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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 new protocols developed under the signalling transport framework[RFC2719]. A description of the main issues regarding the use of the Stream Control Transmission Protocol (SCTP)[RFC2960] and each adaptation layer for transport of telephony signalling information over IP infrastructure is explained.

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1 INTRODUCTION

This document intends to inform how to transport telephony signalling protocols, used in classic telephony systems, over IP networks. The whole architecture is called SIGTRAN (Signalling Transport) as described in RFC2719 and is composed of a transport protocol(SCTP) and several User Adaptation layers(UAL). The transport protocol SCTP has been been developed to fulfill the stringent requirements that telephony signalling networks have. The set of User Adaptation layers have also been introduced to make it possible that different signalling protocols can use the SCTP layer.

1.1 Scope

The scope of this document is to explain the way that user adaptation layers and SCTP protocols have to be used to transport Telephony signalling information over IP.

1.2 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.

SPU: Signalling protocol user, the application on top of the User adaptation layer.

CTSP: Classical Telephony Signalling protocol(examples: MTP level2, MTP level 3, SCCP....).

UAL: User adaptation layer: the protocol that encapsulate the upper layer telephony signalling protocols that are to be transported over SCTP/IP.

ISEP: IP signalling endpoint: a IP node that implements SCTP and a User adapatation layer.

SP: signalling point

1.3 Contributors

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2 SIGTRAN architecture

The SIGTRAN architecture describes the transport of signalling information over IP infrastructure.

Telephony Signalling transport over IP normally uses the following architecture:

```
             Telephony Signalling Application
                          |
         +-----------------------------------+
         |   Signalling Adaptation Layers    |
         +-----------------------------------+
                          |
         +-----------------------------------+
         |Stream Control Transmission Protocol|
         |             (SCTP)                 |
         +-----------------------------------+
                          |
           Internet Protocol (IPv4/IPv6)
```

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.

The telephony signalling protocols to be transported can be:

- SS7 MTP3 users: SCCP, ISUP, TUP...

- SS7 MTP2 users: MTP3

- SS7 SCCP users: RANAP, MAP(+TCAP), INAP(+TCAP)...

- ISDN Q.921 users: Q.931

- V5.2/DSS1

- ....

Every classic telephony protocol can have a corresponding UAL developed.

The user adaptation layers(UALs) are a set of protocols that encapsulate a specific signalling protocol to be transported over SCTP. The adapation is done in a way that the upper signalling protocols that are relayed remain unaware that the lower layers are different to the originail lower telephony signalling layers. In that sense, the upper interface of the user adapatation layers need to be the same as the upper layer interface to its original lower layer. If a MTP user is being relayed over the IP network, the related UAL used to transport the MTP user will have the same upper interface as MTP has.

The Stream Control Transmission protocol was designed to fulfill the stringent transport requirements that classical signalling protocols have and is therefore the recommended transport protocol to use for this purpose.

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

SCTP is used as the transport protocol for telephony signalling applications. Message boundaries are preserved during data transport by SCTP and so each UAL can specify its own message structure within the SCTP user data. The SCTP user data can be delivered by the order of transmission within a stream(in sequence delivery) or unordered.

SCTP can be used to provide redundancy at the transport layer and below. Telephony applications needing this level of redundancy 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

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.

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 application messages which are transported by SCTP. During congestion, messages may be delayed by SCTP, thus sometimes violating the timing requirements of those telephony applications.

In an engineered network (e.g. a private intranet), in which network capacity and maximum traffic are very well understood, some telephony signalling applications may choose to relax the congestion control rules of SCTP in order to satisfy the timing requirements. In order to do this, they should employ their own congestion control mechanisms. 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.

3.2 Detection of failures

Telephony systems often must have no single point of failure in operation.

The UAL must meet certain service availability and performance requirements according to the classical signalling layers they are replacing. Those requirements may be specific for each UAL.

For example, telephony systems 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 quickly so that the UAL can take approriate steps to recover and preserve the 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 might 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 fault detections might increase.

3.3 Shorten end-to-end message delay

Telephony applications often require short end-to-end message delays. The method described in section 3.2.1 on lowering RTO may be considered. The different paths within a single association will have a different RTO, so using the path with the lowest RTO will lead to a shorter end-to-end message delay for the application running on top of the UAL's.

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. User Adaptation Layers

Users Adaptation Layers (UALs) are defined to encapsulate different signalling protocols in order to transport them over SCTP/IP

There are UALs for both access signalling (DSS1) and trunk signalling (SS7). A brief description of the standardized UALs follows in the next sub-sections.

The delivery mechanism in the several UALs

- Supports seamless operation of UALs user peers over an IP network connection.

- Supports the interface boundary that the UAL user had with the traditional lower layer.

- Supports management of SCTP transport associations and traffic between SGs and ISEPs or two ISEPs

- Supports asynchronous reporting of status changes to management.

Signalling User Adaptation Layers have been developed for both: Access and Trunk Telephony Signalling. They are defined as follows.

Access Signalling: This is the signalling that is needed between and access device and an exchange in the core network in order to establish, manage or release the voice or data call paths. There are several protocols that have been developed for this purpose.

Trunk Signalling: This is the signalling that is used between the exchanges inside the core network in order to establish, manage or release the voice or data call paths. The most common protocols used for this purpose are known as the SS7 system that belongs to the Common Channel Signalling (CCS) philosophy. The SS7 protocol stack is depicted below:

```
+------+-----+-------+- -+-------+------+-----+------+
|      |     |       |   |       |  MAP | CAP | INAP |
+      |     + RANAP |...| BSSAP +--------------------+
| ISUP | TUP |       |   |       |       TCAP         |
+      |     +--------------------------------------+
|      |     |                  SCCP                 |
+---------------------------------------------------+
|                           MTP3                     |
+---------------------------------------------------+
|                           MTP2                     |
+---------------------------------------------------+
|                           MTP1                     |
+---------------------------------------------------+
```

The Telephony Signalling Protocols to be transported with the already designed UALS are:

- ISDN Q.921 Users: Q.931
- V5.2/DSS1
- DPNSS/DASS2
- SS7 MTP3 Users: SCCP, ISUP, TUP
- SS7 MTP2 Users: MTP3
- SS7 SCCP Users: TCAP, RANAP, BSSAP, ...

Two main scenarios have been developed to use the different UALS for

IP Signalling Transport:

(1) Intercommunication of traditional Signalling transport nodes and IP based nodes.

```
             Traditional               Telephony
              Telephony                Signalling
*********     Signalling   **********   over IP    ********
*  SEP  *----------------*   SG   *--------------* ISEP *
*********                  **********              ********

+-------+                                          +-------+
|SigProt|                                          |SigProt|
+-------+                  +----+----+             +-------+
|       |                  |    |UAL |             |  UAL  |
|       |                  |    +----+             +-------+
| TTST  |                  |TTST|SCTP|             | SCTP  |
|       |                  |    +----+             +-------+
|       |                  |    | IP |             |  IP   |
+-------+                  +---------+             +-------+

      SEP     -   Signalling Endpoint
      SG      -   Signalling Gateway
      ISEP    -   IP Signalling Endpoint
      SigProt -   Signalling Protocol
      TTSP    -   Traditional Telephony Signalling Protocol
      UAL     -   User Adaptation Layer
      SCTP    -   Stream Control Transport Protocol
```

It is also referred as SG to AS communication. AS is the name that UAL usually gives to the ISEP nodes. It stands for Application Server.

(2) Communication inside the IP network.

```
              Telephony
              Signalling
*********      over IP      *********
* ISEP  *------------------*  ISEP *
*********                   *********

+-------+                   +-------+
|SigProt|                   |SigProt|
+-------+                   +-------+
|  UAL  |                   |  UAL  |
+-------+                   +-------+
| SCTP  |                   | SCTP  |
```

```
                     +-------+                          +-------+
                     |  IP   |                          |  IP   |
                     +-------+                          +-------+
```

This is also referred to as IPSP communication. IPSP stands for IP Signalling Point and describes the role that the UAL plays on a IP-based node.

The first scenario is applied for both types of signalling (access and trunk signalling). On the other hand the peer to peer basis can only be used for trunk signalling.

4.1 Access Signalling

The SIGTRAN WG have developed UALs to transport the following Access Signalling protocols:

- ISDN Q.931
- V5.2
- DPNSS/DASS2

4.1.1 ISDN Q.931 over IP

UAL: IUA (ISDN Q.921 User Adaptation)

This document supports both ISDN Primary Rate Access (PRA) as well as Basic Rate Access (BRA) including the support for both point-to-point and point-to-multipoint modes of communication. This support includes Facility Associated Signalling (FAS), Non-Facility Associated Signalling (NFAS) and NFAS with backup D channel.

It implements the client/server architecture. The default orientation is for the SG to take on the role of server while the ISEP is the client. The SCTP (and UDP/TCP) Registered User Port Number Assignment for IUA is 9900.

Examples of the upper layers to be transported are Q.931 and QSIG.

The main scenario supported by this UAL is the SG to ISEP communication where the ISEP role is typically played by a node called an MGC, as defined in [RFC2719].

```
******   ISDN        ******      IP      *******
*PBX *---------------* SG *--------------* MGC *
******               ******              *******

+-----+                                  +-----+
|Q.931|             (NIF)                |Q.931|
+-----+          +----------+            +-----+
|     |          |     | IUA|            | IUA |
|     |          |     +----+            +-----+
```

```
            |Q.921|           |Q.921|SCTP|           |SCTP |
            |     |           |     +----+           +-----+
            |     |           |     | IP |           | IP  |
            +-----+           +-----+----+           +-----+
```

```
                NIF  - Nodal Interworking Function
                PBX  - Private Branch Exchange
                SCTP - Stream Control Transmission Protocol
                IUA  - ISDN User Adaptation Layer Protocol
```

The SCTP (and UDP/TCP) Registered User Port Number Assignment for IUA is 9900.

The value assigned by IANA for the Payload Protocol Identifier in the SCTP Payload Data chunk is "1".

4.1.2 V5UA over IP

UAL: V5UA (V5.2-User Adaptation)

It is an extension from the IUA layer with the modifications needed to support the differences between Q.921 / Q.931, and V5.2 layer 2 / layer 3. It supports analog telephone access, ISDN basic rate access and ISDN primary rate access over a V5.2 interface. It is typically implemented in an interworking scenario with SG.

```
        ******   V5.2        ******      IP      *******
        * AN *---------------* SG *--------------* MGC *
        ******               ******              *******


        +-----+                                  +-----+
        |V5.2 |              (NIF)               |V5.2 |
        +-----+           +-----------+          +-----+
        |     |           |     |V5UA|           |V5UA |
        |     |           |     +----+           +-----+
        |LAPV5|           |LAPV5|SCTP|           |SCTP |
        |     |           |     +----+           +-----+
        |     |           |     | IP +           | IP  |
        +-----+           +-----+----+           +-----+


        AN    - Access Network
        NIF   - Nodal Interworking Function
        LAPV5 - Link Access Protocol for the V5 channel
        SCTP  - Stream Control Transmission Protocol
```

The SCTP (and UDP/TCP) Registered User Port Number Assignment for V5UA is 5675.

The value assigned by IANA for the Payload Protocol Identifier in the SCTP Payload Data chunk is "6".

4.1.3 DPNSS/DASS2 over IP

UAL: DUA (DPNSS/DASS2 User Adaptation)

The DUA is built on top of IUA and defines the necessary extensions to IUA for a DPNSS/DASS2 transport. DPNSS stands for Digital Private Network Signalling System and DASS2 for Digital Access Signalling System No 2.

```
******   DPNSS       ******      IP      *******
*PBX *---------------* SG *--------------* MGC *
******               ******              *******

+-----+                                  +-----+
|DPNSS|              (NIF)               |DPNSS|
| L3  |                                  | L3  |
+-----+           +-----+----+           +-----+
|     |           |     | DUA|           | DUA |
|DPNSS|           |DPNSS+----+           +-----+
| L2  |           | L2  |SCTP|           |SCTP |
|     |           |     +----+           +-----+
|     |           |     | IP +           | IP  |
+-----+           +-----+----+           +-----+
```

PBX - Private Branch eXchange
NIF - Nodal Interworking function
SCTP - Stream Control Transmission Protocol
DUA - DPNSS User Adaptation Layer Protocol

The value assigned by IANA for the Payload Protocol Identifier in the SCTP Payload Data chunk is "10".

4.2 Network Signalling

The SIGTRAN WG have developed UALs to transport the following SS7 protocols:

- MTP2 Users: MTP3
- MTP3 Users: ISUP, TUP, SCCP
- SCCP Users: TCAP, RNSAP, RANAP, BSSAP, ...

4.2.1   MTP lvl3 over IP

UALs:

- M2UA (SS7 MTP2 User Adaptation)
- M2PA (SS7 MTP2-User Peer-to-Peer Adaptation)

4.2.1.1   M2UA (SS7 MTP2 User Adaptation)

M2UA protocol is typically used between a Signalling Gateway (SG) and Media Gateway Controller (MGC). The SG will terminate up to MTP Level 2 and the MGC will terminate MTP Level 3 and above. In other words, the SG will transport MTP Level 3 messages over an IP network to a MGC.

MTP3 and MTP3b are the the only SS7 MTP2 User protocols that is transported by this UAL.

The SG provides a interworking of transport functions with the IP transport to transfer MTP2-User signalling messages with an Application Server (e.g. MGC) where the peer MTP2-User exists.

```
******    SS7    ******      IP     *******
*SEP *-----------* SG *-------------* MGC *
******           ******             *******

+----+                              +----+
|S7UP|                              |S7UP|
+----+                              +----+
|MTP3|                              |MTP3|
|    |           (NIF)              |    |
+----+         +----+----+          +----+
|    |         |    |M2UA|          |M2UA|
|    |         |    +----+          +----+
|MTP2|         |MTP2|SCTP|          |SCTP|
```

```
              |    |         |    +----+         +----+
              |    |         |    |IP  |         |IP  |
              +----+         +---------+         +----+
```

MGC - Media Gateway Controler
SG   - Signalling Gateway
SEP - SS7 Signalling Endpoint
NIF - Nodal Interworking Function
IP   - Internet Protocol
SCTP - Stream Control Transmission Protocol

The SCTP (and UDP/TCP) Registered User Port Number Assignment for M2UA is 2904.

The value assigned by IANA for the Payload Protocol Identifier in the SCTP Payload Data chunk is "2".

4.2.1.2 M2PA (SS7 MTP2-User Peer-to-Peer Adaptation) Layer

M2PA protocol is used between SS7 Signalling Points employing the MTP Level 3 protocol. The SS7 Signalling Points may also use standard SS7 links using the SS7 MTP Level 2 to provide transport of MTP Level 3 signalling messages.

Both configurations: communication of SS7 and IP with SG and communication between ISEPs are possible.

Communication between two IP nodes:

```
********   IP   ********
* IPSP *--------* IPSP *
********        ********

+------+        +------+
| TCAP |        | TCAP |
+------+        +------+
| SCCP |        | SCCP |
+------+        +------+
| MTP3 |        | MTP3 |
+------+        +------+
| M2PA |        | M2PA |
+------+        +------+
| SCTP |        | SCTP |
+------+        +------+
|  IP  |        |  IP  |
+------+        +------+
```

IP   - Internet Protocol
IPSP - IP Signalling Point

Connection of SS7 and IP nodes:

```
        ********  SS7   ***************   IP   ********
        * SEP  *--------*     SG      *--------* IPSP *
        ********        ***************        ********

        +------+                               +------+
        | TCAP |                               | TCAP |
        +------+                               +------+
        | SCCP |                               | SCCP |
        +------+        +-------------+        +------+
        | MTP3 |        |    MTP3     |        | MTP3 |
        +------+        +------+------+        +------+
        |      |        |      | M2PA |        | M2PA |
        |      |        |      +------+        +------+
        | MTP2 |        | MTP2 | SCTP |        | SCTP |
        |      |        |      +------+        +------+
        |      |        |      | IP   |        | IP   |
        +------+        +------+------+        +------+

                SEP   - SS7 Signalling Endpoint
```

These figures are only an example. Other configurations are possible. For example, IPSPs without traditional SS7 links could use the protocol layers MTP3/M2PA/SCTP/IP to route SS7 messages in a network with all IP links.

Another example is that two SGs could be connected over an IP network to form an SG mated pair similar to the way STPs are provisioned in traditional SS7 networks.

The SCTP (and UDP/TCP) Registered User Port Number Assignment for M2PA is 3565.

The value assigned by IANA for the Payload Protocol Identifier in the SCTP Payload Data chunk is "5".

4.2.1.3 Main differences between M2PA and M2UA:

a. M2PA: IPSP processes MTP3/MTP2 primitives.
   M2UA: MGC transports MTP3/MTP2 primitives between the SG's MTP2
         and the MGC's MTP3 (via the NIF) for processing.

b. M2PA: SG-IPSP connection is an SS7 link.
   M2UA: SG-MGC connection is not an SS7 link. It is an
         extension of MTP to a remote entity.

4.3 MTP lvl3-Users (ISUP, TUP, SCCP) over IP

UAL: M3UA (SS7 MTP3 User Adaptation)

M3UA protocol supports the transport of any SS7 MTP3-User signalling such as TUP, ISUP and SCCP over IP using the services of SCTP.

Interconnection of SS7 and IP nodes:

```
            ********   SS7   *****************   IP   ********
            * SEP  *---------*      SGP      *--------* ASP  *
            ********         *****************        ********

            +------+         +---------------+        +------+
            | ISUP |         |     (NIF)     |        | ISUP |
            +------+         +------+ +------+        +------+
            | MTP3 |         | MTP3 | | M3UA |        | M3UA |
            +------|         +------+-+------+        +------+
            | MTP2 |         | MTP2 | | SCTP |        | SCTP |
            +------+         +------+ +------+        +------+
            |  L1  |         |  L1  | |  IP  |        |  IP  |
            +------+         +------+ +------+        +------+

                SEP  - SS7 Signalling End Point
                SCTP - Stream Control Transmission Protocol
                NIF  - Nodal Interworking Function
```

Communication between two IP nodes:

```
********    IP    ********
* IPSP *----------* IPSP *
********          ********

+------+          +------+
|SCCP- |          |SCCP- |
| User |          | User |
+------+          +------+
| SCCP |          | SCCP |
+------+          +------+
| M3UA |          | M3UA |
+------+          +------+
| SCTP |          | SCTP |
+------+          +------+
|  IP  |          |  IP  |
+------+          +------+
```

M3UA uses a client-server architecture. It is recommended that the ISEP acts as the client and initiate the SCTP assocaitions with the SG. The port reserved by IANA is 2905. This is the port upon which the SG should listen for possible client connections.

The assigned payload protocol identifier for the SCTP DATA chunks is "3".

4.4 SCCP-Users over IP

UAL: SUA (SS7 SCCP User Adaptation)

SUA protocol supports the transport of any SS7 SCCP-User signalling such as MAP, INAP, SMS, BSSAP, RANAP over IP using the services of SCTP. Each of the applications using SUA have their own set of timing requirements that can be found in their respective standards documents.

Possible configurations are showed in the pictures below.

- Interconnection of SS7 and IP:

```
        ********         ***************        ********
        * SEP  *    IP   *             *   IP   *      *
        *  or  *---------*     SG      *--------* ASP  *
        * STP  *         *             *        *      *
        ********         ***************        ********

        +------                                 +------+
        | SUAP |                                | SUAP |
        +------+         +------+------+        +------+
        | SCCP |         | SCCP | SUA  |        | SUA  |
        +------+         +------+------+        +------+
        |      |         |      |      |        |      |
        | MTP3 |         | MTP3 | SCTP |        | SCTP |
        |      |         |      |      |        |      |
        +------+         +------+------+        +------+
        | MTP2 |         | MTP2 |  IP  |        |  IP  |
        +------+         +------+------+        +------+

          SUAP - SCCP/SUA User Protocol (TCAP, for example)
          STP  - SS7 Signalling Transfer Point
```

- IP Node to IP Node communication:

```
                      ********        ********
                      *      *   IP   *      *
                      * IPSP *--------* IPSP *
                      *      *        *      *
                      ********        ********

                      +------+        +------+
                      | SUAP |        | SUAP |
                      +------+        +------+
                      | SUA  |        | SUA  |
                      +------+        +------+
                      | SCTP |        | SCTP |
                      +------+        +------+
                      |  IP  |        |  IP  |
                      +------+        +------+
```

IANA has registered SCTP Port Number 14001 for SUA. It is recommended that SGs use this SCTP port number for listening for new connections. The payload protocol identifier for the SCTP DATA chunks is "4".

5 Security considerations

UALs are designated to carry signalling messages for telephony services. As such, UALs must involve the security needs of several parties: the end users of the services; the network providers and the applications involved. Additional requirements may come from local regulation. While having some overlapping security needs, any security solution should fulfill all of the different parties' needs. See specific Security considerations in each UAL technical specification.

SCTP only tries to increase the availability of a network. SCTP does not contain any protocol mechanisms which are directly related to communication security, i.e. user message authentication, integrity or confidentiality functions. For such features, it depends on security protocols. In the field of system security, SCTP includes mechanisms for reducing the risk of blind denial-of-service attacks as it is described in section 11 in RFC2960.

This document does not add any new components to the protocols included in the discussion. For secure use of the SIGTRAN protocols the readers should go through the "Security Considerations for SIGTRAN protocols" [RFCSIGSEC]). According to that document, the use of the IPsec is the main recommendation to secure SIGTRAN protocols in the Internet, but TLS is also considered as a perfectly valid option to be used in certain scenarios. Recomendations of usage are also included.

6 References and related work

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