SLA CALCULUS FOR END-TO-END QOS OF TCP-BASED APPLICATIONS IN A MULTI-DOMAIN ENVIRONMENT

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Abstract

Next-generation communication services will be offered over distributed information and communication infrastructures consisting of a multitude of administrative domains, owned by different parties. This raises the problem for service providers to provide satisfactory levels of end-to-end Quality of Service (QoS), as experienced by the paying end user, in a cost-effective manner. Motivated by this, we consider the problem of end-to-end QoS provisioning for TCP-based applications that cross multiple network domains. To this end, we construct an analytical model that provides a so-called SLA calculus, i.e. a mapping between per-domain network QoS parameters defined in the involved Service Level Agreements (SLAs) and end-to-end QoS metrics like response times and file download times that determine the QoS perceived by the end users.

1 Introduction

A key topic in future telecommunications is how to provision end-to-end QoS for services/applications offered in multi-domain environments. An effective and increasingly popular means to deal with QoS in multi-domain environments is to negotiate Service Level Agreements (SLAs) between the different domain owners. In this context, a key question to be addressed by a Service Provider (SP) is "What *combination* of SLAs should be agreed upon by the SP and the respective network domain owners to achieve a certain predefined end-to-end QoS level?". Nowadays most attention is focussed on the delivery of time sensitive applications, such as voice, video and transction applications (in banking for example). For instance, in [5] the user perceived QoS for VoIP in a heterogenous multi-domain environment is considered. However, also the performance of TCP-based applications is of an ever increasing importance. For instance, it is mentioned in [12] that online customers who are most satisfied with a site spend 57% more at a site than those who kept shopping but were unhappy about the experience. Therefore in this paper we consider the performance of TCP-based applications in a multi-domain environment, see Figure 1.

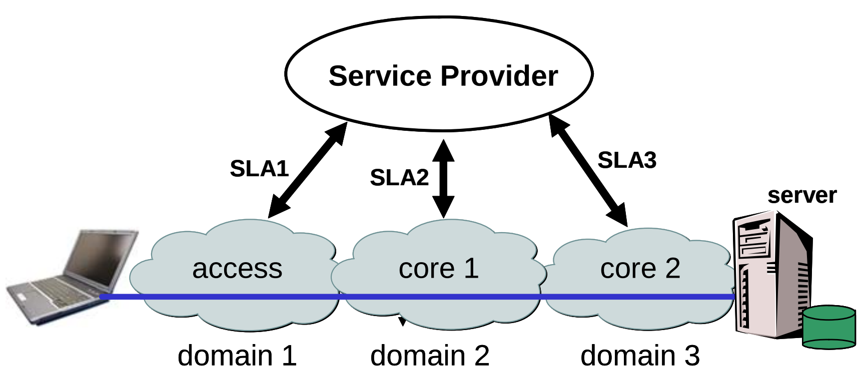


Figure 1: End-to-end TCP connection crossing an access and two core network domains

Note that a service provider would not subscribe to a costly premium service class in one of the intermediate backbone networks, when there appears to be an inevitable bottleneck with respect to the achievable end-to-end QoS in e.g. the wireless access. In more general terms, the service provider should try to distribute the involved 'performance budgets' (e.g. packet loss, delay) over the underlying domains, such that the envisioned end-to-end QoS level can be realized against minimal costs. It is clear that this 'cost optimization' problem requires the ability to determine end-to-end QoS guarantees from the performance parameters specified in the SLAs. This mapping of a set of per-domain SLAs to an associated (achievable) end-to-end QoS level is called *SLA calculus*.

In the literature, most papers on QoS in multi-domain environments are concerned with the role of SLAs or network functionalities (signaling, routing, traffic engineering mechanisms) that are – or may be - needed in order to enable QoS provisioning, see e.g.[2, 4, 7, 9, 10]. However papers that consider end-to-end QoS provisioning in multi-domain environments from a more quantitative point of view are rare. As noted before [5] is an exception to this.

In this work we will show in a multi-domain environment how the the performance parameters specified in SLAs determine QoS metrics for web browsing that are relevant to users. These QoS metrics have been discussed in detail in [3]. In fact, based upon extensive subjective tests, in [3] a relation is established between these QoS metrics and the quality as perceived by users, expressed as a Mean Opinion Score (MOS). The mapping between network performance and these QoS metrics was already described in the single domain setting in [11].

The remainder of this paper is organized as follows. In section 2 the main QoS metrics for web browsing are defined, namely the Response Time and the Total Download Time. In section 3 we discuss the TCP performance models proposed in [11] that can be used to compute these QoS metrics. In section 4 we extend the work of [11] by describing how to apply the TCP performance models for estimating the end-to-end QoS of web browsing in a multi-domain environment. In section 5 we validate our models for the Response Time and Total Download Time in a multi-domain environment through ns-2 simulations. Finally, in section 6 the main conclusions are given.

2 QoS metrics for web browsing that are of interest to the user

Analysis of the perceived QoS for web browsing has shown that the following two metrics are relevant to the end-user, see [3]:

1. Response Time (RT): time from clicking on a link until something appears on the screen,

2. Total Download Time (TDT): time between clicking on a link and time the download is complete.

The RT and TDT are depicted in Figure 2. Here, the parameters T_i have the following meaning: T_1 = time when user clicks, T_2 = server sends first data packet, T_3 = user receives first response, and T_4 = total data transfer complete.

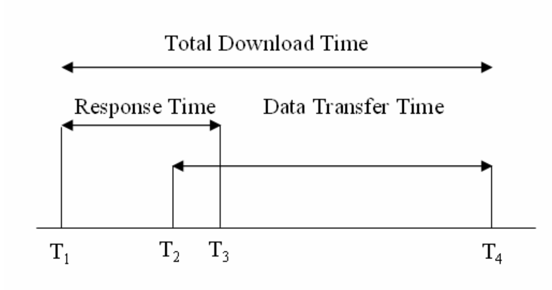


Figure 2: QoS metrics for TCP downloads

It is obvious that

$$RT = T_3 - T_1 \text{ , } DTT = T_4 – T_2 \text{ and } TDT = T_4 - T_1, \tag{1}$$

where DTT stands for the Data Transfer Time (DTT). Note that in general, not only the mean of the RT and TDT are relevant, but also the variability of these performance metrics is of key importance. However, as can be deduced from [3], for web browsing variation in the order of 25% about the mean of these metrics is hardly perceived by users. Therefore, in this paper we will only focus on mean values for the RT and TDT.

3 Analytical models for mean Response Time and Total Download Time

This section will contain a brief description of the analytical models for the mean RT and TDT as reported in [11].

3.1 Response Time

In this subsection we describe the model suggested in [11] for the Response Time (RT), i.e., the time it takes to establish a TCP connection and the additional time it takes to send the first packet containing data. From a user's point of view, the RT is simply the time from clicking on a URL link until the first packet arrives and something appears on the screen. Each TCP connection starts with a "three-way handshake", in which the client and server exchange initial sequence numbers. The RT is determined by the time it takes to send four packets successfully; here, the first three packets are related to the three-way handshake while the fourth packet contains the first data. If it assumed that the packet loss probabilities in forward and reverse direction are the same and denoted by p then according to [11] the mean Response Time RT satisfies:

$$RT = \frac{A\left(b_0 + b_1 A + b_2 A^2 + b_3 A^3 + b_4 A^4 + b_5 A^5 + b_6 A^6 + b_7 A^7\right)}{1 - A} + 2RTT \tag{2}$$

where

$$A = p(2-p), b_0 = T_0 + T_u, b_1 = 3T_0, b_2 = 6T_0 + T_u, b_3 = 14T_0 + 2T_u,$$
$$b_4 = 32T_0 + 4T_u, b_5 = 72T_0 + 8T_u, b_6 = 160T_0 + 16T_u, b_7 = -160T_0 - 32T_u.$$

In this expression T_0 denotes the initial value of the Retransmission Timer, which according to RFC2988 [8] satisfies $T_0 = 3$ seconds. According to [8], upon its first update the Retransmission Timer becomes

$$T_u = \max\{1, RTT + \max\{G, 2RTT\}\},$$

where G denotes the TCP timer granularity. In many TCP implementations G is set to 500ms.

It is also shown in [11] that in case the forward and backward loss probabilities are denoted by p_f and p_b respectively then the mean RT also satisfies (2) with $A = p_f + (1- p_f) p_b$.

3.2 Total Download Time

The Total Download Time (TDT) consists of the time between clicking on a link and the arrival of the last data packet. From section 2 we know that this corresponds with the TCP connection establishment time plus the time to transfer all the data. Because the TCP connection establishment time is the time taken to send three packets successfully, we approximate it by 3/4 times the *RT*. Because the time it takes to send all data equals the Data Transfer Time we get the following relation:

$$TDT = \tfrac{3}{4}\, RT + DTT \tag{3}$$

Cardwell et al [1] proposed a model for the Data Transfer Time under the assumption that packet loss happens only in the direction from sender to receiver. Their model directly depends on the one-way packet loss (from server to client), whereas in reality not only data segments can be lost during a TCP data transmission but also the ACKs of data packets can be dropped in the direction from the client to the server. In [11] we have extended the Cardwell model by including the impact of loss of ACKs.

According to [11] the mean Data Transfer Time can be expressed as follows

$$DTT = f_2(d; W_{\max}, T_0, b, MSS, w_1, T_{delACK}; RTT, p) \tag{4}$$

where the parameters have the following meaning: d: size of downloaded file, W_{max}: maximum TCP window size, T_0: Retransmission Timer, b: number of data packets aknowledged by one ACK, *MSS*: TCP maximum segment size, w_1: initial slow-start window size, T_{delACK}: the delayed ACK timer, *RTT*: round trip time. The parameter p denotes the packet loss as experienced by the TCP source. As mentioned before, in [11] this parameter depends both on the forward packet loss rate (p_f, data packet loss rate) and on the backward packet loss rate (p_b, ACK loss rate). Due to space limitation we refer to [11] for the explicit expression of $f_2(.)$.

4 Response Time and Total Download Time in a multi-domain environment

According to section 3 the mean RT, due to the connection establishment of TCP and sending the first data packet, can be modeled as follows:

$$RT = f_1(T_0; RTT, p). \tag{5}$$

Note that in Equation (5) the Retransmission Timer T_0 is determined by TCP, while the other parameters are network parameters. This difference is reflected in the notation.

The TDT is the sum of ¾ times RT and the Data Transfer Time (DTT), see Equation (3). A symbolic expression for the DTT is given in Equation (4).

To apply Equations (4) and (5) in a multi-domain environment the network parameters *RTT* and p have to be determined, under the condition that the *round trip time* and *packet loss ratio* are known for the individual domains. In addition a possible bandwidth restriction in each domain has to be known, in order to quantify the impact on W_{max}, the maximum TCP window size. This is precisely what is commonly specified in commercial SLAs.

To be more precise, for each of the network domains the following performance measures are specified in the SLA's between service provider and network operators:

p_i : packet loss probability for network i,
RTT_i : round trip time for network i,
B_i : bandwidth limitation for network i.

Now, we will approximate the end-to-end performance measures parameters RTT and p from the packet loss probabilities and round trip times for the individual networks according to the following simple formulas:

$$RTT = \sum_{i=1}^{N} RTT_i, \qquad p = 1 - \prod_{i=1}^{N} (1 - p_i). \tag{6}$$

The parameter W_{max} is derived from the bandwidth limitations B_i of the individual networks and the sender and receiver buffer limitations. Denoting the latter by W_{send} and $W_{receive}$, we approximate W_{max} as follows:

$$W_{\max} = \min\{W_{send}, W_{receive,} \frac{RTT}{MSS} B_1, \frac{RTT}{MSS} B_2, \ldots, \frac{RTT}{MSS} B_n\}. \tag{7}$$

Combining Equations (3) – (7) gives the desired model for mean end-to-end RT and TDT for TCP based downloads in a multi-domain environment.

5 Validation

In this section we will validate the models proposed in the previous section for the end-to-end RT and TDT in a multi-domain environment by means of extensive simulations with the ns-2 simulator, see [6]. In the ns-2 simulations the default TCP flavor is chosen i.e. TCP Reno.

5.1 Validation of Response Time model

We assume that a user crosses four domains in order to reach a server, see Figure 3.

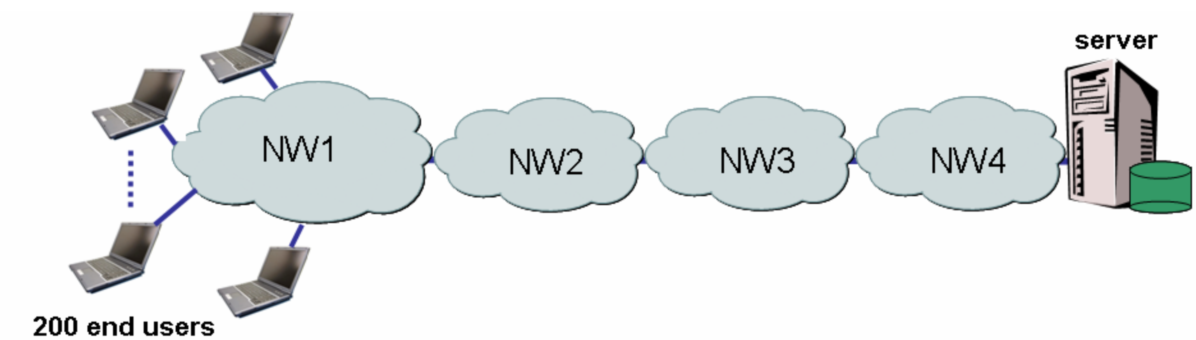


Figure 3: Simulation topology for RT model in a multi-domain environment

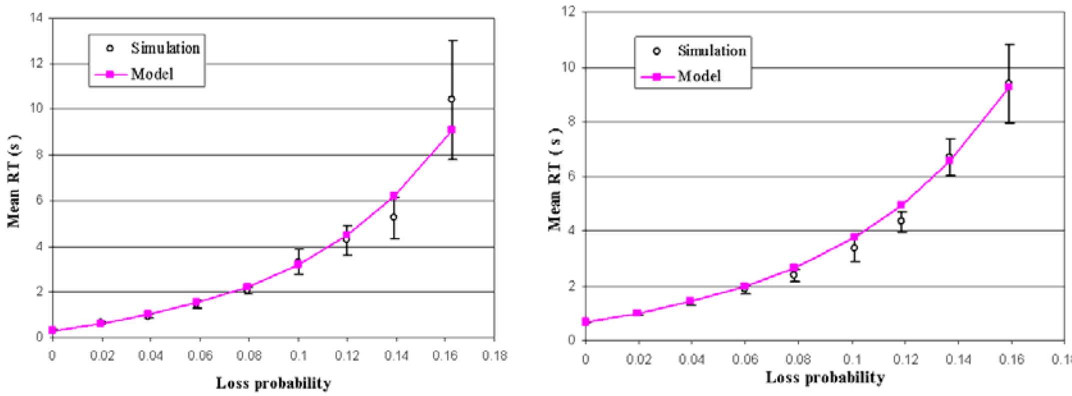
For the TCP parameters we choose the following values: (a) MSS = 1640 Byte, (b) the maximum receiver window size = 32 packets, (c) TCP's retransmission timer at the client side = 1s, (d) b, the number of packets that is acknowledged by each ACK = 1 and (e) w_1, the initial number of packets the TCP source is sending = 1. For the SLA's in the four networks (NWs) we have the following assumptions:

NW1: $p_1 = 0$, $RTT_1 = 20$ ms, $B_1 = 10$ Mbps
NW2: $p_2 = 0.0001$, $RTT_2 = 20$ ms, $B_2 = 1$ Gbps

NW3: $p_3 = 0.0005$, $RTT_3 \in \{2\text{ ms}, 20\text{ ms}, 200\text{ ms}\}$, $B_3 = 1$ Gbps
NW4: $p_4 \in \{0, 0.02,..0,16\}$, $RTT_4 = 100$ ms, $B_4 = 1$ Gbps.

Note that NW1 typically corresponds to a broadband access network with more bandwidth than ADSL, e.g. VDSL. NWs 2, 3 and 4 typically correspond to core networks.
In order to obtain enough data it is assumed for every simulation run that 200 end users connect to NW1. For each scenario we assumed a fixed value for RTT_3 and p_4 and for each scenario 10 simulations were run.

The analytical results for the mean RT in a multi-domain environment and the 95% confidence intervals of the simulated values are depicted in Figure 4 for the cases RTT = 160 ms and RTT = 340 ms. These results correspond with the case RTT_3 = 20 ms and RTT_3 = 200 ms respectively. Denoted on the horizontal axis in Figure 4 is the loss probability in one way.



(a) RTT = 160 ms (b) RTT = 340 ms

Figure 4: Response Time in a multi-domain environment: model versus simulation

On the basis of Figure 4 and various additional simulation results we conclude that the model-based predictions of Response Time in a multi-domain environment is very accurate.

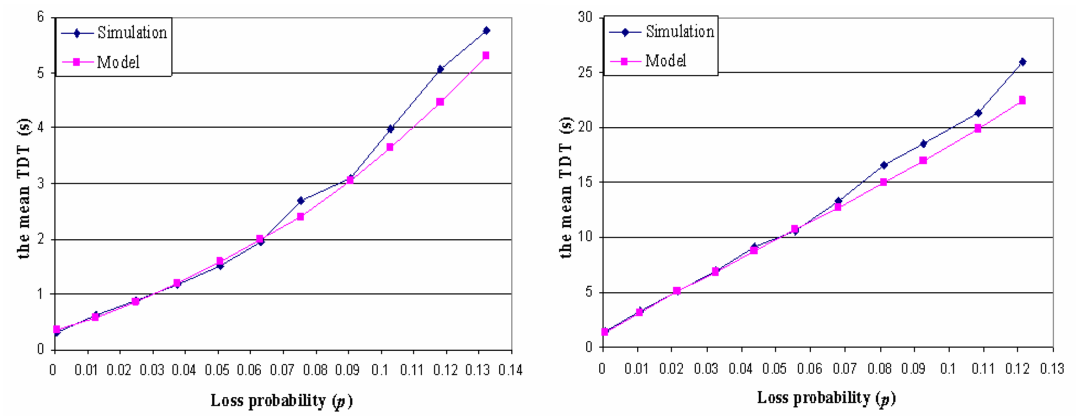
5.2 Validation of Total Download Time model

Again we assume that a user crosses four domains in order to reach a server.
For the TCP parameters we choose the following values: (a) MSS = 1640 Byte, (b) *b*, the number of packets that is acknowledged by each ACK = 1 and (c) w_1, the initial number of packets the TCP source is sending = 1.
For the SLA's in the four networks (NWs) we have the following assumptions:

NW1: $p_1 = 0$, $RTT_1 = 20$ ms, $B_1 = 10$ Mbps
NW2: $p_2 = 0.0001$, $RTT_2 = 10$ ms, $B_2 = 1$ Gbps
NW3: $p_3 = 0.0005$, $RTT_3 \in \{20\text{ ms}, 60\text{ ms}, 260\text{ ms}, 460\text{ms}\}$, $B_3 = 1$ Gbps
NW4: $p_4 \in \{0, 0.01,..0,1\}$, $RTT_4 = 10$ ms, $B_4 = 1$ Gbps.

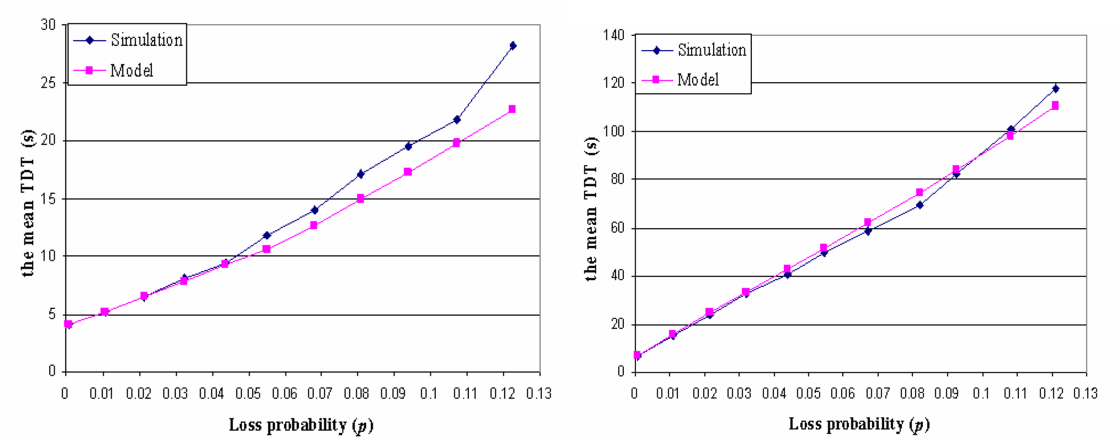
Note that the packet loss in both directions is assumed to be equal. Again each scenario is characterized by a fixed value for RTT_3 and p_4.In addition to these two parameters mentioned above we have also varied the size of the downloaded file ($d \in \{10, 500\}$ packets) and the maximum window size ($W_{max} \in \{8, 32\}$ packets).

The analytical results for the mean TDT in a multi-domain environment and the simulated values are depicted in Figure 5 and Figure 6 for several cases. Denoted on the horizontal axis in Figure 5 and Figure 6 is the packet loss as experienced by the TCP source.



(a) file size = 10 packets (b) file size = 500 packets

Figure 5: TDT in a multi-domain environment; RTT = 60 ms, W_{max} = 32: model versus simulation



(a) RTT = 60 ms, W_{max} = 8 (b) RTT = 300 ms, W_{max} = 32

Figure 6: TDT in a multi-domain environment; file size = 500 packets: model versus simulation

On the basis of Figure 5 and Figure 6 and various additional simulation results we conclude that the model-based predictions of Total Download Time in a multi-domain environment are very accurate for small packet loss ratio's (up to 5%) and quite accurate for packet loss ratio up to 10%.

6 Conclusions

We have addressed the problem of predicting end-to-end QoS of TCP-based applications in a multi-domain environment, a scenario that is typical for next-generation networks in unbundled telecommunication infrastructures. To this end, we have used recent analytical models for response times and total download times. Both models have been made suitable for application within a multi-domain setting which is characterized by per-domain SLAs. The response time and total download time are of main importance for the performance experienced by the end user. We have proposed simple rules for an SLA-calculus for predicting the end-to-end performance of TCP for connections crossing multiple domains, each with a per-domain SLA. Experimental results show a very good match between predictions based on the analytical model and simulations, demonstrating the fact that these simple calculation rules are well applicable to predict end-to-end performance in multi-domain infrastructures.

In a future work we will consider variances of Response Time and Total Download Time under stationary 'worst case' network conditions, i.e. under the assumption that the packet loss probabilities and roundtrip times are equal to the upperbounds p_i and RTT_i given in the SLAs. In particular, under these assumptions, the analysis of the mean response time in Section 3.1 can be extended to response time variances. The analysis of the variance of the total download time will, in general, be much harder. However, for small flows consisting of only a few packets this seems possible: in that case one can easily keep track of the possible occurrences during the slow start phase. For the congestion avoidance phase it seems not possible to extend the analysis of mean download times in [1] to the download time variance. However, one may expect that for very large downloads, the variability of the download time will be very small (relative to the mean time needed for these large downloads).

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