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Within the scope of its powers and responsibilities the Czech Telecommunication Office (hereinafter referred to as "Office") measures and evaluates selected parameters of data networks. The measurement and evaluation of selected parameters of electronic communication networks is unified in fixed and mobile networks in a general methodology entitled:

Measurement of Data Parameters of Networks Using the TCP Protocol, Version 2.0, which is published and applied by the Office in the case of inspection measurements in fixed and mobile networks.

The measurements are performed using the Office's own measuring devices (terminals) with clearly defined parameters, in fixed as well as in mobile networks. The measurement method applied is based on the recommendation of RFC 6349, "Framework for TCP Throughput Testing".

I. Introduction

The purpose of this document (hereinafter referred to as "Methodology") is to describe and unify the procedure for the measurement of representative data parameters of fixed, mobile, wireless and other electronic communication networks using the TCP protocol. The Methodology is intentionally kept on general level to allow generalization of the measurement of the data parameters and to free the measurement of the physical layer of the network protocol and thus also the technology. The physical layer of the network traffic (including different interfaces, connection points, terminals, etc.) will be described for each technology and covered in a separate annex if necessary. The Methodology, which focuses on measurement in the transport layer of the ISO/OSI model, will also clearly show that the data parameters which, due to their nature and importance, can substantially influence the quality and effectiveness of data transfer include available information rate for download and upload, frame delay, inter-frame delay variation, and mainly frame loss ratio.

A necessary condition for the measurement of TCP throughput is the availability of network sources (IP addresses, ports, services) and related transparency of the network paths (in accordance with net neutrality).

The document fully respects or acknowledges the international recommendations of IETF RFC 6349, RFC 2697, RFC 1191, RFC 1981, RFC 2544, RFC 2681, RFC 2923, RFC 4443, RFC 4656, RFC 4821, RFC 4898, RFC 5136, RFC 5357, RFC 7323, and also standards ITU-T Y.1563 and ITU-T Y.1564.

II. Definition of the measuring sides and the network under test

1. Measuring server

Measuring server (MS) shall mean the measuring side which provides the opposite side with services (data) upon request in the case of data download. In general, measuring server is a device connected to the Internet at a defined point. The measuring server should have sufficient performance and independence of the data connection so that sufficient throughput and guarantee of the data parameters is ensured, even in the case of multiple connections of the measuring devices at a time. The measuring server is a part of the Measuring System of Electronic Communications (hereinafter referred to as “MSEK”) managed by the Office.

2. Measuring device (terminal)

Measuring device, terminal (MT) shall mean the measuring side which functions as the recipient of the service (data) in the case of data download. Measuring device shall mean a terminal with the respective service software which is capable of performing measurements according to the applicable methodological procedures of the Office and whose computing and network performance is high enough that it does not affect negatively the measurement results. The measuring device must be capable of monitoring and saving basic and extended set of data parameters of fixed electronic communication networks, exporting them in a standardized format suitable for machine or other suitable processing, and subsequently allowing the transfer of the resulting values to the central storage of the MSEK or storing them in internal memory during the measuring process.

3. Network under test

Network under test (NUT) shall mean such sequence of transmission nodes that there is a connection between every two consecutive transmission nodes and, at the same time, the first transmission node is MT and the last transmission node is MS. The electronic communication network measured is a network that is part of the network under test to which the measuring device (terminal) was connected during the measurement.

III. Measurement process

The following procedure describes a sequence of steps that are necessary for obtaining correct measurement data. Preceding section 5 which only covers the measurement of TCP throughput itself, sections 1 through 3 describe the necessary conditions which must be fulfilled before the measurement according to section 5. If this procedure is not followed, the result of the measurement might and will very likely be distorted by the incorrect set-up of the measuring sides (mainly in terms of their download and upload capacities, respectively).

1. Introductory arrangements and risks

The TCP protocol cannot be used for reliable measurement of non-functional electronic communication networks (i.e., networks which are exposed to high packet loss or large packet delay variation). According to RFC 6349, it is possible to use the threshold of 5% packet loss and packet delay variation of 150 ms as a reference value. These or even higher values suggest a failure or emergency status of the network (e.g., overloading, insufficient network capacity), in particular in the environment of data networks on the territory of the Czech Republic. It is not possible to measure networks reliably where the parameters change relatively quickly in time (parameters according to sections 2 and 3).

It is necessary to ensure compliance with and respecting the following arrangements:

- Taking into account “traffic shaping” – in this case, a delay of traffic of certain services or reduction of the overall throughput can occur.
- Taking into account “traffic policing” - in this case, monitoring of the traffic and subsequent reduction or exclusion of traffic can occur when the agreed limit is exceeded; described in RFC 2697.
- Availability of services on one port may not mean availability of services on other ports. It is therefore advisable to add a comparative test of measurement of the ports – availability of known ports to the test of TCP throughput according to section 5.
- Independence of the measurement must be ensured at each point of the measurement (test) – it means that during each measurement no other data stream, which is not part of the measurement may be implemented, or the available data flow must be sufficient enough to prevent significant influence on the measurement results.

2. Identification of MTU

Identification of MTU of the network under test is essential for correct set-up of the measuring system to prevent fragmentation and so that it is possible to measure the capacity of the network under test as accurately as possible; the following must apply:

$$\mathrm{MTU\ (TCP\ TTD) = MTU\ (NUT);\ [B;\ B]}. \tag{1}$$

Several methods can be used for identification of MTU of the network under test; the methods differ from each other mainly by the network area in which they ca be deployed. The following methods can be used for correct identification of MTU of the network under test:

- Identification according to RFC 1191,
- Identification according to RFC 1981,
- Identification according to RFC 4821.

The following sections 2.1 through 2.4 briefly describe the individual methods of identification of MTU of the network under test; for details see the relevant recommendations of IETF RFC.

2.1. Identification according to RFC 1191

The RFC 1191 recommendation offers the easiest and quickest method for determining MTU for IPv4. It is about using the characteristics of IPv4 packets with fixed selection of the size of MTU and with a set attribute $\mathrm{DF} = 1$ (do not fragment). If the set value of MTU is too large for the respective network under test or, more precisely, for a certain network element on the route, the network element will discard the IP datagram and will reply back to the sender with an ICMP message that the datagram cannot pass through and that the fragmentation option is blocked using the DF attribute. This method can be used only in situations where the administrator of the network under test is not blocking the use of ICMP messages within the network.

2.2. Identification according to RFC 1981

The RFC 1981 recommendation offers a similar principle of identification of MTU for IPv6 as the RFC 1191 recommendation. Due to the essence of the IPv6 protocol, however, it is not possible to use the settings of the bit of the attribute $\mathrm{DF} = 1$. When this option is missing the method uses the principle of sending an ICMPv6 message (with content “packet too big” according to RFC 4443) through the network element which is not able to transfer the packet of the given size. From this message it is possible to clearly identify the maximum size of MTU of the respective network element. This method, however, can again be used only in cases where the network administrator is not blocking the use of ICMPv6 messages within the network.

2.3. Identification according to RFC 4821

This procedure deals with situations where for some reason (section 2.4) it is not possible to use the previous two procedures for identification of MTU. These are in particular the cases where for some reason the option of sending ICMPv4 or ICMPv6 messages is blocked. Windows and Linux operating systems make it possible to use the implementation of the standardized technique of PMTUD (Path MTU Discovery) using the "black hole detection" (BHD) option.

2.4. Problems with the determination of the size of MTU of the network under test

Problems with the determination of the size of MTU of the network under test are covered by the RFC 2923 recommendation.

3. Measurement of delay (Delay)

It is possible to imagine Delay as a time elapsed between the sending of the first bit of the TCP segment and the receipt of the last bit corresponding to the confirmation of the TCP segment. Measurement of delay, as well as the identification of MTU, can be performed in multiple ways which differ by accuracy and robustness. It is recommended that the initial measurement is performed in the process of the test interval. Within the test interval, it is recommended to set the value of parameter bDelay which corresponds to the lowest measured value of delay not burdened with the established TCP session and also the value of parameter $\mathrm{minDelay}$ which corresponds to the lowest measured value of Delay during the established TCP session. Parameter bDelay is applied when determining the TCP metrics of BD, parameter $\mathrm{minDelay}$ is necessary for the subsequent calculation of the further defined parameters such as BDP, $\mathrm{TCP\ RWNDmin}$ and also the size of the so-called socket buffers. The resulting values will be subsequently used to ensure sufficient capacity of the data receiver as well as data transceiver prior to the measurement.

3.1. ICMP ping

The use of ICMP ping can be considered adequate method of estimating the value of delay, provided that the datagram size is taken into account. Nevertheless, with respect to the nature of ICMP ping it is not possible to describe this method as sufficiently accurate (problems on the side of network elements, prioritization of QoS), and therefore it is not recommended.

3.2. Use of extended MIB statistics

Use of statistics available in MIB for the measurement of the value of delay according to the recommendation of RFC 4898.

3.3. Use of suitable tools

It is appropriate to use iperf, FTP, or other tools based on the principle of capturing packets from the test TCP sessions to measure delay. It is important to realize that results based on SYN → SYN-ACK messages at the beginning of a TCP session should not be used to measure the value of Delay.

3.4. Use of the TWAMP protocol

The most robust and suitable method for measuring delay is to proceed according to RFC 5357 where the use of TWAMP protocol for the measurement is recommended.

4. Measurement of BB

Prior to the measurement of TCP throughput, it is necessary to measure the lowest value of the capacity of the measured network under test, BB, or to derive its value based on contract terms during local investigation. From the point of view of the ISO/OSI model, the value of BB corresponds to the physical layer (L 1).

If there is doubt about the value of BB or if the value of BB is unknown it is necessary to use one of the measurement methods by means of a stateless protocol (e.g., UDP) to determine BB. It is advisable to perform the measurement in both directions, in particular if it concerns asymmetric technology of the electronic communication network. It is recommended to perform the measurement repeatedly, i.e., in different time intervals and at off-peak times to achieve relevant values and that the values are to the strict minimum affected by local or time-variable fluctuations in the availability of network sources. It is also necessary to keep in mind that BB is affected not only by the capacity of the network under test of the given data connection or the services purchased from the provider but also, for example, by unsuitable equipment of the end-user (slow end router, receiving terminal, etc.), or by the use of an unsuitable access method (e.g., wireless network with high interference, setting a slow transfer mode, insufficient bandwidth, or even unsuitable encryption). To measure BB, it is possible to use several methods according to the IETF recommendation:

- Measurement of BB according to RFC 2544,
- Measurement of BB according to RFC 5136.

4.1. Measurement of BB according to RFC 2544

This measurement method is suitable for a qualified estimate of BB, but it is necessary to keep in mind that this method of measurement of BB has been designed for testing network elements in laboratory conditions.

4.2. Measurement of BB according to RFC 5136

This is measurement according to RFC 5136 which focuses on measurement in real conditions, and therefore measurement according to this standard should become the standard method for estimating BB. Unfortunately, this recommendation does not contain any specific procedures how BB should be measured; it only defines general mathematical calculations, and therefore its applicability today is minimal.

5. Mathematical apparatus

Prior to the measurement of TCP throughput, it is necessary to perform required calculations and set important parameters which include BDP, buffer size BS, and the size of $\mathrm{TCP\ RWND}$. For these calculations it is necessary to use acquired value of $\mathrm{minDelay}$ or, more precisely, the initial value of the delay measured according to the methods mentioned in section 3 as well as the set parameter BB according to section 4.

5.1. Calculation of BDP

BDP shall be calculated by multiplying the acquired values of $\mathrm{minDelay}$ and BB:

$$\mathrm{BDP} = \mathrm{minDelay} \cdot \mathrm{BB};\ [\mathrm{b};\ \mathrm{s},\ \mathrm{b/s}]. \tag{2}$$

5.2. Calculation of the buffer size BS

The buffer size (BS) needs to be set according to:

$$\mathrm{BS} \geq \mathrm{BDP};\ [\mathrm{b};\ \mathrm{b}]. \tag{3}$$

5.3. Setting the size of $\mathrm{TCP\ RWND}$

The setting of the size of $\mathrm{TCP\ RWND}$ window on the data receiver side is based on the value of $\mathrm{TCP\ RWNDmin}$ parameter which can be determined using the following relation:

$$\mathrm{TCP\ RWNDmin} = \frac{\mathrm{BDP}}{8};\ [\mathrm{B};\ \mathrm{b}]. \tag{4}$$

General setting of BS and $\mathrm{TCP\ RWND}$ to high value can lead to overloading of the buffer of the network element when the values of BB, which will generate in the first phase a large number of segments toward $\mathrm{TCP\ TTD}$ which the network equipment cannot send over BB, and therefore packets will be thrown away unnecessarily due to the network element buffer size.

5.4. Single or multiple TCP sessions

The decision whether a single or multiple TCP sessions should be used in the measurement depends on the size of BDP or, more precisely, on the value of $\mathrm{TCP\ RWNDmin}$, in relation to the set value of the $\mathrm{TCP\ RWND}$ window on the data receiver side (e.g., $64\ \mathrm{kB}$). The purpose of the use of multiple TCP sessions is maximally credible coverage of the entire capacity of the network under test. If it is true that:

$$\mathrm{TCP\ RWNDmin} > \mathrm{TCP\ RWND};\ [\mathrm{B};\mathrm{B}], \tag{5}$$

the number of TCP sessions should correspond to the result of the equation (rounded up to the nearest whole number):

$$\mathrm{n} = \left\lceil \frac{\mathrm{TCP\ RWNDmin}}{\mathrm{TCP\ RWND}} \right\rceil;\ [\text{-};\mathrm{B},\mathrm{B}], \tag{6}$$

where n is the number of TCP sessions and $\mathrm{TCP\ RWND}$ represents the real set size of the window on the data receiver side. An example can be a situation where the subscriber can access electronic communication network with the capacity of the network under test $\mathrm{BB} = 500\ \mathrm{Mb}/s$ and $\mathrm{minDelay} = 5\ \mathrm{ms}$. The BDP parameter can be determined according to the equation (2), or specifically $312.5\ \mathrm{kB}$. Within each sequence of the tests, it is necessary to establish respective number of TCP sessions such as to achieve maximum utilization of the capacity of the network under test. If we set $\mathrm{TCP\ RWND} = 64\ \mathrm{kB}$, which corresponds to the basic maximum used, the number of TCP sessions should be $\mathrm{n} = 5$.

General recommendation:

- It is more suitable to perform measurements for multiple TCP sessions, even if the measurement with multiple TCP sessions according to the equation (5) is apparently not necessary. The reason is that a greater capacity of the network under test can be allocated with respect to the settings of the parameters of electronic communication networks. It is therefore recommended to use $\mathrm{n} \geq 2$.
- $\mathrm{TCP\ RWND}$ with size greater than $64\ \mathrm{kB}$ does not have to be always available because it can only be set in the case of using TCP extension (so-called "TCP window scale option"). Moreover, a situation can occur in actual implementations where the set size of the window can be ignored by the program or re-configured to the default value (e.g., $64\ \mathrm{kB}$).
- In the case of use of any application measuring device it is necessary to have access to the configuration and reports of both measuring sides. The default setting values may not be sufficient and may lead to erroneous results.
- It is necessary to identify whether the measuring tool uses the fixed $\mathrm{TCP\ RWND}$, or whether it determines the value of $\mathrm{TCP\ RWND}$ itself based on the NUT status before starting the measurement and maintains it constant during the measurement or whether it continuously changes this value during the measurement. This fact has significant impact on the measurement.

5.5. Calculation of the value of TCP throughput

The RFC 6349 recommendation defines two different methods of calculation of the parameters determining the value of TCP throughput. The first calculation method is theoretical, based on the structure of the individual layers of the ISO/OSI model, and determines ideal value of TCP throughput $\mathrm{TCP\ iTR}$. The second method is practical and is based on the current status of the NUT. The result of this method is the current value of TCP throughput $\mathrm{TCP\ aTR}$.

An example can be a technology corresponding to standard 100BASE-TX where speed of $100\ \mathrm{Mb}/s$ (NBR; "net bit rate") is achieved on the first layer of the ISO/OSI model. The maximum achievable information rate IR of the link layer of the ISO/OSI model is limited by the maximum number of frames FPS ("frames per second") according to the equation (Ethernet II frame):

$$\mathrm{FPS} = \frac{\mathrm{NBR}}{(\mathrm{IFG} + \mathrm{Preamble} + \mathrm{MAC\ DST} + \mathrm{MAC\ SRC} + 802.1\mathrm{Q}\ (802.1\mathrm{ad}) + \mathrm{Ethertyp} + \mathrm{Payload} + \mathrm{FCS}) \cdot 8}; [1/\mathrm{s}; \mathrm{b/s}, \mathrm{B}]. \quad (7)$$

In the above-mentioned case, if we assume that $\mathrm{IFG} = 12\ \mathrm{B}$, $\mathrm{Preamble} = 8\ \mathrm{B}$, $\mathrm{MAC\ DST} = 6\ \mathrm{B}$, $\mathrm{MAC\ SCR} = 6\ \mathrm{B}$, $802.1\mathrm{Q}\ (802.1\mathrm{ad}) = 0\ \mathrm{B}$, $\mathrm{Ethertyp} = 2\ \mathrm{B}$, $\mathrm{Payload} = = \mathrm{MTU} = 1500\ \mathrm{B}$, and $\mathrm{FCS} = 4\ \mathrm{B}$, the 100BASE-TX technology achieves the value of $\mathrm{FPS} = 8127\ 1/\mathrm{s}$ according to the relation (7). The value of parameter $\mathrm{TCP\ iTR}$ on the transport layer of the ISO/OSI model, when using the IPv4 protocol as a network layer protocol without optional parts of the header (20 B) and TCP header without any extensions (20 B), is determined according to the following equation:

$$\mathrm{TCP\ iTR} = (\mathrm{MTU}\ - \mathrm{IP}_{\mathrm{header}}\ - \mathrm{TCP}_{\mathrm{header}}) \cdot 8 \cdot \mathrm{FPS}; [\mathrm{b/s}; \mathrm{B}, 1/\mathrm{s}]. \quad (8)$$

In the above-mentioned case, the value $\mathrm{TCP\ iTR} = 94.92\ \mathrm{Mb/s}$. If an extended TCP/IP header (20 to 60 B) is used in the process of measurement of TCP throughput it is necessary to consider this extended header in the formula (8). The method of determination of the current value of TCP throughput $\mathrm{TCP\ aTR}$ is based on continuous measurement of Delay and the subsequent determination of the average value of this delay, $\mathrm{Delay(avg)}$, during the test in question. The average value of delay, $\mathrm{Delay(avg)}$, can therefore be defined as:

$$\mathrm{Delay(avg)} = \frac{1}{\mathrm{t}} \sum_{\mathrm{i}=0}^{\mathrm{N}-1} \mathrm{Delay}_{\mathrm{i}}; [\mathrm{s}; \mathrm{s}, \mathrm{s}], \quad (9)$$

where $\mathrm{Delay}_{\mathrm{i}}$ means individual values of Delay which are measured continuously with $1\ \mathrm{s}$ period and saved during the test, and parameter t means total duration of the test. The resulting current value of TCP throughput $\mathrm{TCP\ aTR}$ of the transport layer of the ISO/OSI model can be written as follows:

$$\mathrm{TCP\ aTR} = \frac{\mathrm{TCP\ RWND} \cdot 8}{\mathrm{Delay(avg)}}; [\mathrm{b/s}; \mathrm{B}, \mathrm{s}]. \quad (10)$$

5.6. Calculation of TCP metrics

The RFC 6349 recommendation defines three basic TCP metrics which may be used for better understanding and comparison of the individual results of the measurement. In addition, these metrics make it possible to compare the TCP data stream in different network conditions and settings of the measuring sides, and for these reasons they should be determined during each test. The necessary condition is that all three basic TCP metrics must be determined separately for each direction.

5.6.1. TCP transfer time ratio

TCP transfer time ratio ($\mathrm{TCP\ TTR}$) is a ratio between the actual value of $\mathrm{TCP\ aTT}$ (actual transfer rate value) and the ideal form thereof ($\mathrm{TCP\ iTT}$). This TCP metric which defines how many times the actual TCP transfer time is greater than its ideal value can be determined according to the following equation:

$$\mathrm{TCP\ TTR} = \frac{\mathrm{TCP\ aTT}}{\mathrm{TCP\ iTT}}; [\text{-}; \mathrm{s}, \mathrm{s}], \quad (11)$$

where $\mathrm{TCP\ aTT}$ is the real time of transfer of a data set time through the TCP session, whereas the ideal value $\mathrm{TCP\ iTT}$ is the predicted time in which the data set should be transferred through the TCP session. The ideal time $\mathrm{TCP\ iTT}$ is derived from the ideally achievable TCP throughput ($\mathrm{TCP\ iTR}$) on the transport layer of the ISO/OSI model. Ideal time of data set transfer time $\mathrm{TCP\ iTT}$ can be determined according to the following equation:

$$\mathrm{TCP\ iTT} = \frac{\mathrm{SD}}{\mathrm{TCP\ iTR}}; [\mathrm{s}; \mathrm{b}, \mathrm{b/s}], \quad (12)$$

where SD means the size of the data set intended for transfer.

5.6.2. TCP efficiency

TCP efficiency ($\mathrm{TCP\ EFF}$) represents the percentage of successfully transferred bits without the need to resend them. This metric gives an idea of the error rate of the entire TCP session and the necessity to resend. The calculation of the effectiveness of TCP transfer can be performed using the following equation:

$$\mathrm{TCP\ EFF} = \frac{\mathrm{TB} - \mathrm{rTB}}{\mathrm{TB}} \cdot 100;\ [\%; \mathrm{b}, \mathrm{b}], \tag{13}$$

where TB indicates the number of bits transferred and rTB indicates the number of bits which had to be resent after detected error.

5.6.3. Buffer delay

Buffer delay (BD) represents the relationship between the increase of the average value of delay, $\mathrm{Delay(avg)}$, during the test in question and the initial value of delay, bDelay, determined prior to the start of the test. The resulting value of BD can be defined as follows:

$$\mathrm{BD} = \frac{\mathrm{Delay(avg)} - \mathrm{bDelay}}{\mathrm{bDelay}} \cdot 100;\ [\%; \mathrm{s}, \mathrm{s}]. \tag{14}$$

6. Measurement of TCP throughput

This section defines the techniques of measurement of TCP throughput in such a manner as to make it possible to verify the maximum achievable value thereof. If the TCP protocol is not using dynamic regulation techniques for optimal utilization of the network under test (automatic setting of $\mathrm{TCP\ RWND}$), it is necessary to know the parameters $\mathrm{minDelay}$ and BB for the given network under test and thus to have the required calculations provided in section 5 finished, and to have fulfilled the necessary condition specified in section 2.

As the measurement of TCP throughput using this methodology is conditional upon correct function of the lower network layers, it is necessary to check prior to the commencement of the measurement the functionality, capacity of the network under test, and other parameters on the second and in particular third layer of the reference ISO/OSI model. The recommended steps before the start of the measurement of TCP throughput are as follows:

- The basic check, e.g., using the available testing tools which can suggest the expected values. In order to determine the parameters of the measurement it is recommended to check with a packet capturing program, e.g., Wireshark, what is actually going on at the network interface (what is the actual $\mathrm{TCP\ RWND}$, whether there are repeated packet transfers, and whether or not $\mathrm{TCP\ RWND}$ is exhausted during the transfer, etc.).
- Check whether or not the traffic is prioritized based on IP address of the standard (generally known) measuring servers. It is therefore advisable to perform the initial measurement of TCP throughput against reference measuring servers.
- Good approach is also checking the fulfillment of net neutrality rules, i.e., whether or not the traffic of certain service is prioritized. In this case, whether e.g., there is prioritization of the ports which require greater capacity of the network under test. Special case may be prioritization of ports which are used by the measuring devices (terminals). In such case, the result would certainly be substantially distorted.
- If it is very likely that traffic is deliberately prioritized toward standard measuring servers, either based on the IP address, or port, it is necessary to perform comparative measurement according to the above steps. If the results of the standard measurement and the comparative measurement significantly differ it is necessary to report this fact in the measurement results accordingly.
- It is advisable to perform supplemental indicative measurement by means of a publicly available tool for the measurement of the current quality of Internet access service, e.g., NetMetr (measuring server within MSEK).

6.1. Measuring tools

There are a number of measuring tools that are capable of performing measurements of TCP throughput. These measuring tools must be implemented in each of the two measuring sides where one acts as a client and one as a server. The tool must allow manual or automatic setting of the sending buffer size BS as well as the size of $\mathrm{TCP\,RWND}$, on both sides. The achievable TCP throughput of the data stream should be subsequently measured one-way as well as bidirectionally.

It is necessary to consider the performance of both measuring sides in order to prevent degradation of the measurement. Due to the qualitative development of Internet access service it is required that the measuring tool include an interface making it possible to perform measurements up to the maximum speed of $\mathrm{NBR} \leq 1000\ \mathrm{Mb/s}$ (on the side of the measuring server up to $\mathrm{NBR} \leq 10\ \mathrm{Gb/s}$). Due to the performance requirements of the measuring processes of the selected tools when measuring the data parameters with speed $\mathrm{NBR} > 100\ \mathrm{Mb/s}$ it is recommended to use measuring tools with dedicated hardware. When using the end-user's equipment, e.g., for indicative measurements, it is always necessary to consider the nominal performance of the equipment, loading with common applications as well as the age of the equipment. In such cases it is possible that even measurements at speed $\mathrm{NBR} \approx 50\ \mathrm{Mb/s}$ can be beyond the capabilities of the end-user's equipment.

6.2. Sequence of measurements

Access, sequence and evaluation of the results of TCP throughput are different for measurements in fixed electronic communication networks and for measurements in mobile electronic communication networks. In the case of mobile electronic communication networks, the sequences of measurements are further divided into measurement at a stationary point and mobile measurement. The following chapters provide the characteristics of the different measurement methods.

6.2.1. Measurements in fixed electronic communication networks

Measurement in fixed electronic communication networks in terms of the location of the measuring device (terminal) corresponds to stationary measurement. For all measurements at a stationary point it is recommended to perform repeated measurements with sufficient time and operational diversity.

It is recommended to perform three main independent measurements, including compliance with sufficient time diversity, i.e., at least one measurement at the peak hour and at least one measurement during an off-peak time. With respect to the time intensity of the process of measurement of TCP throughput, it is permissible to perform all three main measurements during the peak hours.

One measurement should not exceed the timeframe of 20 minutes during which a sequence of three tests will be performed. Because the resulting data parameters of the measuring process can be included in the set of basic data parameters, i.e., TCP throughput (upload) $\mathrm{TCP\ aTR_{up}}$, TCP throughput (download) $\mathrm{TCP\ aTR_{down}}$ and Delay or, as the case may be, $\mathrm{Delay(avg)}$, the term basic test (hereinafter referred to as "testB") is introduced. One test of the testB category must guarantee the length of measurement of TCP throughput in the following interval:

$$60\ \mathrm{s} < \mathrm{T_{TCP}} < 120\ \mathrm{s}, \tag{15}$$

whereas $\mathrm{T_{TCP}} = 90\ \mathrm{s}$ can be considered a recommended value of the length of measurement of TCP throughput. The reason for determining this value is detection of a regularly recurring discrepancy from normally available speed (BDR). With respect to the processing of the values measured ($\mathrm{T_{proc}}$) by the measuring tools used the total duration of one test should not exceed $\mathrm{T_{testB}}$ (see Figure 1):

$$\mathrm{T_{testB}} = \mathrm{T_{TCP}} + \mathrm{T_{proc}} \leq 150\ \mathrm{s}. \tag{16}$$

The resulting measurement process should consist of the following steps (see Figure 1):

- Step 1 – one-way test of TCP throughput (upload) $\mathrm{TCP\ aTR_{up}}$ including the value of $\mathrm{Delay(avg)}$ with total duration of the test $\mathrm{T_{testB}} \leq 150\ \mathrm{s}$,
- Step 2 – a break (saving the previous test results) with duration $\mathrm{T_{break}} \leq 120\ \mathrm{s}$,
- Step 3 – one-way test of TCP throughput (download) $\mathrm{TCP\ aTR_{down}}$ včetně hodnoty zpoždění $\mathrm{Delay(avg)}$ with total duration of the test $\mathrm{T_{testB}} \leq 150\ \mathrm{s}$,
- Step 4 – a break (saving the previous test results) with duration $\mathrm{T_{break}} \leq 120\ \mathrm{s}$,
- Step 5 – bidirectional test of TCP throughput (upload + download) $\mathrm{TCP\ aTR_{up}}$ and $\mathrm{TCP\ aTR_{down}}$ including the value of $\mathrm{Delay(avg)}$ with total duration of the test $\mathrm{T_{testB}} \leq 150\ \mathrm{s}$,
- Step 6 – a break until the start of the next sequence of measurements corresponding to the lapse of time (saving the previous test results, preparation for the next test) with duration $\mathrm{T_{break}} \leq 120\ \mathrm{s}$.

If the measuring tool does not make it possible to set the order of the sequence of tests as recommended it is possible to change the order without breaching the integrity of the measurement. In the same way, it is possible to omit the bidirectional test of TCP throughput (step 5) or the sequence of breaks between individual tests (steps 2, 4 and 6). The minimum admissible form of the process of measurement of TCP throughput must consist of one-way upload test (step 1) and one-way download test (step 3) of TCP throughput.

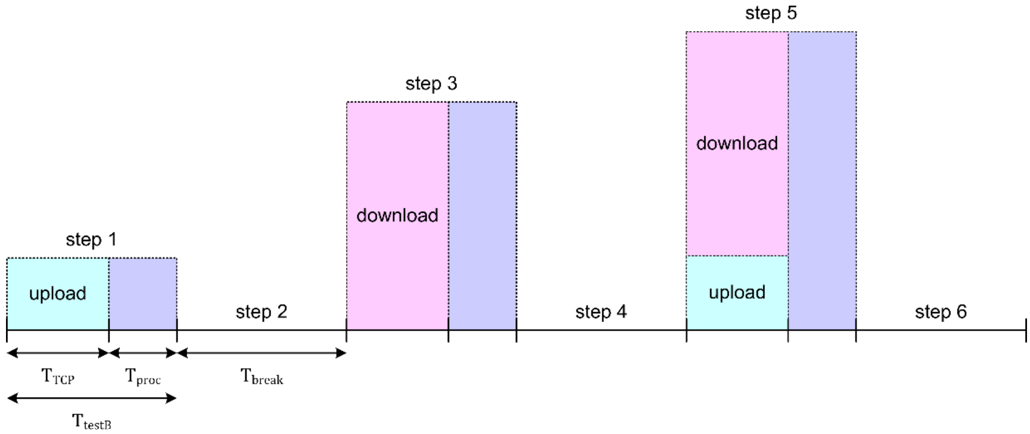

Figure 1: Recommended form of the process of measurement of TCP throughput

Possible combinations of the implementation of the minimum permissible form of the process of measurement depends on the measuring tools used. The theoretically possible combinations are provided on Figure 2, and they only differ by the processing of the values measured.

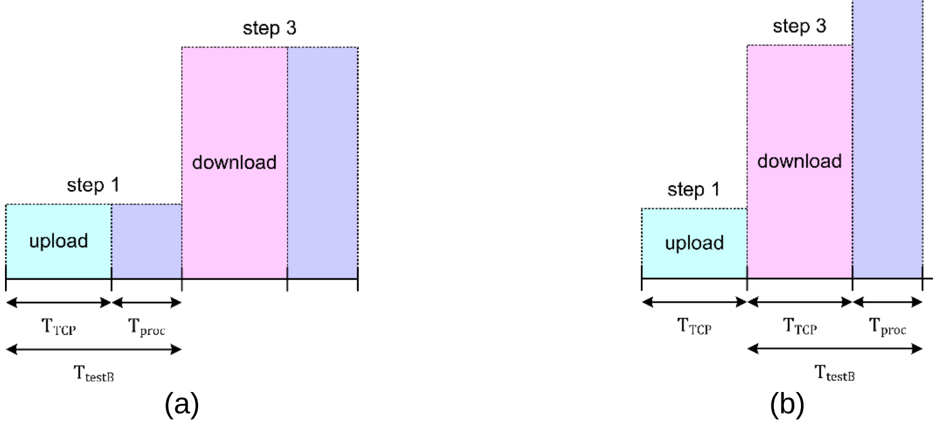

Figure 2: Versions of the minimum permissible form of the process of measurement of TCP throughput: (a) processing each one-way test separately, (b) processing all one-way tests at the conclusion of the measurement process

The measurement must be performed within specific demarcation points (DeP x) which will be specified in detail in the Methodology for measurement and evaluation of the data parameters of fixed electronic communication networks. The most common case we can imagine is measurement performed on the side of the end-user directly on the demarcation point of the service. Primarily it is necessary to use the converter (terminal unit) which is supplied to the customer upon activation of the service. If the situation requires it is possible to use another converter which is suitable for the particular type of service and technology. In all cases, however, it is necessary to check whether only the measuring device (terminal) is connected to the converter, at all demarcation points.

6.2.2. Measurements in mobile electronic communication networks

Measurement in mobile electronic communication networks in terms of the location of the measuring device (terminal) can correspond to stationary as well as mobile measurement. For all measurements at a stationary point it is recommended to perform repeated measurements with sufficient time and operating diversity. In cases where it is necessary to measure services of a mobile nature it is possible to use the so-called “drive test” or “walk test”). The typical purpose is ensuring coverage of the area in question with mobile data electronic communication network. In such case the measurement is continuous with a predefined measurement period (e.g., $1\ \mathrm{s}$), metrics (e.g., combination of the radio signal level and the data flow value at a specified point) and an evaluation network (e.g., square of $100 \times 100\ \mathrm{m}$). The current position of measurement while driving is determined using a GPS receiver, or approximated using other means (when GPS signal is not available), and the receiving antenna must be positioned such as to minimize negative effects of the vehicle.

When performing mobile measurements, it is necessary to keep several facts in mind:

- The “drive test” or “walk test” may only be performed at places where it is possible (i.e., in the case of a vehicle: on motorways, streets or roads; In the case of a manual (“handy”) measurement it is possible to include commercial areas, trains or otherwise inaccessible areas),
- The measurement must be ensured in the physical conditions of the equipment in question, mainly with regard to the speed of movement and the related issue of Doppler effect,
- The measurement of data speeds while driving is described in detail in a document entitled “Procedure to Measure the Data Transmission Speed in Mobile Networks Accordance with the LTE Standard” published in connection with the announcement of the competitive bidding procedure for the purpose of granting rights to use radio frequencies for the provision of public communication network in frequency bands 800 MHz, 1800 MHz and 2600 MHz.

IV. Evaluation and interpretation of results

The result and output of the entire measurement of TCP throughput should be a Measurement Report which shall include at least:

- Information on the time and place of measurement, measured technology, procedure and chronology of measurement.
- Information on the settings of the measuring system (measuring device), i.e., at least in the form of basic parameters such as BB, $\mathrm{minDelay}$, $\mathrm{TCP\ RWND}$ and MTU.
- Values of TCP throughput or, more precisely, the ideal value of TCP throughput $\mathrm{TCP\ iTR}$ and the current value of TCP throughput $\mathrm{TCP\ aTR}$ for each direction corresponding to the specific value of $\mathrm{TCP\ RWND}$ or the dynamically set size of $\mathrm{TCP\ RWND}$, always together with the information on the resulting $\mathrm{Delay(avg)}$. In addition, results of TCP metrics specified in the subsection Calculation of TCP metrics, the minimum permissible version in the form of at least $\mathrm{TCP\ EFF}$ and BD, for each direction.

In the case of a detected outage of the service or discrepancies from the expected values, it is necessary to consider possible causes. The details of the process of evaluation and interpretation of the results of the measuring process will be provided in the main section of the document and the respective annexes to the Methodology for measurement and evaluation of the data parameters of fixed electronic communication networks.

1. Evaluation process

As mentioned in the subsection Sequence of measurements, the process of evaluation of the measured results of TCP throughput is different for measurements in fixed electronic communication networks and for measurements in mobile electronic communication networks.

1.1. Fixed electronic communication networks

According to the subsection Measurements in fixed electronic communication networks, it is recommended to perform three main independent measurements of TCP throughput, whereas one measurement should not exceed the timeframe of 20 minutes during which a sequence of three tests will be performed.

Within the recommended form of the process of measurement of TCP throughput, the measurement result should be the following resulting values of parameters which can be included in the set of basic data parameters of the fixed electronic communication networks:

- upload test of TCP throughput $\mathrm{TCP\ aTR_{up}}$ including the value of $\mathrm{Delay(avg)}$, a part of the minimum admissible form of the process of measurement, step 1,
- download test of TCP throughput $\mathrm{TCP\ aTR_{down}}$ including the value of $\mathrm{Delay(avg)}$, a part of the minimum admissible form of the process of measurement, step 3,
- bidirectional test of TCP throughput $\mathrm{TCP\ aTR_{up}}$ and $\mathrm{TCP\ aTR_{down}}$ including the value of $\mathrm{Delay(avg)}$, step 5.

For the sake of better visibility, the results can be represented in the form of box plot. In the case of testing availability of major (known) ports (services) it is advisable to include this fact in a well-organized table.

A more detailed process of evaluation of the measured results of TCP throughput with regard to the Regulation (EU) 2015/2120 of the European Parliament and of the Council and the related Statement of the Czech Telecommunication Office on selected issues relating to open Internet access and European net neutrality rules is provided in the Methodology for measurement and evaluation of the data parameters of fixed electronic communication networks.

1.2. Mobile electronic communication networks

A more detailed process of evaluation of the measured results of TCP throughput with regard to the Regulation (EU) 2015/2120 of the European Parliament and of the Council and the related Statement of the Czech Telecommunication Office on selected issues relating to open Internet access and European net neutrality rules is provided in the Methodology for measurement and evaluation of the data parameters of mobile electronic communication networks.

2. Reasons for discrepancies from ideal values

There can be various reasons for unexpected results, from incorrect set-up of the measuring system up to insufficient capacity of the network and unavailability of network sources. Details of the reasons for the discrepancies can be found in the RFC 6349 recommendation, but substantial help in clarifying the discrepancies can be supplemental measurement based on ITU-T Y. 1564 or determination of the qualitative data parameters of the NUT in question (frame delay FD, inter-frame delay variation IFDV and frame loss ratio FLR).

3. Security considerations

Because the measurement of BB requires the use of stateless protocols this behavior in the measurement process can be perceived by network operators (providers) as an attempt for DoS or a DDoS attack. Therefore, the testing of TCP data flow may require coordination with the Internet connection provider.

3.1. Measurements in the networks with IPv6 and NAT

With respect to the possibility to encapsulate the TCP protocol in an IPv6 packet, significant differences in the measurement of TCP throughput can occur between IPv6 and IPv4 today in the electronic communication network with native IPv6 support. It is therefore advisable to check whether IPv6 connectivity is available, and if so, perform the measurement even in a situation where the TCP session will be encapsulated in IPv6 packets.

3.1.1. Measurements in the environment of private IP addresses and stateful firewalls

If, for some reason, the initiation of network session in the direction from server ("remote") → client ("local") is impossible, it is necessary to use such measuring tool that enables reverse initiation of network session when measuring TCP throughput (download). This situation may occur e.g., in electronic communication networks with NAT or with a set stateful firewall which blocks the TCP segment with SYN attribute (establishment of session) from the outside.

3.2. Physical and technological parameters

The measurement of TCP throughput should be performed in the configuration of client ("local") → server ("remote").

The server part should be located in the central (backbone) node of the data connection of all (either directly or through a mediator) providers of data services of electronic communications (hereinafter referred to as "provider"). The condition is the fulfillment of independence of the server part from all the providers so that the error of measurement of TCP throughput of a particular provider is as low as possible.

The client part should be located as close as possible to the interface which is declared by the provider as the place of provision of the provider's services (demarcation point) while fulfilling the condition of measuring TCP throughput at a place usual for the service subscriber or at a place determined by the contract between the provider and the subscriber. If the placement of the client part at the above-mentioned place is not possible, either for physical, technological, or other reasons, the measurement shall be performed at the closest possible network point.

V. Terms, definitions and abbreviations

BB (bottleneck bandwidth) – means the lowest value of the capacity of the network under test (b/s)

BDP (bandwidth-delay product) – means the result of the multiple of the capacity of the network under test (b/s) and the delay between both terminals of this network under test

BDR – normally available speed

bDelay (baseline Delay) – means the lowest measured value of Delay not burdened by the established TCP session during the initial testing interval

BS (socket buffer) – means the buffer of the data receiver or the data transceiver

Delay – is the time elapsed between the sending of the first bit of the TCP segment and the receipt of the last bit corresponding to the confirmation of the TCP segment

DF (don't fragment) – bit attribute

$\mathrm{Ethertyp}$ – determines the type of a higher protocol for Ethernet II

FCS (frame check sequence) – it is a 4 B cyclic redundant sum which makes it possible to detect damaged frames (CRC32 residue with value 0xC704DD7B)

FPS (frames per second) – parameter of the 2nd layer of the ISO/OSI model defining the number of frames transferred per second

IFG (inter-frame gap) – an obligatory gap between two frames, (100BASE-TX $= 0.96\ \mu\mathrm{s} = = 12\ \mathrm{B}$)

IR – information rate meaning the transfer rate on the connection layer (L 2) according to the ISO/OSI model

$\mathrm{MAC\ DST}$ – means the MAC address of the destination network interface with length 6 B

$\mathrm{MAC\ SRC}$ – means the MAC address of the source network interface with length 6 B

MIB (management information base) – represents an object-oriented set of SNMP objects, sessions and operations on and between objects. It is divided into 5 areas, while the area of performance management (monitoring of availability, response, throughput and use of individual vehicles) is necessary for the purposes of the Methodology

$\mathrm{minDelay}$ – means the lowest measured value of Delay during an established TCP session during the initial testing interval

MTU (maximum transmission unit) – means the maximum size of the IP datagram (TCP segment) which can newly be sent through the given network interface

n – number of TCP sessions

NAT - network address translation

NBR (net bit rate) – transmission speed on the physical layer (L 1) according to the ISO/OSI model

NUT - network under test

PMTUD (path MTU discovery) – standardized technique for determining the size of MTU

PPP (point-to-point protocol) – protocol of the link layer of the ISO/OSI model (L 2) enabling authentication, encryption and compression of the data transferred

$\mathrm{Preamble}$ – means 7 × 10101010 + 1 × 10101011 and is used for synchronization of the recipient's clock (Ethernet II)

packet delay variation – variation in the delay between the delivery of individual packets (jitter) rTB – means the number of bits which had to be resent after an error

SD – data sets

TB – number of transferred bits

TCP TTD (TCP throughput test device) – means a device which generates the traffic metrics and performs measurements as defined in the recommendation of IETF RFC 6349

TCP RWND (TCP receive window) – means the size of the TCP window of the data receiver

TCP RWNDmin (minimum TCP receive window) – means the calculated value of TCP RWND based on the value of parameter BDP

TCP window scale option – makes it possible, according to the RFC 7323 recommendation, “TCP extensions for high performance”, to increase the size of TCP RWND up to $< 2^{30}$, i.e., up to the value < 1GB

traffic policing – means of monitoring traffic of the electronic communication network in order to limit the maximum transfer rate by means of traffic trimming

traffic shaping – means of management of the traffic of the electronic communication network in order to distribute it and to regulate the transmission speed

TWAMP (a two-way active measurement protocol) – means an open protocol for measuring two-way metrics of the network under test. It is based on the architecture of the OWAMP protocol (RFC 4656) and uses also the same architecture and design

802.1Q – means VLAN Tagging or, more precisely, makes it possible to divide one physical Ethernet into multiple logical networks (so-called VLAN) by means of extending the header of the Ethernet frame to include additional items

802.1ad – means the concept of double VLAN Tagging