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Delay Limits and Multiplexing Policies to be employed with Tunneling
Compressing and Multiplexing Traffic Flows
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Abstract

This document contains recommendations to be taken into account when using methods which optimize bandwidth utilization through Tunneling Compressing and Multiplexing (TCM) traffic flows over a network path. Different multiplexing policies and implementation issues which are service and link specific are discussed. Additionally, this document describes policies which can be used for detecting, classifying, and choosing the network flows suitable for optimization by using TCM. Finally, recommendations of maximum tolerable delays to be added by optimization techniques are reported. Recommendations are presented only for network services for which such bandwidth optimization techniques are applicable (i.e., services with low payload to header size ratio, which will also be denoted as "small-packet flows").

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1. Introduction

This document extends the draft [TCM] with a set of recommendations regarding the processes of compressing, multiplexing, and tunneling. These recommendations are needed because the techniques proposed in [TCM], while saving bandwidth, may cause network impairments.

Network delay is one of the main factors which can degrade the Quality of Experience (QoE) of real-time network services RFC 6390 [RFC6390] [TGPP_TR26.944]. In order to prevent the perceived quality degradation of the services when using TCM, a policy defining a multiplexing period can be employed.

First, the document describes different multiplexing policies which can be employed for defining which native packets are multiplexed together. A policy combining a multiplexing period and a packet size limit is proposed in order to put an upper bound on the added delay.

Additionally, this document describes the policies that can be employed for detecting, classifying, and choosing the network flows suitable for TCM optimization.

Finally, values for maximum tolerable delays presented here from the base of the proposed multiplexing policy. The recommendations are presented for both real-time and non real-time network services in which TCM bandwidth optimization is applicable (i.e., services which have low payload-to-header-size ratio, which results in high protocol overhead, which will also be denoted as small-packet flows).

1.1. Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [RFC2119].

2. Terminology

This document uses a number of terms to refer to the roles played by participants in, and objects of, the TCM sessions.

TCM optimizer

The host where TCM optimization is deployed. If that hosts corresponds to the ingress of the tunnel where native packets are included it is called TCM-ingress optimizer (TCM-IO).

The host where TCM multiplexed packets are received and rebuilt to their native form is called TCM-egress optimizer (TCM-IO). It corresponds to the tunnel egress.

policy manager

A network entity which makes the decisions about TCM parameters: multiplexing period; flows to be multiplexed together, depending on their IP addresses, ports, etc. It is connected with a number of TCM optimizers, and orchestrates the optimization that takes place between them.

native packet

A packet sent by an application, belonging to a flow that can be optimized by means of TCM.

TCM-optimized packet

A packet including a number of multiplexed and header-compressed native ones, and also a tunneling header shared by all the packets, as detailed by TCM.

3. Considered services

The services considered suitable for being optimized by TCM are those that generate long-term flows of small packets, with a low payload to header size ratio. Some real-time and some non real-time services are suitable for optimization by means of TCM.

3.1. Real-time services

Under the term "real-time network services" we consider both conversational and streaming service classes as defined in [TGPP_TS]. Interactive and background services are considered non real-time. Fundamental requirements of real-time network services include conversational pattern (stringent and low delay) and preservation of the time relation (variation) between the information entities of the stream.

We identify the following real-time network services with low payload to header size ratio as candidates for the bandwidth optimization techniques presented in TCM:

- o Voice over IP
- o Online games
- o Remote desktop services

While video services are considered real-time, they are not suitable for bandwidth optimization techniques proposed in [TCM], due to their high payload to header size ratio. Due to the same reason, we do not take into account cloud gaming services. In such gaming services all the calculations of the game state are deployed at the server and a real-time video stream is sent to the client. In these cases, TCM optimization is neither interesting nor applicable.

3.2. Non real-time services

On the other hand, TCM can be applied for some non real-time services such as streaming audio, and instant messaging. These applications are suitable for TCM in terms of payload to header size ratio, but different studies have shown that acceptable delays for these services are up to several seconds [ITU-T_G.1010]. Also, some types of machine to machine (M2M) traffic (e.g., metering messages from various sensors) may have traffic properties suitable for TCM. Acceptable delays for these services can be go up to an hour [Liu_M2M]. We list limitations for these services as well, although in the practical application TCM should not introduce delays which would be noticeable in comparison with delays of such magnitude (i.e., seconds and more).

4. Multiplexing policies in TCM

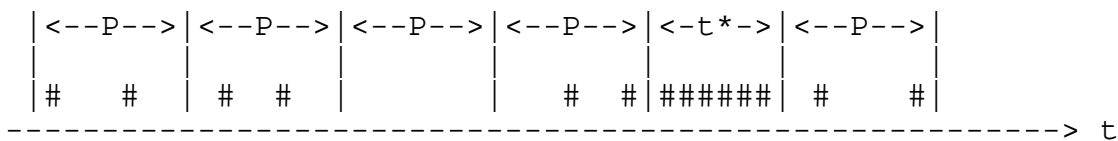
A multiplexing policy defines the decision process for determining which native packet goes in which multiplexed packet. The policies proposed for TCM are:

- o Fixed number of packets - once a fixed number of packets (N) has arrived, a multiplexed packet is created and sent.
- o Size limit - once a size limit is reached (e.g., next to the MTU of the underlying network), a multiplexed packet is sent.
- o Period - a multiplexed packet is sent every time period T.
- o Timeout - sends a multiplexed packet if a native one arrives and the time since the last multiplexed packet departure is above a defined timeout value.

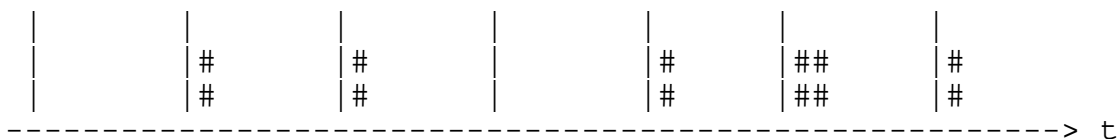
Only the two latter policies are able to control the additional delay introduced by multiplexing. In addition, different policies can be combined.

In this document we focus on the combination of "size limit" and "period" policies, as shown in Figure 1. A multiplexed packet is sent at the end of each "period". However, if the size limit is reached, then a multiplexed packet is sent immediately, and the period is "reset". Thus, the added delay is for the worst case scenario equal to the defined period.

native traffic:



multiplexed traffic:



* period reset ($t < P$) because size limit is reached

Combined "period" and "size limit" policies

Figure 1

It should be noted that the number of aggregated flows and their packet rate will have an influence on the average multiplexing delay added. If the number of flows is high, then the MTU size will be reached before the end of the period in most cases, so the additional delay will be smaller than the period. The recommendations presented in this document present the maximum period values to be used as a limit, in order to avoid delays which could impair the quality of the service.

5. Detecting, classifying, and choosing network flows to be optimized

Three basic issues need to be solved in order to employ TCM optimization. First, the flows which are candidates for optimization need to be detected from the overall traffic mix. Secondly, the flows need to be classified into one of the defined categories so an adequate multiplexing period can be assigned. Finally, the decision whether a specific flow will be optimized or not using TCM needs to be made.

5.1. Optimization within an administrative domain

Several scenarios can be considered for the use of TCM. If the optimization is deployed within an administrative domain, then all the data of the end hosts, the service class, etc., are known by the TCM optimizers.

Two examples of this are 1) the end-to-end optimization and aggregation of a number of flows between two offices of the same

company and 2) the agreement between a network operator and a game provider in order to multiplex all the packets generated in an aggregation network with destination on a game server. In these cases, the detection and classification of the desired flows will be straightforward, since the TCM optimizer can simply inspect the destination IP address and port, and apply the traffic category according to the kind of service.

5.2. Optimization based on statistics

If the optimization is not performed within an administrative domain, then the detection and classification of the flows, and the decision about multiplexing them, will have to be based on statistics of the traffic and heuristics. The intelligence of the flow identification method can be improved according to the statistics of already classified flows. E.g., if a number of small-packet flows sharing the same IP destination address are found, then this destination host can be classified as a frequent receiver of small-packet flows, and a tunnel including all the packets addressed to it can be set within a common network path.

In addition, statistics can be enriched by the assignment of the traffic class, taking into account that some services, in addition to well-known ports, also have well-known IP addresses. E.g., the traffic travelling to the IP address of a popular online game server, can be associated with the proper traffic class; or the ports corresponding to certain services can also be identified and used in order to improve the classification.

The detection of the flows candidates for TCM optimization should be done based on flow characteristics, primarily on the packet payload to header ratio and on the packet rate. As these properties cannot be established from statistics of just one packet, it is needed to gather a certain number of packets for each new flow arriving at the TCM optimizer, and to use some heuristics in order to decide whether to multiplex a certain flow or not.

The classification method employed for the TCM needs to identify only the flows which are candidates for the TCM optimization. Therefore, the classification problem is significantly simplified by removal of peer to peer (P2P) downloading applications, video streaming, data transfer, and all other services which in general, utilize large packets. This is especially important as P2P applications are known to use various non assigned ports which may greatly ruin the performance of simple traffic classification methods. For the purposes of TCM optimization there is no need to identify a particular application, only the delay sensitive class in which that application falls. Also, the traffic classification methods employed

by TCM need to be able to assign flows to respective delay sensitive classes in real time. Current network traffic classification methods can be grouped into [Nguyen_TCSurvey]:

- o Port based - through lookup of port numbers of endpoints in the Internet Assigned Numbers Authority (IANA)'s list of registered ports.
- o Payload based - through stateful reconstruction of session and application information from each packet's content.
- o Statistical - through comparison of the statistical properties of the traffic at the network layer.

While payload inspection does give the best results, and is often used as ground truth in classification of network traffic, it is demanding computation wise. Also, these techniques may be interpreted as a violation of privacy. For the purposes of TCM we recommend using a combination of port based classification (which can yield very good results for games based on a client-server architecture and remote desktop services), and inspection of statistical properties of the flows. One such algorithm has been employed for classification of different types of game flows and showed good results [Han_GameClassification]. TCM should use metadata information regarding the traffic class of particular flow where such information is available as that significantly simplifies the detection and classification problem.

The decision whether the flow should be optimized with TCM can be made based on the following policies (configurations of the multiplexing node):

- o A static configuration - predefined rule set for each new occurring flow, so the TCM optimizer makes a decision.
- o A policy manager which dynamically enforces the rule set.
- o The TCM optimizer demands instructions for each new flow from the policy manager.

6. Delay recommendations

The three normally considered network impairments in the studies related to subjective quality in real-time interactive games are:

- o delay - can be reported as one-way-delay (OWD) [RFC2679] and two-way-delay (Round Trip Time) [RFC2681]. In this document, under the term latency, one way end-to-end delay is considered.

- o delay variation - which is a statistical variance of the data packet inter-arrival time, in other words the variation of the delay as defined in RFC 3393 [RFC3393].
- o packet loss - more important for certain applications, while other include very good algorithms for concealing it (e.g., some game genres).

In this document we give recommendations for overall tolerable delays to be taken into account when optimizing network services by means of TCM. In an interactive service, the total delay is composed by the addition of delays as defined in 3GPP TR 26.944 [TGPP_TR26.944].

- o Transfer delay - from Host1 to Host2 at time T is defined by the statement: Host1 sent the first bit of a unit data to Host2 at wire-time T and that Host2 received the last bit of that packet at wire-time T+dT. Thus, it includes the transmission delay (the amount of time Host1 requires to push all of the packet's bits into the wire) and the propagation delay in the network (the amount of time it takes for the head of the packet to travel from Host1 to Host2).
- o Transaction delay - the sum of the time for a data packet to wait in queue and receive the service during the server transaction.

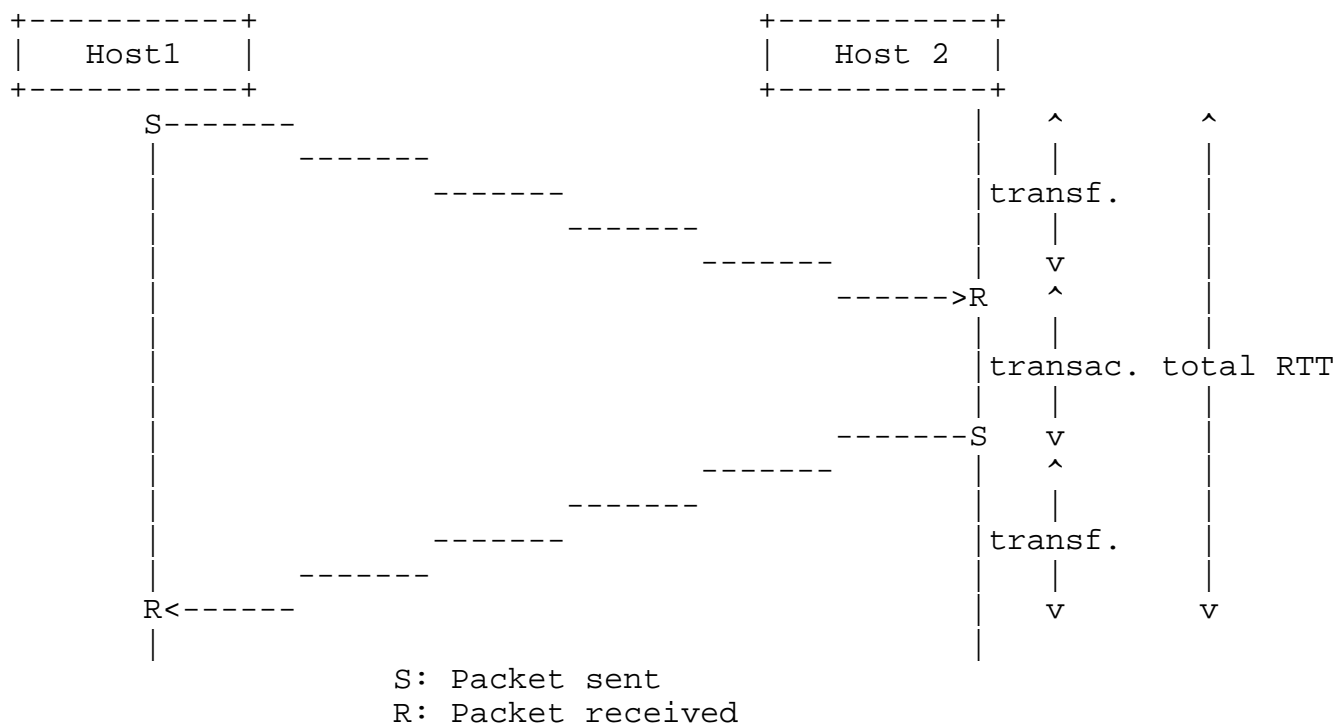


Figure 2

Figure 2 illustrates these delays. The labeled times (S and R) designate the times in which the packet is sent and received, respectively, by the network card interface.

The use of TCM requires the addition of TCM optimizers in the scenario. A number of flows are multiplexed together before being sent through the network. The packets are demultiplexed and rebuilt before being forwarded to the application server. A scheme of TCM is included in Figure 3:

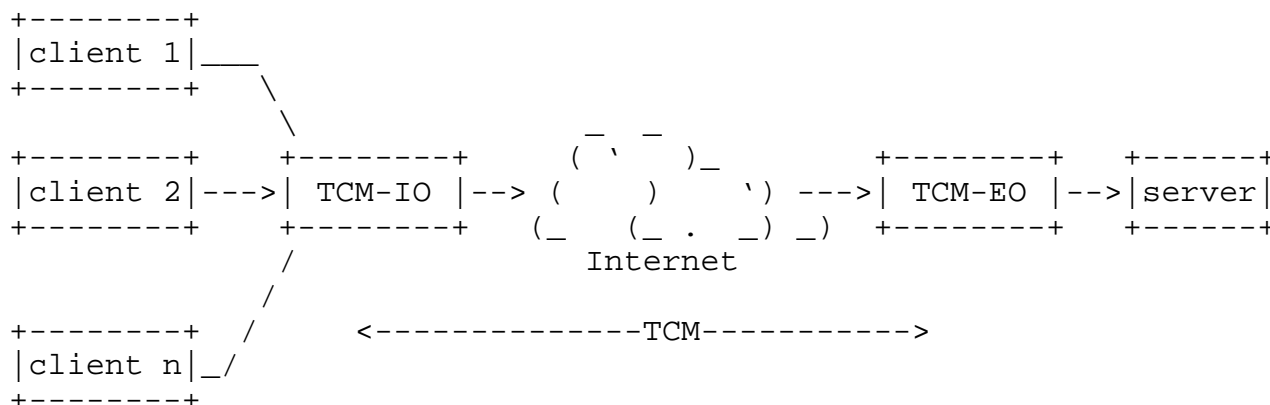


Figure 3

This technique groups packets in order to build a multiplexed one. As previously stated, the focus of this document is on "multiplexing period" policy for creating the multiplexed packet combined with size limit policy. Multiplexing period is a time frame in which the TCM optimizer waits for native packets to arrive in order to send them as one multiplexed packet. If the multiplexed packet size limit is reached before the multiplexing period has run out (i.e., if enough native packets arrive to fill the limit), the multiplexed packet is sent right away. In this way a certain amount of delay caused by the TCM optimization is added in the communication. It is important to emphasize that multiplexing delay can't exceed the multiplexing period, and that it will only reach the value of multiplexing period on a link with a low traffic load. Multiplexing delay can be classified as caused by the middlebox presence as defined in RFC 6390 [RFC6390]. The delay in the TCM-I/O includes the time during which the packets are retained until the bundled packet is sent, plus processing time. In the TCM-E/O however, the packets are not retained, so only the processing time is considered.

Figure 4 shows the total delay, when a TCM optimizers are added. It should be noted that multiplexing can be deployed independently in both directions, or only in one of them.

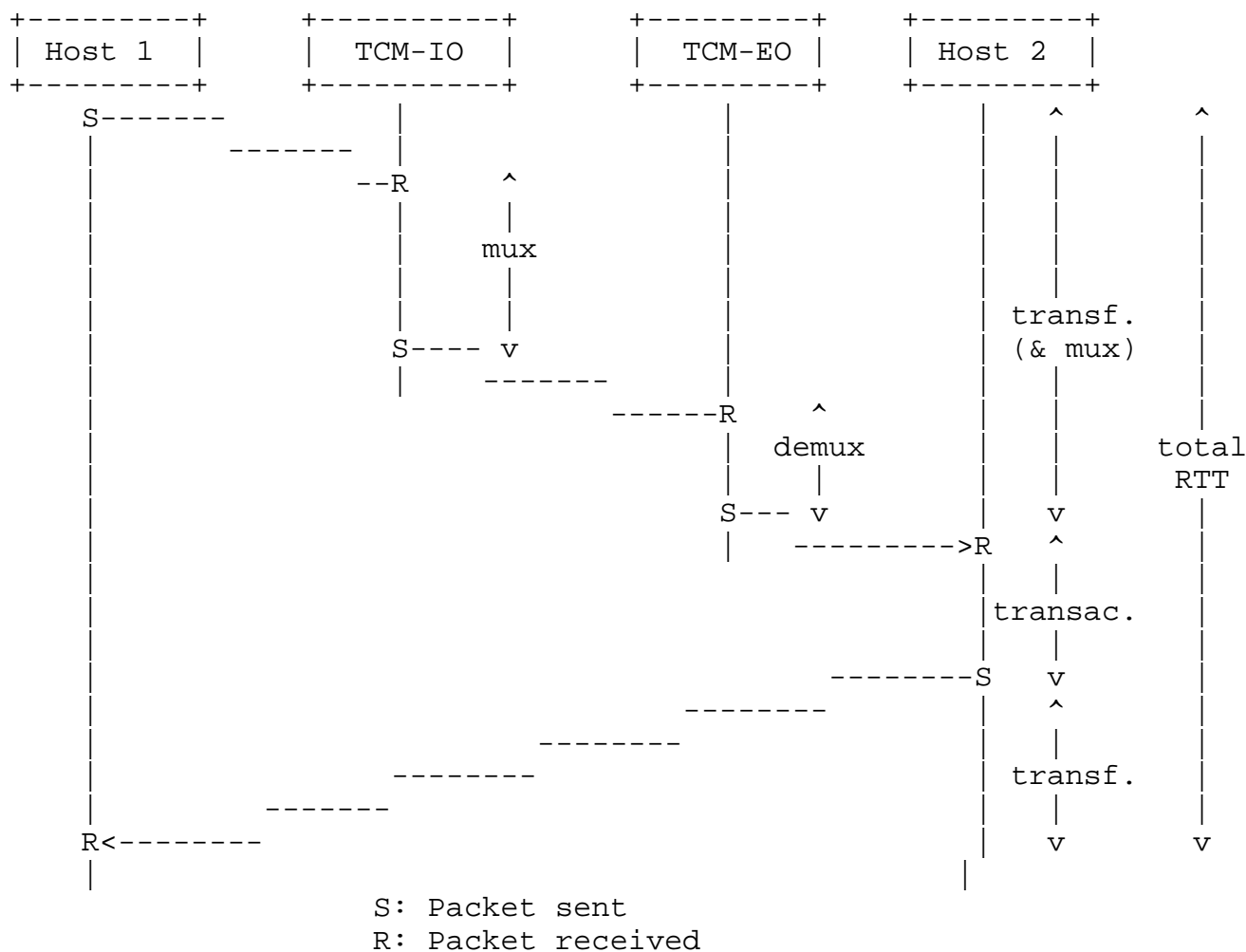


Figure 4

With respect to efficiency in terms of use of the bandwidth, a tradeoff appears: the longer the multiplexing period, the higher the number of packets which can be grouped, thus obtaining better bandwidth savings. So in order to calculate the maximum multiplexing period, the rest of the delays have to be considered: if the sum of transaction, and transfer delays is under the maximum tolerable delay, then multiplexing will be possible without harming the user experience. The overall delay may be calculated according to the ITU-T Y.1541 recommendation [ITU-T_Y.1541]. Subtracting propagation, processing, and transmission delay from the tolerable delay for specific service results in the maximum value of the multiplexing period.

Next, we will report the maximum tolerable latency for the previously listed real-time network services.

6.1. VoIP

For conversational audio, the International Telecommunication Union recommends [ITU-T_G.114] less than 150 millisecond one-way end-to-end delay for high-quality real time traffic, but delays between 150 ms and 400 ms are acceptable. When considering conversational audio it should be noted that this delay limits include jitter buffers and codec processing. For streaming audio, delay constraints are much looser, the delay should be less than 10 s [ITU-T_G.1010]. Tunneling Multiplexed Compressed RTP (TCRTP) [RFC4170] already considers tunneling, compressing and multiplexing VoIP packets.

6.2. Online games

Online games comprise game genres which have different latency requirements. This draft focuses on real-time online games and endorses the general game categorization proposed in [Claypool_Latency] in which online games have been divided into:

- o Omnipresent, with the threshold of acceptable latency (i.e., latency in which performance is above 75% of the unimpaired performance) of 1000 ms. The most representative genre of omnipresent games are Real-Time Strategies.
- o Third Person Avatar, with the threshold of acceptable latency of 500 ms. These games include Role Playing Games (RPG) and Massively Multiplayer Online Role-Playing Games (MMORPG).
- o First Person Avatar, in which threshold of acceptable latency is 100 ms. The most popular subgenre of them are First Person Shooters, such as "Call of Duty" or "Halo" series.

As remarked in [Bernier_Latency] and [Oliveira_online], different methods can be employed to combat delay in online games. The so-called "client-side prediction" has been largely used in First Person Shooters. It can be divided into "input prediction" and "dead reckoning," where input prediction hides the latency for the client-controlled actions while dead reckoning hides the latency of other participating players.

The study [Claypool_Latency] evaluated players' performance in certain tasks, while increasing latency, and reported values at which the performance dropped below 75% of the performance under unimpaired network conditions. While measuring objective performance metrics, this method highly underestimates the impact of delays on players' QoE. Further studies accessing a particular game genre reported much lower latency thresholds for unimpaired gameplay.

Other approach some studies have taken is to perform "objective measurements" [Kaiser_objective] a number of identical "bots", i.e. virtual avatars controlled by Artificial Intelligence, are placed in the same virtual scenario and a number of parties between them are performed. If the number of parties is high enough, then the score will be the same for all the bots. Then, different network impairments (latency, jitter, packet loss) are added to one of the bots, and another set of tests is performed. The performance degradation of the network-impaired bot can then be statistically characterized.

A survey using a large number of First Person Shooter games has been carried out in [Dick_Analysis]. They state that latency about 80 ms could be considered as acceptable, since the games have been rated as "unimpaired". Besides service QoE, it has been shown that delay has great impact on the user's decision to join a game, but significantly less on the decision to leave the game [Henderson_QoS].

A study on Mean Opinion Score (MOS) evaluation, based on variation of delay and jitter for MMORPGs, suggested that MOS drops below 4 for delays greater than 120 ms [Ries_QoEMMORPG]. The MOS score of 5 indicates excellent quality, while MOS score of 1 indicates bad quality. Another study focused on extracting the duration of play sessions for MMORPGs from the network traffic traces showed that the session durations start to decline sharply when round trip time is between 150 ms and 200 ms [Chen_HowSensitive].

While original classification work [Claypool_Latency] states that latency up to 1 second is tolerated by omnipresent games, other studies argued that only latency up to 200 ms is tolerated by players of RTS games [Cajada_RTS].

6.3. Remote desktop access

For the remote computer access services, the delays are dependent on the task performed through the remote desktop. Tasks may include operations with audio, video and data (e.g., reading, web browsing, document creation). A QoE study indicates that for audio latency below 225 ms and for data latency below 200 ms is tolerated [Dusi_Thin].

6.4. Non real-time service

Traffic flows of several types of non real-time services can be optimized using TCM. Under this category we include services for M2M metering information, streaming audio, and instant messaging. M2M metering services are suitable for TCM optimization not only due to their very loose delay requirements, but also because of the one way

nature of the communication (i.e., most information travels from sensors to the central server) [Liu_M2M]. The signalling information related to M2M can also be optimized. Internet of Things application layer protocols such as CoAP RFC 7252 [RFC7252], used in Constrained RESTful Environments (CoRE)[RFC6690], work over UDP and send small packets. The ACK_TIMEOUT period in CoAP is set to 2 seconds. Instant messaging (despite "instant" in its name) has been categorized as data service by the ITU-T, and it has been designated with acceptable delays of up to a few seconds [ITU-T_G.1010].

6.5. Summary

We group all the results in the Table 1 indicating the maximum allowed latency and proposed multiplexing periods. Proposed multiplexing periods are guidelines, since the exact values are dependant of the existing delay in the network. It should be noted that reported tolerable latency is based on values of preferred delays, and delays in which QoE estimation is not significantly degraded. Multiplexing periods of about 1 second can be considered as sufficient for non real-time services (e.g., streaming audio).

Service	Tolerable latency (OWD)	Mux. period
Voice communication	< 150ms	< 30ms
Omnipresent games	< 200ms	< 40ms
First person avatar games	< 80ms	< 15ms
Third person avatar games	< 120ms	< 25ms
Remote desktop	< 200ms	< 40ms
Instant messaging	< 5s	< 1s
M2M (metering)	< 1hour	< 1s

Table 1: Final recommendations

7. Acknowledgements

8. IANA Considerations

This memo includes no request to IANA.

9. Security Considerations

No relevant security considerations have been identified

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