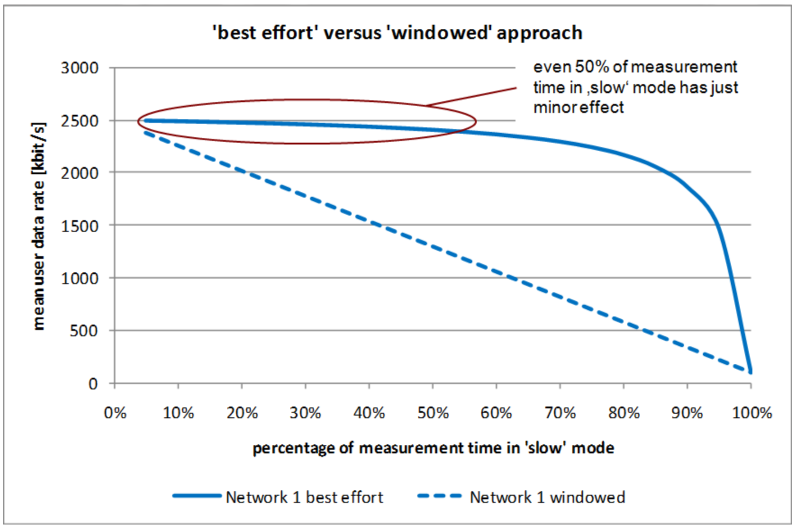


Figure 5: Comparison of "best effort" and "windowed" approaches

For the sake of simplicity assume a network providing only two dedicated speeds: 100 kbit/s and 2 500 kbit/s (e.g. EDGE and HSPA). The graph shows the mean value of the user data rate calculated from (virtual) measurements in this network with a certain percentage x of time in "slow" mode (100 kbit/s) and (1-x) in "fast" mode (2 500 kbit/s).

Observations:

- With the "best effort" approach the slow parts of the network affect the mean user data rate only in a significant way if more than half of the time was spent in "slow" mode.
- With the "windowed" approach slow parts of the network affect the mean user data rate proportionally to the time spent in "slow" mode.

4.7.5 Conclusions from first example

With the "best effort" approach the mean user data rate measured in networks providing "medium" and "high" speed (e.g. pure 3G network) can be lower than the mean user data rate measured in networks providing "slow" and "high" speed (e.g. 2G/3G network).

The reason is that even if both networks are in "high" speed mode for the same duration the higher number of "medium" speed samples collected in the remaining measurement time in one network contribute more to the mean value than the lower number of "slow" speed samples in the other network.

4.7.6 Second example

The second example compares "best effort" and "windowed" approaches for two networks, as shown in Figure 6.

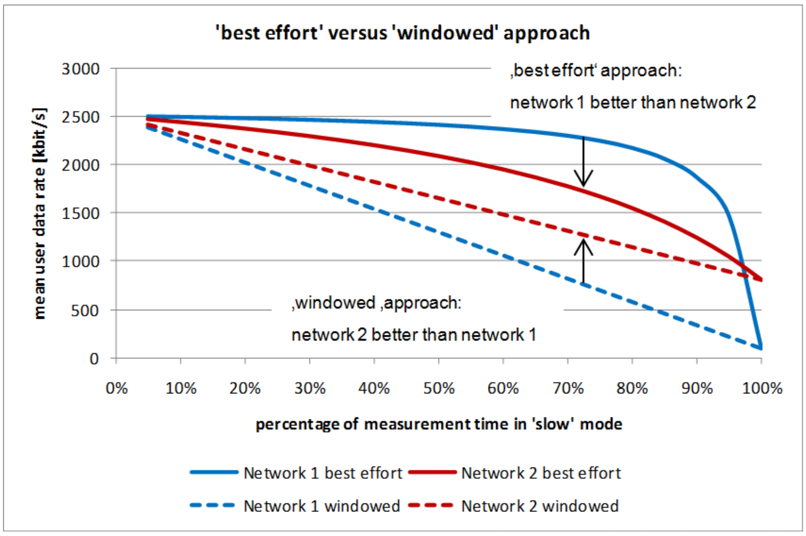


Figure 6: Comparison of approaches for two networks with different speeds

For the sake of simplicity assume the same network (1) as before providing only two dedicated speeds: 100 kbit/s and 2 500 kbit/s (e.g. network with EDGE and HSPA). Assume a second network (2) also with only two speeds: 800 kbit/s and 2 500 kbit/s (e.g. only HSPA).

The graph shows the mean value of the user data rate calculated from (virtual) measurements in these networks with a certain percentage x of time in "slow" mode (100 kbit/s for network 1 and 800 kbit/s for network 2).

Observations:

- With the "best effort" approach, network 2 having a complete 3G HSPA coverage and showing the same amount of time the same high speed is evaluated worse than network 1 having EDGE/HSPA coverage.
- With the "windowed" approach network 2 is evaluated better than network 1.

4.7.7 Conclusions from second example

The "best effort" approach is common and best practice for assessing end-to-end Quality of Service parameters. A big advantage is the fact that end-user-focused test cases can be created easily, e.g. download of a file or web page with typical content. Thus the "best effort" approach is the best practice method for testing particular services (web browsing, video sharing websites, email, etc.).

But one should act with caution when concluding from "best effort" measurements to user experience or just to expected performance of other services than measured. The "windowed" approach should be used in addition to the "best effort" approach in order to assess overall network performance and to conclude upon the expected service-independent overall user experience.

4.7.8 Combining "best effort" and "windowed" approach

There is the possibility to combine the implementation ease of the "best effort" approach and the fairness of the "windowed" approach by applying a "windowing" or "time binning" to the data collected from a best-effort measurement. Both over- and under-sampling approaches are possible, as long as the final goal of assessing the downloaded data amount within a certain time window is reached.

NOTE: Another way of combining the advantage of "fair calculation" by averaging transfer times and having "meaningful data rates" and "service failure rates" is to perform a best effort approach, average the transfer times achieved and recalculate the average throughput from the average transfer time in the sense of an adjusted mean. Please also refer to clause 6.1.

4.8 Upper bounds for TCP throughput measurements

In many cases the maximum achievable throughput can be estimated. For instance, in case of TCP the maximum throughput per socket is limited by the following equation:

$$Throughput \leq \frac{TCP\ Receive\ Window\ Size}{RTT}$$

This formula reflects the statement that there cannot be more unacknowledged traffic in the network for one socket than the maximum TCP receive window size. For further details, please refer to e.g. RFC 793 [i.11], RFC 1323 [i.12] and RFC 3481 [i.13].

A better estimation for the stationary TCP throughput per socket can be gained from delay, packet loss, and the knowledge of congestion avoidance procedures. In the following equation, MSS stands for maximum segment size and P_{loss} stands for the probability of packet loss. For further details, please refer to the paper [i.14].

$$Throughput < \frac{MSS}{RTT\sqrt{P_{loss}}}$$

5 Measurement environment aspects

5.1 Client/Server hardware

Any hardware being used to perform throughput measurements might have an impact on the overall results affecting the achievable throughput. This holds especially true for the used measurement system, including any client and server side hardware.

Depending on the used hard- and software components of the measurement system, throughput measurements can be impacted by the amounts of measurement data that are written to a storage medium. Extreme throughput can cause high CPU load as well as a high CPU load can have an impact on the maximum achievable throughput. Each hardware component (CPU, hard disk, motherboard, etc.) can influence the total performance of a measurement system.

Thus the measurement system used should provide the means to send, collect and store all relevant measurement data, including related logging data of the system itself, in a reliable way.

In case of passive measurements, storage medium and/or memory speed and/or size of the measurement equipment limits the number of data that can be stored during data acquisition.

In case of active measurements the data upload and/or download is usually more IO-limited than CPU-limited. E.g. when the throughput value gets close to the hard disk speed, the whole upload and/or download can be slowed down because of the speed limitations of the hard disk.

In such a scenario, a possible solution could be to avoid writing any received file to a hard disk on the client or the server side. Of course if the sender cannot send the data with enough speed, this has to be solved before any measurement takes place. In very high speed networks caching the measurement files in memory (RAM disks) can solve the problem.

NOTE: In order to overcome IO-limitations, another option is to decrease the amount of data that is written to the hard disk on the fly or by totally switching off certain logging information if the measurement system allows for such optimization. Especially in cases where Performance Enhancement Proxies (PEP, also called accelerators) are involved and used together with a performance enhancement client being part of the measurement system, TCP tracing can usually be switched off. The reason for this is that these software components often implement protocol optimization and encryption techniques making the TCP traces useless with respect to throughput evaluation based on the definitions given in TS 102 250-2 [i.2].

Another option in case that high throughput values are expected during measurement is to use several storage media in order to distribute hard disk activity. In such a scenario one hard disk could be used to store the received data whereas another hard disk could be used to store logging information of the local measurement system itself. Further information with respect to requirements for Quality of Service measurement equipment can be found in TS 102 250-4 [i.4] where the minimum requirements of QoS measurement equipment for digital wireless networks are defined. The assessment of the QoS parameters, defined in TS 102 250-2 [i.2], should be done following the procedures defined in TS 102 250-3 [i.3].

5.2 Operating System

Depending on the platform and operating system of the measurement system the TCP/IP stack might be implemented differently with different TCP settings available. Different parameterization has an impact on the TCP/IP stack behaviour and thus on the throughput. This applies for PC platforms as well as for mobile platforms, e.g. smart phones.

Network operators often provide operating system-specific drivers or custom-tailored connection software, a so-called dashboard. Besides offering the network operator's customer an easy way to establish and maintain an internet connection this software also often tunes TCP/IP stack implementation on the user's system with respect to the operator's network. If such dashboard software is used as part of a measurement system, the throughput might be affected due to the changes made to the TCP/IP stack as well as through additionally generated traffic. This can be traffic related to advertisements displayed within the Dashboard software or transfer of user-specific data between the client and the operator's server.

For further details and examples of changes commonly applied by such software refer to TR 102 607 [i.7].

5.3 Performance Enhancement Proxies

On application level it is possible to improve the performance perceived from a user's perspective with Performance Enhancement Proxies (PEP, also called accelerators or speed proxies).

There are many solutions to speed up the transmission, one being to decrease the signalling time by using permanent sockets. Another approach is to decrease the amount of bytes that need to be transferred in order to decrease the time needed to download the content. For calculation of the throughput, the real downloaded data bytes transferred has to be used instead of the expected number of bytes.

For further information and guidelines applicable regarding Performance Enhancement Proxies, please refer to TS 102 250-2 [i.2], clause 4.2.1.

5.4 Shared medium

When using a shared medium the active measurement shows the achievable throughput of the (N+1) user of the medium used, because the active measurement itself affects the achievable throughput of the others. The importance of this fact is even more increased when the access network gets into congestion because of the measurement being performed. Thus, the traffic generated by the measurement should be the lowest possible fraction of the whole traffic in the related geographical area / medium.

NOTE: The load of the cells should be considered for reporting. Furthermore, it should be taken into account that for a representative measurement representative areas have to be included into the measurement, e.g. mix of urban and suburban areas.

5.5 All traffic versus filtered traffic

5.5.1 User traffic

In the present framework the focus on QoS lies on the complete end-to-end view from the user's perspective. This is also discussed in depth in clause 5 "QoS Background" of TS 102 250-1 [i.1].

The above statement also holds true for throughput measurements being performed from an end-to-end perspective. Here the user usually accesses one specific IP service using a dedicated device (e.g. data card or USB data stick) and is focusing on the performance of that specific service. In parallel to the usage of this service, other IP traffic can be generated by the user equipment as well, which then affects performance for the specific service under test.

The main difference between QoS from a user's perspective and network performance is that QoS from a user's perspective provides quality information on an end-to-end and service-related basis whereas network performance reflects the technical operativeness of telecommunication systems, i.e. network and terminal elements or of network sections.

With respect to this, the question which should be answered before planning throughput measurements is whether the full transmission capacity of a channel/connection or only one service should be measured.

If the full transmission capacity is intended to be measured, every data packet should be taken into the throughput calculation. On the other hand, if only a single service is to be tested, it should be ensured that IP traffic generated by the service under test is not affected by any other traffic, e.g. by the user or the used measurement equipment.

5.5.2 Multithreaded applications

There are services which use more than one single data socket to transmit data in order to achieve a better network utilisation. By using multiple sockets a higher share of the available transmission capacity becomes usable, thus increasing the efficiency of the connection. These services usually implement several threads in order to transmit and/or receive data in parallel via dedicated data sockets. Thus, the number of data sockets used during the transmission can have an impact on the performance of the service and the corresponding throughput. Furthermore, in networks with high RTT or higher packet loss ratio, using more sockets can achieve better application layer throughput.

For instance, during web browsing the web browser application usually utilizes several sockets to download specific parts of the webpage in parallel. The maximum number of concurrent sockets depends on the web browser's implementation. Therefore, the maximum possible number of sockets and the number of sockets actually used during a measurement should be taken into account when comparing measurement results.

Other examples for services utilizing several data sockets are FTP, which is capable of transferring specific parts of a single file in parallel, or applications implementing RTP streaming clients.

With respect to throughput measurements, this means that throughput values measured for a certain service on application layer cannot be directly compared since the measurement results might highly depend on the configuration of the service being used. Figure 7 shows a FTP application using two data sockets.

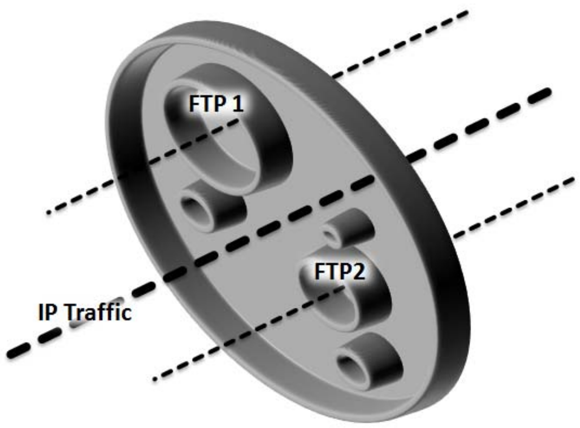
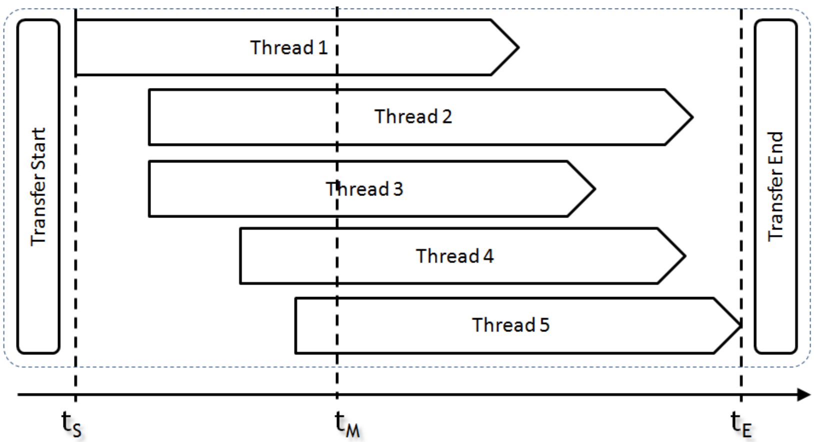


Figure 7: Multithreaded FTP traffic within IP traffic

As an example the FTP throughput measured for a FTP application configured to use three data sockets can provide different throughput values compared to a measurement performed using the same application configured to use only one socket. Some possible reasons for this difference are for instance limitations of the access network, e.g. incorrect shaping parameters, high RTTs or a high packet loss probability.



NOTE: Users are interested in throughput achieved over time t_E-t_S while operators may also be interested in the instantaneous max throughput achieved at time $t_{M.}$

Figure 8: Multithreaded Download Example

Different approaches can be considered with regard to multithreaded throughput shown in Figure 8. From an operator's/access provider's perspective the maximum throughput or the average throughput is important when all the sockets are up and generating traffic. From a user's perspective the average throughput is interesting while any of the sockets are up and generating traffic, which means the transfer is still ongoing.

NOTE: It should be reported how many sockets were used to transfer the content.

5.5.3 Aggregate traffic

From a network perspective all traffic generated in the network can be interpreted as basis for a throughput measure. The network operator can also be interested in the throughput of the aggregate traffic generated by one specific type of service only, e.g. HTTP web browsing, but by all users.

6 Campaign planning and evaluation

A range of statistical aspects has to be considered when planning and evaluating throughput measurements. The following clauses discuss some of the most relevant of these aspects as well as providing advice for campaign planning and evaluation.

6.1 Mean user data rate versus mean transfer time measurements

Data rate can be calculated either by measuring the time it takes to transfer a defined amount of data or by measuring the amount of data transferred in a defined period of time. In both cases start and end trigger points are the same. For a single measurement the data rate can be calculated from the transfer time and vice versa by knowing the amount of data transmitted. However, the mean value of a number of data rates cannot be calculated from the mean value of the respective transfer times.

If the data rate is calculated from the time it takes to transfer a defined amount of data then short transfer times have more impact on the mean value of the data rates than long transfer times, i.e. the mean value of the data rates is biased by the short transfer times ("quick" downloads).

$$MeanTransferTime = \frac{1}{N}\sum_{i=1}^{N} t_i$$

(N: number of samples; t: time to transfer data)

In the calculation of the mean value of the transfer time each time value counts equally. If the transferred amount of data is constant then the data rate has a 1/x relationship to the transfer time.

$$MeanDataRate = \frac{D}{N}\sum_{i=1}^{N} \frac{1}{t_i}$$

(D: amount of transferred data (constant); N: number of samples; t: time to transfer data)

In the calculation of the mean value of the data rate each time value does not count equally but in a 1/x relation. Here, short times have bigger influence than long times. This is illustrated in Figure 9.

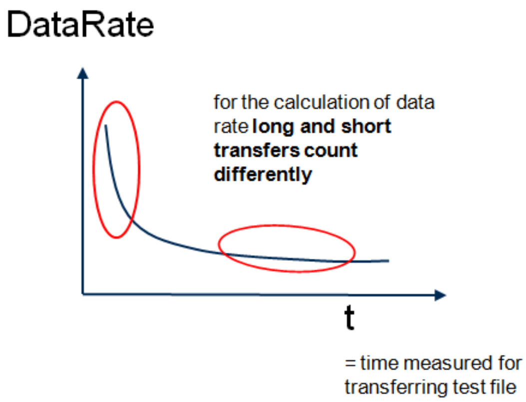


Figure 9: Relationship between the time measured for a transfer and the data rate

6.1.1 Example

Figure 10 shows the CDF of the mean user data rate of two networks used in this example.

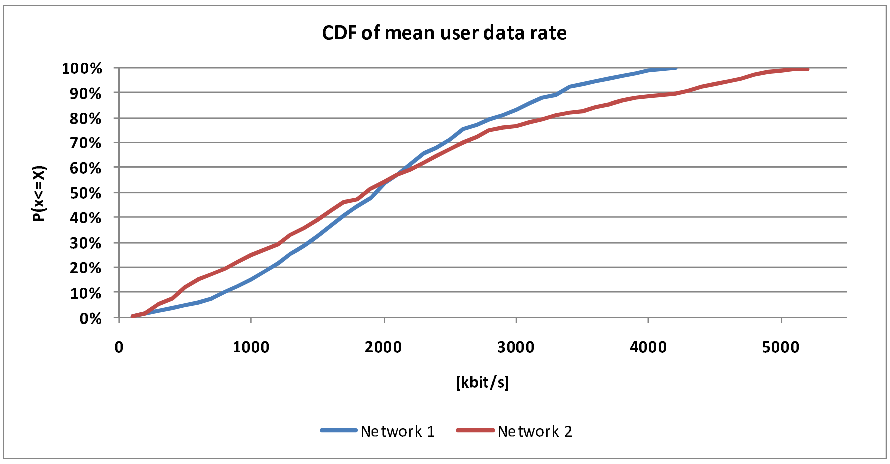


Figure 10: CDF of mean user data rate for two networks

Network 1 has a narrower distribution of data rate, i.e. a lower portion of slow and a lower portion of fast downloads than network 2. Network 2 shows a wider spread of data rates and a more equal distribution.

	Mean User Data Rate [kbit/s]	Mean Transfer Time [s]
Network 1	1984,5	13,4
Network 2	2058,9	16,8

Figure 11: Mean User Data Rate versus Mean Transfer Time

Calculating the mean values of data rates and transfer times as seen in Figure 11 shows that network 2 reaches higher mean user data rates than network 1 but network 1 has shorter mean transfer times than network 2 because in terms of data rate the short downloads overcompensate the influence of the long lasting downloads on the mean value of the data rates.

	Median User Data Rate [kbit/s]	Median Transfer Time [s]
Network 1	1937,5	8,7
Network 2	1884,0	8,9

Figure 12: Median User Data Rate vs. Median Transfer Time

The median value is not affected by this effect as seen in Figure 12.

The effect is becoming more severe with increasing offered data rates, as shown in Figure 13 and Figure 14. At high data rates a variance of a few milliseconds leads to a very large variation of the mean user data rate while at low data rates the effect of the same variance would be hardly noticeable for the data rate. At the same time it is fair to state that variances of a few milliseconds in transfer time will not affect the user experience for a bulk download at all.

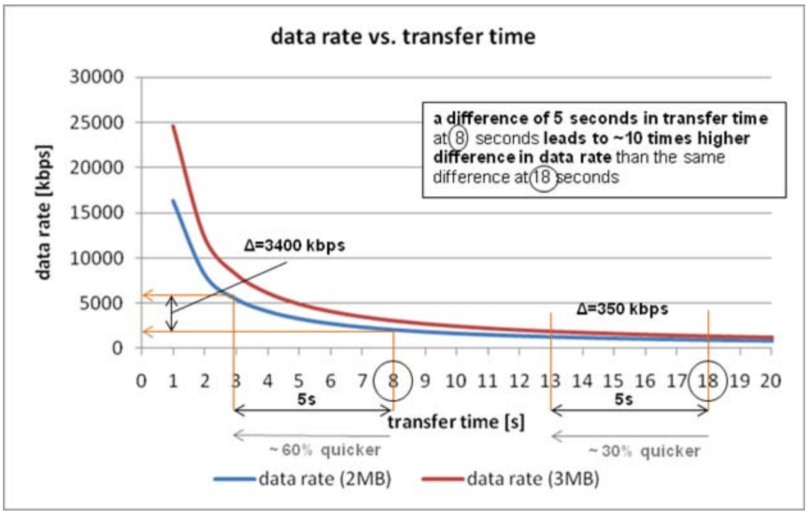


Figure 13: A variance of a few milliseconds can lead to very large variations of the mean user data rate

The faster the networks under test are the more obvious is the difference. Figure 14 shows the limits of UMTS and HSPA at 80 % utilized capacity of the possible physical maximum as dotted lines. The curves show the different values for different file sizes.

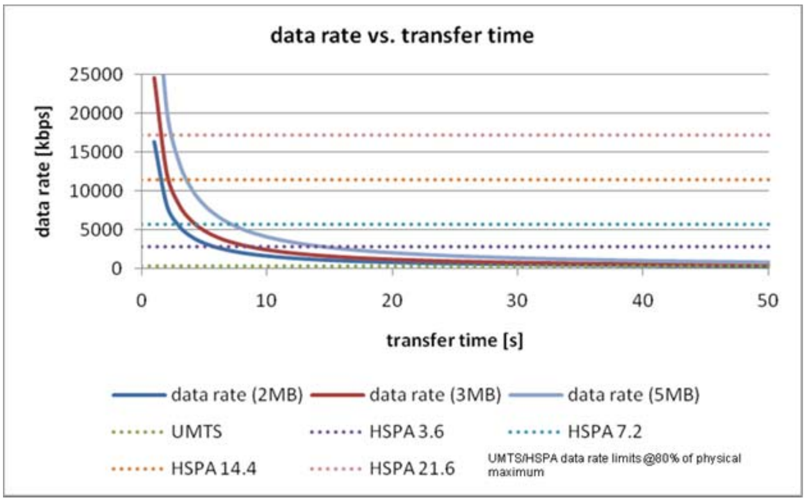


Figure 14: Data rate versus transfer time in relation to different physical data rate limits

6.1.2 Conclusion

If the average data rate is calculated from the time it takes to transfer a defined fixed amount of data, then short download times overcompensate the influence of long download times on the mean value of data rates.

The mean value of data rates is a biased value and cannot be calculated from the mean value of the corresponding transfer times.

Therefore, and because users have a sense of time, the user experience might be better reflected by transfer times related to an amount of data transferred rather than data rates.

6.2 Throughput calculation based on time or traffic

Throughput measurements can be based on a predefined time period or a predefined amount of traffic.

6.2.1 Non-sampled averaging

During throughput measurements only one sample is taken. For instance, FTP throughput can be measured by downloading a file that is big enough or it can be measured by downloading a file for a fixed duration, assuming that the file is big enough to last for the whole download time. The start time or the start event of the measurement can differ from zero.

6.2.2 Sampled averaging

During throughput measurements, several samples can be measured and the sampling can be time- or traffic-based. The traffic-based sample generates a sample after reaching some traffic or packet count, the time-based sampling generated a sample after reaching some time period. The start time or the start event of the measurement can differ from zero.

In case of sampled averaging, the samples cannot be handled as independent, identical distribution samples, since a sample created during an existing download is not independent from the previous sample.

6.3 Busy hours or peak-off

The throughput varies over time because the users of the network generate different amount of traffic during the day. When creating a measurement campaign, the traffic profile of the network has to be known or the samples of the measurement campaign have to be equally distributed over the day.

6.4 Working days or weekends

The throughput varies over time on a long scale, also the users of the network generate different amount of traffic during the week. When creating a measurement campaign, the traffic profile of the network has to be known or the samples of the measurement campaign have to be equally distributed over the weekday.

6.5 Locations

When creating a measurement campaign, the profile of the area covered has to be considered and later reported together with the measurement result.

6.5.1 Mobility aspects

The type of measurement with respect to mobility (e.g. stationary or drive test) has to be considered and later reported together with the measurement results.

6.5.2 Area categories

Area categories have to be defined based on geographical areas or based on population density, e.g. big cities, major cities, small towns, rural areas, major roads, minor roads or villages.

As an alternative just stochastically select latitude and longitude values and measure there. In this case the locations should be limited to the service area.

6.6 Calculating sample mean

When creating a measurement campaign, a sufficient number of samples for calculating the sample mean, a confidence interval and significance level have to be considered and later reported together with the measurement result. For further details, please refer to TS 102 250-6 [i.5].

7 Throughput measurement checklists

In the following, all above mentioned aspects are considered by providing checklists for starting a measurement campaign and evaluating the results, respectively.

Table 1 provides a checklist, which is suitable for performing campaigns in a single network. Each of the listed steps should be considered.

Table 1: Checklist for performing campaigns in a single network

Step	Entity for throughput measurement		
	Network	Service/Application	Device
Select the network part under test See note 1	Access, IP, Core - according to the purpose of the measurement		
Select the purpose of the measurement	Access network / IP Network (Subscriber-reachable / Bottleneck search) / E2E	Endpoint defined by service	Access network / IP Network (Subscriber-reachable / Bottleneck search) / E2E
Select the PCO See note 2	Depends on the purpose of the measurement	Select the PCO closest to the service level	Select the PCO closest to device level
Select the type of the measurement See note 3	Active or passive measurement		
Select the protocol (PCO, network part) See note 4	Choose the protocol	Select the protocol defined by the service	Choose the protocol
Plan the campaign from a statistical point of view	Identify depth of information needed – average, maximum, minimum, confidence interval, threshold measurement		
Plan the campaign from a sampling point of view See note 5	Access network - many locations / distributed in time / best effort or windowed approach IP Network - some locations / distributed in time / best effort or windowed approach E2E - many locations / distributed in time / best effort or windowed approach	Many or only one location / distributed in time / best effort or windowed approach	Access network - many locations / distributed in time / best effort or windowed approach IP Network - some locations / distributed in time / best effort or windowed approach E2E - many locations / distributed in time / best effort or windowed approach
Plan the locations See note 6	Define the locations, areas, stationary or drive test		
Plan the measurement hours See note 7	Define measurement hours depending on the information needed, e.g. - best available throughput - peakoff, worst available in peak hours		
Plan the campaign from a measurement equipment point of view See note 8	Check CPU / HDD / Memory / Network interface speeds / Operating system - choose the best available to get the best results or choose the average to get the mean values depends on the purpose of the measurement	Check CPU / HDD / Memory / Network interface speeds / Operating system - choose the best suits for the application to get the best results or choose the average to get the mean values depends on the purpose of the measurement	Check CPU / HDD / Memory / Network interface speeds / Operating system - choose the best suits for the device to get the best results or choose the average to get the mean values depends on the purpose of the measurement
Plan the measurement configuration See note 9	Measurement types, measurement parameters depends on the purpose of the test - e.g. file sizes, timeouts, test pages	Measurement type defined by the service, measurement parameters closest to the service parameters	Measurement types, measurement parameters depends on the purpose of the test - e.g. file sizes, timeouts, test pages
Set the protocol stack	In case of TCP measurement: Check TCP parameters in TS 102 250-7 [i.6] In case of UDP measurement: check send speed on server side	In case of TCP measurement: Check TCP parameters in TS 102 250-7 [i.6] only at client side In case of UDP measurement: server side configuration should be the same as for users	

Step	Entity for throughput measurement		
	Network	Service/Application	Device
Set the end point of the measurement See note 10	Access - closest to the access IP network - place many locations along the service line and test each of them E2E - at the end of the service chain	Location of the server defined by the service	Access - closest to the access IP network - place many locations along the service line and test each of them E2E - at the end of the service chain In case device testing E2E is commonly used
Plan how the measurement device accesses the network	Access network test - radio conditions (attenuation / diversity / mobility) depends on the purpose of the measurement IP network test - best achievable radio conditions E2E test - radio conditions (attenuation / diversity / mobility) depends on the purpose of the measurement		
Plan the device for testing	Access network test - used device depends on the purpose of the measurement IP Network test - the best achievable device E2E Test - used device depends on the purpose of the measurement		Device is defined by the measurement
Plan the subscription	SIM / service configuration - max bit rate / precedence / fair use policy Access network measurement - used subscription depends on the purpose of the measurement IP Network measurement - the best achievable subscription E2E measurement - used subscription depends on the purpose of the measurement	Subscription recommended for the service	SIM / service configuration - max bit rate / precedence / fair use policy Access network measurement - used subscription depends on the purpose of the measurement IP Network measurement - the best achievable subscription E2E Test - used subscription depends on the purpose of the measurement
Plan how to get the SIMs	Anonymity of the SIM is not mandatory		
Keep in mind the affected transparent L4/L7 devices See note 11	Used L4/L7 network elements depends on the endpoint of the measurement	Used L4/L7 network elements depends on measured service	Used L4/L7 network elements depends on the endpoint of the measurement
Plan the measurement configuration	Plan measurement configuration measurement types, measurement parameters depends on the purpose of the measurement	Plan measurement configuration measurement type defined by the service, measurement parameters closest to the service parameters	Plan measurement configuration measurement types, measurement parameters depends on the purpose of the measurement
NOTE 1: Please refer to clause 4.3 for further details. NOTE 2: The concept of PCOs is explained in clause 7.1 of TS 102 250-1 [i.1] and in clause 5.1 of TR 102 807 [i.9]. NOTE 3: Please refer to clause 4.5 for further details. NOTE 4: Please refer to clause 4.8 for further details. NOTE 5: Please refer to clause 4.7 for further details. NOTE 6: Please refer to clause 6.5 for further details. NOTE 7: Please refer to clauses 6.3 and 6.4 for further details. NOTE 8: Please refer to clauses 5.1 and 5.2 for further details. NOTE 9: Please refer to clauses 5.5.2 and 4.6 for further details. NOTE 10: Please refer to clause 4.4 for further details. NOTE 11: Please refer to clause 5.3 for further details.			

Table 2 provides a checklist, which is suitable for performing benchmarking campaigns. Each of the listed steps should be considered.

Table 2: Checklist suitable for performing benchmarking campaigns

Step	Entity for throughput measurement		
	Network	Service/Application	Device
Select the network part under test See note 1	Only E2E measurement is possible		
Select the purpose of the measurement	Only E2E measurement is possible		
Select the PCO See note 2	Same PCO for all networks		
Select the type of the measurement See note 3	Active measurement		
Select the protocol (PCO, network part) See note 4	Same protocol for all networks	Protocol defined by service	Same protocol for all networks
Plan the campaign from a statistical point of view	Identify depth of information needed – average, maximum, minimum, confidence interval, threshold measurement Measure the same for all networks		
Plan the campaign from a sampling point of view See note 5	Only E2E measurement is possible - many locations, distributed in time, best effort or windowed approach, but the same for all networks under test		
Plan the locations See note 6	Define the locations, areas, stationary or drive test, select the same locations and measurement type for all networks		
Plan the measurement hours See note 7	Define measurement hours depending on the information needed, e.g. - best available throughput - peakoff, worst available in peak hours, measure in the same hours for all networks		
Plan the campaign from a measurement equipment point of view See note 8	Measurement environment - CPU / HDD / Memory / Network interface speeds / Operating system same for all networks, should not limit the QoS provided by the network		Measurement environment - CPU / HDD / Memory / Network interface speeds - do not tune, use their own device settings for each network
Plan the measurement configuration See note 9	Plan measurement configuration measurement types, measurement parameters should be the same for all networks	Plan measurement configuration measurement type defined by the service, measurement parameters closest to the service parameters	Plan measurement configuration measurement types, measurement parameters the same for all networks
Set the protocol stack	In case of TCP measurement: Check TCP parameters in TS 102 250-7 [i.6] - should be set same for all networks In case of UDP measurement: check send speed on server side	In case of TCP measurement: Check TCP parameters in TS 102 250-7 [i.6] set only at client side and same for all networks or to the recommended value	In case of TCP measurement: Check TCP parameters in TS 102 250-7 [i.6] only at server side In case of UDP measurement: check send speed on server side
Set the end point of the measurement See note 10	Only E2E is possible, no access to all operator IP network, same end server for all networks, place it for instance at internet exchange for no bias for any networks		
Plan how the measurement device accesses the network	E2E measurement - same attenuations / diversity / mobility for all networks or the best configuration for all networks		
Plan the device for testing	E2E Measurement - same device or the best device for each network	Access to network - device E2E Measurement - same device or the best device for each network	Device is defined by the measurement

Step	Entity for throughput measurement		
	Network	**Service/Application**	**Device**
Plan the subscription	SIM / service configuration - max bitrate / precedence / Fair use policy E2E Measurement - same / closest configuration for each network	SIM / service configuration - max bitrate / precedence Subscription recommended for the service at each network	SIM / service configuration - max bitrate / precedence E2E Measurement - same / closest configuration for each network
Plan how to get the SIMs	Anonymity of the SIM is not mandatory		
Keep in mind the affected transparent L4/L7 devices See note 11	Network configuration use only those L4/L7 network elements which are valid for all networks	Network configuration use of L4/L7 network elements as defined by the service	Network configuration use only those L4/L7 network elements which are valid for all networks
Plan the measurement configuration	Plan measurement configuration measurement types, measurement parameters should be the same for all networks	Plan measurement configuration measurement type defined by the service, measurement parameters closest to the service parameters	Plan measurement configuration measurement types, measurement parameters the same for all networks
NOTE 1: Please refer to clause 4.3 for further details. NOTE 2: The concept of PCOs is explained in clause 7.1 of TS 102 250-1 [i.1] and in clause 5.1 of TR 102 807 [i.9]. NOTE 3: Please refer to clause 4.5 for further details. NOTE 4: Please refer to clause 4.8 for further details. NOTE 5: Please refer to clause 4.7 for further details. NOTE 6: Please refer to clause 6.5 for further details. NOTE 7: Please refer to clauses 6.3 and 6.4 for further details. NOTE 8: Please refer to clauses 5.1 and 5.2 for further details. NOTE 9: Please refer to clauses 5.5.2 and 4.6 for further details. NOTE 10: Please refer to clause 4.4 for further details. NOTE 11: Please refer to clause 5.3 for details.			

Annex A: Analysing IP traces under different aspects

In the following it is assumed that a proper IP trace is at hand from the corresponding PCO.

A.1 Layer-based analysis

During trace analysis the selected stack layer packet length is used to calculate the amount of generated traffic. To calculate all IP throughput on a specified IP interface, each IP packet length seen by the capture interface has to be aggregated and divided by the duration (e.g. from first packet to last packet).

To select only the traffic that belongs to one specific service, the trace analyser has to identify the socket the application used for transferring packets:

- For FTP, the trace analyser has to detect the control socket and then, based on the information gathered on control socket, the data socket has to be identified. The length of all packets received on the data socket has to be aggregated and divided by the duration.
- For HTTP, the used sockets can be filtered by content type, content length or based on any application layer parameter.
- In case of RTP streaming the used sockets can be filtered based on SDP.

For every IP service the traffic analyser can find the socket which was used to transmit packets for the service.

The trigger points defined in TS 102 250-2 [i.2] for every IP service can be used for throughput calculation, since the triggers are defined on packet level and can be easily found in the IP trace. The trigger points are defined on application layer, but they can easily be extended to lower layer protocol stacks, since the upper layer protocol elements are transmitted with lower layer protocol elements. E.g. a TCP SYN or a FTP GET command is transmitted in an IP packet as well.

A.2 User-based analysis

For further study.

A.3 Volume-based analysis

Traffic-based analysis means that for throughput measurements the amount of traffic or the number of packets is fixed instead of the time.

$$Throughput = \frac{Amount\ Of\ Traffic}{Fixed\ Duration}$$

A traffic measurement can start at any time after the transmission has started, i.e. the start time does not necessarily have to be zero.

In TS 102 250-2 [i.2] all of the throughput and data rate definitions are volume-based QoS parameters.

A.4 Time-based analysis

Time-based analysis means that for throughput measurement the amount of time and not the traffic is taken into account.

$$Throughput = \frac{Fixed\ Amount\ Of\ Traffic}{Duration}$$

A traffic measurement can start at any time after the transmission has started, i.e. the start time does not necessarily have to be zero.

In TR 102 678 [i.8] this kind of calculation method is explained in-depth.

A.5 Examples

For further study.

History

Document history		
V1.1.1	February 2012	Membership Approval Procedure MV 20120424: 2012-02-24 to 2012-04-24