Session Initiation Protocol for Telephones (SIP-T):
Context and Architectures

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Abstract

   The popularity of gateways that interwork between the PSTN (Public
   Switched Telephone Network) and SIP networks has motivated the
   publication of a set of common practices that can assure consistent
   behavior across implementations. This document taxonomizes the uses
   of PSTN-SIP gateways, provides uses cases, and identifies mechanisms
   necessary for interworking. The mechanisms detail how SIP provides
   for both 'encapsulation' (bridging the PSTN signaling across a SIP
   network) and 'translation' (gatewaying).

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1. Introduction

   The Session Initiation Protocol (SIP [1]) is an application-layer
   control protocol that can establish, modify and terminate multimedia
   sessions or calls. These multimedia sessions include multimedia
   conferences, Internet telephony and similar applications. SIP is one
   of the key protocols used to implement Voice over IP (VoIP).
   Although performing telephony call signaling and transporting the
   associated audio media over IP yields significant advantages over
   traditional telephony, a VoIP network cannot exist in isolation from
   traditional telephone networks. It is vital for a SIP telephony
   network to interwork with the PSTN.

   The popularity of gateways that interwork between the PSTN and SIP
   networks has motivated the publication of a set of common practices
   that can assure consistent behavior across implementations. The
   scarcity of SIP expertise outside the IETF suggests that the IETF is
   the best place to stage this work, especially since SIP is in a
   relative state of flux compared to the core protocols of the PSTN.
   Moreover, the IETF working groups that focus on SIP (SIP and SIPPING)
   are best positioned to ascertain whether or not any new extensions to
   SIP are justified for PSTN interworking. This framework addresses
   the overall context in which PSTN-SIP interworking gateways might be
   deployed, provides use cases and identifies the mechanisms necessary
   for interworking.

   An important characteristic of any SIP telephony network is feature
   transparency with respect to the PSTN. Traditional telecom services
   such as call waiting, freephone numbers, etc., implemented in PSTN
   protocols such as Signaling System No. 7 (SS7 [6]) should be offered
   by a SIP network in a manner that precludes any debilitating
   difference in user experience while not limiting the flexibility of
   SIP. On the one hand, it is necessary that SIP support the
   primitives for the delivery of such services where the terminating
   point is a regular SIP phone (see definition in Section 2 below)
   rather than a device that is fluent in SS7. However, it is also
   essential that SS7 information be available at gateways, the points

of SS7-SIP interconnection, to ensure transparency of features not
otherwise supported in SIP. If possible, SS7 information should be
available in its entirety and without any loss to trusted parties in
the SIP network across the PSTN-IP interface; one compelling need to
do so also arises from the fact that certain networks utilize
proprietary SS7 parameters to transmit certain information through
their networks.

Another important characteristic of a SIP telephony network is
routability of SIP requests - a SIP request that sets up a telephone
call should contain sufficient information in its headers to enable
it to be appropriately routed to its destination by proxy servers in
the SIP network. Most commonly this entails that parameters of a
call like the dialed number should be carried over from SS7 signaling
to SIP requests. Routing in a SIP network may in turn be influenced
by mechanisms such as TRIP [8] or ENUM [7].

The SIP-T (SIP for Telephones) effort provides a framework for the
integration of legacy telephony signaling into SIP messages. SIP-T
provides the above two characteristics through techniques known as
'encapsulation' and 'translation' respectively. At a SIP-ISUP
gateway, SS7 ISUP messages are encapsulated within SIP in order that
information necessary for services is not discarded in the SIP
request. However, intermediaries like proxy servers that make
routing decisions for SIP requests cannot be expected to understand
ISUP, so simultaneously, some critical information is translated from
an ISUP message into the corresponding SIP headers in order to
determine how the SIP request will be routed.

While pure SIP has all the requisite instruments for the
establishment and termination of calls, it does not have any baseline
mechanism to carry any mid-call information (such as the ISUP INF/INR
query) along the SIP signaling path during the session. This mid-
call information does not result in any change in the state of SIP
calls or the parameters of the sessions that SIP initiates. A
provision to transmit such optional application-layer information is
also needed.

Problem definition: To provide ISUP transparency across SS7-SIP
interworking

SS7-SIP Interworking Requirements	SIP-T Functions
Transparency of ISUP Signaling	Encapsulation of ISUP in the SIP body
Routability of SIP messages with dependencies on ISUP	Translation of ISUP information into the SIP header
Transfer of mid-call ISUP signaling messages	Use of the INFO Method for mid-call signaling

Table 1: SIP-T features that fulfill PSTN-IP inter-connection
         Requirements

While this document specifies the requirements above, it provide
mechanisms to satisfy them - however, this document does serve as an
framework for the documents that do provide these mechanisms, all of
which are referenced in Section 5.

Note that many modes of signaling are used in telephony (SS7 ISUP,
BTNUP, Q.931, MF etc.). This document focuses on SS7 ISUP and aims
to specify the behavior across ISUP-SIP interfaces only. The scope
of the SIP-T enterprise may, over time, come to encompass other
signaling systems as well.

2. SIP-T for ISUP-SIP Interconnections

SIP-T is not a new protocol - it is a set of mechanisms for
interfacing traditional telephone signaling with SIP. The purpose of
SIP-T is to provide protocol translation and feature transparency
across points of PSTN-SIP interconnection. It intended for use where
a VoIP network (a SIP network, for the purposes of this document)
interfaces with the PSTN.

Using SIP-T, there are three basic models for how calls interact with
gateways. Calls that originate in the PSTN can traverse a gateway to
terminate at a SIP endpoint, such as an IP phone. Conversely, an IP
phone can make a call that traverses a gateway to terminate in the
PSTN. Finally, an IP network using SIP may serve as a transit
network between gateways - a call may originate and terminate in the
PSTN, but cross a SIP-based network somewhere in the middle.

The SS7 interfaces of a particular gateway determine the ISUP
variants that that gateway supports. Whether or nor a gateway
supports a particular version of ISUP determines whether it can
provide feature transparency while terminating a call.

The following are the primary agents in a SIP-T-enabled network.

o PSTN (Public Switched Telephone Network): This refers to the
   entire interconnected collection of local, long-distance and
   international phone companies. In the examples below, the term
   Local Exchange Carrier (LEC) is used to denote a portion (usually,
   a regional division) of the PSTN.

o IP endpoints: Any SIP user agent that can act as an originator or
   recipient of calls. Thus, the following devices are classified as
   IP endpoints:

   * Gateways: A telephony gateway provides a point of conversion
      between signaling protocols (such as ISUP and SIP) as well as
      circuit-switch and packet-switched audio media. The term Media
      Gateway Controller (MGC) is also used in the examples and
      diagrams in this document to denote large-scale clusters of
      decomposed gateways and control logic that are frequently
      deployed today. So for example, a SIP-ISUP gateway speaks ISUP
      to the PSTN and SIP to the Internet and is responsible for
      converting between the types of signaling, as well as
      interchanging any associated bearer audio media.

   * SIP phones: The term used to represent all end-user devices
      that originate or terminate SIP VoIP calls.

   * Interface points between networks where administrative policies
      are enforced (potentially middleboxes, proxy servers, or
      gateways).

o Proxy Servers: A proxy server is a SIP intermediary that routes
   SIP requests to their destinations. For example, a proxy server
   might direct a SIP request to another proxy, a gateway or a SIP
   phone.

```
                    *********************
                 ***                   ***
                *                         *
               *      -------             *
              *      |proxy|               *
             *        -------               *
          ----|                          ----|
         /|MGC1|       VoIP Network      |MGC2|\
        / ----           * *              ---- \
   SS7 /         *                      *       \ SS7
      /           *      -------       *         \
     /             *    |proxy|       *           \
 --------           *    -------     *         ---------
| LEC1 |             **            **         | LEC2  |
 --------        *********************         ---------
```
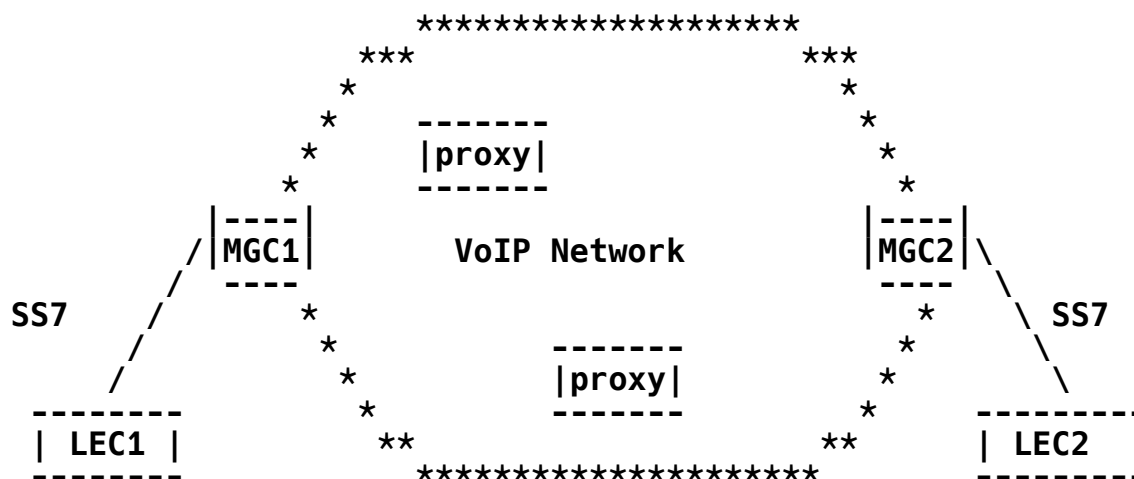
Figure 1: Motivation for SIP-T in ISUP-SIP interconnection

In Figure 2 a VoIP cloud serves as a transit network for telephone
calls originating in a pair of LECs, where SIP is employed as the
VoIP protocol used to set up and tear down these VoIP calls. At the
edge of the depicted network, an MGC converts the ISUP signals to SIP
requests, and sends them to a proxy server which in turn routes
calls on other MGCs. Although this figure depicts only two MGCs,
VoIP deployments would commonly have many such points of
interconnection with the PSTN (usually to diversify among PSTN rate
centers). For a call originating from LEC1 and be terminating in
LEC2, the originator in SIP-T is the gateway that generates the SIP
request for a VoIP call, and the terminator is the gateway that is
the consumer of the SIP request; MGC1 would thus be the originator
and MGC2, the terminator. Note that one or more proxies may be used
to route the call from the originator to the terminator.

In this flow, in order to seamlessly integrate the IP network with
the PSTN, it is important to preserve the received SS7 information
within SIP requests at the originating gateway and reuse this SS7
information when signaling to the PSTN at the terminating gateway.
By encapsulating ISUP information in the SIP signaling, a SIP network
can ensure that no SS7 information that is critical to the
instantiation of features is lost when SIP bridges calls between two
segments of the PSTN.

That much said, if only the exchange of ISUP between gateways were
relevant here, any protocol for the transport of signaling
information may be used to achieve this, obviating the need for SIP
and consequently that of SIP-T. SIP-T is employed in order to
leverage the intrinsic benefits of utilizing SIP: request routing and
call control leveraging proxy servers (including the use of forking),

ease of SIP service creation, SIP's capability negotiation systems,
and so on. Translation of information from the received ISUP message
parameters to SIP header fields enables SIP intermediaries to
consider this information as they handle requests. SIP-T thus
facilitates call establishment and the enabling of new telephony
services over the IP network while simultaneously providing a method
of feature-rich interconnection with the PSTN.

Finally, the scenario in Figure 2 is just one of several flows in
which SIP-T can be used - voice calls do not always both originate
and terminate in the PSTN (via gateways); SIP phones can also be
endpoints in a SIP-T session. In subsequent sections, the following
possible flows will be further detailed:

1. PSTN origination - PSTN termination: The originating gateway
    receives ISUP from the PSTN and it preserves this information
    (via encapsulation and translation) in the SIP messages that it
    transmits towards the terminating gateway. The terminator
    extracts the ISUP content from the SIP message that it receives
    and it reuses this information in signaling sent to the PSTN.

2. PSTN origination - IP termination: The originating gateway
    receives ISUP from the PSTN and it preserves this ISUP
    information in the SIP messages (via encapsulation and
    translation) that it directs towards the terminating SIP user
    agent. The terminator has no use for the encapsulated ISUP and
    ignores it.

3. IP origination - PSTN termination: A SIP phone originates a VoIP
    call that is routed by one or more proxy servers to the
    appropriate terminating gateway. The terminating gateway
    converts to ISUP signaling and directs the call to an appropriate
    PSTN interface, based on information that is present in the
    received SIP header.

4. IP origination - IP termination: This is a case for pure SIP.
    SIP-T (either encapsulation or translation of ISUP) does not come
    into play as there is no PSTN interworking.

3. SIP-T Flows

The follow sections explore the essential SIP-T flows in detail.
Note that because proxy servers are usually responsible for routing
SIP requests (based on the Request-URI) the eventual endpoints at
which a SIP request will terminate is generally not known to the
originator. So the originator does not select from the flows

described in this section, as a matter of static configuration or on
a per-call basis - rather, each call is routed by the SIP network
independently, and it may instantiate any of the flows below as the
routing logic of the network dictates.

3.1 SIP Bridging (PSTN - IP - PSTN)

```
                          ********************
              ***                              ***
            *                                      *
          *         -------                           *
         *         |proxy|                             *
        *           -------                             *
       |---|                                          |---|
      / |MGC|           VoIP Network                  |MGC|\
     /   ---                                           ---   \
    /      *                                           *       \
   /        *         -------                         *          \
  /          *       |proxy|                         *            \
 --------      *      -------                       *       ---------
| PSTN |          ***                           ***        | PSTN  |
--------            ********************                    ---------
```
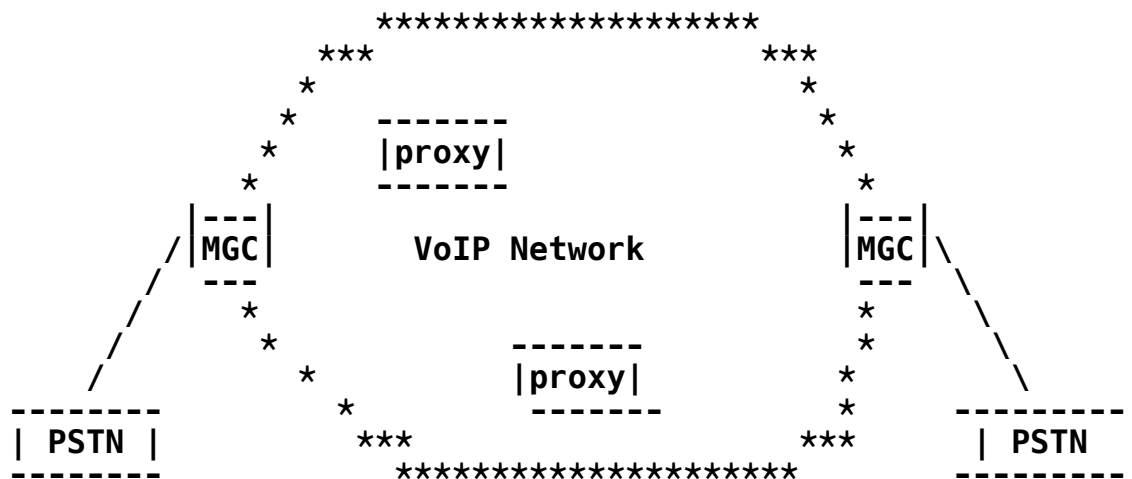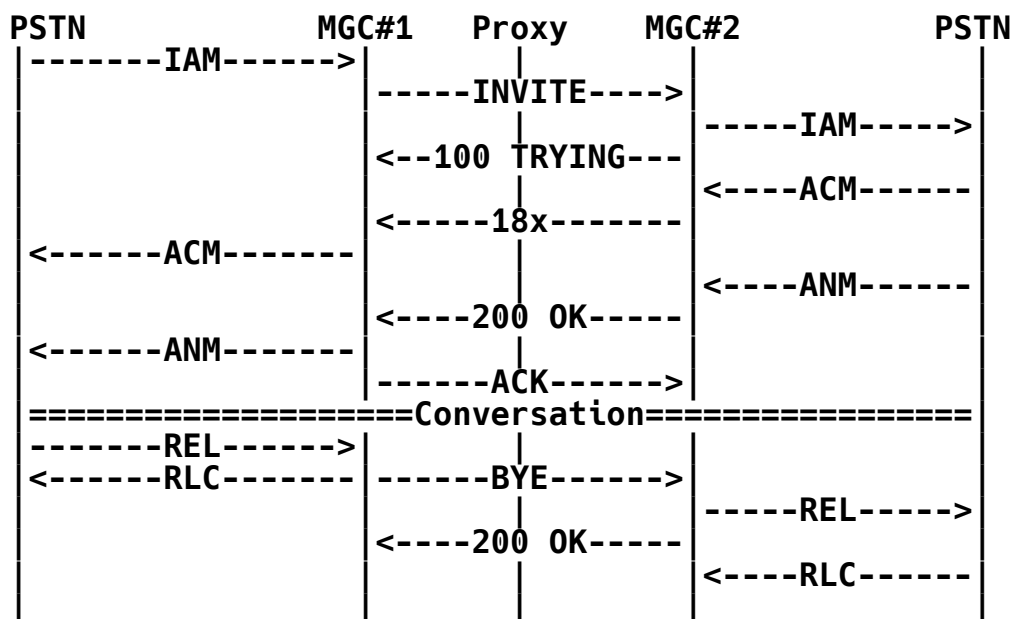
Figure 2: PSTN origination - PSTN termination (SIP Bridging)

A scenario in which a SIP network connects two segments of the PSTN
is referred to as 'SIP bridging'. When a call destined for the SIP
network originates in the PSTN, an SS7 ISUP message will eventually
be received by the gateway that is the point of interconnection with
the PSTN network. This gateway is from the perspective of the SIP
protocol the user agent client for this call setup request.
Traditional SIP routing is used in the IP network to determine the
appropriate point of termination (in this instance a gateway) and to
establish a SIP dialog and begin negotiation of a media session
between the origination and termination endpoints. The egress
gateway then signals ISUP to the PSTN, reusing any encapsulated ISUP
present in the SIP request it receives as appropriate.

A very elementary call-flow for SIP bridging is shown below.

```
      PSTN            MGC#1   Proxy   MGC#2          PSTN
       |------IAM------>|       |       |             |
       |                |-----INVITE---->|            |
       |                |       |        |            |
       |                |<--100 TRYING---|            |
       |                |       |        |-----IAM----->|
       |                |       |        |            |
       |                |<-----18x-------|<----ACM------|
       |<------ACM-------|       |        |            |
       |                |       |        |            |
       |                |<----200 OK-----|<----ANM------|
       |<------ANM-------|       |        |            |
       |                |------ACK------>|             |
       |================================Conversation=================
       |-------REL------>|       |        |            |
       |<------RLC-------|------BYE------>|            |
       |                |       |        |-----REL----->|
       |                |<----200 OK-----|            |
       |                |       |        |<----RLC------|
       |                |       |        |             |
```
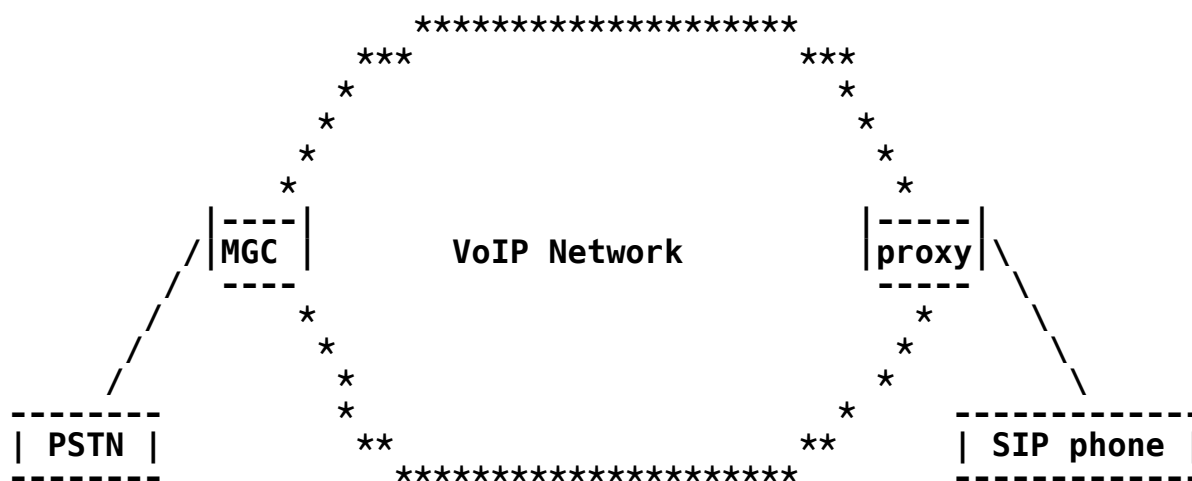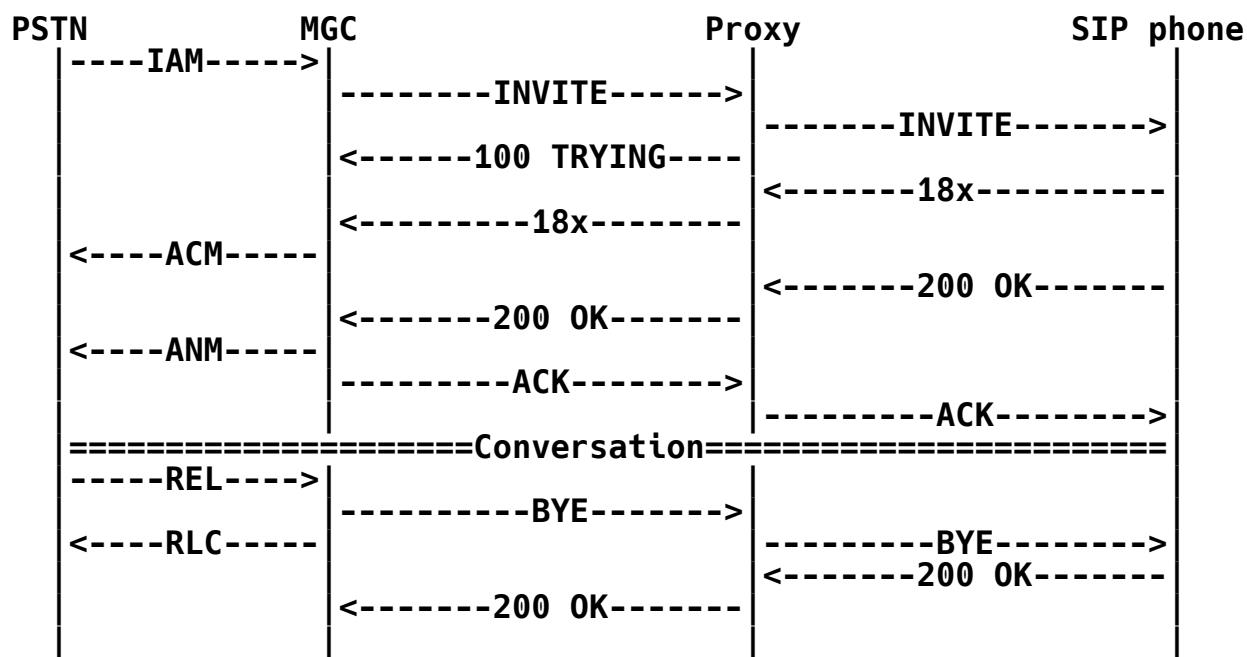
3.2 PSTN origination - IP termination

Figure 3: PSTN origination - IP termination

A call originates from the PSTN and terminates at a SIP phone. Note
that in Figure 5, the proxy server acts as the registrar for the SIP
phone in question.

A simple call-flow depicting the ISUP and SIP signaling for a PSTN-
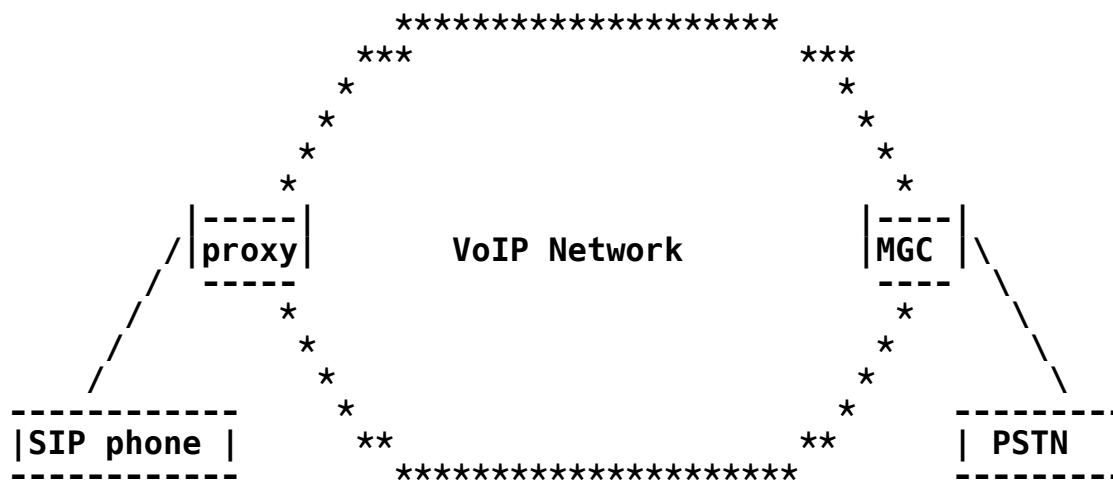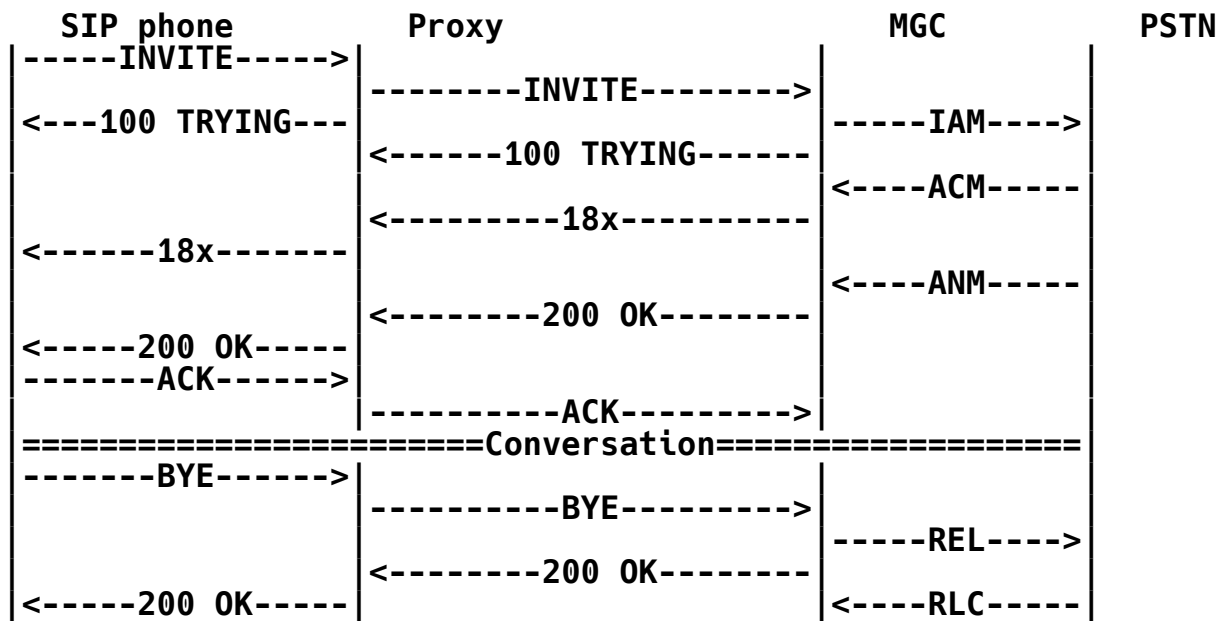originated call terminating at a SIP endpoint follows:

```
PSTN                  MGC                       Proxy            SIP phone
 |----IAM----->|
               |--------INVITE------>|
               |<------100 TRYING----|
                                     |-------INVITE------->|
               |<---------18x--------|
                                     |<-------18x----------|
<----ACM-----|
                                     |<-------200 OK-------|
               |<-------200 OK-------|
<----ANM-----|
               |---------ACK-------->|
                                     |---------ACK-------->|
===================Conversation=======================
 |-----REL---->|
               |---------BYE------->|
                                     |---------BYE-------->|
<----RLC-----|                       |<-------200 OK-------|
               |<-------200 OK-------|
 |            |                       |                    |
```

3.3 IP origination - PSTN termination

```
                      ********************
                    ***                  ***
                   *                        *
                  *                          *
                 *                            *
                *                              *
               |------|                      |----|
              /|proxy |    VoIP Network      |MGC |\
             / |------|                      |----| \
            /      *                           *     \
           /       *                           *      \
          /         *                         *        \
         ------------        *              *     ---------
        |SIP phone |        **             **    | PSTN  |
         ------------        ******************    ---------
```

Figure 4: IP origination - PSTN termination

A call originates from a SIP phone and terminates in the PSTN.
Unlike the previous two flows, there is therefore no ISUP
encapsulation in the request - the terminating gateway therefore only
performs translation on the SIP headers to derive values for ISUP
parameters.

A simple call-flow illustrating the different legs in the call is as
shown below.

```
        SIP phone        Proxy                MGC          PSTN
        |-----INVITE----->|                    |            |
        |                 |--------INVITE-------->|          |
        |<---100 TRYING---|                    |-----IAM---->|
        |                 |<------100 TRYING------|          |
        |                 |                    |<----ACM-----|
        |                 |<---------18x----------|          |
        |<------18x-------|                    |            |
        |                 |                    |<----ANM-----|
        |                 |<--------200 OK--------|          |
        |<-----200 OK-----|                    |            |
        |-------ACK------>|                    |            |
        |                 |---------ACK---------->|          |
        |=====================Conversation===================|
        |-------BYE------>|                    |            |
        |                 |---------BYE---------->|          |
        |                 |                    |-----REL---->|
        |                 |<--------200 OK--------|          |
        |<-----200 OK-----|                    |<----RLC-----|
```

4. SIP-T Roles and Behavior

   There are three distinct sorts of elements (from a functional point
   of view) in a SIP VoIP network that interconnects with the PSTN:

   1. The originators of SIP signaling

   2. The terminators of SIP signaling

   3. The intermediaries that route SIP requests from the originator to
       the terminator

   Behavior for the Section 4.1, Section 4.2 and Section 4.3
   intermediary roles in a SIP-T call are described in the following
   sections.

4.1 Originator

   The function of the originating user agent client is to generate the
   SIP Call setup requests (i.e., INVITEs). When a call originates in
   the PSTN, a gateway is the UAC; otherwise some native SIP endpoint is
   the UAC. In either case, note that the originator generally cannot
   anticipate what sort of entity the terminator will be, i.e., whether
   final destination of the request is in a SIP network or the PSTN.

In the case of calls originating in the PSTN (see Figure 3 and Figure
5), the originating gateway takes the necessary steps to preserve the
ISUP information by encapsulating it in the SIP request it creates.
The originating gateway is entrusted with the responsibility of
identifying the version of the ISUP (ETSI, ANSI, etc.) that it has
received and providing this information in the encapsulated ISUP
(usually by adding a multipart MIME body with appropriate MIME
headers). It then formulates the headers of the SIP INVITE request
from the parameters of the ISUP that it has received from the PSTN as
appropriate (see Section 5). This might, for instance, entail
setting the 'To:' header field in the INVITE to the reflect dialed
number (Called Party Number) of the received ISUP IAM.

In other cases (like Figure 7), a SIP phone is the originator of a
VoIP call. Usually, the SIP phone sends requests to a SIP proxy that
is responsible for routing the request to an appropriate destination.
There is no ISUP to encapsulate at the user agent client, as there is
no PSTN interface. Although the call may terminate in the telephone
network and need to signal ISUP in order for that to take place, the
originator has no way to anticipate this and it would be foolhardy to
require that all SIP VoIP user agents have the capability to generate
ISUP. It is therefore not the responsibility of an IP endpoints like
a SIP phone to generate encapsulated ISUP. Thus, an originator must
generate the SIP signaling while performing ISUP encapsulation and
translation when possible (meaning when the call has originated in
the PSTN).

Originator requirements: encapsulate ISUP, translate information from
ISUP to SIP, multipart MIME support (for gateways only)

4.2 Terminator

The SIP-T terminator is a consumer of the SIP calls. The terminator
is a standard SIP UA that can be either a gateway that interworks
with the PSTN or a SIP phone.

In case of PSTN terminations (see Figure 3 and Figure 7) the egress
gateway terminates the call to its PSTN interface. The terminator
generates the ISUP appropriate for signaling to the PSTN from the
incoming SIP message. Values for certain ISUP parameters may be
gleaned from the SIP headers or extracted directly from an
encapsulated ISUP body. Generally speaking, a gateway uses any
encapsulated ISUP as a template for the message it will send, but it
overwrites parameter values in the template as it translates SIP
headers or adds any parameter values that reflect its local policies
(see Appendix A item 1).

In case of an IP termination (Figure 5), the SIP UAS that receives
SIP messages with encapsulated ISUP typically disregards the ISUP
message. This does introduce a general requirement, however, that
devices like SIP phones handle multipart MIME messages and unknown
MIME types gracefully (this is a baseline SIP requirement, but also a
place where vendors have been known to make shortcuts).

Terminator requirements: standard SIP processing, interpretation of
encapsulated ISUP (for gateways only), support for multipart MIME,
graceful handling of unknown MIME content (for non-gateways only)

4.3 Intermediary

Intermediaries like proxy servers are entrusted with the task of
routing messages to one another, as well as gateways and SIP phones.
Each proxy server makes a forwarding decision for a SIP request based
on values of various headers, or 'routable elements' (including the
Request-URI, route headers, and potentially many other elements of a
SIP request).

SIP-T does introduce some additional considerations for forwarding a
request that could lead to new features and requirements for
intermediaries. Feature transparency of ISUP is central to the
notion of SIP-T. Compatibility between the ISUP variants of the
originating and terminating PSTN interfaces automatically leads to
feature transparency. Thus, proxy servers might take an interest in
the variants of ISUP that are encapsulated with requests - the
variant itself could become a routable element. The termination of a
call at a point that results in greater proximity to the final
destination (rate considerations) is also an important consideration.
The preference of one over the other results in a trade-off between
simplicity of operation and cost. The requirement of procuring a
reasonable rate may dictate that a SIP-T call spans dissimilar PSTN
interfaces (SIP bridging across different gateways that don't support
any ISUP variants in common). In order to optimize for maximum
feature transparency and rate, some operators of intermediaries might
want to consider practices along the following lines:

a) The need for ISUP feature transparency may necessitate ISUP
   variant translation (conversion), i.e., conversion from one
   variant of ISUP to another in order to facilitate the termination
   of that call over a gateway interface that does not support the
   ISUP variant of the originating PSTN interface. (See Appendix A
   item 2.) Although in theory conversion may be performed at any
   point in the path of the request, it is optimal to perform it at a
   point that is at the greatest proximity to the terminating
   gateway. This could be accomplished by delivering the call to an
   application that might perform the conversion between variants.
   Feature transparency in this case is contingent on the
   availability of resources to perform ISUP conversion, and it
   incurs an increase in the call-set up time.

b) An alternative would be to sacrifice ISUP transparency by handing
   the call off to a gateway that does not support the version of the
   originating ISUP. The terminating MGC would then just ignore the
   encapsulated ISUP and use the information in the SIP header to
   terminate the call.

So, it may be desirable for proxy servers to have the intelligence to
make a judicious choice given the options available to it.

Proxy requirements: ability to route based on choice of routable
elements

4.4 Behavioral Requirements Summary

If the SIP-T originator is a gateway that received an ISUP request,
it must always perform both encapsulation and translation ISUP,
regardless of where the originator might guess that the request will
terminate.

If the terminator does not understand ISUP, it ignores it while
performing standard SIP processing. If the terminator does
understand ISUP, and needs to signal to the PSTN, it should reuse the
encapsulated ISUP if it understands the variant. The terminator
should perform the following steps:

o Extract the ISUP from the message body, and use this ISUP as a
   message template. Note that if there is no encapsulated ISUP in
   the message, the gateway should use a canonical template for the
   message type in question (a pre-populated ISUP message configured
   in the gateway) instead.

   o Translate the headers of the SIP request into ISUP parameters,
      overwriting any values in the message template.

   o Apply any local policies in populating parameters.

   An intermediary must be able to route a call based on the choice of
   routable elements in the SIP headers.

5. Components of the SIP-T Protocol

   The mechanisms described in the following sections are the components
   of SIP-T that provide the protocol functions entailed by the
   requirements.

5.1 Core SIP

   SIP-T uses the methods and procedures of SIP as defined by RFC 3261.

5.2 Encapsulation

   Encapsulation of the PSTN signaling is one of the major requirements
   of SIP-T. SIP-T uses multipart MIME bodies to enable SIP messages to
   contain multiple payloads (Session Description Protocol or SDP [5],
   ISUP, etc.). Numerous ISUP variants are in existence today; the ISUP
   MIME type enable recipients too recognize the ISUP type (and thus
   determine whether or not they support the variant) in the most
   expeditious possible manner. One scheme for performing ISUP
   encapsulation using multi-part MIME has been described in [2].

5.3 Translation

   Translation encompasses all aspects of signaling protocol conversion
   between SIP and ISUP. There are essentially two components to the
   problem of translation:

   1. ISUP SIP message mapping: This describes a mapping between ISUP
       and SIP at the message level. In SIP-T deployments gateways are
       entrusted with the task of generating a specific ISUP message for
       each SIP message received and vice versa. It is necessary to
       specify the rules that govern the mapping between ISUP and SIP
       messages (i.e., what ISUP messages is sent when a particular SIP
       message is received: an IAM must be sent on receipt of an INVITE,
       a REL for BYE, and so on). A potential mapping between ISUP and
       SIP messages has been described in [10].

2. ISUP parameter-SIP header mapping: A SIP request that is used to
    set up a telephone call should contain information that enables
    it to be appropriately routed to its destination by proxy servers
    in the SIP network - for example, the telephone number dialed by
    the originating user. It is important to standardize a set of
    practices that defines the procedure for translation of
    information from ISUP to SIP (for example, the Called Party
    Number in an ISUP IAM must be mapped onto the SIP 'To' header
    field and Request-URI, etc.). This issue becomes inherently more
    complicated by virtue of the fact that the headers of a SIP
    request (especially an INVITE) may be transformed by
    intermediaries, and that consequently, the SIP headers and
    encapsulated ISUP bodies come to express conflicting values -
    effectively, a part of the encapsulated ISUP may be rendered
    irrelevant and obsolete.

5.4 Support for mid-call signaling

Pure SIP does not have any provision for carrying any mid-call
control information that is generated during a session. The INFO [3]
method should be used for this purpose. Note however that INFO is
not suitable for managing overlap dialing (for one way of
implementing overlap dialing see [11]). Also note that the use of
INFO for signaling mid-call DTMF signals is not recommended (see
RFC2833 [9] for a recommended mechanism).

6. SIP Content Negotiation

The originator of a SIP-T request might package both SDP and ISUP
elements into the same SIP message by using the MIME multipart
format. Traditionally in SIP, if the terminating device does not
support a multipart payload (multipart/mixed) and/or the ISUP MIME
type, it would then reject the SIP request with a 415 Unsupported
Media Type specifying the media types it supports (by default,
'application/SDP'). The originator would subsequently have to re-
send the SIP request after stripping out the ISUP payload (i.e. with
only the SDP payload) and this would then be accepted.

This is a rather cumbersome flow, and it is thus highly desirable to
have a mechanism by which the originator could signify which bodies
are required and which are optional so that the terminator can
silently discard optional bodies that it does not understand
(allowing a SIP phone to ignore an ISUP payload when processing ISUP
is not critical). This is contingent upon the terminator having
support for a Content-type of multipart/mixed and access to the
Content-Disposition header to express criticality.

1. Support for ISUP is optional. Therefore, UA2 accepts the INVITE
    irrespective of whether it can process the ISUP.

```
UA1                    UA2
INVITE-->
   (Content-type:multipart/mixed;
    Content-type: application/sdp;
    Content-disposition: session; handling=required;
    Content-type: application/isup;
    Content-disposition: signal; handling=optional;)

                    <--18x
```

2. Support for ISUP is preferred. UA2 does not support the ISUP and
    rejects the INVITE with a 415 Unsupported Media Type. UA1 strips
    off the ISUP and re-sends the INVITE with SDP only and this is
    the accepted.

```
UA1                    UA2
INVITE--> (Content-type:multipart/mixed;
    Content-type: application/sdp;
    Content-disposition: session; handling=required;
    Content-type: application/isup;
    Content-disposition: signal; handling=required;)


                    <--415
                 (Accept: application/sdp)

ACK-->

INVITE-->
(Content-type: application/sdp)

                    <--18x
```

3. Support for ISUP is mandatory for call establishment. UA2 does
    not support the ISUP and rejects the INVITE with a 415
    Unsupported Media type. UA1 then directs its request to UA3.

```
   UA1                    UA2
   INVITE--> (Content-type:multipart/mixed;
      Content-type: application/sdp;
      Content-disposition: session; handling=required;
      Content-type: application/isup;
      Content-disposition: signal; handling=required;)

                    <--415
              (Accept: application/sdp)

   ACK-->

   UA1                    UA3
   INVITE--> (Content-type:multipart/mixed;
       Content-type: application/sdp;
       Content-disposition: session; handling=required;
       Content-type: application/isup;
       Content-disposition: signal; handling=required;)
```

Note that the exchanges of messages above are not complete; only the messages relevant to this discussion are shown. Specifics of the ISUP MIME type can be obtained from [2]. The 'version' and 'base' parameters are not shown here, but must be used in accordance with the rules of [2].

7. Security Considerations

   SIP-T can be employed as an interdomain signaling mechanism that may be subject to pre-existing trust relationships between administrative domains. In many legal environments, distribution of ISUP is restricted to licensed carriers; SIP-T introduces some challenges in so far as it bridges carrier signaling with end-user signaling. Any administrative domain implementing SIP-T should have an adequate security apparatus (including elements that manage any appropriate policies to manage fraud and billing in an interdomain environment) in place to ensure that the transmission of ISUP information does not result in any security violations.

   Transporting ISUP in SIP bodies may provide opportunities for abuse, fraud, and privacy concerns, especially when SIP-T requests can be generated, inspected or modified by arbitrary SIP endpoints. ISUP MIME bodies should be secured (preferably with S/MIME [4]) to alleviate this concern, as is described in the Security Considerations of the core SIP specification [1]. Authentication properties provided by S/MIME would allow the recipient of a SIP-T message to ensure that the ISUP MIME body was generated by an

authorized entity. Encryption would ensure that only carriers possessing a particular decryption key are capable of inspecting encapsulated ISUP MIME bodies in a SIP request.

SIP-T endpoints MUST support S/MIME signatures (CMS SignedData), and SHOULD support encryption (CMS EnvelopedData).

8. IANA Considerations

This document introduces no new considerations for IANA.

Normative References

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[4]   Ramsdell, B., "S/MIME Version 3 Message Specification", RFC 2633, June 1999.

[5]   Handley, M. and V. Jacobson, "SDP: Session Description Protocol", RFC 2327, April 1998.

Non-Normative References

[6]   International Telecommunications Union, "Signaling System No. 7; ISDN User Part Signaling procedures", ITU-T Q.764, September 1997, <http://www.itu.int>.

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[10]   Camarillo, G., Roach, A., Peterson, J. and L. Ong, "ISUP to SIP Mapping", Work in Progress.

   [11] Camarillo, G., Roach, A., Peterson, J. and L. Ong, "Mapping of
         ISUP Overlap Signaling to SIP", Work in Progress.

Appendix A. Notes

   1. Some terminating MGCs may alter the encapsulated ISUP in order to
       remove any conditions specific to the originating circuit; for
       example, continuity test flags in the Nature of Connection
       Indicators, etc.

   2. Even so, the relevance of ANSI-specific information in an ETSI
       network (or vice versa) is questionable. Clearly, the strength
       of SIP-T is realized when the encapsulated ISUP involves the
       usage of proprietary parameters.

Appendix B. Acknowledgments

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