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L. Coene  
Siemens  
J. Pastor-Balbas  
Ericsson  
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Telephony Signalling Transport over  
Stream Control Transmission Protocol (SCTP) Applicability Statement

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Abstract

This document describes the applicability of the several protocols developed under the signalling transport framework. A description of the main issues regarding the use of the Stream Control Transmission Protocol (SCTP) and an explanation of each adaptation layer for transport of telephony signalling information over IP infrastructure are given.

## Table of Contents

1. Introduction .....	2
1.1. Scope .....	2
1.2. Terminology .....	3
1.3. Contributors .....	3
2. SIGTRAN Architecture .....	3
3. Issues for Transporting Telephony Signalling over SCTP .....	5
3.1. Congestion Control .....	5
3.2. Detection of Failures .....	6
3.2.1. Retransmission TimeOut (RTO) Calculation .....	6
3.2.2. Heartbeat .....	7
3.2.3. Maximum Number of Retransmissions .....	7
3.3. Shorten End-to-End Message Delay .....	7
3.4. Bundling Considerations .....	7
3.5. Stream Usage .....	7
4. User Adaptation Layers .....	7
4.1. Access Signalling .....	10
4.1.1. IUA (ISDN Q.921 User Adaptation) .....	10
4.1.2. V5UA (V5.2-User Adaptation) Layer .....	12
4.1.3. DUA (DPNSS/DASS User adaptation) Layer .....	13
4.2. Network Signalling .....	13
4.2.1. MTP lvl3 over IP .....	14
4.2.2. M3UA (SS7 MTP3 User Adaptation) Layer .....	17
4.2.3. SUA (SS7 SCCP User Adaptation) Layer .....	18
5. Security Considerations .....	20
6. Informative References .....	20

## 1. Introduction

This document is intended to describe how to transport telephony signalling protocols, used in classic telephony systems, over IP networks. As described in [RFC2719], the whole architecture is called SIGTRAN (Signalling Transport) and is composed of a transport protocol (SCTP) and several User Adaptation Layers (UALs). The transport protocol SCTP has been developed to fulfill the stringent requirements of telephony signalling networks [RFC3257]. The set of UALs has also been introduced to make it possible for different signalling protocols to use the SCTP layer.

## 1.1. Scope

The scope of this document is the SIGTRAN user adaptation layers and SCTP protocols and how they are used to transport telephony signalling information over IP networks.

## 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 include: MTP level 2, MTP level 3, and SCCP).

UAL: User Adaptation Layer, the protocol that encapsulates the upper layer telephony signalling protocols that are to be transported over SCTP/IP.

ISEP: IP Signalling Endpoint, an IP node that implements SCTP and a User adaptation layer.

SP: Signalling Point.

## 1.3. 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.

## 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:

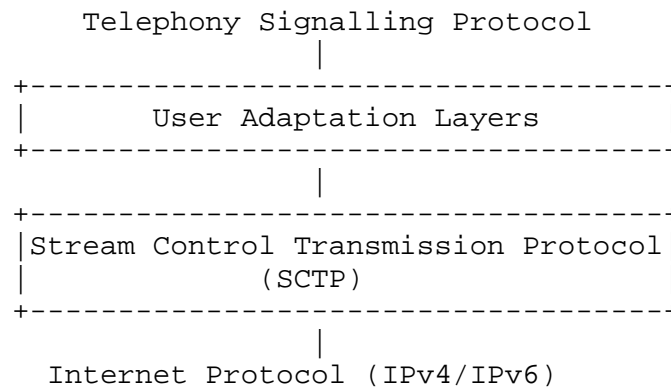


Figure 1: Telephony SIGNalling TRANsport Protocol Stack

The components of the protocol stack are:

1. Adaptation layers used when the telephony application needs to preserve an existing primitive interface (e.g., management indications or 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:

- o [RFC3332] SS7 MTP3 users: SCCP, ISUP, TUP...
- o [RFC3331] SS7 MTP2 users: MTP3
- o [RFC3868] SS7 SCCP users: RANAP, MAP(+TCAP), INAP(+TCAP)...
- o [RFC3057] ISDN Q.921 users: Q.931
- o [RFC3807] V5.2 / DSS1
- o ....

The user adaptation layers (UALs) are a set of protocols that encapsulate a specific signalling protocol to be transported over SCTP. The adaption is done in a way that the upper signalling protocols, which are relayed, remain unaware that the lower layers are different from the original lower telephony signalling layers. In that sense, the upper interface of the user adaptation layers needs to be the same as the upper layer interface is 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.

SCTP provides the following functions:

- o Reliable Data Transfer
- o Multiple streams to help avoid head-of-line blocking
- o Ordered and unordered data delivery on a per-stream basis
- o Bundling and fragmentation of user data
- o Congestion and flow control
- o Support for continuous monitoring of reachability
- o Graceful termination of association
- o Support of multi-homing for added reliability
- o Protection against blind denial-of-service attacks
- o 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, 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, an implementation must take special care to meet the performance, timing, and failure management requirements.

#### 3.1. Congestion Control

The basic mechanism of congestion control in SCTP has 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 controlled, 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. This must be done without destabilizing the network; otherwise, it 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

Often, telephony systems 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 appropriate steps to recover and preserve the calls. This poses special requirements on SCTP to discover unreachability 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 on the public Internet.

It should be noted that not using delayed SACK will also increase the speed of 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 for 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 UALs.

### 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

User Adaptation Layers (UALs) are defined to encapsulate different signalling protocols for transport 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 several UALs supports:

- o Seamless operation of UALs user peers over an IP network connection.
- o The interface boundary that the UAL user had with the traditional lower layer.
- o Management of SCTP transport associations and traffic between SGs and ISEPs or two ISEPs
- o 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 an access device and an exchange in the core network in order to establish, manage, or release the voice or data call paths. Several protocols 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, which belongs to the Common Channel Signalling (CCS) philosophy. The SS7 protocol stack is depicted below:

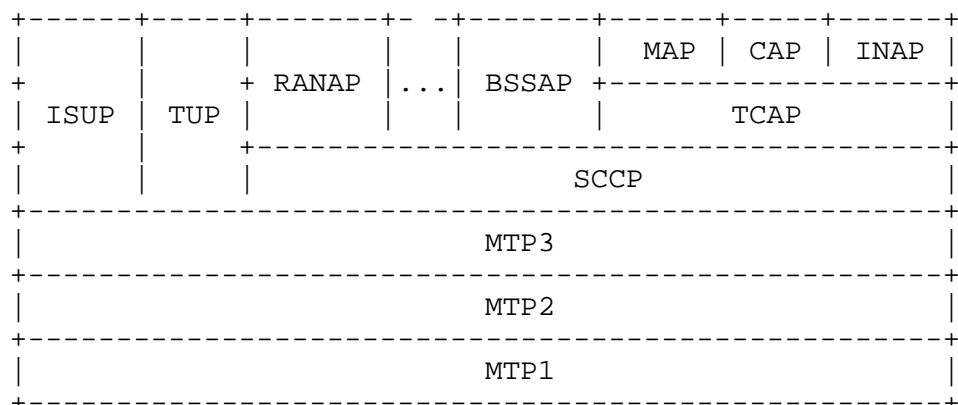


Figure 2: SS7 Protocol Stack



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

- o ISDN Q.921 Users: Q.931
- o V5.2/DSS1
- o DPNSS/DASS2 [RFC4129]
- o SS7 MTP3 Users: SCCP, ISUP, TUP
- o SS7 MTP2 Users: MTP3
- o 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.

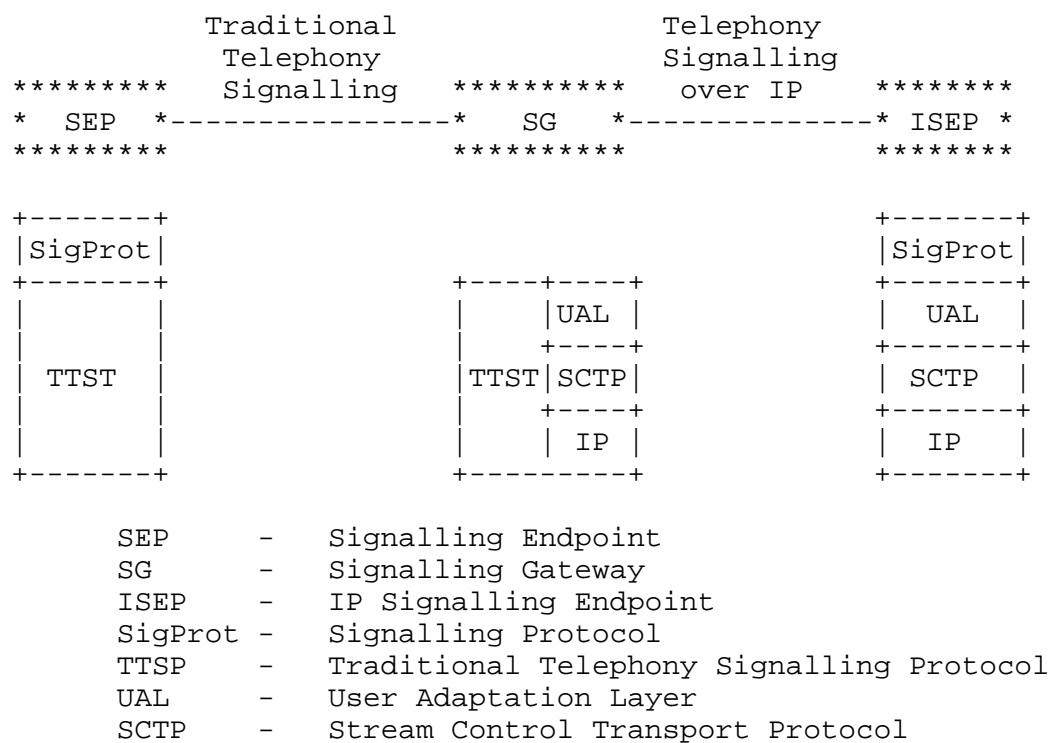


Figure 3: General Architecture of SS7-IP Interworking

This is also referred to 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.

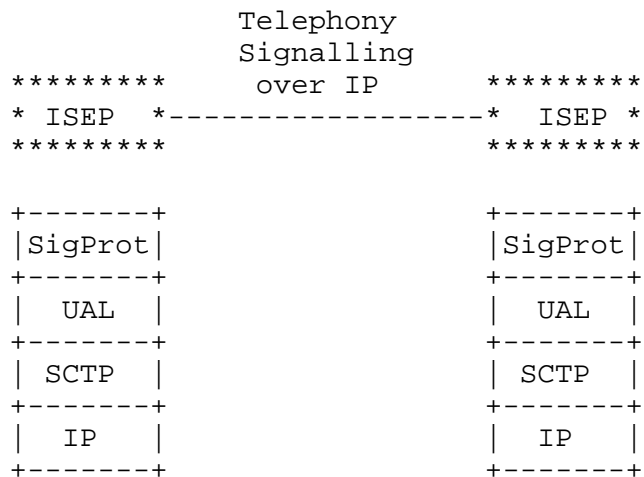


Figure 4: General Architecture of Intra-IP Communication

This is also referred to as IPSP communication. IPSP stands for IP Signalling Point and describes the role that the UAL plays on an 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 has developed UALs to transport the following Access Signalling protocols:

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

#### 4.1.1. IUA (ISDN Q.921 User Adaptation)

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-ISP communication where the ISEP role is typically played by a node called an MGC, as defined in [RFC2719].

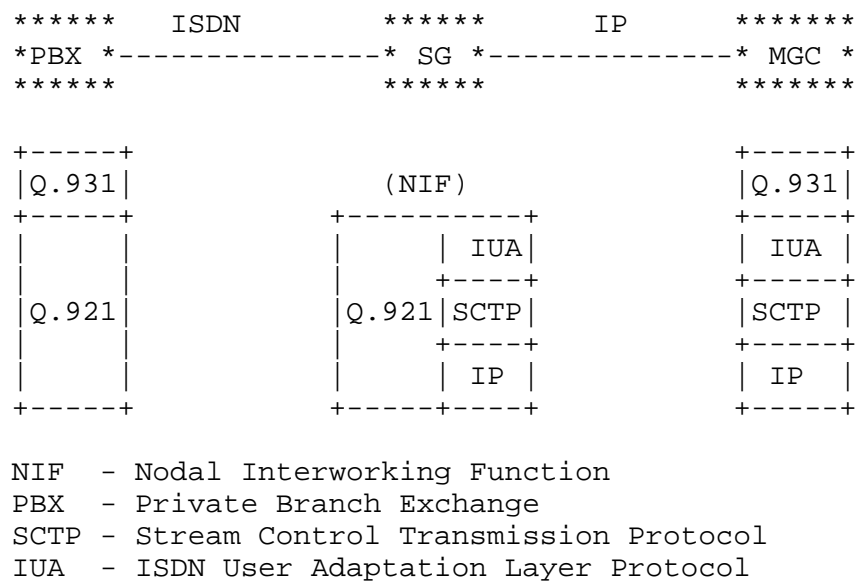


Figure 5: ISDN-IP Interworking using IUA

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".

## 4.1.1.2. V5UA (V5.2-User Adaptation) Layer

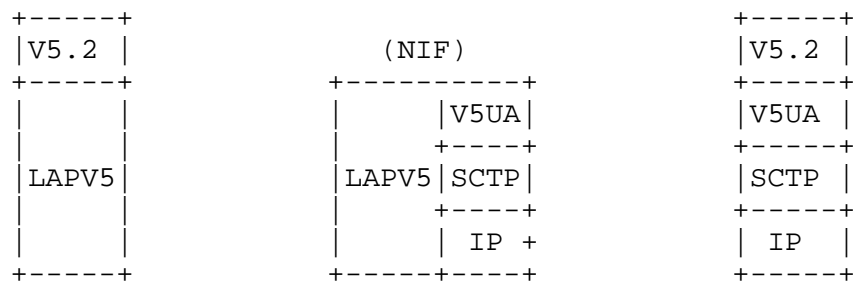
UAL: V5UA (V5.2-User Adaptation)

V5UA 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 *
*****                      *****                      *****

```



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

Figure 6: V5.2-IP Interworking using V5UA

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. DUA (DPNSS/DASS User adaptation) Layer

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 2.

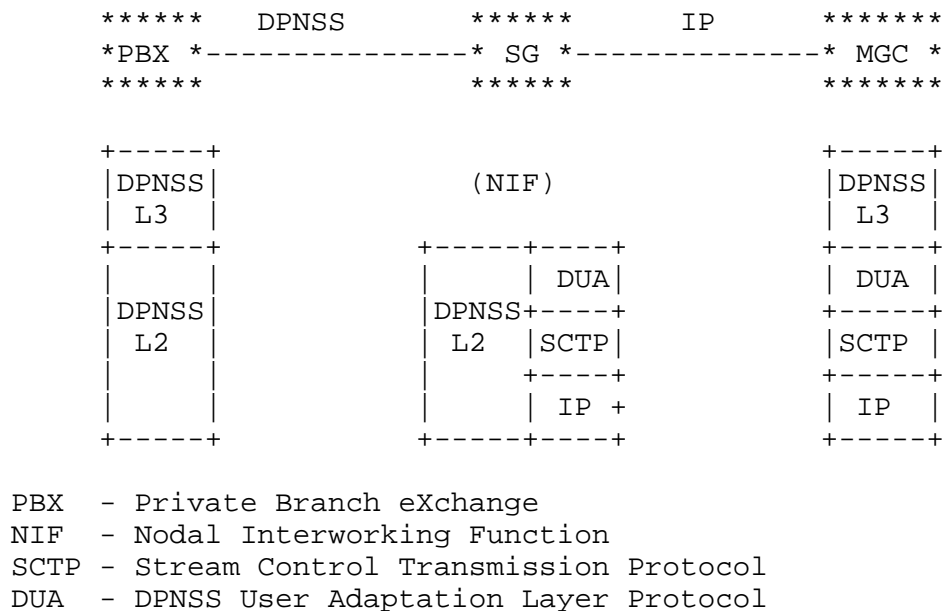


Figure 7: DPNSS-IP Interworking using DUA

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 has developed UALs to transport the following SS7 protocols:

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

## 4.2.1. MTP lvl3 over IP

UALs:

- o M2UA (SS7 MTP2 User Adaptation [RFC3331])
- o M2PA (SS7 MTP2-User Peer-to-Peer Adaptation [RFC4165])

## 4.2.1.1. M2UA (SS7 MTP2-User Adaptation) Layer

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 an MGC.

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

The SG provides an 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.

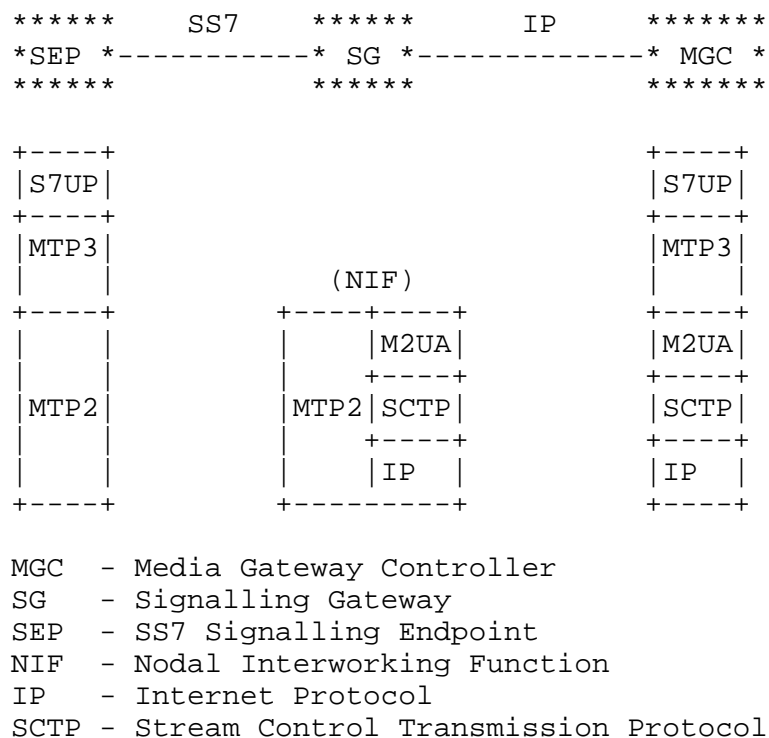


Figure 8: SS7-IP Interworking using M2UA

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)

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.

Connection of SS7 and IP nodes:

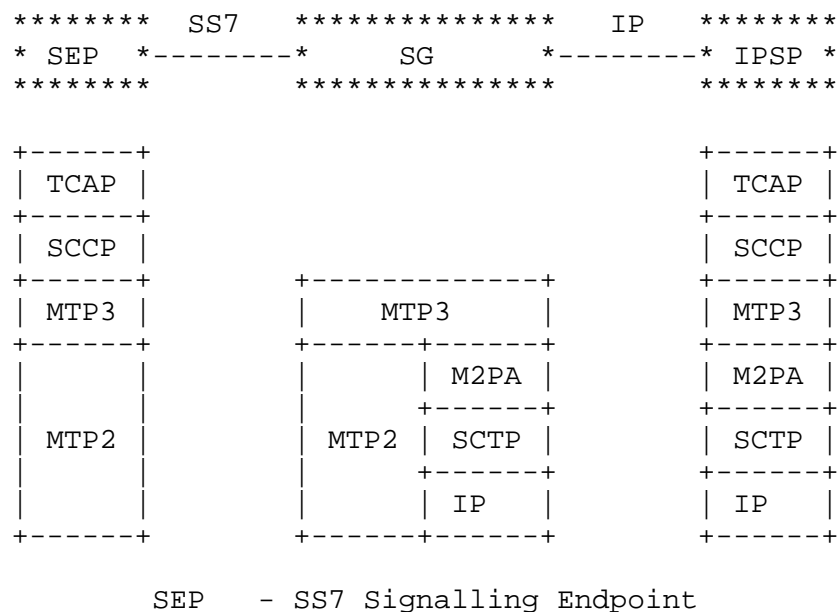
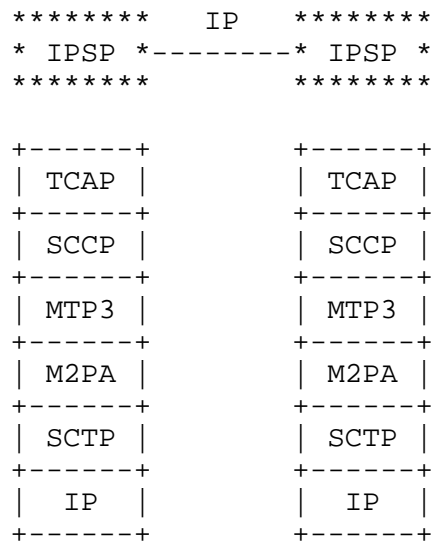


Figure 9: SS7-IP Interworking with M2PA

Communication between two IP nodes:



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

Figure 10: Intra-IP Communication using M2PA

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

- o M2PA: IPSP processes MTP3/MTP2 primitives.
- o M2UA: MGC transports MTP3/MTP2 primitives between the SG's MTP2 and the MGC's MTP3 (via the NIF) for processing.
- o M2PA: SG-IPSP connection is an SS7 link.
- o M2UA: SG-MGC connection is not an SS7 link. It is an extension of MTP to a remote entity.

## 4.2.2. M3UA (SS7 MTP3 User Adaptation) Layer

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:

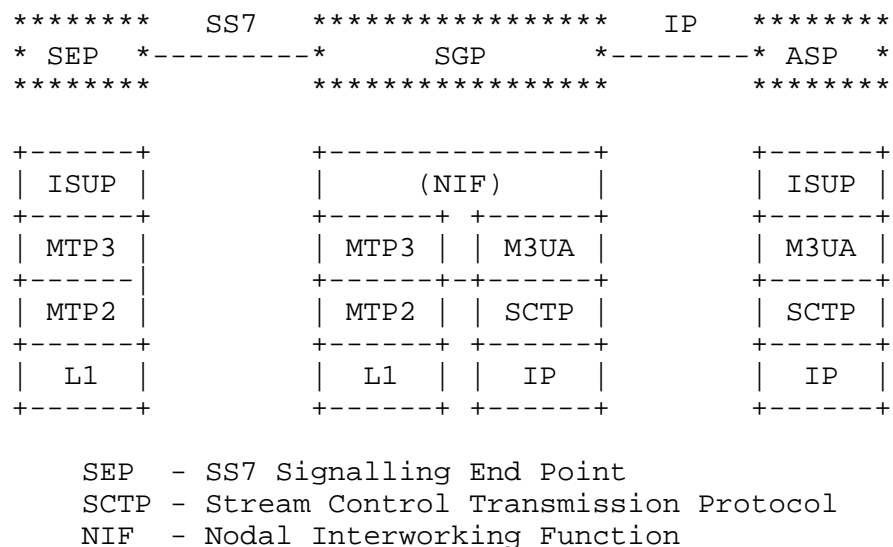


Figure 11: SS7-IP Interworking using M3UA

Communication between two IP nodes:

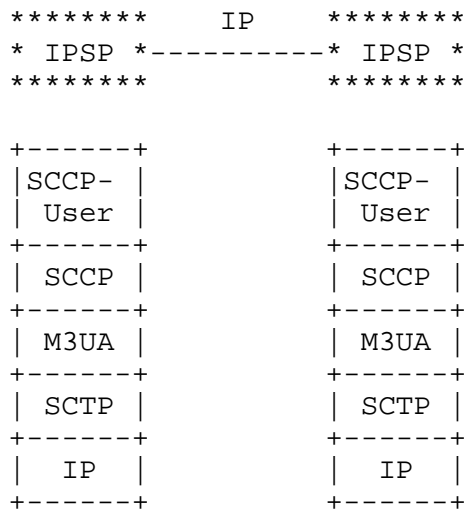


Figure 12: Intra-IP Communication using M3UA

M3UA uses a client-server architecture. It is recommended that the ISEP acts as the client and initiate the SCTP associations 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".

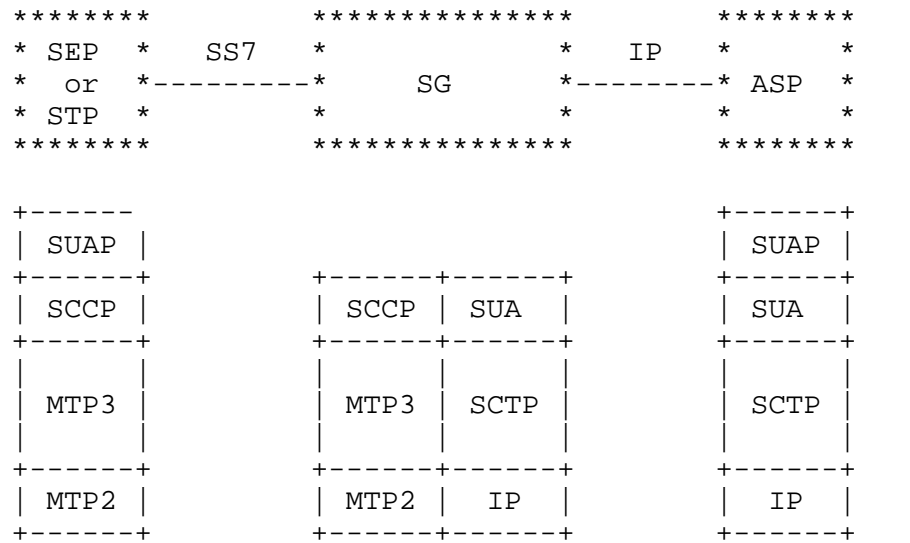
#### 4.2.3. SUA (SS7 SCCP User Adaptation) Layer

UAL: SUA (SS7 SCCP User Adaptation)

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

Possible configurations are showed in the pictures below.

- Interconnection of SS7 and IP:



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

Figure 13: SS7-IP Interworking using SUA

- IP Node to IP Node communication:

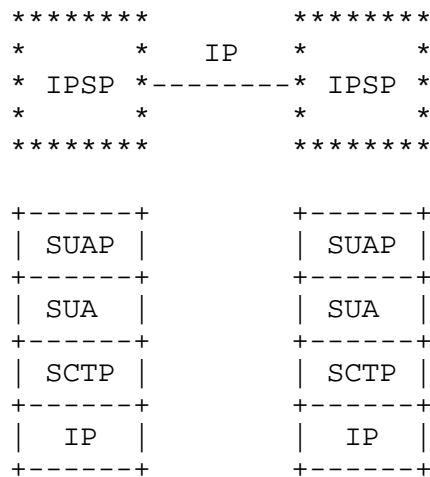


Figure 14: Intra-IP Communication using SUA

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. Although some security needs overlap, any security solution should fulfill all the different parties' needs. See specific Security Considerations in each UAL Technical specification for details (for general security principles of SIGTRAN, see [RFC3788]).

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

This document does not add any new components to the protocols included in the discussion. For secure use of the SIGTRAN protocols, readers should go through the "Security Considerations for SIGTRAN Protocols" [RFC3788]). According to that document, the use of the IPsec is the main requirement to secure SIGTRAN protocols in the Internet, but Transport Layer Security (TLS) is also considered a perfectly valid option for use in certain scenarios (see [RFC3436] for more information on using TLS with SCTP). Recommendations of usage are also included.

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## Authors' Addresses

Lode Coene  
Siemens  
Atealaan 34  
Herentals B-2200  
Belgium

Phone: +32-14-252081  
EMail: lode.coene@siemens.com

Javier Pastor-Balbas  
Ericsson  
Via de los Poblados 13  
Madrid 28033  
Spain

Phone: +34 91 339 1397  
EMail: J.Javier.Pastor@ericsson.com

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