INTERNET-DRAFT
Internet Engineering Task Force
Issued: February 2002
Expires: August 2002

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Telephony Signalling Transport over SCTP applicability statement <draft-ietf-sigtran-signalling-over-sctp-applic-04.txt>

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 Contributors

The following people contributed to the document: L. Coene(Editor), M. Tuexen, G. Verwimp, J. Loughney, R.R. Stewart, Qiaobing Xie, M. Holdrege, M.C. Belinchon, A. Jungmaier, J. Pastor and L. Ong.

1.3 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:

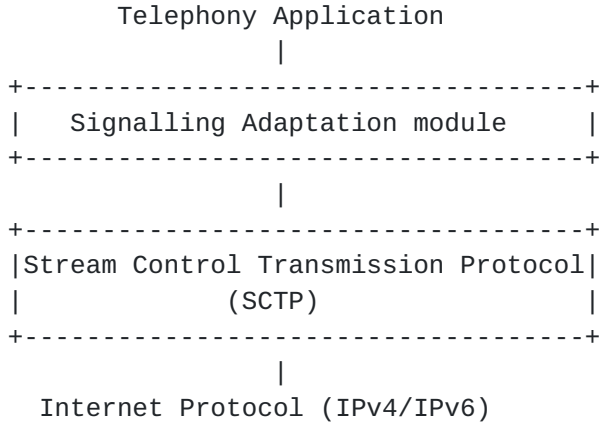


Figure 1.1: Telephony signalling transport protocol stack

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 User Adaptation Layers

Users Adaptation Layers are defined to substitute the telephony signaling protocol that is below of the telephony signaling protocol to be relayed.

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

The delivery mechanism in the several UALs

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

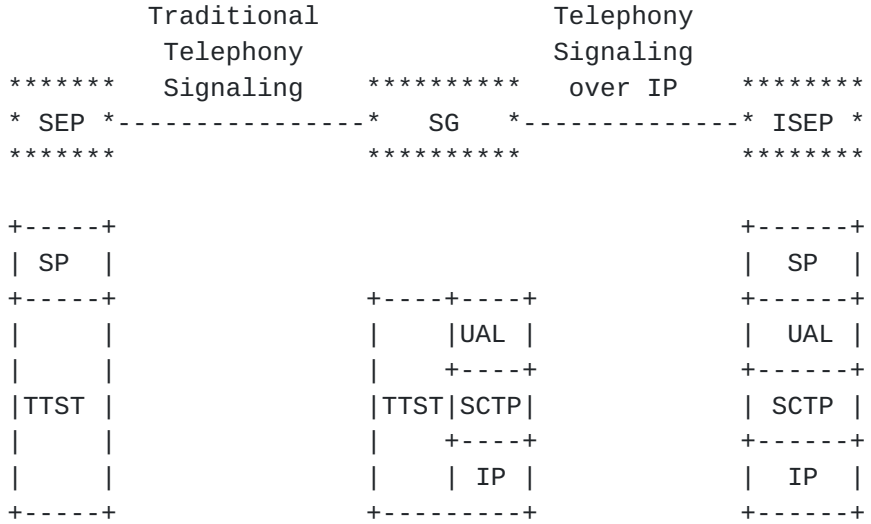
- Support the interface boundary that the UAL user had with the traditional lower layer.

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

- Support asynchronous reporting of status changes to management.

Two main scenarios have been developed for Signaling Transport:

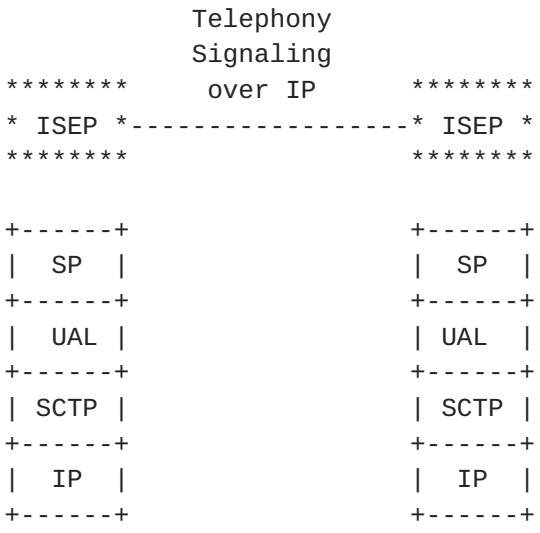
- Intercommunication of traditional Signaling transport nodes and IP based nodes.



SEP: Signaling Endpoint
SG: Signaling Gateway
ISEP: IP Signaling Endpoint
SP: Signaling Protocol
TTST: Traditional Telephony Signaling Transport
UAL: User Adaptation Layer

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

- Communication inside the IP networks.



It is also referred as IPSP communication. IPSP is the name of the role that an IP-based node plays UAL usually gives to the ISEP nodes. It stands for IP Signaling Point.

4.1 IUA (ISDN Q.921 User Adaptation)

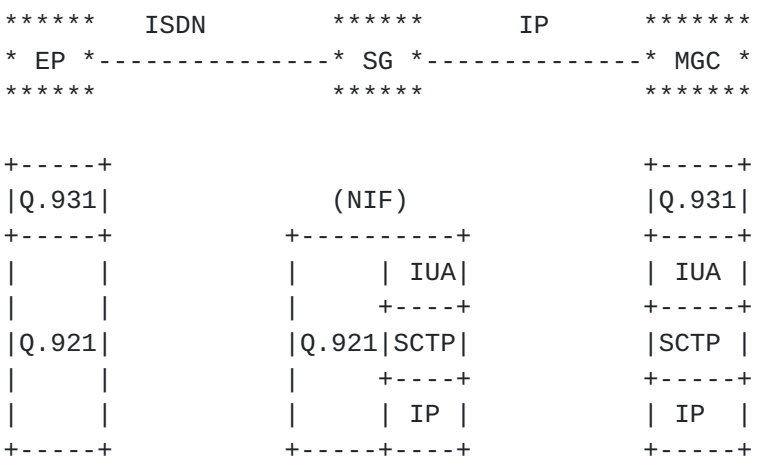
This document supports both ISDN Primary Rate Access (PRA) as well as Basic Rate Access (BRA) including the support for both point-to-point and point-to-multipoint modes of communication. This support includes Facility Associated Signaling (FAS), Non-Facility Associated Signaling (NFAS) and NFAS with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

It implements the client/server architecture. The default orientation would be 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 would be 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 MGC defined in [RFC2719].



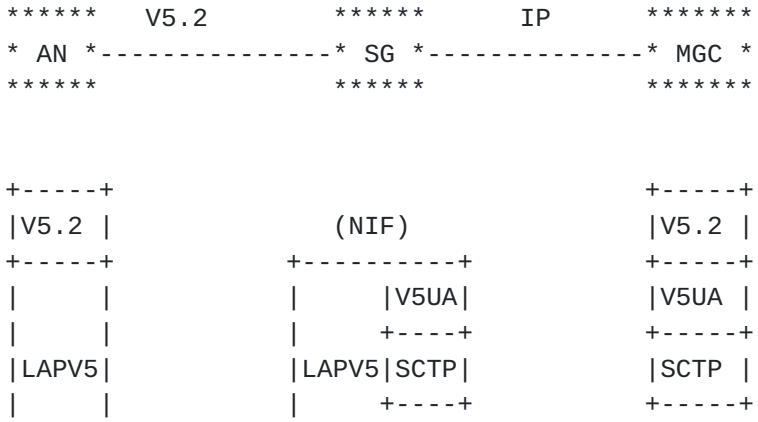
NIF - Nodal Interworking Function
EP - ISDN End Point
SCTP - Stream Control Transmission Protocol
IUA - ISDN User Adaptation Layer Protocol

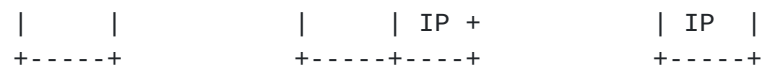
The SCTP (and UDP/TCP) Registered User Port Number Assignment for IUA is 9900.

The value assigned by IANA for the Payload Protocol Identifier in the SCTP Payload Data chunk is ô1ö

4.2 V5UA (V5.2-User Adaptation) Layer

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 basically implemented in an interworking scenario with SG.




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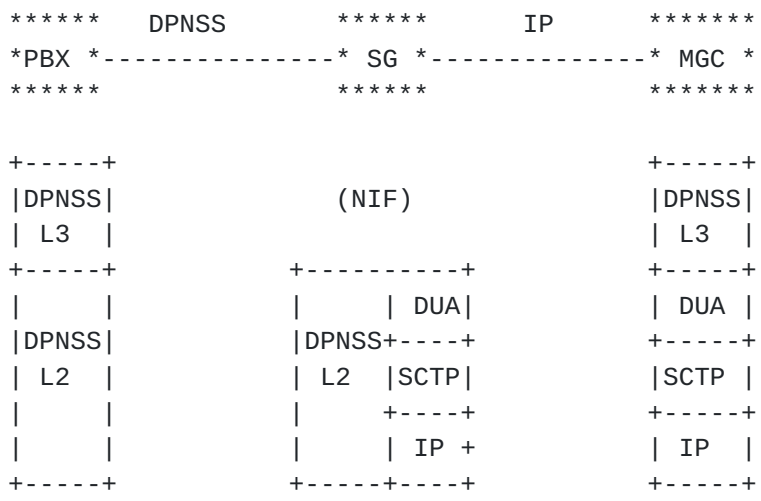
AN û Access Network
NIF û Nodal Interworking Function
LAPV5 û Link Access Protocol for the V5 channel
SCTP - Stream Control Transmission Protocol

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 ô6ö

4.3 DUA (DPNSS/DASS 2 User Adaptation) Layer

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



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 ôTBDö.

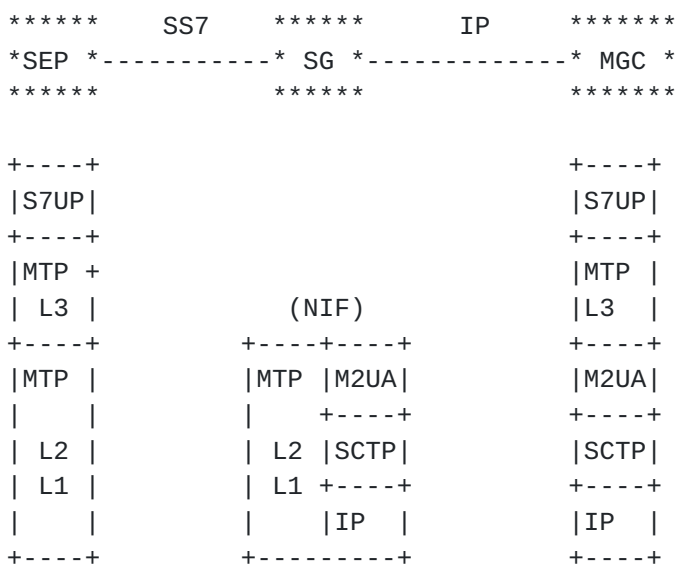
4.4 M2UA (SS7 MTP2 User Adaptation) Layer

This protocol would be mainly used between a Signaling Gateway (SG)

and Media Gateway Controler (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.

The only SS7 MTP2 User is MTP3 that is the protocol transported by this UAL.

The SG provides a interworking of transport functions with the IP transport, in order to transfer the MTP2-User signaling messages to and from an Application Server (e.g. MGC) where the peer MTP2-User protocol layer exists.



MGC - Media Gateway Controler
SG - Signaling Gateway
SEP - SS7 Signaling 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.

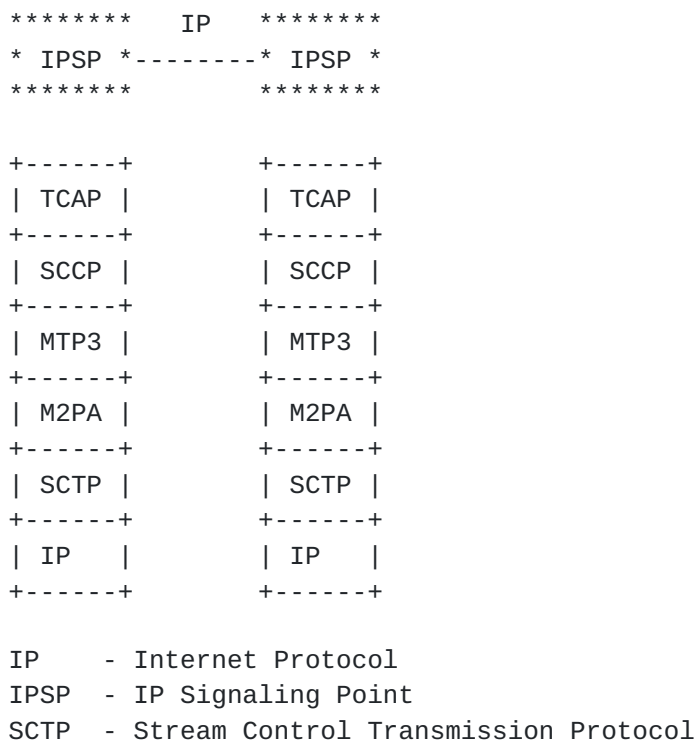
The value assigned by IANA for the Payload Protocol Identifier in the SCTP Payload Data chunk is ô2ö

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

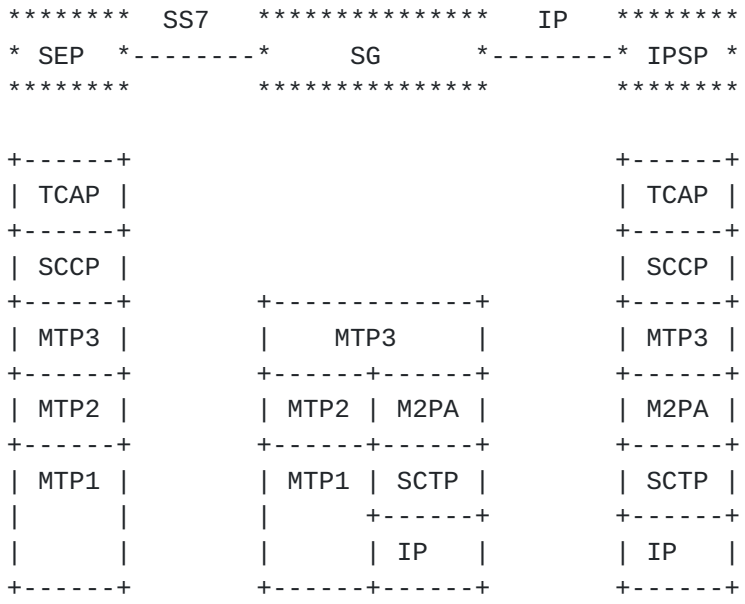
This protocol is used between SS7 Signaling Points employing the MTP Level 3 protocol. The SS7 Signaling Points may also employ standard SS7 links using the SS7 MTP Layer 2 to provide transport of MTP Layer

3 signaling messages.

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



IP - Internet Protocol
IPSP - IP Signaling Point
SCTP - Stream Control Transmission Protocol



SEP - SS7 Signaling 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 TBD.

The value assigned by IANA for the Payload Protocol Identifier in the SCTP Payload Data chunk is TBD

Differences between M2PA and M2UA include:

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.

c. M2PA: SG is an SS7 node with a point code.
   M2UA: SG is not an SS7 node and has no point code.

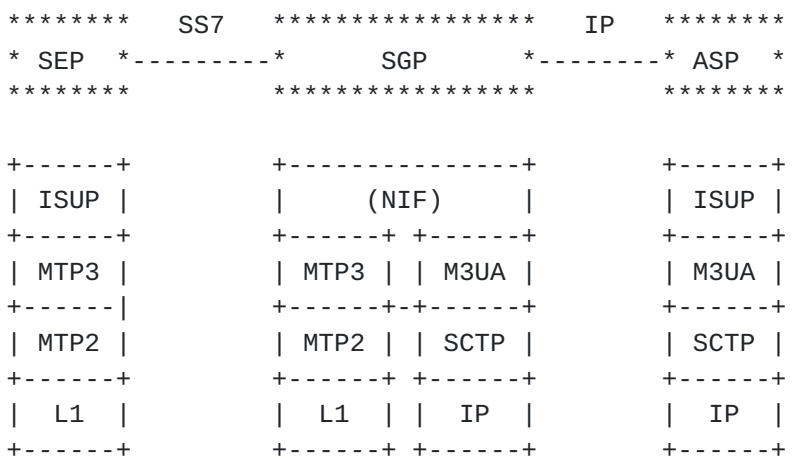
d. M2PA: SG can have upper SS7 layers, e.g., SCCP.
   M2UA: SG does not have upper SS7 layers since it has no MTP3.

e. M2PA: relies on MTP3 for management procedures.
   M2UA: uses M2UA management procedures.

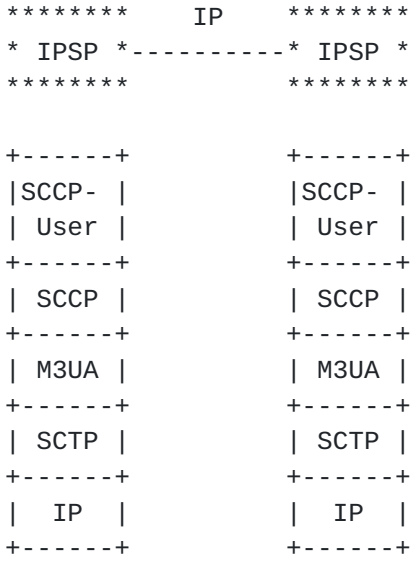
4.6 M3UA (SS7 MTP3 User Adaptation) Layer

This adaptation layer supports the transport of any SS7 MTP3-User signaling such as TUP, ISUP and SCCP over IP using the services of SCTP.

This protocol allows both:
- Interconnection of SS7 and IP nodes
- Communication between two IP nodes



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



It works using the client-server philosophy. ISEP is recommended to be client when talking with a SG. The reserved port by IANA is 2905 to listen to possible client connections.

The assigned payload protocol identifier for the SCTP DATA chunks is ô3ö.

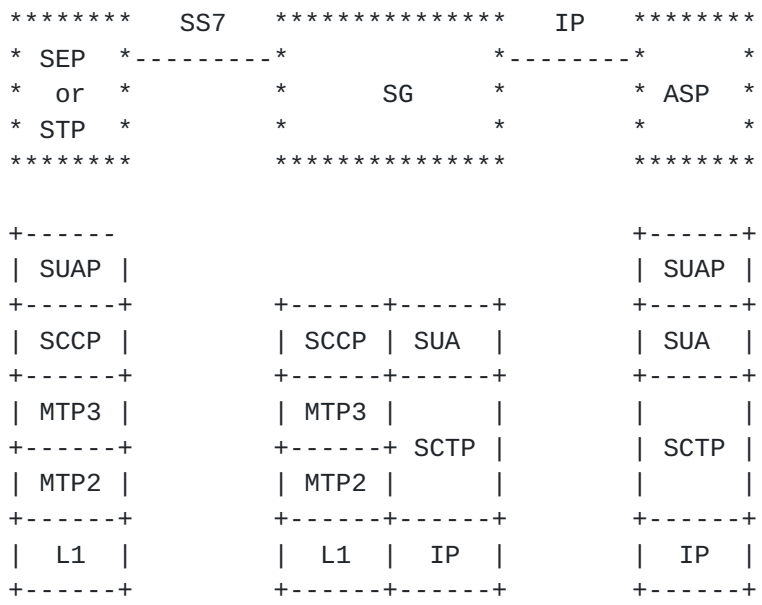
4.7 SUA (SS7 SCCP User Adaptation) Layer

This adaptation layer supports the transport of any SS7 SCCP-User signaling such as MAP, INAP, SMS, BSSAP, RANAP over IP using the services of SCTP. SUA can support only non-call related signaling.

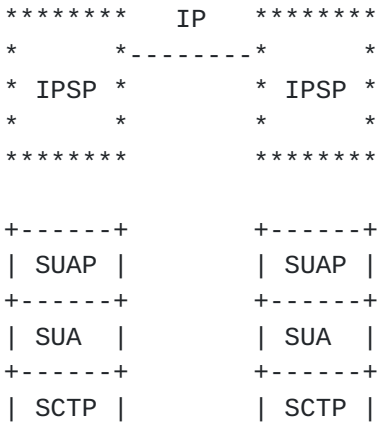
SUA does not pose stringent timing constraints on SCTP due to the fact that SUA applications have broad timing requirement (from 10 of seconds to hours) which the applications guard themselves and the timing supervision of the application is end-to-end, not hop-by-hop(as with ISUP).

Possible configurations showed in the pictures below:

- Interconnection of SS7 and IP
- IP Node to IP Node communication



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




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 ô4ö.

5 Security considerations

UALs are designated to carry signaling 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 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.

6 References and related work

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R. R., Xie, Q., Holdrege, M., Belinchon, M.C., and Jungmayer, A., "Stream Control Transmission Protocol Applicability statement", <draft-ietf-sigtran-sctp-applicability-03.txt>, December 2000. Work

In Progress.

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7 Acknowledgments

This document was initially developed by a design team consisting of Lode Coene, John Loughney, Michel Tuexen, Randall R. Stewart, Qiaobing Xie, Matt Holdrege, Maria-Carmen Belinchon, Andreas Jungmaier, Gery Verwimp and Lyndon Ong.

The authors wish to thank Renee Revis, H.J. Schwarzbauer, T. Taylor, G. Sidebottom, K. Morneault, T. George, M. Stillman and many others for their invaluable comments.

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