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Telephony Signalling Transport over SCTP applicability statement

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Status of this Memo

Abstract

This document describes the applicability of the Stream Control Transmission Protocol (SCTP)[RFC2960] for transport of telephony signalling information over IP infrastructure. Special considerations for using SCTP to meet the requirements of transporting telephony signalling [RFC2719] are discussed.

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1 INTRODUCTION

Transport of telephony signalling requires special considerations. In order to use SCTP, special care must be taken to meet the performance, timing and failure management requirements.

1.1 Terminology

The following terms are commonly identified in related work:

Association: SCTP connection between two endpoints.

Stream: A uni-directional logical channel established within an

association, within which all user messages are delivered in sequence except for those submitted to the unordered delivery service.

1.2 Overview

SCTP provides a general purpose, reliable transport between two endpoints.

The following functions are provided by SCTP:

- Reliable Data Transfer

- Multiple streams to help avoid head-of-line blocking

- Ordered and unordered data delivery on a per-stream basis

- Bundling and fragmentation of user data

- Congestion and flow control

- Support continuous monitoring of reachability

- Graceful termination of association

- Support of multi-homing for added reliability

- Protection against blind denial-of-service attacks

- Protection against blind masquerade attacks

Telephony Signalling transport over IP normally uses the following architecture:

```
                     Telephony Application
                               |
              +------------------------------------+
              |   Signalling Adaptation module     |
              +------------------------------------+
                               |
              +------------------------------------+
              |Stream Control Transmission Protocol|
              |              (SCTP)                |
              +------------------------------------+
                               |
                Internet Protocol (IPv4/IPv6)
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Figure 1.1: Telephony signalling transport protocol stack

The components of the protocol stack are :

(1) Adaptation modules are used when the telephony application needs to preserve an existing primitive interface. (e.g. management indications, data operation primitives, ... for a particular user/application protocol).

(2) SCTP, specially configured to meet the telephony application performance requirements.

(3) The standard Internet Protocol.

2 Applicability of Telephony Signalling transport using SCTP

SCTP can be used as the transport protocol for telephony applications. Message boundaries are preserved during data transport and so no message delineation is needed. The user data can be delivered by the order of transmission within a stream(in sequence delivery) or the order of arrival.

SCTP can be used to provide redundancy and fault tolerance at the transport layer and below. Telephony applications needing this level of fault tolerance can make use of SCTP's multi-homing support.

SCTP can be used for telephony applications where head-of-line blocking is a concern. Such an application should use multiple streams to provide

independent ordering of telephony signalling messages.

3 Issues for transporting telephony signalling over SCTP

3.1 Congestion Control

The basic mechanism of congestion control in SCTP have been described in [RFC2960]. SCTP congestion control sometimes conflicts with the timing requirements of telephony signalling transport.

In an engineered network (e.g. a private intranet), in which network capacity and maximum traffic is very well understood, some telephony signalling applications may choose to relax the congestion control rules in order to satisfy the timing requirements. But this should be done without destabilising the network, otherwise this would lead to potential congestion collapse of the network.

Some telephony signalling applications may have their own congestion control and flow control techniques. These techniques may interact with the congestion control procedures in SCTP. Additionally, telephony applications may use SCTP stream based flow control [SCTPFLOW].

3.2 Detection of failures

Telephony systems often must achieve high availability in operation. For example, they are often required to be able to preserve stable calls during a component failure. Therefore error situations at the transport layer and below must be detected very fast so that the application can take approriate steps to recover and preserve the stable calls. This poses special requirements on SCTP to discover unreachablility of a destination address or a peer.

3.2.1 Retransmission TimeOut (RTO) calculation

The SCTP protocol parameter RTO.Min value has a direct impact on the calculation of the RTO itself. Some telephony applications want to lower the value of the RTO.Min to less than 1 second. This would allow the message sender to reach the maximum number-of-retransmission threshold faster in the case of network failures. However, lowering RTO.Min may have a negative impact on network behaviour [ALLMAN99].

In some rare cases, telephony applications might not want to use the exponential timer back-off concept in RTO calculation in order to speed

up failure detection. The danger of doing this is that, when network congestion occurs, not backing off the timer may worsen the congestion situation. Therefore, this strategy should never be used in public Internet.

It should be noted that not using delayed SACK will also help faster failure detection.

3.2.2 Heartbeat

For faster detection of (un)availability of idle paths, the telephony application may consider lowering the SCTP parameter HB.interval. It should be noted this will result in a higher traffic load.

3.2.3 Maximum number of retransmissions

Setting Path.Max.Retrans and Association.Max.Retrans SCTP parameters to lower values will speed up both destination address and peer failure detection. However, if these values are set too low, the probability of false detections will increase.

3.3 Shorten end-to-end message delay

Telephony applications often require short end-to-end message delays. The methods described in section 3.2.1 on lowering RTO and not using delayed SACK may be considered.

3.4 Bundling considerations

Bundling small telephony signalling messages at transmission helps improve the bandwidth usage efficiency of the network. On the downside, bundling may introduce additional delay to some of the messages. This should be taken into consideration when end-to-end delay is a concern.

3.5 Stream Usage

Telephony signalling traffic is often composed of multiple, independent message sequences. It is highly desirable to transfer those independent message sequences in separate SCTP streams. This reduces the probability of head-of-line blocking in which the retransmission of a lost message affects the delivery of other messages not belonging to the same message sequence.

4 Security considerations

SCTP only tries to increase the availability of a network. SCTP does not contain any protocol mechanisms which are directly related to user message authentication, integrity and confidentiality functions. For such features, it depends on the IPSEC protocols and architecture and/or on security features of its user protocols.

Mechanisms for reducing the risk of blind denial-of-service attacks and masquerade attacks are built into SCTP protocol. See RFC2960, section 11 for detailed information.

Currently the IPSEC working group is investigating the support of multihoming by IPSEC protocols. At the present time to use IPSEC, one must use 2 * N * M security associations if one endpoint uses N addresses and the other M addresses.

5 References and related work

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