

Internet Engineering Task Force (IETF)  
Request for Comments: 7765  
Category: Experimental  
ISSN: 2070-1721

P. Hurtig  
A. Brunstrom  
Karlstad University  
A. Petlund  
Simula Research Laboratory AS  
M. Welzl  
University of Oslo  
February 2016

## TCP and Stream Control Transmission Protocol (SCTP) RT0 Restart

### Abstract

This document describes a modified sender-side algorithm for managing the TCP and Stream Control Transmission Protocol (SCTP) retransmission timers that provides faster loss recovery when there is a small amount of outstanding data for a connection. The modification, RT0 Restart (RTOR), allows the transport to restart its retransmission timer using a smaller timeout duration, so that the effective retransmission timeout (RTO) becomes more aggressive in situations where fast retransmit cannot be used. This enables faster loss detection and recovery for connections that are short lived or application limited.

### Status of This Memo

This document is not an Internet Standards Track specification; it is published for examination, experimental implementation, and evaluation.

This document defines an Experimental Protocol for the Internet community. This document is a product of the Internet Engineering Task Force (IETF). It represents the consensus of the IETF community. It has received public review and has been approved for publication by the Internet Engineering Steering Group (IESG). Not all documents approved by the IESG are a candidate for any level of Internet Standard; see Section 2 of RFC 5741.

Information about the current status of this document, any errata, and how to provide feedback on it may be obtained at <http://www.rfc-editor.org/info/rfc7765>.

## Copyright Notice

Copyright (c) 2016 IETF Trust and the persons identified as the document authors. All rights reserved.

This document is subject to BCP 78 and the IETF Trust's Legal Provisions Relating to IETF Documents (<http://trustee.ietf.org/license-info>) in effect on the date of publication of this document. Please review these documents carefully, as they describe your rights and restrictions with respect to this document. Code Components extracted from this document must include Simplified BSD License text as described in Section 4.e of the Trust Legal Provisions and are provided without warranty as described in the Simplified BSD License.

## Table of Contents

1. Introduction . . . . .	3
2. Terminology . . . . .	4
3. RT0 Overview and Rationale for RTOR . . . . .	4
4. RTOR Algorithm . . . . .	6
5. Discussion . . . . .	7
5.1. Applicability . . . . .	7
5.2. Spurious Timeouts . . . . .	7
5.3. Tracking Outstanding and Previously Unsent Segments . . . . .	8
6. Related Work . . . . .	9
7. SCTP Socket API Considerations . . . . .	10
7.1. Data Types . . . . .	10
7.2. Socket Option for Controlling the RT0 Restart Support (SCTP_RT0_RESTART) . . . . .	10
8. Security Considerations . . . . .	11
9. References . . . . .	11
9.1. Normative References . . . . .	11
9.2. Informative References . . . . .	13
Acknowledgements . . . . .	14
Authors' Addresses . . . . .	15

## 1. Introduction

TCP and SCTP use two almost identical mechanisms to detect and recover from data loss, specified in [RFC6298] and [RFC5681] for TCP and [RFC4960] for SCTP. First, if transmitted data is not acknowledged within a certain amount of time, a retransmission timeout (RT0) occurs and the data is retransmitted. While the RT0 is based on measured round-trip times (RTTs) between the sender and receiver, it also has a conservative lower bound of 1 second to ensure that delayed data are not mistaken as lost. Second, when a sender receives duplicate acknowledgments or similar information via selective acknowledgments, the fast retransmit algorithm suspects data loss and can trigger a retransmission. Duplicate (and selective) acknowledgments are generated by a receiver when data arrives out of order. As both data loss and data reordering cause out-of-order arrival, fast retransmit waits for three out-of-order notifications before considering the corresponding data as lost. In some situations, however, the amount of outstanding data is not enough to trigger three such acknowledgments, and the sender must rely on lengthy RT0s for loss recovery.

The amount of outstanding data can be small for several reasons:

- (1) The connection is limited by congestion control when the path has a low total capacity (bandwidth-delay product) or the connection's share of the capacity is small. It is also limited by congestion control in the first few RTTs of a connection or after an RT0 when the available capacity is probed using slow-start.
- (2) The connection is limited by the receiver's available buffer space.
- (3) The connection is limited by the application if the available capacity of the path is not fully utilized (e.g., interactive applications) or is at the end of a transfer.

While the reasons listed above are valid for any flow, the third reason is most common for applications that transmit short flows or use a bursty transmission pattern. A typical example of applications that produce short flows are web-based applications. [RJ10] shows that 70% of all web objects, found at the top 500 sites, are too small for fast retransmit to work. [FDT13] shows that about 77% of all retransmissions sent by a major web service are sent after RT0 expiry. Applications with bursty transmission patterns often send data in response to actions or as a reaction to real life events. Typical examples of such applications are stock-trading systems, remote computer operations, online games, and web-based applications

using persistent connections. What is special about this class of applications is that they are often time dependent, and extra latency can reduce the application service level [P09].

The RT0 Restart (RTOR) mechanism described in this document makes the effective RT0 slightly more aggressive when the amount of outstanding data is too small for fast retransmit to work, in an attempt to enable faster loss recovery while being robust to reordering. While RTOR still conforms to the requirement for when a segment can be retransmitted, specified in [RFC6298] for TCP and [RFC4960] for SCTP, it could increase the risk of spurious timeouts. To determine whether this modification is safe to deploy and enable by default, further experimentation is required. Section 5 discusses experiments still needed, including evaluations in environments where the risk of spurious retransmissions are increased, e.g., mobile networks with highly varying RTTs.

The remainder of this document describes RTOR and its implementation for TCP only, to make the document easier to read. However, the RTOR algorithm described in Section 4 is applicable also for SCTP. Furthermore, Section 7 details the SCTP socket API needed to control RTOR.

## 2. Terminology

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].

This document introduces the following variables:

- o The number of previously unsent segments (prevunsnt): The number of segments that a sender has queued for transmission, but has not yet sent.
- o RT0 Restart threshold (rrthresh): RTOR is enabled whenever the sum of the number of outstanding and previously unsent segments (prevunsnt) is below this threshold.

## 3. RT0 Overview and Rationale for RTOR

The RT0 management algorithm described in [RFC6298] recommends that the retransmission timer be restarted when an acknowledgment (ACK) that acknowledges new data is received and there is still outstanding data. The restart is conducted to guarantee that unacknowledged segments will be retransmitted after approximately RT0 seconds. The standardized RT0 timer management is illustrated in Figure 1, where a TCP sender transmits three segments to a receiver. The arrival of

the first and second segment triggers a delayed ACK (delACK) [RFC1122], which restarts the RT0 timer at the sender. The RT0 is restarted approximately one RTT after the transmission of the third segment. Thus, if the third segment is lost, as indicated in Figure 1, the effective loss detection time becomes "RT0 + RTT" seconds. In some situations, the effective loss detection time becomes even longer. Consider a scenario where only two segments are outstanding. If the second segment is lost, the time to expire the delACK timer will also be included in the effective loss detection time.

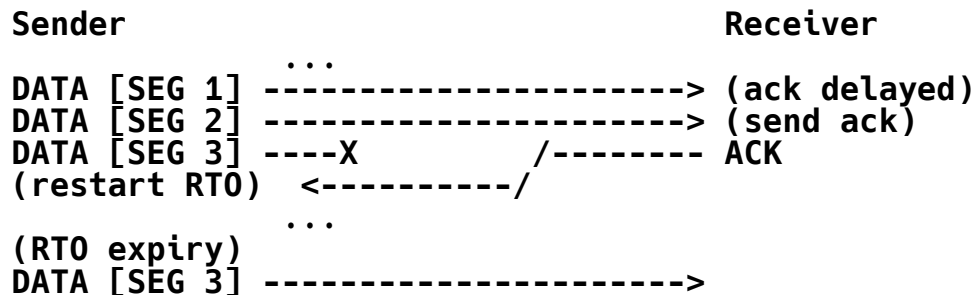


Figure 1: RT0 Restart Example

For bulk traffic, the current approach is beneficial -- it is described in [EL04] to act as a "safety margin" that compensates for some of the problems that the authors have identified with the standard RT0 calculation. Notably, the authors of [EL04] also state that "this safety margin does not exist for highly interactive applications where often only a single packet is in flight." In general, however, as long as enough segments arrive at a receiver to enable fast retransmit, RT0-based loss recovery should be avoided. RT0s should only be used as a last resort, as they drastically lower the congestion window as compared to fast retransmit.

Although fast retransmit is preferable, there are situations where timeouts are appropriate or are the only choice. For example, if the network is severely congested and no segments arrive, RT0-based recovery should be used. In this situation, the time to recover from the loss(es) will not be the performance bottleneck. However, for connections that do not utilize enough capacity to enable fast retransmit, RT0-based loss detection is the only choice, and the time required for this can become a performance bottleneck.

#### 4. RTOR Algorithm

To enable faster loss recovery for connections that are unable to use fast retransmit, RTOR can be used. This section specifies the modifications required to use RTOR. By resetting the timer to "RT0 - T\_earliest", where T\_earliest is the time elapsed since the earliest outstanding segment was transmitted, retransmissions will always occur after exactly RT0 seconds.

This document specifies an OPTIONAL sender-only modification to TCP and SCTP, which updates step 5.3 in Section 5 of [RFC6298] (and a similar update in Section 6.3.2 of [RFC4960] for SCTP). A sender that implements this method MUST follow the algorithm below:

When an ACK is received that acknowledges new data:

- (1) Set T\_earliest = 0.
- (2) If the sum of the number of outstanding and previously unsent segments (prevunsnt) is less than an RTOR threshold (rrthresh), set T\_earliest to the time elapsed since the earliest outstanding segment was sent.
- (3) Restart the retransmission timer so that it will expire after (for the current value of RT0):
  - (a)  $RT0 - T_{\text{earliest}}$ , if  $RT0 - T_{\text{earliest}} > 0$ .
  - (b) RT0, otherwise.

The RECOMMENDED value of rrthresh is four, as this value will ensure that RTOR is only used when fast retransmit cannot be triggered. With this update, TCP implementations MUST track the time elapsed since the transmission of the earliest outstanding segment (T\_earliest). As RTOR is only used when the amount of outstanding and previously unsent data is less than rrthresh segments, TCP implementations also need to track whether the amount of outstanding and previously unsent data is more, equal, or less than rrthresh segments. Although some packet-based TCP implementations (e.g., Linux TCP) already track both the transmission times of all segments and also the number of outstanding segments, not all implementations do. Section 5.3 describes how to implement segment tracking for a general TCP implementation. To use RTOR, the calculated expiration time MUST be positive (step 3(a) in the list above); this is required to ensure that RTOR does not trigger retransmissions prematurely when previously retransmitted segments are acknowledged.

## 5. Discussion

Although RTOR conforms to the requirement in [RFC6298] that segments must not be retransmitted earlier than RTT seconds after their original transmission, RTOR makes the effective RTT more aggressive. In this section, we discuss the applicability and the issues related to RTOR.

### 5.1. Applicability

The currently standardized algorithm has been shown to add at least one RTT to the loss recovery process in TCP [LS00] and SCTP [HB11] [PBP09]. For applications that have strict timing requirements (e.g., interactive web) rather than throughput requirements, using RTOR could be beneficial because the RTT and the `delACK` timer of receivers are often large components of the effective loss recovery time. Measurements in [HB11] have shown that the total transfer time of a lost segment (including the original transmission time and the loss recovery time) can be reduced by 35% using RTOR. These results match those presented in [PGH06] and [PBP09], where RTOR is shown to significantly reduce retransmission latency.

There are also traffic types that do not benefit from RTOR. One example of such traffic is bulk transmission. The reason why bulk traffic does not benefit from RTOR is that such traffic flows mostly have four or more segments outstanding, allowing loss recovery by fast retransmit. However, there is no harm in using RTOR for such traffic as the algorithm is only active when the amount of outstanding and unsent segments are less than `rrthresh` (default 4).

Given that RTOR is a mostly conservative algorithm, it is suitable for experimentation as a system-wide default for TCP traffic.

### 5.2. Spurious Timeouts

RTOR can in some situations reduce the loss detection time and thereby increase the risk of spurious timeouts. In theory, the retransmission timer has a lower bound of 1 second [RFC6298], which limits the risk of having spurious timeouts. However, in practice, most implementations use a significantly lower value. Initial measurements show slight increases in the number of spurious timeouts when such lower values are used [RHB15]. However, further experiments, in different environments and with different types of traffic, are encouraged to quantify such increases more reliably.

Does a slightly increased risk matter? Generally, spurious timeouts have a negative effect on the network as segments are transmitted needlessly. However, recent experiments do not show a significant

increase in network load for a number of realistic scenarios [RHB15]. Another problem with spurious retransmissions is related to the performance of TCP/SCTP, as the congestion window is reduced to one segment when timeouts occur [RFC5681]. This could be a potential problem for applications transmitting multiple bursts of data within a single flow, e.g., web-based HTTP/1.1 and HTTP/2.0 applications. However, results from recent experiments involving persistent web traffic [RHB15] revealed a net gain using RTOR. Other types of flows, e.g., long-lived bulk flows, are not affected as the algorithm is only applied when the amount of outstanding and unsent segments is less than `rrthresh`. Furthermore, short-lived and application-limited flows are typically not affected as they are too short to experience the effect of congestion control or have a transmission rate that is quickly attainable.

While a slight increase in spurious timeouts has been observed using RTOR, it is not clear whether or not the effects of this increase mandate any future algorithmic changes -- especially since most modern operating systems already include mechanisms to detect [RFC3522] [RFC3708] [RFC5682] and resolve [RFC4015] possible problems with spurious retransmissions. Further experimentation is needed to determine this and thereby move this specification from Experimental to the Standards Track. For instance, RTOR has not been evaluated in the context of mobile networks. Mobile networks often incur highly variable RTTs (delay spikes), due to e.g., handovers, and would therefore be a useful scenario for further experimentation.

### 5.3. Tracking Outstanding and Previously Unsent Segments

The method of tracking outstanding and previously unsent segments will probably differ depending on the actual TCP implementation. For packet-based TCP implementations, tracking outstanding segments is often straightforward and can be implemented using a simple counter. For byte-based TCP stacks, it is a more complex task. Section 3.2 of [RFC5827] outlines a general method of tracking the number of outstanding segments. The same method can be used for RTOR. The implementation will have to track segment boundaries to form an understanding as to how many actual segments have been transmitted but not acknowledged. This can be done by the sender tracking the boundaries of the `rrthresh` segments on the right side of the current window (which involves tracking `rrthresh + 1` sequence numbers in TCP). This could be done by keeping a circular list of the segment boundaries, for instance. Cumulative ACKs that do not fall within this region indicate that at least `rrthresh` segments are outstanding, and therefore RTOR is not enabled. When the outstanding window becomes small enough that RTOR can be invoked, a full understanding of the number of outstanding segments will be available from the `rrthresh + 1` sequence numbers retained. (Note: the implicit sequence



number consumed by the TCP FIN bit can also be included in the tracking of segment boundaries.)

Tracking the number of previously unsent segments depends on the segmentation strategy used by the TCP implementation, not whether it is packet based or byte based. In the case where segments are formed directly on socket writes, the process of determining the number of previously unsent segments should be trivial. In the case that unsent data can be segmented (or resegmented) as long as it is still unsent, a straightforward strategy could be to divide the amount of unsent data (in bytes) with the Sender Maximum Segment Size (SMSS) to obtain an estimate. In some cases, such an estimation could be too simplistic, depending on the segmentation strategy of the TCP implementation. However, this estimation is not critical to RTOR. The tracking of prevunsnt is only made to optimize a corner case in which RTOR was unnecessarily disabled. Implementations can use a simplified method by setting prevunsnt to rrthresh whenever previously unsent data is available, and set prevunsnt to zero when no new data is available. This will disable RTOR in the presence of unsent data and only use the number of outstanding segments to enable/disable RTOR.

## 6. Related Work

There are several proposals that address the problem of not having enough ACKs for loss recovery. In what follows, we explain why the mechanism described here is complementary to these approaches:

The limited transmit mechanism [RFC3042] allows a TCP sender to transmit a previously unsent segment for each of the first two duplicate acknowledgements (dupACKs). By transmitting new segments, the sender attempts to generate additional dupACKs to enable fast retransmit. However, limited transmit does not help if no previously unsent data is ready for transmission. [RFC5827] specifies an early retransmit algorithm to enable fast loss recovery in such situations. By dynamically lowering the number of dupACKs needed for fast retransmit (dupthresh), based on the number of outstanding segments, a smaller number of dupACKs is needed to trigger a retransmission. In some situations, however, the algorithm is of no use or might not work properly. First, if a single segment is outstanding and lost, it is impossible to use early retransmit. Second, if ACKs are lost, early retransmit cannot help. Third, if the network path reorders segments, the algorithm might cause more spurious retransmissions than fast retransmit. The recommended value of RTOR's rrthresh variable is based on the dupthresh, but it is possible to adapt to allow tighter integration with other experimental algorithms such as early retransmit.

Tail Loss Probe [TLP] is a proposal to send up to two "probe segments" when a timer fires that is set to a value smaller than the RT0. A "probe segment" is a new segment if new data is available, else it is a retransmission. The intention is to compensate for sluggish RT0 behavior in situations where the RT0 greatly exceeds the RTT, which, according to measurements reported in [TLP], is not uncommon. Furthermore, TLP also tries to circumvent the congestion window reset to one segment by instead enabling fast recovery. The probe timeout (PT0) is normally two RTTs, and a spurious PT0 is less risky than a spurious RT0 because it would not have the same negative effects (clearing the scoreboard and restarting with slow-start). TLP is a more advanced mechanism than RTOR, requiring e.g., SACK to work, and is often able to further reduce loss recovery times. However, it also noticeably increases the amount of spurious retransmissions, as compared to RTOR [RHB15].

TLP is applicable in situations where RTOR does not apply, and it could overrule (yielding a similar general behavior, but with a lower timeout) RTOR in cases where the number of outstanding segments is smaller than four and no new segments are available for transmission. The PT0 has the same inherent problem of restarting the timer on an incoming ACK and could be combined with a strategy similar to RTOR's to offer more consistent timeouts.

## 7. SCTP Socket API Considerations

This section describes how the socket API for SCTP defined in [RFC6458] is extended to control the usage of RT0 restart for SCTP.

Please note that this section is informational only.

### 7.1. Data Types

This section uses data types from [IEEE.9945]: `uintN_t` means an unsigned integer of exactly N bits (e.g., `uint16_t`). This is the same as in [RFC6458].

### 7.2. Socket Option for Controlling the RT0 Restart Support (SCTP\_RT0\_RESTART)

This socket option allows the enabling or disabling of RT0 Restart for SCTP associations.

Whether or not RT0 restart is enabled per default is implementation specific.

This socket option uses IPPROTO\_SCTP as its level and SCTP\_RT0\_RESTART as its name. It can be used with getsockopt() and setsockopt(). The socket option value uses the following structure defined in [RFC6458]:

```
struct sctp_assoc_value {
    sctp_assoc_t assoc_id;
    uint32_t assoc_value;
};
```

**assoc\_id:** This parameter is ignored for one-to-one style sockets. For one-to-many style sockets, this parameter indicates upon which association the user is performing an action. The special sctp\_assoc\_t SCTP\_{FUTURE|CURRENT|ALL}\_ASSOC can also be used in assoc\_id for setsockopt(). For getsockopt(), the special value SCTP\_FUTURE\_ASSOC can be used in assoc\_id, but it is an error to use SCTP\_{CURRENT|ALL}\_ASSOC in assoc\_id.

**assoc\_value:** A non-zero value encodes the enabling of RT0 restart whereas a value of 0 encodes the disabling of RT0 restart.

sctp\_opt\_info() needs to be extended to support SCTP\_RT0\_RESTART.

## 8. Security Considerations

This document specifies an experimental sender-only modification to TCP and SCTP. The modification introduces a change in how to set the retransmission timer's value when restarted. Therefore, the security considerations found in [RFC6298] apply to this document. No additional security problems have been identified with RT0 Restart at this time.

## 9. References

### 9.1. Normative References

- [RFC1122] Braden, R., Ed., "Requirements for Internet Hosts - Communication Layers", STD 3, RFC 1122, DOI 10.17487/RFC1122, October 1989, <<http://www.rfc-editor.org/info/rfc1122>>.
- [RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, DOI 10.17487/RFC2119, March 1997, <<http://www.rfc-editor.org/info/rfc2119>>.

- [RFC3042] Allman, M., Balakrishnan, H., and S. Floyd, "Enhancing TCP's Loss Recovery Using Limited Transmit", RFC 3042, DOI 10.17487/RFC3042, January 2001, <<http://www.rfc-editor.org/info/rfc3042>>.
- [RFC3522] Ludwig, R. and M. Meyer, "The Eifel Detection Algorithm for TCP", RFC 3522, DOI 10.17487/RFC3522, April 2003, <<http://www.rfc-editor.org/info/rfc3522>>.
- [RFC3708] Blanton, E. and M. Allman, "Using TCP Duplicate Selective Acknowledgement (DSACKs) and Stream Control Transmission Protocol (SCTP) Duplicate Transmission Sequence Numbers (TSNs) to Detect Spurious Retransmissions", RFC 3708, DOI 10.17487/RFC3708, February 2004, <<http://www.rfc-editor.org/info/rfc3708>>.
- [RFC4015] Ludwig, R. and A. Gurtov, "The Eifel Response Algorithm for TCP", RFC 4015, DOI 10.17487/RFC4015, February 2005, <<http://www.rfc-editor.org/info/rfc4015>>.
- [RFC4960] Stewart, R., Ed., "Stream Control Transmission Protocol", RFC 4960, DOI 10.17487/RFC4960, September 2007, <<http://www.rfc-editor.org/info/rfc4960>>.
- [RFC5681] Allman, M., Paxson, V., and E. Blanton, "TCP Congestion Control", RFC 5681, DOI 10.17487/RFC5681, September 2009, <<http://www.rfc-editor.org/info/rfc5681>>.
- [RFC5682] Sarolahti, P., Kojo, M., Yamamoto, K., and M. Hata, "Forward RT0-Recovery (F-RT0): An Algorithm for Detecting Spurious Retransmission Timeouts with TCP", RFC 5682, DOI 10.17487/RFC5682, September 2009, <<http://www.rfc-editor.org/info/rfc5682>>.
- [RFC5827] Allman, M., Avrachenkov, K., Ayesta, U., Blanton, J., and P. Hurtig, "Early Retransmit for TCP and Stream Control Transmission Protocol (SCTP)", RFC 5827, DOI 10.17487/RFC5827, May 2010, <<http://www.rfc-editor.org/info/rfc5827>>.
- [RFC6298] Paxson, V., Allman, M., Chu, J., and M. Sargent, "Computing TCP's Retransmission Timer", RFC 6298, DOI 10.17487/RFC6298, June 2011, <<http://www.rfc-editor.org/info/rfc6298>>.

## 9.2. Informative References

- [EL04] Ekstroem, H. and R. Ludwig, "The Peak-Hopper: A New End-to-End Retransmission Timer for Reliable Unicast Transport", IEEE INFOCOM 2004, DOI 10.1109/INFCOM.2004.1354671, March 2004.
- [FDT13] Flach, T., Dukkkipati, N., Terzis, A., Raghavan, B., Cardwell, N., Cheng, Y., Jain, A., Hao, S., Katz-Bassett, E., and R. Govindan, "Reducing Web Latency: the Virtue of Gentle Aggression", Proc. ACM SIGCOMM Conf., DOI 10.1145/2486001.2486014, August 2013.
- [HB11] Hurtig, P. and A. Brunstrom, "SCTP: designed for timely message delivery?", Springer Telecommunication Systems 47 (3-4), DOI 10.1007/s11235-010-9321-3, August 2011.
- [IEEE.9945] IEEE/ISO/IEC, "International Standard - Information technology Portable Operating System Interface (POSIX) Base Specifications, Issue 7", IEEE 9945-2009, <<http://standards.ieee.org/findstds/standard/9945-2009.html>>.
- [LS00] Ludwig, R. and K. Sklower, "The Eifel retransmission timer", ACM SIGCOMM Comput. Commun. Rev., 30(3), DOI 10.1145/382179.383014, July 2000.
- [P09] Petlund, A., "Improving latency for interactive, thin-stream applications over reliable transport", Unipub PhD Thesis, Oct 2009.
- [PBP09] Petlund, A., Beskow, P., Pedersen, J., Paaby, E., Griwodz, C., and P. Halvorsen, "Improving SCTP retransmission delays for time-dependent thin streams", Springer Multimedia Tools and Applications, 45(1-3), DOI 10.1007/s11042-009-0286-8, October 2009.
- [PGH06] Pedersen, J., Griwodz, C., and P. Halvorsen, "Considerations of SCTP Retransmission Delays for Thin Streams", IEEE LCN 2006, DOI 10.1109/LCN.2006.322082, November 2006.
- [RFC6458] Stewart, R., Tuexen, M., Poon, K., Lei, P., and V. Yasevich, "Sockets API Extensions for the Stream Control Transmission Protocol (SCTP)", RFC 6458, DOI 10.17487/RFC6458, December 2011, <<http://www.rfc-editor.org/info/rfc6458>>.

- [RHB15] Rajiullah, M., Hurtig, P., Brunstrom, A., Petlund, A., and M. Welzl, "An Evaluation of Tail Loss Recovery Mechanisms for TCP", ACM SIGCOMM CCR 45 (1), DOI 10.1145/2717646.2717648, January 2015.
- [RJ10] Ramachandran, S., "Web metrics: Size and number of resources", May 2010, <<https://goo.gl/0a6Q9A>>.
- [TLP] Dukkipati, N., Cardwell, N., Cheng, Y., and M. Mathis, "Tail Loss Probe (TLP): An Algorithm for Fast Recovery of Tail Losses", Work in Progress, draft-dukkipati-tcpm-tcp-loss-probe-01, February 2013.

## Acknowledgements

The authors wish to thank Michael Tuexen for contributing the SCTP Socket API considerations and Godred Fairhurst, Yuchung Cheng, Mark Allman, Anantha Ramaiah, Richard Scheffenegger, Nicolas Kuhn, Alexander Zimmermann, and Michael Scharf for commenting on the document and the ideas behind it.

All the authors are supported by RITE (<http://riteproject.eu/>), a research project (ICT-317700) funded by the European Community under its Seventh Framework Program. The views expressed here are those of the author(s) only. The European Commission is not liable for any use that may be made of the information in this document.

**Authors' Addresses**

Per Hurtig  
Karlstad University  
Universitetsgatan 2  
Karlstad 651 88  
Sweden

Phone: +46 54 700 23 35  
Email: per.hurtig@kau.se

Anna Brunstrom  
Karlstad University  
Universitetsgatan 2  
Karlstad 651 88  
Sweden

Phone: +46 54 700 17 95  
Email: anna.brunstrom@kau.se

Andreas Petlund  
Simula Research Laboratory AS  
P.O. Box 134  
Lysaker 1325  
Norway

Phone: +47 67 82 82 00  
Email: apetlund@simula.no

Michael Welzl  
University of Oslo  
PO Box 1080 Blindern  
Oslo N-0316  
Norway

Phone: +47 22 85 24 20  
Email: michawe@ifi.uio.no