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Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping

Status of this Memo

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Abstract

This document describes a way to perform the mapping between two signaling protocols: the Session Initiation Protocol (SIP) and the Integrated Services Digital Network (ISDN) User Part (ISUP) of Signaling System No. 7 (SS7). This mechanism might be implemented when using SIP in an environment where part of the call involves interworking with the Public Switched Telephone Network (PSTN).

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1. Introduction

SIP [1] is an application layer protocol for establishing, terminating and modifying multimedia sessions. It is typically carried over IP. Telephone calls are considered a type of multimedia sessions where just audio is exchanged.

Integrated Services Digital Network (ISDN) User Part (ISUP) [12] is a level 4 protocol used in Signaling System No. 7 (SS7) networks. It typically runs over Message Transfer Part (MTP) although it can also run over IP (see SCTP [19]). ISUP is used for controlling telephone calls and for maintenance of the network (blocking circuits, resetting circuits etc.).

A module performing the mapping between these two protocols is usually referred to as Media Gateway Controller (MGC), although the terms 'softswitch' or 'call agent' are also sometimes used. An MGC has logical interfaces facing both networks, the network carrying ISUP and the network carrying SIP. The MGC also has some capabilities for controlling the voice path; there is typically a Media Gateway (MG) with E1/T1 trunking interfaces (voice from Public Switched Telephone Network - PSTN) and with IP interfaces (Voice over IP - VoIP). The MGC and the MG can be merged together in one physical box or kept separate.

These MGCs are frequently used to bridge SIP and ISUP networks so that calls originating in the PSTN can reach IP telephone endpoints and vice versa. This is useful for cases in which PSTN calls need to take advantage of services in IP world, in which IP networks are used as transit networks for PSTN-PSTN calls, architectures in which calls originate on desktop 'softphones' but terminate at PSTN terminals, and many other similar next-generation telephone architectures.

This document describes logic and procedures which an MGC might use to implement the mapping between SIP and ISUP by illustrating the correspondences, at the message level and parameter level, between the protocols. It also describes the interplay between parallel state machines for these two protocols as a recommendation for implementers to synchronize protocol events in interworking architectures.

2. Scope

This document focuses on the translation of ISUP messages into SIP messages, and the mapping of ISUP parameters into SIP headers. For ISUP calls that traverse a SIP network, the purpose of translation is to allow SIP elements such as proxy servers (which do not typically understand ISUP) to make routing decisions based on ISUP criteria such as the called party number. This document consequently provides a SIP mapping only for those ISUP parameters which might be used by intermediaries in the routing of SIP requests. As a side effect of this approach, translation also increases the overall interoperability by providing critical information about the call to SIP endpoints that cannot understand encapsulated ISUP, or perhaps which merely cannot understand the particular ISUP variant encapsulated in a message.

This document also only takes into account the call functionality of ISUP. Maintenance messages dealing with PSTN trunks are treated only as far as they affect the control of an ongoing call; otherwise these messages neither have nor require any analog in SIP.

Messages indicating error or congestion situations in the PSTN (MTP-3) and the recovery mechanisms used such as User Part Available and User Part Test ISUP messages are outside the scope of this document

There are several flavors of ISUP. International Telecommunication Union Telecommunication Standardization Sector (ITU-T) International ISUP [12] is used through this document; some differences with the American National Standards Institute (ANSI) [11] ISUP and the Telecommunication Technology Committee (TTC) ISUP are also outlined. ITU-T ISUP is used in this document because it is the most widely known of all the ISUP flavors. Due to the small number of fields that map directly from ISUP to SIP, the signaling differences between ITU-T ISUP and specific national variants of ISUP will generally have little to no impact on the mapping. Note, however, that the ITU-T has not substantially standardized practices for Local Number Portability (LNP) since portability tends to be grounded in national numbering plan practices, and that consequently LNP must be described on a virtually per-nation basis. The number portability practices described in this document are presented as an optional mechanism.

Mapping of SIP headers to ISUP parameters in this document focuses largely on the mapping between the parameters found in the ISUP Initial Address Message (IAM) and the headers associated with the SIP INVITE message; both of these messages are used in their respective protocols to request the establishment of a call. Once an INVITE has been sent for a particular session, such headers as the To and From field become essentially fixed, and no further translation will be required during subsequent signaling, which is routed in accordance with Via and Route headers. Hence, the problem of parameter-to-header mapping in SIP-T is confined more or less to the IAM and the INVITE. Some additional detail is given in the population of parameters in the ISUP messages Address Complete Message (ACM) and Release Message (REL) based on SIP status codes.

This document describes when the media path associated with a SIP call is to be initialized, terminated, modified, etc., but it does not go into details such as how the initialization is performed or which protocols are used for that purpose.

3. Terminology

In this document, the key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "NOT RECOMMENDED", "MAY", and "OPTIONAL" are to be interpreted as described in RFC2119 [2] and indicate requirement levels for compliant SIP implementations.

4. Scenarios

There are several scenarios where ISUP-SIP mapping takes place. The way the messages are generated is different depending on the scenario.

When there is a single MGC and the call is from a SIP phone to a PSTN phone, or vice versa, the MGC generates the ISUP messages based on the methods described in this document.

```
+-------------+       +-----+       +-------------+
| PSTN switch +-------+ MGC +-------+ SIP UAC/UAS |
+-------------+       +-----+       +-------------+
```

The scenario where a call originates in the PSTN, goes into a SIP network and terminates in the PSTN again is known as "SIP bridging". SIP bridging should provide ISUP transparency between the PSTN switches handling the call. This is achieved by encapsulating the incoming ISUP messages in the body of the SIP messages (see [3]). In this case, the ISUP messages generated by the egress MGC are the ones present in the SIP body (possibly with some modifications; for example, if the called number in the request Uniform Resource Identifier - URI - is different from the one present in the ISUP due to SIP redirection, the ISUP message will need to be adjusted).

```
+------+   +-------------+   +-----+   +------------+   +------+
| PSTN +---+ Ingress MGC +---+ SIP +---+ Egress MGC +---+ PSTN |
+------+   +-------------+   +-----+   +------------+   +------+
```

SIP is used in the middle of both MGCs because the voice path has to be established through the IP network between both MGs; this structure also allows the call to take advantage of certain SIP services. ISUP messages in the SIP bodies provide further information (such as cause values and optional parameters) to the peer MGC.

In both scenarios, the ingress MGC places the incoming ISUP messages in the SIP body by default. Note that this has security implications; see Section 15. If the recipient of these messages (typically a SIP User Agent Client/User Agent Server - UAC/UAS) does not understand them, a negotiation using the SIP 'Accept' and 'Require' headers will take place and they will not be included in the next SIP message exchange.

There can be a Signaling Gateway (SG) between the PSTN and the MGC. It encapsulates the ISUP messages over IP in a manner such as the one described in [19]. The mapping described in this document is not affected by the underlying transport protocol of ISUP.

Note that overlap dialing mechanisms (use of the Subsequent Address Message - SAM) are outside the scope of this document. This document assumes that gateways facing ISUP networks in which overlap dialing is used will implement timers to insure that all digits have been collected before an INVITE is transmitted to a SIP network.

In some instances, gateways may receive incomplete ISUP messages which indicate message segmentation due to excessive message length. Commonly these messages will be followed by a Segmentation Message (SGM) containing the remainder of the original ISUP message. An incomplete message may not contain sufficient parameters to allow for a proper mapping to SIP; similarly, encapsulating (see below) an incomplete ISUP message may be confusing to terminating gateways. Consequently, a gateway MUST wait until a complete ISUP message is received (which may involve waiting until one or more SGMs arrive) before sending any corresponding INVITE.

5. SIP Mechanisms Required

For a correct mapping between ISUP and SIP, some SIP mechanisms above and beyond those available in the base SIP specification are needed. These mechanisms are discussed below. If the SIP UAC/UAS involved in the call does not support them, it is still possible to proceed, but the behavior in the establishment of the call may be slightly different than that expected by the user (e.g., other party answers before receiving the ringback tone, user is not informed about the call being forwarded, etc.).

5.1 'Transparent' Transit of ISUP Messages

To allow gateways to take advantage of the full range of services afforded by the existing telephone network when placing calls from PSTN to PSTN across a SIP network, SIP messages MUST be capable of transporting ISUP payloads from gateway to gateway. The format for encapsulating these ISUP messages is defined in [3].

SIP user agents which do not understand ISUP are permitted to ignore these optional MIME bodies.

5.2 Understanding MIME Multipart Bodies

In most PSTN interworking situations, SIP message bodies will be required to carry session information (Session Description Protocol - SDP) in addition to ISUP and/or billing information.

PSTN interworking nodes MUST understand the MIME type of "multipart/mixed" as defined in RFC2046 [4]. Clients express support for this by including "multipart/mixed" in an "Accept" header.

5.3 Transmission of Dual-Tone Multifrequency (DTMF) Information

How DTMF tones played by the user are transmitted by a gateway is completely orthogonal to how SIP and ISUP are interworked; however, as DTMF carriage is a component of a complete gatewaying solution some guidance is offered here.

Since the codec selected for voice transmission may not be ideally suited for carrying DTMF information, a symbolic method of transmitting this information in-band is desirable (since out-of-band transmission alone would provide many challenges for synchronization of the media stream for tone re-insertion). This transmission MAY be performed as described in RFC2833 [5].

5.4 Reliable Transmission of Provisional Responses

Provisional responses (in the 1xx class) are used in the transmission of call progress information. PSTN interworking in particular relies on these messages for control of the media channel and timing of call events.

When interworking with the PSTN, SIP messages MUST be sent reliably end-to-end; reliability of requests is guaranteed by the base protocol. One application-layer provisional reliability mechanism for responses is described in [18].

5.5 Early Media

Early media denotes the capability to play media (audio for telephony) before a SIP session has been established (before a 2xx response code has been sent). For telephony, establishment of media in the backwards direction is desirable so that tones and announcements can be played, especially when interworking with a network that cannot signal call status out of band (such as a legacy MF network). In cases where interworking has not been encountered, use of early media is almost always undesirable since it consumes inter-machine trunk recourses to play media for which no revenue is collected. Note that since an INVITE almost always contains the SDP required to send media in the backwards direction, and requires that user agents prepare themselves to receive backwards media as soon as an INVITE transmitted, the baseline SIP protocol has enough support to enable rudimentary unidirectional early media systems. However, this mechanism has a number of limitations - for example, media streams offered in the SDP of the INVITE cannot be modified or declined, and bidirectional RTCP required for session maintenance cannot be established.

Therefore gateways MAY support more sophisticated early media systems as they come to be better understood. One mechanism that provides a way of initiating a fully-featured early media system is described in [20].

Note that in SIP networks not just switches but also user agents can generate the 18x response codes and initiate early backwards media, and that therefore some gateways may wish to enforce policies that restrict the use of backwards media from arbitrary user agents (see Section 15).

5.6 Mid-Call Transactions which do not change SIP state

When interworking with the PSTN, there are situations when gateways will need to send messages to each other over SIP that do not correspond to any SIP operations.

In support of mid-call transactions and other ISUP events that do not correspond to existing SIP methods, SIP gateways MUST support the INFO method, defined in RFC2976 [6]. Note that this document does not prescribe or endorse the use of INFO to carry DTMF digits.

Gateways MUST accept "405 Method Not Allowed" and "501 Not Implemented" as non-fatal responses to INFO requests - that is, any call in progress MUST NOT be torn down if a destination so rejects an INFO request sent by a gateway.

5.7 Privacy Protection

ISUP has a concept of presentation restriction - a mechanism by which a user can specify that they would not like their telephone number to be displayed to the person they are calling (presumably someone with Caller ID). When a gateway receives an ISUP request that requires presentation restriction, it must therefore shield the identity of the caller in some fashion.

The base SIP protocol supports a method of specifying that a user is anonymous. However, this system has a number of limitations - for example, it reveals the identity of the gateway itself, which could be a privacy-impacting disclosure. Therefore gateways MAY support more sophisticated privacy systems. One mechanism that provides a way of supporting fully-featured privacy negotiation (which interacts well with identity management systems) is described in [9B].

5.8 CANCEL causes

There is a way in ISUP to signal that you would like to discontinue an attempt to set up a call - the general-purpose REL is sent in the forwards direction. There is a similar concept in SIP - that of a CANCEL request that is sent in order to discontinue the establishment of a SIP dialog. For various reasons, however, CANCEL requests cannot contain message bodies, and therefore in order to carry the important information in the REL (the cause code) end-to-end in sip bridging cases, ISUP encapsulation cannot be used.

Ordinarily, this is not a big problem, because for practical purposes the only reason that a REL is ever issued to cancel a call setup attempt is that a user hangs up the phone while it is still ringing (which results in a "Normal clearing" cause code). However, under exceptional conditions, like catastrophic network failure, a REL may be sent with a different cause code, and it would be handy if a SIP network could carry the cause code end-to-end. Therefore gateways MAY support a mechanism for end-to-end delivery of such failure reasons. One mechanism that provides this capability is described in [9].

6. Mapping

The mapping between ISUP and SIP is described using call flow diagrams and state machines. One state machine handles calls from SIP to ISUP and the second from ISUP to SIP. There are details, such as some retransmissions and some states (waiting for the Release Complete Message - RLC, waiting for SIP ACK etc.), that are not shown in the figures in order to make them easier to follow.

The boxes represent the different states of the gateway, and the arrows show changes in the state. The event that triggers the change in the state and the actions to take appear on the arrow: event / section describing the actions to take.

For example, 'INVITE / 7.2.1' indicates that an INVITE request has been received by the gateway, and the procedure upon reception is described in the section 7.2.1 of this document.

It is RECOMMENDED that gateways implement functional equivalence with the call flows detailed in Section 7.1 and Section 8.1. Deviations from these flows are permissible in support of national ISUP variants, or any of the conservative policies recommended in Section 15.

7. SIP to ISUP Mapping

7.1 SIP to ISUP Call flows

The following call flows illustrate the order of messages in typical success and error cases when setting up a call initiated from the SIP network. "100 Trying" acknowledgements to INVITE requests are not displayed below although they are required in many architectures.

In these diagrams, all call signaling (SIP, ISUP) is going to and from the MGC; media handling (e.g., audio cut-through, trunk freeing) is being performed by the MG, under the control of the MGC. For the purpose of simplicity, these are shown as a single node, labeled "MGC/MG."

7.1.1 En-bloc Call Setup (no auto-answer)

```
SIP                       MGC/MG                      PSTN
 1|---------INVITE---------->|                          |
  |<----------100------------|                          |
  |                          |------------IAM---------->|2
  |                          |<=========Audio===========|
  |                          |<-----------ACM-----------|3
 4|<----------18x------------|                          |
  |<=========Audio===========|                          |
  |                          |<-----------CPG-----------|5
 6|<----------18x------------|                          |
  |                          |<-----------ANM-----------|7
  |                          |<=========Audio==========>|
 8|<----------200------------|                          |
  |<=========Audio==========>|                          |
 9|-----------ACK----------->|                          |
```

1. When a SIP user wishes to begin a session with a PSTN user, the SIP node issues an INVITE request.

2. Upon receipt of an INVITE request, the gateway maps it to an IAM message and sends it to the ISUP network.

3. The remote ISUP node indicates that the address is sufficient to set up a call by sending back an ACM message.

4. The "called party status" code in the ACM message is mapped to a SIP provisional response (as described in Section 7.2.5 and Section 7.2.6) and returned to the SIP node. This response may contain SDP to establish an early media stream (as shown in the diagram). If no SDP is present, the audio will be established in both directions after step 8.

5. If the ISUP variant permits, the remote ISUP node may issue a variety of Call Progress (CPG) messages to indicate, for example, that the call is being forwarded.

6. Upon receipt of a CPG message, the gateway will map the event code to a SIP provisional response (see Section 7.2.9) and send it to the SIP node.

7. Once the PSTN user answers, an Answer (ANM) message will be sent to the gateway.

8. Upon receipt of the ANM, the gateway will send a 200 message to the SIP node.

9. The SIP node, upon receiving an INVITE final response (200), will send an ACK to acknowledge receipt.

7.1.2 Auto-answer call setup

```
SIP                        MGC/MG                       PSTN
 1|---------INVITE---------->|                           |
  |<----------100------------|                           |
  |                          |------------IAM----------->|2
  |                          |<=========Audio===========|
  |                          |<-----------CON-----------|3
  |                          |<=========Audio==========>|
 4|<----------100------------|                           |
  |<=========Audio==========>|                           |
 5|-----------ACK----------->|                           |
```

Note that this flow is not supported in ANSI networks.

1. When a SIP user wishes to begin a session with a PSTN user, the SIP node issues an INVITE request.

2. Upon receipt of an INVITE request, the gateway maps it to an IAM message and sends it to the ISUP network.

3. Since the remote node is configured for automatic answering, it will send a Connect Message (CON) upon receipt of the IAM. (For ANSI, this message will be an ANM).

4. Upon receipt of the CON, the gateway will send a 200 message to the SIP node.

5. The SIP node, upon receiving an INVITE final response (200), will send an ACK to acknowledge receipt.

7.1.3 ISUP T7 Expires

```
       SIP                       MGC/MG                       PSTN
        1|---------INVITE---------->|                           |
         |<----------100------------|                           |
         |                          |------------IAM---------->|2
         |                          |<=========Audio===========|
         |                          |    *** T7 Expires ***     |
         |             ** MG Releases PSTN Trunk **             |
        4|<----------504------------|------------REL---------->|3
        5|-----------ACK----------->|                           |
```

1. When a SIP user wishes to begin a session with a PSTN user, the SIP node issues an INVITE request.

2. Upon receipt of an INVITE request, the gateway maps it to an IAM message and sends it to the ISUP network. The ISUP timer T7 is started at this point.

3. The ISUP timer T7 expires before receipt of an ACM or CON message, so a REL message is sent to cancel the call.

4. A gateway timeout message is sent back to the SIP node.

5. The SIP node, upon receiving an INVITE final response (504), will send an ACK to acknowledge receipt.

7.1.4 SIP Timeout

```
       SIP                       MGC/MG                     PSTN
        1|---------INVITE---------->|                          |
         |<----------100------------|                          |
         |                          |------------IAM---------->|2
         |                          |<=========Audio===========|
         |                          |<-----------CON-----------|3
         |                          |<=========Audio==========>|
        4|<----------200------------|                          |
         |     *** T1 Expires ***   |                          |
         |<----------200------------|                          |
         |     *** T1 Expires ***   |                          |
         |<----------200------------|                          |
         |     *** T1 Expires ***   |                          |
         |<----------200------------|                          |
         |     *** T1 Expires ***   |                          |
         |<----------200------------|                          |
         |     *** T1 Expires ***   |                          |
         |<----------200------------|                          |
         |     *** T1 Expires ***   |                          |
        5|<----------200------------|                          |
         |     *** T1 Expires ***   |                          |
         |             ** MG Releases PSTN Trunk **            |
        7|<----------BYE------------|------------REL---------->|6
         |                          |<-----------RLC-----------|8
```

1. When a SIP user wishes to begin a session with a PSTN user, the SIP node issues an INVITE request.

2. Upon receipt of an INVITE request, the gateway maps it to an IAM message and sends it to the ISUP network.

3. Since the remote node is configured for automatic answering, it will send a CON message upon receipt of the IAM. In ANSI flows, rather than a CON, an ANM (without ACM) would be sent.

4. Upon receipt of the ANM, the gateway will send a 200 message to the SIP node and set SIP timer T1.

5. The response is retransmitted every time the SIP timer T1 expires.

6. After seven retransmissions, the call is torn down by sending a REL to the ISUP node, with a cause code of 102 (recover on timer expiry).

7. A BYE is transmitted to the SIP node in an attempt to close the call. Further handling for this clean up is not shown, since the SIP node's state is not easily known in this scenario.

8. Upon receipt of the REL message, the remote ISUP node will reply with an RLC message.

7.1.5 ISUP Setup Failure

```
SIP                       MGC/MG                        PSTN
 1|---------INVITE---------->|                            |
  |<----------100------------|                            |
  |                          |------------IAM---------->|2
  |                          |<-----------REL-----------|3
  |                          |------------RLC---------->|4
 5|<----------4xx+-----------|                            |
 6|-----------ACK----------->|                            |
```

1. When a SIP user wishes to begin a session with a PSTN user, the SIP node issues an INVITE request.

2. Upon receipt of an INVITE request, the gateway maps it to an IAM message and sends it to the ISUP network.

3. Since the remote ISUP node is unable to complete the call, it will send a REL.

4. The gateway releases the circuit and confirms that it is available for reuse by sending an RLC.

5. The gateway translates the cause code in the REL to a SIP error response (see Section 7.2.4) and sends it to the SIP node.

6. The SIP node sends an ACK to acknowledge receipt of the INVITE final response.

7.1.6 Cause Present in ACM Message

```
       SIP                       MGC/MG                      PSTN
        1|---------INVITE---------->|                           |
         |<----------100------------|                           |
         |                          |------------IAM---------->|2
         |                          |<=========Audio===========|
         |                          |<---ACM with cause code---|3
        4|<------183 with SDP-------|                           |
         |<=========Audio===========|                           |
                     ** Interwork timer expires **
        5|<----------4xx+-----------|                           |
         |                          |------------REL---------->|6
         |                          |<-----------RLC-----------|7
        8|-----------ACK----------->|                           |
```

1. When a SIP user wishes to begin a session with a PSTN user, the SIP node issues an INVITE request.

2. Upon receipt of an INVITE request, the gateway maps it to an IAM message and sends it to the ISUP network.

3. Since the ISUP node is unable to complete the call and wants to generate the error tone/announcement itself, it sends an ACM with a cause code. The gateway starts an interwork timer.

4. Upon receipt of an ACM with cause (presence of the CAI parameter), the gateway will generate a 183 message towards the SIP node; this contains SDP to establish early media cut-through.

5. A final INVITE response, based on the cause code received in the earlier ACM message, is generated and sent to the SIP node to terminate the call. See Section 7.2.4.1 for the table which contains the mapping from cause code to SIP response.

6. Upon expiration of the interwork timer, a REL is sent towards the PSTN node to terminate the call. Note that the SIP node can also terminate the call by sending a CANCEL before the interwork timer expires. In this case, the signaling progresses as in Section 7.1.7.

7. Upon receipt of the REL message, the remote ISUP node will reply with an RLC message.

8. The SIP node sends an ACK to acknowledge receipt of the INVITE final response.

7.1.7 Call Canceled by SIP

```
       SIP                       MGC/MG                       PSTN
        1|---------INVITE---------->|                          |
         |<----------100------------|                          |
         |                          |------------IAM---------->|2
         |                          |<=========Audio===========|
         |                          |<-----------ACM-----------|3
        4|<----------18x------------|                          |
         |<=========Audio===========|                          |
         |            ** MG Releases IP Resources **           |
        5|----------CANCEL--------->|                          |
        6|<----------200------------|                          |
         |             ** MG Releases PSTN Trunk **            |
         |                          |------------REL---------->|7
        8|<----------487------------|                          |
         |                          |<-----------RLC-----------|9
       10|-----------ACK----------->|                          |
```

1. When a SIP user wishes to begin a session with a PSTN user, the SIP node issues an INVITE request.

2. Upon receipt of an INVITE request, the gateway maps it to an IAM message and sends it to the ISUP network.

3. The remote ISUP node indicates that the address is sufficient to set up a call by sending back an ACM message.

4. The "called party status" code in the ACM message is mapped to a SIP provisional response (as described in Section 7.2.5 and Section 7.2.6) and returned to the SIP node. This response may contain SDP to establish an early media stream.

5. To cancel the call before it is answered, the SIP node sends a CANCEL request.

6. The CANCEL request is confirmed with a 200 response.

7. Upon receipt of the CANCEL request, the gateway sends a REL message to terminate the ISUP call.

8. The gateway sends a "487 Call Cancelled" message to the SIP node to complete the INVITE transaction.

9. Upon receipt of the REL message, the remote ISUP node will reply with an RLC message.

10. Upon receipt of the 487, the SIP node will confirm reception with an ACK.

7.2 State Machine

Note that REL can be received in any state; the handling is the same for each case (see Section 10).

```
                               +---------+
      +----------------------->|  Idle   |<---------------------+
      |                        +----+----+                      |
      |                             |                           |
      |                             | INVITE/6.2.1              |
      |                             V                           |
      |      T7/6.2.2   +-------------------------+   REL/6.2.4 |
      +<----------------+         Trying          +------------>+
      |                 +-+--------+------+-------+             |
      |    CANCEL/6.2.3 | |        |      |                     |
      +<----------------+ | E.ACM/ | ACM/ | CON/ANM             |
      |                   | 6.2.5  |6.2.6 | 6.2.7               |
      |                   V        |      |                     |
      | T9/6.2.8  +--------------+ |      |                     |
      +<----------+ Not alerting | |      |                     |
      |           +-------+------+ |      |                     |
      |  CANCEL/6.2.3 |   |        |      |                     |
      |<--------------+   | CPG/   |      |                     |
      |                   | 6.2.9  |      |                     |
      |                   V        V      |                     |
      |    T9/6.2.8     +---------------+ |    REL/6.2.4        |
      +<----------------+    Alerting   |-|-------------------->|
      |<----------------+--+-----+------+ |                     |
      |  CANCEL/6.2.3      |  ^  |        |                     |
      |               CPG/ |  |  | ANM/   |                     |
      |              6.2.9 +--+  | 6.2.7  |                     |
      |                          V        V                     |
      |                 +-------------------------+    REL/9.2  |
      |                 |     Waiting for ACK     |------------>|
      |                 +-------------+-----------+             |
      |                               |                         |
      |                               | ACK/6.2.10              |
      |                               V                         |
      |     BYE/9.1     +-------------------------+    REL/9.2  |
      +<----------------+        Connected        +------------>+
                        +-------------------------+
```

7.2.1 INVITE received

When an INVITE request is received by the gateway, a "100 Trying" response MAY be sent back to the SIP network indicating that the gateway is handling the call.

The necessary hardware resources for the media stream MUST be reserved in the gateway when the INVITE is received, since an IAM message cannot be sent before the resource reservation (especially TCIC selection) takes place. Typically the resources consist of a time slot in an E1/T1 and an RTP/UDP port on the IP side. Resources might also include any quality-of-service provisions (although no such practices are recommended in this document).

After sending the IAM the timer T7 is started. The default value of T7 is between 20 and 30 seconds. The gateway goes to the ’Trying’ state.

7.2.1.1 INVITE to IAM procedures

This section details the mapping of the SIP headers in an INVITE message to the ISUP parameters in an Initial Address Message (IAM). A PSTN-SIP gateway is responsible for creating an IAM when it receives an INVITE.

Five mandatory parameters appear within the IAM message: the Called Party Number (CPN), the Nature of Connection Indicator (NCI), the Forward Call Indicators (FCI), the Calling Party’s Category (CPC), and finally a parameter that indicates the desired bearer characteristics of the call - in some ISUP variants the Transmission Medium Requirement (TMR) is required, in others the User Service Information (USI) (or both). All IAM messages MUST contain these five parameters at a minimum. Thus, every gateway must have a means of populating each of those five parameters when an INVITE is received. Many of the values that will appear in these parameters (such as the NCI or USI) will most likely be the same for each IAM created by the gateway. Others (such as the CPN) will vary on a call-by-call basis; the gateway extracts information from the INVITE in order to properly populate these parameters.

There are also quite a few optional parameters that can appear in an IAM message; Q.763 [17] lists 29 in all. However, each of these parameters need not to be translated in order to achieve the goals of SIP-ISUP mapping. As is stated above, translation allows SIP network elements to understand the basic PSTN context of the session (who it is for, and so on) if they are not capable of deciphering any encapsulated ISUP. Parameters that are only meaningful to the PSTN will be carried through PSTN-SIP- PSTN networks via encapsulation -

translation is not necessary for these parameters. Of the aforementioned 29 optional parameters, only the following are immediately useful for translation: the Calling Party's Number (CIN, which is commonly present), Transit Network Selection (TNS), Carrier Identification Parameter (CIP, present in ANSI networks), Original Called Number (OCN), and the Generic Digits (known in some variants as the Generic Address Parameter (GAP)).

When a SIP INVITE arrives at a PSTN gateway, the gateway SHOULD attempt to make use of encapsulated ISUP (see [3]), if any, within the INVITE to assist in the formulation of outbound PSTN signaling, but SHOULD also heed the security considerations in Section 15. If possible, the gateway SHOULD reuse the values of each of the ISUP parameters of the encapsulated IAM as it formulates an IAM that it will send across its PSTN interface. In some cases, the gateway will be unable to make use of that ISUP - for example, if the gateway cannot understand the ISUP variant and must therefore ignore the encapsulated body. Even when there is comprehensible encapsulated ISUP, the relevant values of SIP header fields MUST 'overwrite' through the process of translation the parameter values that would have been set based on encapsulated ISUP. In other words, the updates to the critical session context parameters that are created in the SIP network take precedence, in ISUP-SIP-ISUP bridging cases, over the encapsulated ISUP. This allows many basic services, including various sorts of call forwarding and redirection, to be implemented in the SIP network.

For example, if an INVITE arrives at a gateway with an encapsulated IAM with a CPN field indicating the telephone number +12025332699, but the Request-URI of the INVITE indicates 'tel:+15105550110', the gateway MUST use the telephone number in the Request-URI, rather than the one in the encapsulated IAM, when creating the IAM that the gateway will send to the PSTN. Further details of how SIP header fields are translated into ISUP parameters follow.

Gateways MUST be provisioned with default values for mandatory ISUP parameters that cannot be derived from translation(such as the NCI or TMR parameters) for those cases in which no encapsulated ISUP is present. The FCI parameter MUST also have a default, as only the 'M' bit of the default may be overwritten during the process of translation if the optional number portability translation mechanisms described below are used.

The first step in the translation of the fields of an INVITE message to the parameters of an IAM is the inspection of the Request-URI.

If the optional number portability practices are supported by the gateway, then the following steps related to handling of the 'npdi' and 'rn' parameters of the Request-URI should be followed.

If there is no 'npdi=yes' field within the Request-URI, then the primary telephone number in the tel URL (the digits immediately following 'tel:') MUST be converted to ISUP format, following the procedures described in Section 12, and used to populate the CPN parameter.

If the 'npdi=yes' field exists in the Request-URI, then the FCI parameter bit for 'number translated' within the IAM MUST reflect that a number portability dip has been performed.

If in addition to the 'npdi=yes' field there is no 'rn=' field present, then the main telephone number in the tel URL MUST be converted to ISUP format (see Section 12) and used to populate the CPN parameter. This indicates that a portability dip took place, but that the called party's number was not ported.

If in addition to the 'npdi=yes' field an 'rn=' field is present, then in ANSI ISUP the 'rn=' field MUST be converted to ISUP format and used to populate the CPN. The main telephone number in the tel URL MUST be converted to ISUP format and used to populate the Generic Digits Parameter (or GAP in ANSI). In some other ISUP variants, the number given in the 'rn=' field would instead be prepended to the main telephone number (with or without a prefix or separator) and the combined result MUST be used to populate the CPN. Once the 'rn=' and 'npdi=' parameters have been translation, the number portability translation practices are complete.

The following mandatory translation practices are performed after number portability translations, if any.

If number portability practices are not supported by the gateway, then the primary telephone number in the tel URL (the digits immediately following 'tel:') MUST be converted to ISUP format, following the procedures described in Section 12, and used to populate the CPN parameter.

If the primary telephone number in the Request-URI and that of the To header are at variance, then the To header SHOULD be used to populate an OCN parameter. Otherwise the To header SHOULD be ignored.

Some optional translation procedures are provided for carrier-based routing. If the 'cic=' parameter is present in the Request-URI, the gateway SHOULD consult local policy to make sure that it is appropriate to transmit this Carrier Identification Code (CIC, not to be confused with the MTP3 'circuit identification code') in the IAM; if the gateway supports many independent trunks, it may need to choose a particular trunk that points to the carrier identified by the CIC, or a tandem through which that carrier is reachable. Policies for such trunks (based on the preferences of the carriers with which the trunks are associated and the ISUP variant in use) SHOULD dictate whether the CIP or TNS parameter is used to carry the CIC. In the absence of any pre-arranged policies, the TNS should be used when the CPN parameter is in an international format (i.e., the tel URL portion of the Request-URI is preceded by a '+', which will generate a CPN in international format), and (where supported) the CIP should be used in other cases.

When a SIP call has been routed to a gateway, then the Request-URI will most likely contain a tel URL (or a SIP URI with a tel URL user portion) - SIP-ISUP gateways that receive Request-URIs that do not contain valid telephone numbers SHOULD reject such requests with an appropriate response code. Gateways SHOULD however continue to process requests with a From header field that does not contain a telephone number, as will sometimes be the case if a call originated at a SIP phone that employs a SIP URI user@host convention. The CIN parameter SHOULD be omitted from the outbound IAM if the From field is unusable. Note that as an alternative, gateway implementers MAY consider some non-standard way of mapping particular SIP URIs to telephone numbers.

When a gateway receives a message with (comprehensible) encapsulated ISUP, it MUST set the FCI indicator in the generated IAM so that all interworking-related bits have the same values as their counterparts in the encapsulated ISUP. In most cases, these indicators will state that no interworking was encountered, unless interworking has been encountered somewhere else in the call path. If usable encapsulated ISUP is not present in an INVITE received by the gateway, it is STRONGLY RECOMMENDED that the gateway set the Interworking Indicator bit of the FCI to 'no interworking' and the ISDN User Part Indicator to 'ISUP used all the way'; the gateway MAY also set the Originating Access indicator to 'Originating access non-ISDN' (generally, it is not safe to assume that SIP phones will support ISDN endpoint services, and the procedures in this document do not detail mappings to translate all such services).

Note that when 'interworking encountered' is set in the FCI parameter of the IAM, this indicates that ISUP is interworking with a network which is not capable of providing as many services as ISUP does. ISUP networks will therefore not employ certain features they otherwise normally would, including potentially the use of ISDN cause codes in failure conditions (as opposed to sending ACMs followed by audible announcements). If desired, gateway vendors MAY provide a configurable option, usable at the discretion of service providers, that will signal in the FCI that interworking has been encountered (and that ISUP is not used all the way) when encapsulated ISUP is not present; however, doing so may significantly limit the efficiency and transparency of SIP-ISUP translation.

Claiming to be an ISDN node might make the callee request ISDN user to user services. Since user to user services 1 and 2 must be requested by the caller, they do not represent a problem (see [14]). User to user service 3 can be requested by the callee also. In non-SIP bridging situations, the MGC should be capable of rejecting this service request.

7.2.2 ISUP T7 expires

Since no response was received from the PSTN all the resources in the MG are released. A '504 Server Timeout' SHOULD be sent back to the SIP network. A REL message with cause value 102 (protocol error, recovery on timer expiry) SHOULD be sent to the PSTN. Gateways can expect the PSTN to respond with RLC and the SIP network to respond with an ACK indicating that the release sequence has been completed.

7.2.3 CANCEL or BYE received

If a CANCEL or BYE request is received before a final SIP response has been sent, a '200 OK' MUST be sent to the SIP network to confirm the CANCEL or BYE; a 487 MUST also be sent to terminate the INVITE transaction. All the resources are released and a REL message SHOULD be sent to the PSTN with cause value 16 (normal clearing). Gateways can expect an RLC from the PSTN to be received indicating that the release sequence is complete.

In SIP bridging situations, a REL might be encapsulated in the body of a BYE request. Although BYE is usually mapped to cause code 16 (normal clearing), under exceptional circumstances the cause code in the REL message might be different. Therefore the Cause Indicator parameter of the encapsulated REL should be re-used in the REL sent to the PSTN.

Note that a BYE or CANCEL request may contain a Reason header that SHOULD be mapped to the Cause Indicator parameter (see Section 5.8). If a BYE contains both a Reason header and encapsulated ISUP, the value in the Reason header MUST be preferred.

All the resources in the gateway SHOULD be released before the gateway sends any REL message.

7.2.4 REL received

This section applies when a REL is received before a final SIP response has been sent. Typically, this condition arises when a call has been rejected by the PSTN.

Any gateway resources SHOULD be released immediately and an RLC MUST be sent to the ISUP network to indicate that the circuit is available for reuse.

If the INVITE that originated this transaction contained a legitimate and comprehensible encapsulated ISUP message (i.e., an IAM using a variant supported by the gateway, preferably with a digital signature), then encapsulated ISUP SHOULD be sent in the response to the INVITE when possible (since this suggests an ISUP-SIP-ISUP bridging case) - therefore, the REL message just received SHOULD be included in the body of the SIP response. The gateway SHOULD NOT return a response with encapsulated ISUP if the originator of the INVITE did not enclose ISUP itself.

Note that the receipt of certain maintenance messages in response to IAM such as Blocking Message (BLO) or Reset Message (RSC) (or their circuit group message equivalents) may also result in the teardown of calls in this phase of the state machine. Behavior for maintenance messages is given below in Section 11.

7.2.4.1 ISDN Cause Code to Status Code Mapping

The use of the REL message in the SS7 network is very general, whereas SIP has a number of specific tools that, collectively, play the same role as REL - namely BYE, CANCEL, and the various status/response codes. An REL can be sent to tear down a call that is already in progress (BYE), to cancel a previously sent call setup request that has not yet been completed (CANCEL), or to reject a call setup request (IAM) that has just been received (corresponding to a SIP status code).

Note that it is not necessarily appropriate to map some ISDN cause codes to SIP messages because these cause codes are only meaningful to the ISUP interface of a gateway. A good example of this is cause code 44 "Request circuit or channel not available." 44 signifies that the CIC for which an IAM had been sent was believed by the receiving equipment to be in a state incompatible with a new call request - however, the appropriate behavior in this case is for the originating switch to re-send the IAM for a different CIC, not for the call to be torn down. Clearly, there is not (nor should there be) an SIP status code indicating that a new CIC should be selected - this matter is internal to the originating gateway. Hence receipt of cause code 44 should not result in any SIP status code being sent; effectively, the cause code is untranslatable.

If a cause value other than those listed below is received, the default response '500 Server internal error' SHOULD be used.

Finally, in addition to the ISDN Cause Code, the CAI parameter also contains a cause 'location' that gives some sense of which entity in the network was responsible for terminating the call (the most important distinction being between the user and the network). In most cases, the cause location does not affect the mapping to a SIP status code; some exceptions are noted below. A diagnostic field may also be present for some ISDN causes; this diagnostic will contain additional data pertaining to the termination of the call.

The following mapping values are RECOMMENDED:

Normal event

ISUP Cause value	SIP response
1 unallocated number	404 Not Found
2 no route to network	404 Not found
3 no route to destination	404 Not found
16 normal call clearing	--- (*)
17 user busy	486 Busy here
18 no user responding	408 Request Timeout
19 no answer from the user	480 Temporarily unavailable
20 subscriber absent	480 Temporarily unavailable
21 call rejected	403 Forbidden (+)
22 number changed (w/o diagnostic)	410 Gone
22 number changed (w/ diagnostic)	301 Moved Permanently
23 redirection to new destination	410 Gone
26 non-selected user clearing	404 Not Found (=)
27 destination out of order	502 Bad Gateway
28 address incomplete	484 Address incomplete
29 facility rejected	501 Not implemented
31 normal unspecified	480 Temporarily unavailable

(*) ISDN Cause 16 will usually result in a BYE or CANCEL

(+) If the cause location is 'user' than the 6xx code could be given rather than the 4xx code (i.e., 403 becomes 603)

(=) ANSI procedure - in ANSI networks, 26 is overloaded to signify 'misrouted ported number'. Presumably, a number portability dip should have been performed by a prior network. Otherwise cause 26 is usually not used in ISUP procedures.

A REL with ISDN cause 22 (number changed) might contain information about a new number where the callee might be reachable in the diagnostic field. If the MGC is able to process this information it SHOULD be added to the SIP response (301) in a Contact header.

Resource unavailable

This kind of cause value indicates a temporary failure. A 'Retry-After' header MAY be added to the response if appropriate.

ISUP Cause value	SIP response
34 no circuit available	503 Service unavailable
38 network out of order	503 Service unavailable
41 temporary failure	503 Service unavailable
42 switching equipment congestion	503 Service unavailable
47 resource unavailable	503 Service unavailable

Service or option not available

This kind of cause value indicates that there is a problem with the request, rather than something that will resolve itself over time.

ISUP Cause value	SIP response
55 incoming calls barred within CUG	403 Forbidden
57 bearer capability not authorized	403 Forbidden
58 bearer capability not presently available	503 Service unavailable

Service or option not available

ISUP Cause value	SIP response
65 bearer capability not implemented	488 Not Acceptable Here
70 only restricted digital avail	488 Not Acceptable Here
79 service or option not implemented	501 Not implemented

Invalid message

ISUP Cause value	SIP response
87 user not member of CUG	403 Forbidden
88 incompatible destination	503 Service unavailable

Protocol error

ISUP Cause value	SIP response
102 recovery of timer expiry	504 Gateway timeout
111 protocol error	500 Server internal error

Interworking

ISUP Cause value	SIP response
127 interworking unspecified	500 Server internal error

7.2.5 Early ACM received

An ACM message is sent in certain situations to indicate that the call is in progress in order to satisfy ISUP timers, rather than to signify that the callee is being alerted. This occurs for example in mobile networks, where roaming can delay call setup significantly. The early ACM is sent before the user is alerted to reset T7 and start T9. An ACM is considered an 'early ACM' if the Called Party's Status Indicator is set to 00 (no indication).

After sending an early ACM, the ISUP network can be expected to indicate the further progress of the call by sending CPGs.

When an early ACM is received the gateway SHOULD send a 183 Session Progress response (see [1]) to the SIP network. In SIP bridging situations (where encapsulated ISUP was contained in the INVITE that initiated this call) the early ACM SHOULD also be included in the response body.

Note that sending 183 before a gateway has confirmation that the address is complete (ACM) creates known problems in SIP bridging cases, and it SHOULD NOT therefore be sent.

7.2.6 ACM received

Most commonly, on receipt of an ACM a provisional response (in the 18x class) SHOULD be sent to the SIP network. If the INVITE that initiated this session contained legitimate and comprehensible encapsulated ISUP, then the ACM received by the gateway SHOULD be encapsulated in the provisional response.

If the ACM contains a Backward Call Indicators parameter with a value of 'subscriber free', the gateway SHOULD send a '180 Ringing' response. When a 180 is sent, it is assumed, in the absence of any early media extension, that any necessary ringback tones will be generated locally by the SIP user agent to which the gateway is responding (which may in turn be a gateway).

If the Backward Call Indicators (BCI) parameter of the ACM indicates that interworking has been encountered (generally designating that the ISUP network sending the ACM is interworking with a less sophisticated network which cannot report its status via out-of-band signaling), then there may be in-band announcements of call status such as an audible busy tone or caller intercept message, and if possible a backwards media transmission SHOULD be initiated. Backwards media SHOULD also be transmitted if the Optional Backward Call Indicators parameter field for in-band media is set. For more information on early media (before 200 OK/ANM) see Section 5.5. After early media transmission has been initiated, the gateway SHOULD send a 183 Session Progress response code.

Gateways MAY have some means of ascertaining the disposition of in-band audio media; for example, a way of determining by inspecting signaling in some ISUP variants, or by listening to the audio, that ringing, or a busy tone, is being played over the circuit. Such gateways MAY elect to discard the media and send the corresponding response code (such as 180 or 486) in its stead. However, the implementation of such a gateway would entail overcoming a number of known challenges that are outside the scope of this document.

When they receive an ACM, switches in many ISUP networks start a timer known as "T9" which usually lasts between 90 seconds and 3 minutes (see [13]). When early media is being played, this timer permits the caller to hear backwards audio media (in the form ringback, tones or announcements) from a remote switch in the ISUP network for that period of time without incurring any charge for the connection. The nearest possible local ISUP exchange to the callee generates the ringback tone or voice announcements. If longer announcements have to be played, the network has to send an ANM, which initiates bidirectional media of indefinite duration. In common ISUP network practice, billing commences when the ANM is received. Some networks do not support timer T9.

7.2.7 CON or ANM Received

When an ANM or CON message is received, the call has been answered and thus '200 OK' response SHOULD be sent to the SIP network. This 200 OK SHOULD contain an answer to the media offered in the INVITE. In SIP bridging situations (when the INVITE that initiated this call contained legitimate and comprehensible encapsulated ISUP), the ISUP message is included in the body of the 200 OK response. If it has not done so already, the gateway MUST establish a bidirectional media stream at this time.

When there is interworking with some legacy networks, it is possible for an ISUP switch to receive an ANM immediately after an early ACM (without CPG or any other backwards messaging), or without receiving any ACM at all (when an automaton answers the call). In this situation the SIP user will never have received a 18x provisional response, and consequently they will not hear any kind of ringtone before the callee answers. This may result in some clipping of the initial forward media from the caller (since forward media transmission cannot commence until SDP has been acquired from the destination). In ISDN (see [12]) this is solved by connecting the voice path backwards before sending the IAM.

7.2.8 Timer T9 Expires

The expiry of this timer (which is not used in all networks) signifies that an ANM has not arrived a significant period of time after alerting began (with the transmission of an ACM) for this call. Usually, this means that the callee's terminal has been alerted for many rings but has not been answered. It may also occur in interworking cases when the network is playing a status announcement (such as one indicating that a number is not in service) that has cycled several times. Whatever the cause of the protracted incomplete call, when this timer expires the call MUST be released. All of the gateway resources related to the media path SHOULD be released. A '480 Temporarily Unavailable' response code SHOULD be sent to the SIP network, and an REL message with cause value 19 (no answer from the user) SHOULD be sent to the ISUP network. The PSTN can be expected to respond with an RLC and the SIP network to respond with an ACK indicating that the release sequence has been completed.

7.2.9 CPG Received

A CPG is a provisional message that can indicate progress, alerting or in-band information. If a CPG suggests that in-band information is available, the gateway SHOULD begin to transmit early media and cut through the unidirectional backwards media path.

In SIP bridging situations (when the INVITE that initiated this session contained legitimate and comprehensible encapsulated ISUP), the CPG SHOULD be sent in the body of a particular 18x response, determined from the CPG Event Code as follows:

ISUP event code	SIP response
1 Alerting	180 Ringing
2 Progress	183 Session progress
3 In-band information	183 Session progress
4 Call forward; line busy	181 Call is being forwarded
5 Call forward; no reply	181 Call is being forwarded
6 Call forward; unconditional	181 Call is being forwarded
- (no event code present)	183 Session progress

Note that if the CPG does not indicate "Alerting," the current state will not change.

7.3 ACK received

At this stage, the call is fully connected and the conversation can take place. No ISUP message should be sent by the gateway when an ACK is received.

8. ISUP to SIP Mapping

8.1 ISUP to SIP Call Flows

The following call flows illustrate the order of messages in typical success and error cases when setting up a call initiated from the PSTN network. "100 Trying" acknowledgements to INVITE requests are not depicted, since their presence is optional.

In these diagrams, all call signaling (SIP, ISUP) is going to and from the MGC; media handling (e.g., audio cut-through, trunk freeing) is being performed by the MG, under the control of the MGC. For the purpose of simplicity, these are shown as a single node, labeled "MGC/MG".

8.1.1 En-bloc call setup (non auto-answer)

```
       SIP                       MGC/MG                       PSTN
         |                          |<-----------IAM-----------|1
         |                          |==========Audio==========>|
        2|<--------INVITE-----------|                          |
         |-----------100----------->|                          |
        3|-----------18x----------->|                          |
         |==========Audio==========>|                          |
         |                          |=========================>|
         |                          |------------ACM---------->|4
        5|-----------18x----------->|                          |
         |                          |------------CPG---------->|6
        7|-----------200-(I)------->|                          |
         |<=========Audio==========>|                          |
         |                          |------------ANM---------->|8
         |                          |<=========Audio==========>|
        9|<----------ACK------------|                          |
```

1. When a PSTN user wishes to begin a session with a SIP user, the PSTN network generates an IAM message towards the gateway.

2. Upon receipt of the IAM message, the gateway generates an INVITE message, and sends it to an appropriate SIP node.

3. When an event signifying that the call has sufficient addressing information occurs, the SIP node will generate a provisional response of 180 or greater.

4. Upon receipt of a provisional response of 180 or greater, the gateway will generate an ACM message. If the response is not 180, the ACM will carry a "called party status" value of "no indication."

5. The SIP node may use further provisional messages to indicate session progress.

6. After an ACM has been sent, all provisional responses will translate into ISUP CPG messages as indicated in Section 8.2.3.

7. When the SIP node answers the call, it will send a 200 OK message.

8. Upon receipt of the 200 OK message, the gateway will send an ANM message towards the ISUP node.

9. The gateway will send an ACK to the SIP node to acknowledge receipt of the INVITE final response.

8.1.2 Auto-answer call setup

```
      SIP                        MGC/MG                      PSTN
        |                           |<-----------IAM-----------|1
        |                           |==========Audio==========>|
       2|<--------INVITE-----------|                           |
       3|-----------200----------->|                           |
        |<=========Audio==========>|                           |
        |                           |------------CON---------->|4
        |                           |<=========Audio==========>|
       5|<----------ACK------------|                           |
```

1. When a PSTN user wishes to begin a session with a SIP user, the PSTN network generates an IAM message towards the gateway.

2. Upon receipt of the IAM message, the gateway generates an INVITE message and sends it to an appropriate SIP node based on called number analysis.

3. Since the SIP node is set up to automatically answer the call, it will send a 200 OK message.

4. Upon receipt of the 200 OK message, the gateway will send a CON message towards the ISUP node.

5. The gateway will send an ACK to the SIP node to acknowledge receipt of the INVITE final response.

8.1.3 SIP Timeout

```
       SIP                       MGC/MG                      PSTN
         |                          |<-----------IAM-----------|1
         |                          |==========Audio==========>|
        2|<--------INVITE-----------|                          |
         |     *** T1 Expires ***   |                          |
        3|<--------INVITE-----------|                          |
         |     *** T1 Expires ***   |                          |
         |<--------INVITE-----------|                          |
         |                          |    *** T11 Expires ***   |
         |                          |------------ACM---------->|4
         |     *** T1 Expires ***   |                          |
         |<--------INVITE-----------|                          |
         |     *** T1 Expires ***   |                          |
         |<--------INVITE-----------|                          |
         |     *** T1 Expires ***   |                          |
         |<--------INVITE-----------|                          |
         |     *** T1 Expires ***   |                          |
         |<--------INVITE-----------|                          |
         |     *** T1 Expires ***   |                          |
         |             ** MG Releases PSTN Trunk **            |
         |                          |------------REL---------->|5
        6|<--------CANCEL-----------|                          |
         |                          |<-----------RLC-----------|7
```

1. When a PSTN user wishes to begin a session with a SIP user, the PSTN network generates an IAM message towards the gateway.

2. Upon receipt of the IAM message, the gateway generates an INVITE message, and sends it to an appropriate SIP node based on called number analysis. The ISUP timer T11 and SIP timer T1 are set at this time.

3. The INVITE message will continue to be sent to the SIP node each time the timer T1 expires. The SIP standard specifies that INVITE transmission will be performed 7 times if no response is received.

4. When T11 expires, an ACM message will be sent to the ISUP node to prevent the call from being torn down by the remote node’s ISUP T7. This ACM contains a ’Called Party Status’ value of ’no indication.’

5. Once the maximum number of INVITE requests has been sent, the gateway will send a REL (cause code 18) to the ISUP node to terminate the call.

6. The gateway also sends a CANCEL message to the SIP node to terminate any initiation attempts.

7. Upon receipt of the REL, the remote ISUP node will send an RLC to acknowledge.

8.1.4 ISUP T9 Expires

```
SIP                        MGC/MG                       PSTN
  |                           |<-----------IAM-----------|1
  |                           |==========Audio==========>|
 2|<--------INVITE-----------|                          |
  |    *** T1 Expires ***     |                          |
 3|<--------INVITE-----------|                          |
  |    *** T1 Expires ***     |                          |
  |<--------INVITE-----------|                          |
  |                           |    *** T11 Expires ***   |
  |                           |------------ACM---------->|4
  |    *** T1 Expires ***     |                          |
  |<--------INVITE-----------|                          |
  |    *** T1 Expires ***     |                          |
  |<--------INVITE-----------|                          |
  |    *** T1 Expires ***     |                          |
  |<--------INVITE-----------|                          |
  |                           |    *** T9 Expires ***    |
  |             ** MG Releases PSTN Trunk **             |
  |                           |<-----------REL-----------|5
  |                           |------------RLC---------->|6
 7|<--------CANCEL-----------|                          |
```

1. When a PSTN user wishes to begin a session with a SIP user, the PSTN network generates an IAM message towards the gateway.

2. Upon receipt of the IAM message, the gateway generates an INVITE message, and sends it to an appropriate SIP node based on called number analysis. The ISUP timer T11 and SIP timer T1 are set at this time.

3. The INVITE message will continue to be sent to the SIP node each time the timer T1 expires. The SIP standard specifies that INVITE transmission will be performed 7 times if no response is received. Since SIP T1 starts at 1/2 second or more and doubles each time it is retransmitted, it will be at least a minute before SIP times out the INVITE request; since SIP T1 is allowed to be larger than 500 ms initially, it is possible that 7 x SIP T1 will be longer than ISUP T11 + ISUP T9.

4. When T11 expires, an ACM message will be sent to the ISUP node to prevent the call from being torn down by the remote node's ISUP T7. This ACM contains a 'Called Party Status' value of 'no indication.'

5. When ISUP T9 in the remote PSTN node expires, it will send a REL.

6. Upon receipt of the REL, the gateway will send an RLC to acknowledge.

7. The REL will trigger a CANCEL request, which gets sent to the SIP node.

8.1.5 SIP Error Response

```
SIP                        MGC/MG                       PSTN
  |                          |<-----------IAM-----------|1
  |                          |==========Audio==========>|
 2|<--------INVITE-----------|                          |
 3|-----------4xx+---------->|                          |
 4|<----------ACK------------|                          |
  |             ** MG Releases PSTN Trunk **            |
  |                          |------------REL---------->|5
  |                          |<-----------RLC-----------|6
```

1. When a PSTN user wishes to begin a session with a SIP user, the PSTN network generates an IAM message towards the gateway.

2. Upon receipt of the IAM message, the gateway generates an INVITE message, and sends it to an appropriate SIP node based on called number analysis.

3. The SIP node indicates an error condition by replying with a response with a code of 400 or greater.

4. The gateway sends an ACK message to acknowledge receipt of the INVITE final response.

5. An ISUP REL message is generated from the SIP code, as specified in Section 8.2.6.1.

6. The remote ISUP node confirms receipt of the REL message with an RLC message.

8.1.6 SIP Redirection

```
SIP node 1                 MGC/MG                      PSTN
   |                          |<-----------IAM-----------|1
   |                          |==========Audio==========>|
  2|<--------INVITE-----------|                          |
  3|-----------3xx+---------->|                          |
   |                          |------------CPG---------->|4
  5|<----------ACK------------|                          |
                              |                          |
                              |                          |
SIP node 2                    |                          |
  6|<--------INVITE-----------|                          |
  7|-----------18x----------->|                          |
   |<=========Audio===========|                          |
   |                          |------------ACM---------->|8
  9|-----------200-(I)------->|                          |
   |<=========Audio==========>|                          |
   |                          |------------ANM---------->|10
   |                          |<=========Audio==========>|
 11|<----------ACK------------|                          |
```

1. When a PSTN user wishes to begin a session with a SIP user, the PSTN network generates an IAM message towards the gateway.

2. Upon receipt of the IAM message, the gateway generates an INVITE message, and sends it to an appropriate SIP node based on called number analysis.

3. The SIP node indicates that the resource which the user is attempting to contact is at a different location by sending a 3xx message. In this instance we assume the Contact URL specifies a valid URL reachable by a VoIP SIP call.

4. The gateway sends a CPG with event indication that the call is being forwarded upon receipt of the 3xx message. Note that this translation should be able to be disabled by configuration, as some ISUP nodes do not support receipt of CPG messages before ACM messages.

5. The gateway acknowledges receipt of the INVITE final response by sending an ACK message to the SIP node.

6. The gateway re-sends the INVITE message to the address indicated in the Contact: field of the 3xx message.

7. When an event signifying that the call has sufficient addressing information occurs, the SIP node will generate a provisional response of 180 or greater.

8. Upon receipt of a provisional response of 180 or greater, the gateway will generate an ACM message with an event code as indicated in Section 8.2.3.

9. When the SIP node answers the call, it will send a 200 OK message.

10. Upon receipt of the 200 OK message, the gateway will send an ANM message towards the ISUP node.

11. The gateway will send an ACK to the SIP node to acknowledge receipt of the INVITE final response.

8.1.7 Call Canceled by ISUP

```
SIP                       MGC/MG                     PSTN
  |                          |<-----------IAM-----------|1
  |                          |==========Audio==========>|
 2|<--------INVITE-----------|                          |
 3|-----------18x----------->|                          |
  |==========Audio==========>|                          |
  |                          |------------ACM---------->|4
  |             ** MG Releases PSTN Trunk **            |
  |                          |<-----------REL-----------|5
  |                          |------------RLC---------->|6
 7|<---------CANCEL----------|                          |
  |            ** MG Releases IP Resources **           |
 8|-----------200----------->|                          |
 9|-----------487----------->|                          |
10|<----------ACK------------|                          |
```

1. When a PSTN user wishes to begin a session with a SIP user, the PSTN network generates an IAM message towards the gateway.

2. Upon receipt of the IAM message, the gateway generates an INVITE message, and sends it to an appropriate SIP node based on called number analysis.

3. When an event signifying that the call has sufficient addressing information occurs, the SIP node will generate a provisional response of 180 or greater.

4. Upon receipt of a provisional response of 180 or greater, the gateway will generate an ACM message with an event code as indicated in Section 8.2.3.

5. If the calling party hangs up before the SIP node answers the call, a REL message will be generated.

6. The gateway frees the PSTN circuit and indicates that it is available for reuse by sending an RLC.

7. Upon receipt of a REL message before an INVITE final response, the gateway will send a CANCEL towards the SIP node.

8. Upon receipt of the CANCEL, the SIP node will send a 200 response.

9. The remote SIP node will send a "487 Call Cancelled" to complete the INVITE transaction.

10. The gateway will send an ACK to the SIP node to acknowledge receipt of the INVITE final response.

8.2 State Machine

Note that REL may arrive in any state. Whenever this occurs, the actions in section Section 8.2.7. are taken. Not all of these transitions are shown in this diagram.

```
                                 +---------+
        +----------------------->|  Idle   |<---------------------+
        |                        +----+----+                      |
        |                             |                           |
        |                             | IAM/7.2.1                 |
        |                             V                           |
        |    REL/7.2.7    +-------------------------+ 400+/7.2.6  |
        +<----------------+         Trying          |------------>|
        |                 +-+--------+------+-------+             |
        |                   |        |      |                     |
        |                   | T11/   | 18x/ | 200/                |
        |                   | 7.2.8  |7.2.3 | 7.2.4               |
        |                   V        |      |                     |
        | REL/7.2.7 +--------------+ |      |      400+/7.2.6     |
        |<----------| Progressing  |-|------|-------------------->|
        |           +--+----+------+ |      |                     |
        |              |    |        |      |                     |
        |        200/  |    | 18x/   |      |                     |
        |        7.2.4 |    | 7.2.3  |      |                     |
        |              |    V        V      |                     |
        |  REL/7.2.7   |  +---------------+ |      400+/7.2.6     |
        |<-------------|--|    Alerting   |-|-------------------->|
        |              |  +--------+------+ |                     |
        |              |           |        |                     |
        |              |           | 200/   |                     |
        |              |           | 7.2.4  |                     |
        |              V           V        V                     |
        |     BYE/9.1 +-----------------------------+    REL/9.2  |
        +<------------+          Connected          +------------>+
                      +-----------------------------+
```

8.2.1 Initial Address Message received

Upon receipt of an IAM, the gateway SHOULD reserve appropriate internal resources (Digital Signal Processors - DSPs - and the like) necessary for handling the IP side of the call. It MAY make any necessary preparations to connect audio in the backwards direction (towards the caller).

8.2.1.1 IAM to INVITE procedures

When an IAM arrives at a PSTN-SIP gateway, a SIP INVITE message MUST be created for transmission to the SIP network. This section details the process by which a gateway populates the fields of the INVITE based on parameters found within the IAM.

The context of the call setup request read by the gateway in the IAM will be mapped primarily to two URIs in the INVITE, one representing the originator of the session and the other its destination. The former will always appear in the From header (after it has been converted from ISUP format by the procedure described in Section 12), and the latter is almost always used for both the To header and the Request-URI.

Once the address of the called party number has been read from the IAM, it SHOULD be translated into a destination tel URL that will serve as the Request-URI of the INVITE. Alternatively, a gateway MAY first attempt a Telephone Number Mapping (ENUM) [8] query to resolve the called party number to a URI. Some additional ISUP fields MAY be added to the tel URL after translation has been completed, namely:

o If the gateway supports carrier-based routing (which is optional in this specification), it SHOULD ascertain if either the CIP (in ANSI networks) or TNS parameter is present in the IAM. If a value is present, the CIC SHOULD be extracted from the given parameter and analyzed by the gateway. A 'cic=' field with the value of the CIC SHOULD be appended to the destination tel URL, if doing so is in keeping with local policy (i.e., provided that the CIC does not indicate the network which owns the gateway or some similar condition). Note that if it is created, the 'cic=' parameter MUST be prefixed with the country code used or implied in the called party number, so that CIC '5062' becomes, in the United States, '+1-5062'. For further information on the 'cic=' tel URL field see [21].

o If the gateway supports number portability-based routing (which is optional in this specification), then the gateway will need to look at a few other fields. To correctly map the FCI 'number translated' bit indicating that an LNP dip had been performed in the PSTN, an 'npdi=yes' field SHOULD be appended to the tel URL. If a GAP is present in the IAM, then the contents of the CPN (the Location Routing Number - LRN) SHOULD be translated from ISUP format (as described in Section 12) and copied into an 'rn=' field which must be appended to the tel URL, whereas the GAP itself should be translated to ISUP format and used to populate the primary telephone number field of the tel URL. Note that in some national numbering plans, both the LRN and the dialed number may be stored in the CPN parameter, in which case they must be separated out into different fields to be stored in the tel URL. Note that LRNs are necessarily national in scope, and consequently they MUST NOT be preceded by a '+' in the 'rn=' field. For further information on these tel URL fields see [21].

In most cases, the resulting destination tel URL SHOULD be used in both the To field and Request-URI sent by the gateway. However, if the OCN parameter is present in the IAM, the To field SHOULD be constructed from the translation (from ISUP format following Section 12 of the OCN parameter, and hence the Request-URI and To field MAY be different.

The construction of the From header field is dependent on the presence of a CIN parameter. If the CIN is not present, then the gateway SHOULD create a dummy From header field containing a SIP URI without a user portion which communicates only the hostname of the gateway (e.g., 'sip:gw.sipcarrier.com). If the CIN is available, then it SHOULD be translated (in accordance with the procedure described above) into a tel URL which should populate the From header field. In either case, local policy or requests for presentation restriction (see Section 12.1) MAY result in a different value for the From header field.

8.2.2 100 received

A 100 response SHOULD NOT trigger any PSTN interworking messages; it only serves the purpose of suppressing INVITE retransmissions.

8.2.3 18x received

Upon receipt of a 18x provisional response, if no ACM has been sent and no legitimate and comprehensible ISUP is present in the 18x message body, then the ISUP message SHOULD be generated according to the following table. Note that if an early ACM is sent, the call MUST enter state "Progressing" instead of state "Alerting."

Response received	Message sent by the MGC
180 Ringing	ACM (BCI = subscriber free)
181 Call is being forwarded	Early ACM and CPG, event=6
182 Queued	ACM (BCI = no indication)
183 Session progress message	ACM (BCI = no indication)

If an ACM has already been sent and no ISUP is present in the 18x message body, an ISUP message SHOULD be generated according to the following table.

Response received	Message sent by the MGC
180 Ringing	CPG, event = 1 (Alerting)
181 Call is being forwarded	CPG, event = 6 (Forwarding)
182 Queued	CPG, event = 2 (Progress)
183 Session progress message	CPG, event = 2 (Progress)

Upon receipt of a 180 response, the gateway SHOULD generate the ringback tone to be heard by the caller on the PSTN side (unless the gateway knows that ringback will be provided by the network on the PSTN side).

Note however that a gateway might receive media at any time after it has transmitted an SDP offer that it has sent in an INVITE, even before a 18x provisional response is received. Therefore the gateway MUST be prepared to play this media to the caller on the PSTN side (if necessary, ceasing any ringback tone that it may have begun to generate and then playing media). Note that the gateway may also receive SDP offers in responses for an early media session using some SIP extension, see Section 5.5. If a gateway receives a 183 response while it is playing backwards media, then when it generates a mapping for this response, if no encapsulated ISUP is present, the gateway SHOULD indicate that in-band information is available (for example, with the Event Information parameter of the CPG message or the Optional Backward Call Indicators parameter of the ACM).

When an ACM is sent, the mandatory Backward Call Indicators parameter must be set, as well as any optional parameters as gateway policy dictates. If legitimate and comprehensible ISUP is present in the 18x response, the gateway SHOULD re-use the appropriate parameters of the ISUP message contained in the response body, including the value of the Backward Call Indicator parameter, as it formulates a message that it will send across its PSTN interface. In the absence of a usable encapsulated ACM, the BCI parameter SHOULD be set as follows:

```
Message type:                            ACM

Backward Call Indicators
Charge indicator:                      10 charge
Called party's status indicator:       01 subscriber free or
                                       00 no indication
Called party's category indicator:     01 ordinary subscriber
End-to-end method indicator:           00 no end-to-end method
Interworking indicator:                0  no interworking
End-to-end information indicator:      0  no end-to-end info
ISDN user part indicator:              1  ISUP used all the way
Holding indicator:                     0  no holding
ISDN access indicator:                 0  No ISDN access
Echo control device indicator:         It depends on the call
SCCP method indicator:                 00 no indication
```

Note that when the ISUP Backward Call Indicator parameter Interworking indicator field is set to 'interworking encountered', this indicates that ISDN is interworking with a network which is not capable of providing as many services as ISDN does. ISUP therefore may not employ certain features it otherwise normally uses. Gateway vendors MAY however provide a configurable option, usable at the discretion of service providers when they require additional ISUP services, that in the absence of encapsulated ISUP will signal in the BCI that interworking has been encountered, and that ISUP is not used all the way, for those operators that as a matter of policy would rather operate in this mode. For more information on the effects of interworking see Section 7.2.1.1.

8.2.4 2xx received

Response received	Message sent by the MGC
200 OK	ANM, ACK

After receiving a 200 OK response the gateway MUST establish a directional media path in the gateway and send an ANM to the PSTN as well as an ACK to the SIP network.

If the 200 OK response arrives before the gateway has sent an ACM, a CON is sent instead of the ANM, in those ISUP variants that support the CON message.

When a legitimate and comprehensible ANM is encapsulated in the 200 OK response, the gateway SHOULD re-use any relevant ISUP parameters in the ANM it sends to the PSTN.

Note that gateways may sometimes receive 200 OK responses for requests other than INVITE (for example, those used in managing provisional responses, or the INFO method). The procedures described in this section apply only to 200 OK responses received as a result of sending an INVITE. The gateway SHOULD NOT send any PSTN messages if it receives a 200 OK in response to non-INVITE requests it has sent.

8.2.5 3xx Received

When any 3xx response (a redirection) is received, the gateway SHOULD try to reach the destination by sending one or more new call setup requests using URIs found in any Contact header field(s) present in the response, as is mandated in the base SIP specification. Such 3xx responses are typically sent by a redirect server, and can be thought of as similar to a location register in mobile PSTN networks.

If a particular URI presented in the Contact header of a 3xx is best reachable (according to the gateway’s routing policies) via the PSTN, the gateway SHOULD send a new IAM and from that moment on act as a normal PSTN switch (no SIP involved) - usually this will be the case when the URI in the Contact header is a tel URL, one that the gateway cannot reach locally and one for which there is no ENUM mapping.

Alternatively, the gateway MAY send a REL message to the PSTN with a redirection indicator (23) and a diagnostic field corresponding to the telephone number in the URI. If, however, the new location is best reachable using SIP (if the URI in the Contact header contains no telephone number at all), the MGC SHOULD send a new INVITE with a Request-URI possibly a new IAM generated by the MGC in the message body.

While it is exploring a long list of Contact header fields with SIP requests, a gateway MAY send a CPG message with an event code of 6 (Forwarding) to the PSTN in order to indicate that the call is proceeding (where permitted by the ISUP variant in question).

All redirection situations have to be treated very carefully because they involved special charging situations. In PSTN the caller typically pays for the first leg (to the gateway) and the callee pays the second (from the forwarding switch to the destination).

8.2.6 4xx-6xx Received

When a response code of 400 or greater is received by the gateway, then the INVITE previously sent by the gateway has been rejected. Under most circumstances the gateway SHOULD release the resources in the gateway, send a REL to the PSTN with a cause value and send an

ACK to the SIP network. Some specific circumstances are identified below in which a gateway MAY attempt to rectify a SIP-specific problem communicated by a status code without releasing the call by retrying the request. When a REL is sent to the PSTN, the gateway expects the arrival of an RLC indicating that the release sequence is complete.

8.2.6.1 SIP Status Code to ISDN Cause Code Mapping

When a REL message is generated due to a SIP rejection response that contains an encapsulated REL message, the Cause Indicator (CAI) parameter in the generated REL SHOULD be set to the value of the CAI parameter received in the encapsulated REL. If no encapsulated ISUP is present, the mapping below between status code and cause codes are RECOMMENDED.

Any SIP status codes not listed below (associated with SIP extensions, versions of SIP subsequent to the issue of this document, or simply omitted) should be mapping to cause code 31 "Normal, unspecified". These mappings cover only responses; note that the BYE and CANCEL requests, which are also used to tear down a dialog, SHOULD be mapped to 16 "Normal clearing" under most circumstances (although see Section 5.8).

By default, the cause location associated with the CAI parameter should be encoded such that 6xx codes are given the location ’user’, whereas 4xx and 5xx codes are given a ’network’ location. Exceptions are marked below.

Just as there are certain ISDN cause codes that are ISUP-specific and have no corollary SIP action, so there are SIP status codes that should not simply be translated to ISUP - some SIP-specific action should be attempted first. See the note on the (+) tag below.

Response received	Cause value in the REL
400 Bad Request	41 Temporary Failure
401 Unauthorized	21 Call rejected (*)
402 Payment required	21 Call rejected
403 Forbidden	21 Call rejected
404 Not found	1 Unallocated number
405 Method not allowed	63 Service or option unavailable
406 Not acceptable	79 Service/option not implemented (+)
407 Proxy authentication required	21 Call rejected (*)
408 Request timeout	102 Recovery on timer expiry
410 Gone	22 Number changed (w/o diagnostic)
413 Request Entity too long	127 Interworking (+)
414 Request-URI too long	127 Interworking (+)
415 Unsupported media type	79 Service/option not implemented (+)
416 Unsupported URI Scheme	127 Interworking (+)
420 Bad extension	127 Interworking (+)
421 Extension Required	127 Interworking (+)
423 Interval Too Brief	127 Interworking (+)
480 Temporarily unavailable	18 No user responding
481 Call/Transaction Does not Exist	41 Temporary Failure
482 Loop Detected	25 Exchange - routing error
483 Too many hops	25 Exchange - routing error
484 Address incomplete	28 Invalid Number Format (+)
485 Ambiguous	1 Unallocated number
486 Busy here	17 User busy
487 Request Terminated	--- (no mapping)
488 Not Acceptable here	--- by Warning header
500 Server internal error	41 Temporary failure
501 Not implemented	79 Not implemented, unspecified
502 Bad gateway	38 Network out of order
503 Service unavailable	41 Temporary failure
504 Server time-out	102 Recovery on timer expiry
504 Version Not Supported	127 Interworking (+)
513 Message Too Large	127 Interworking (+)
600 Busy everywhere	17 User busy
603 Decline	21 Call rejected
604 Does not exist anywhere	1 Unallocated number
606 Not acceptable	--- by Warning header

(*) In some cases, it may be possible for a SIP gateway to provide credentials to the SIP UAS that is rejecting an INVITE due to authorization failure. If the gateway can authenticate itself, then obviously it SHOULD do so and proceed with the call; only if the gateway cannot authenticate itself should cause code 21 be sent.

(+) If at all possible, a SIP gateway SHOULD respond to these protocol errors by remedying unacceptable behavior and attempting to re-originate the session. Only if this proves impossible should the SIP gateway fail the ISUP half of the call.

When the Warning header is present in a SIP 606 or 488 message, there may be specific ISDN cause code mappings appropriate to the Warning code. This document recommends that '31 Normal, unspecified' SHOULD by default be used for most currently assigned Warning codes. If the Warning code speaks to an unavailable bearer capability, cause code '65 Bearer Capability Not Implemented' is a RECOMMENDED mapping.

8.2.7 REL Received

This circumstance generally arises when the user on the PSTN side hangs up before the call has been answered; the gateway therefore aborts the establishment of the session. A CANCEL request MUST be issued (a BYE is not used, since no final response has arrived from the SIP side). A 200 OK for the CANCEL can be expected by the gateway, and finally a 487 for the INVITE arrives (which the gateway ACKs in turn).

The gateway SHOULD store state information related to this dialog for a certain period of time, since a 200 final response for the INVITE originally sent might arrive (even after the reception of the 200 OK for the CANCEL). In this situation, the gateway MUST send an ACK followed by an appropriate BYE request.

In SIP bridging situations, the REL message cannot be encapsulated in a CANCEL message (since CANCEL cannot have a message body). Usually, the REL message will contain a CAI value of 16 "Normal clearing". If the value is other than a 16, the gateway MAY wish to use some other means of communicating the cause value (see Section 5.8).

8.2.8 ISUP T11 Expires

In order to prevent the remote ISUP node's timer T7 from expiring, the gateway MAY keep its own supervisory timer; ISUP defines this timer as T11. T11's duration is carefully chosen so that it will always be shorter than the T7 of any node to which the gateway is communicating.

To clarify timer T11's relevance with respect to SIP interworking, Q.764 [12] explains its use as: "If in normal operation, a delay in the receipt of an address complete signal from the succeeding network is expected, the last common channel signaling exchange will originate and send an address complete message 15 to 20 seconds [timer (T11)] after receiving the latest address message." Since SIP nodes have no obligation to respond to an INVITE request within 20 seconds, SIP interworking inarguably qualifies as such a situation.

If the gateway supports this optional mechanism, then if its T11 expires, it SHOULD send an early ACM (i.e., called party status set to "no indication") to prevent the expiration of the remote node's T7 (where permitted by the ISUP variant). See Section 8.2.3 for the value of the ACM parameters.

If a "180 Ringing" message arrives subsequently, it SHOULD be sent in a CPG, as shown in Section 8.2.3.

See Section 8.1.3 for an example callflow that includes the expiration of T11.

9. Suspend/Resume and Hold

9.1 Suspend (SUS) and Resume (RES) Messages

In ISDN networks, a user can generate a SUS (timer T2, user initiated) in order to unplug the terminal from the socket and plug it in another one. A RES is sent once the terminal has been reconnected and the T2 timer has not expired. SUS is also frequently used to signaling an on-hook state for a remote terminal before timers leading to the transmission of a REL message are sent (this is the more common case by far). While a call is suspended, no audio media is passed end-to-end.

When a SUS is sent for a call that has a SIP leg, a gateway MAY suspend IP media transmission until a RES is received. Putting the media on hold insures that bandwidth is conserved when no audio traffic needs to be transmitted.

If media suspension is appropriate, then when a SUS arrives from the PSTN, the MGC MAY send an INVITE to request that the far-end's transmission of the media stream be placed on hold. The subsequent reception of a RES from the PSTN SHOULD then trigger a re-INVITE that requests the resumption of the media stream. Note that the MGC may or may not elect to stop transmitting any media itself when it requests the cessation of far-end transmission.

If media suspension is not required by the MGC receiving the SUS from the PSTN, the SIP INFO [6] method MAY be used to transmit an encapsulated SUS rather than a re-INVITE. Note that the recipient of such an INFO request may be a simple SIP phone that does not understand ISUP (and would therefore take no action on receipt of this message); if a prospective destination for an INFO-encapsulated SUS has not used encapsulated ISUP in any messages it has previously sent, the gateway SHOULD NOT relay the INFO method, but rather should handle the SUS and the corresponding RES without signaling their arrival to the SIP network.

In any case, subsequent RES messages MUST be transmitted in the same method that was used for the corresponding SUS (i.e., if an INFO is used for a SUS, INFO should also be used for the subsequent RES).

Regardless of whether the INFO or re-INVITE mechanism is used to carry a SUS message, neither has any implication that the originating side will cease sending IP media. The recipient of an encapsulated SUS message MAY therefore elect to send a re-INVITE themselves to suspend media transmission from the MGC side if desired.

The following example uses the INVITE mechanism. Note that this flow is informative, not proscriptive; compliant gateways are free to implement functionally equivalent flows, as described in the preceding paragraphs.

```
     SIP                       MGC/MG                       PSTN
       |                          |<-----------SUS-----------|1
      2|<--------INVITE-----------|                          |
      3|-----------200----------->|                          |
      4|<----------ACK------------|                          |
       |                          |<-----------RES-----------|5
      6|<--------INVITE-----------|                          |
      7|-----------200----------->|                          |
      8|<----------ACK------------|                          |
```

The handling of a network-initiated SUS immediately prior to call teardown is handled in Section 10.2.2.

9.2 Hold (re-INVITE)

After a call has been connected, a re-INVITE could be sent to a gateway from the SIP side in order to place the call on hold. This re-INVITE will have an SDP offer indicating that the originator of the re-INVITE no longer wishes to receive media.

```
     SIP                         MGC/MG                         PSTN
      1|---------INVITE---------->|                              |
       |                          |------------CPG---------->|2
      3|<----------200------------|                              |
      4|-----------ACK----------->|                              |
```

When such a re-INVITE is received, the gateway SHOULD send a CPG in order to express that the call has been placed on hold. The CPG SHOULD contain a Generic Notification Indicator (or, in ANSI networks, a Notification Indicator) with a value of 'remote hold'.

If, subsequent to the sending of the re-INVITE, the SIP side wishes to take the remote end off hold and begin receiving media again, it SHOULD repeat the flow above with an INVITE that contains an SDP offer with an appropriate media destination. The Generic Notification Indicator would in this instance have a value of 'remote retrieval' (or in some variants 'remote hold released').

Finally, note that a CPG with hold indicators may be received by a gateway from the PSTN. In the interests of conserving bandwidth, the gateway SHOULD stop sending media until the call is resumed and SHOULD send a re-INVITE to the SIP leg of the call requesting that the remote side stop sending media.

10. Normal Release of the Connection

From the perspective of a gateway, either the SIP side or the ISUP side can release a call, regardless of which side initiated the call. Note that cancellation of a call setup request (either from the ISUP or SIP side) is discussed elsewhere in this document (in Section 8.2.7 and Section 7.2.3, respectively).

Gateways SHOULD implement functional equivalence with the flows in this section.

10.1 SIP initiated release

For a normal termination of the dialog (receipt of a BYE request), the gateway MUST immediately send a 200 response. The gateway then MUST release any media resources in the gateway (DSPs, TCIC locks, and so on) and send an REL with a cause code of 16 (normal call clearing) to the PSTN. Release of resources is confirmed by the PSTN side with an RLC message.

In SIP bridging situations, the cause code of any REL encapsulated in the BYE request SHOULD be re-used in any REL that the gateway sends to the PSTN.

```
     SIP                       MGC/MG                      PSTN
      1|-----------BYE----------->|                          |
       |            ** MG Releases IP Resources **           |
      2|<----------200------------|                          |
       |             ** MG Releases PSTN Trunk **            |
       |                          |------------REL---------->|3
       |                          |<-----------RLC-----------|4
```

10.2 ISUP initiated release

If the release of the connection was caused by the reception of a REL, the REL SHOULD be encapsulated in the BYE sent by the gateway. Whether the caller or callee hangs up first, the gateway SHOULD release any internal resources used in support of the call and then MUST confirm that the circuit is ready for re-use by sending an RLC.

10.2.1 Caller hangs up

When the caller hangs up, the SIP dialog MUST be terminated by sending a BYE request (which is confirmed with a 200).

```
     SIP                       MGC/MG                      PSTN
       |                          |<-----------REL-----------|1
       |             ** MG Releases PSTN Trunk **            |
       |                          |------------REL---------->|2
      3|<----------BYE------------|                          |
       |            ** MG Releases IP Resources **           |
      4|-----------200----------->|                          |
```

10.2.2 Callee hangs up (SUS)

In some PSTN scenarios, if the callee hangs up in the middle of a call, the local exchange sends a SUS instead of a REL and starts a timer (T6, SUS is network initiated). When the timer expires, the REL is sent. This necessitates a slightly different SIP flow; see Section 9 for more information on handling suspension. It is RECOMMENDED that gateways implement functional equivalence with the following flow for this case:

```
     SIP                       MGC/MG                       PSTN
       |                          |<-----------SUS-----------|1
      2|<--------INVITE-----------|                          |
      3|-----------200----------->|                          |
      4|<----------ACK------------|                          |
       |                          |    *** T6 Expires ***    |
       |                          |<-----------REL-----------|5
       |             ** MG Releases PSTN Trunk **            |
       |                          |------------RLC---------->|6
      7|<----------BYE------------|                          |
       |            ** MG Releases IP Resources **           |
      8|-----------200----------->|                          |
```

11. ISUP Maintenance Messages

ISUP contains a set of messages used for maintenance purposes. They can be received during any ongoing call. There are basically two kinds of maintenance messages (apart from the continuity check): messages for blocking circuits and messages for resetting circuits.

11.1 Reset messages

Upon reception of an RSC message for a circuit currently being used by the gateway for a call, the call MUST be released immediately (this typically results from a serious maintenance condition). RSC MUST be answered with an RLC after resetting the circuit in the gateway. Group reset (GRS) messages which target a range of circuits are answered with a Circuit Group Reset ACK Message (GRA) after resetting all the circuits affected by the message.

The gateways SHOULD behave as if a REL had been received in order to release the dialog on the SIP side. A BYE or a CANCEL are sent depending of the status of the call. See the procedures in Section 10.

11.2 Blocking messages

There are two kinds of blocking messages: maintenance messages or hardware-failure messages. Maintenance blocking messages indicate that the circuit is to be blocked for any subsequent calls, but these messages do not affect any ongoing call. This allows circuits to be gradually quiesced and taken out of service for maintenance.

Hardware-oriented blocking messages have to be treated as reset messages. They generally are sent only when a hardware failure has occurred. Media transmission for all calls in progress on these circuits would be affected by this hardware condition, and therefore all calls must be released immediately.

BLO is always maintenance oriented and it is answered by the gateway with a Blocking ACK Message (BLA) when the circuit is blocked - this requires no corresponding SIP actions. Circuit Group Blocking (CGB) messages have a "type indicator" inside the Circuit Group Supervision Message Type Indicator. It indicates if the CGB is maintenance or hardware failure oriented. If the CGB results from a hardware failure, then each call in progress in the affected range of circuits MUST be terminated immediately as if a REL had been received, following the procedures in Section 10. CGBs MUST be answered with CGBAs.

11.3 Continuity Checks

A continuity check is a test performed on a circuit that involves the reflection of a tone generated at the originating switch by a loopback at the destination switch. Two variants of the continuity check appear in ISUP: the implicit continuity check request within an IAM (in which case the continuity check takes place as a precondition before call setup begins), and the explicit continuity check signaled by a Continuity Check Request (CCR) message. PSTN gateways in regions that support continuity checking generally SHOULD have some way of accommodating these tests (if they hope to be fielded by providers that interconnect with any major carrier).

When a CCR is received by a PSTN-SIP gateway, the gateway SHOULD NOT send any corresponding SIP messages; the scope of the continuity check applies only to the PSTN trunks, not to any IP media paths beyond the gateway. CCR messages also do not designate any called party number, or any other way to determine what SIP user agent server should be reached.

When an IAM with the Continuity Check Indicator flag set within the NCI parameter is received, the gateway MUST process the continuity check before sending an INVITE message (and proceeding normally with call setup); if the continuity check fails (a COT with Continuity Indicator of 'failed' is received), then an INVITE MUST NOT be sent.

12. Construction of Telephony URIs

SIP proxy servers MAY route SIP messages on any signaling criteria desired by network administrators, but generally the Request-URI is the foremost routing criterion. The To and From headers are also frequently of interest in making routing decisions. SIP-ISUP mapping assumes that proxy servers are interested in at least these three fields of SIP messages, all of which contain URIs.

SIP-ISUP mapping frequently requires the representation of telephone numbers in these URIs. In some instances these numbers will be presented first in ISUP messages, and SS7-SIP gateways will need to translate the ISUP formats of these numbers into SIP URIs. In other cases the reverse transformation will be required.

The most common format used in SIP for the representation of telephone numbers is the tel URL [7]. When converting between formats, the tel URL MAY constitute the entirety of a URI field in a SIP message, or it MAY appear as the user portion of a SIP URI. For example, a To field might appear as:

```
To: tel:+17208881000
```

Or

```
To: sip:+17208881000@level3.com
```

Whether or not a particular gateway or endpoint should formulate URIs in the tel or SIP format is a matter of local administrative policy - if the presence of a host portion would aid the surrounding network in routing calls, the SIP format should be used. A gateway MUST accept either tel or SIP URIs from its peers.

The '+' sign preceding the number in tel URLs indicates that the digits which follow constitute a fully-qualified E.164 [16] number; essentially, this means that a country code is provided before any national-specific area codes, exchange/city codes, or address codes. The absence of a '+' sign MAY signify that the number is merely nationally significant, or perhaps that a private dialing plan is in use. When the '+' sign is not present, but a telephone number is represented by the user portion of the URI, the SIP URI SHOULD contain the optional ';user=phone' parameter; e.g.,

```
To: sip:83000@sip.example.net;user=phone
```

However, it is strongly RECOMMENDED that only internationally significant E.164 numbers be passed between SIP-T gateways, especially when such gateways are in different regions or different administrative domains. In many if not most SIP-T networks, gateways are not responsible for end-to-end routing of SIP calls; practically speaking, gateways have no way of knowing if the call will terminate in a local or remote administrative domain and/or region, and hence gateways SHOULD always assume that calls require an international numbering plan. There is no guarantee that recipients of SIP signaling will be capable of understanding national dialing plans used by the originators of calls - if the originating gateway does not internationalize the signaling, the context in which the digits were dialed cannot be extrapolated by far-end network elements.

In ISUP signaling, a telephone number appears in a common format that is used in several parameters, including the CPN and CIN; when it represents a calling party number it sports some additional information (detailed below). For the purposes of this document, we will refer to this format as 'ISUP format' - if the additional calling party information is present, the format shall be referred to as 'ISUP- calling format'. The format consists of a byte called the Nature of Address (NoA) indicator, followed by another byte which contains the Numbering Plan Indicator (NPI), both of which are prefixed to a variable-length series of bytes that contains the digits of the telephone number in Binary Coded Decimal (BCD) format. In the calling party number case, the NPI's byte also contains bit fields which represent the caller's presentation preferences and the status of any call screening checks performed up until this point in the call.

```
     H G F E D C B A      H G F E D C B A
    +-+-+-+-+-+-+-+-+    +-+-+-+-+-+-+-+-+
    | |    NoA      |    | |    NoA      |
    +-+-+-+-+-+-+-+-+    +-+-+-+-+-+-+-+-+
    | | NPI | spare |    | | NPI |PrI|ScI|
    +-+-+-+-+-+-+-+-+    +-+-+-+-+-+-+-+-+
    | dig...| dig 1 |    | dig...| dig 1 |
    |      ...      |    |      ...      |
    | dig n | dig...|    | dig n | dig...|
    +-+-+-+-+-+-+-+-+    +-+-+-+-+-+-+-+-+

      ISUP format       ISUP calling format

           ISUP numbering formats
```

The NPI field is generally set to the value 'ISDN (Telephony) numbering plan (Recommendation E.164)', but this does not mean that the digits which follow necessarily contain a country code; the NoA field dictates whether the telephone number is in a national or international format. When the represented number is not designated to be in an international format, the NoA generally provides information specific to the national dialing plan - based on this information one can usually determine how to convert the number in question into an international format. Note that if the NPI contains a value other than 'ISDN numbering plan', then the tel URL may not be suitable for carrying the address digits, and the handling for such calls is outside the scope of this document.

12.1 ISUP format to tel URL mapping

Based on the above, conversion from ISUP format to a tel URL is as follows. First, provided that the NPI field indicates that the telephone number format uses E.164, the NoA is consulted. If the NoA indicates that the number is an international number, then the telephone number digits SHOULD be appended unmodified to a 'tel:+' string. If the NoA has the value 'national (significant) number', then a country code MUST be prefixed to the telephone number digits before they are committed to a tel URL; if the gateway performing this conversion interconnects with switches homed to several different country codes, presumably the appropriate country code SHOULD be chosen based on the originating switch or trunk group. If the NoA has the value 'subscriber number', both a country code and any other numbering components necessary for the numbering plan in question (such as area codes or city codes) MAY need to be added in order for the number to be internationally significant - however, such procedures vary greatly from country to country, and hence they cannot be specified in detail here. Only if a country or network-specific value is used for the NoA SHOULD a tel URL not include a '+' sign; in these cases, gateways SHOULD simply copy the provided digits into the tel URL and append a 'user=phone' parameter if a SIP URI format is used. Any non-standard or proprietary mechanisms used to communicate further context for the call in ISUP are outside the scope of this document.

If a nationally-specific parameter is present that allows for the transmission of the calling party's name (such as the Generic Name Parameter in ANSI), then generally, if presentation is not restricted, this information SHOULD be used to populate the display-name portion of the From field.

If ISUP calling format is being converted rather than ISUP format, then two additional pieces of information must be taken into account: presentation indicators and screening indicators. If the presentation indicators are set to 'presentation restricted', then a special URI is created by the gateway which communicates to the far end that the caller's identity has been omitted. This URI SHOULD be a SIP URI with a display-name and username of 'Anonymous', e.g.:

From: Anonymous <sip:anonymous@anonymous.invalid>

For further information about privacy in SIP, see Section 5.7.

If presentation is set to 'address unavailable', then gateways should treat the IAM as if the CIN parameter was omitted. Screening indicators should not be translated, as they are only meaningful end-to-end.

12.2 tel URL to ISUP format mapping

Conversion from tel URLs to ISUP format is simpler. If the URI is in international format, then the gateway SHOULD consult the leading country code of the URI. If the country code is local to the gateway (the gateway has one or more trunks that point to switches which are homed to the country code in question), the gateway SHOULD set the NoA to reflect 'national (significant) number' and strip the country code from the URI before populating the digits field. If the country code is not local to the gateway, the gateway SHOULD set the NoA to 'international number' and retain the country code. In either case the NPI MUST be set to 'ISDN numbering plan'.

If the URI is not in international format, the gateway MAY attempt to treat the telephone number within the URI as if it were appropriate to its national or network-specific dialing plan; if doing so gives rise to internal gateway errors or the gateway does not support such procedures, then the gateway SHOULD respond with appropriate SIP status codes to express that the URI could not be understood (if the URI in question is the Request-URI, a 484).

When converting from a tel URL to ISUP calling format, the procedure is identical to that described in the preceding paragraphs, but additionally, the presentation indicator SHOULD be set to 'presentation allowed' and the screening indicator to 'network provided', unless some service provider policy or user profile specifically disallows presentation.

13. Other ISUP flavors

Other flavors of ISUP different than ITU-T ISUP have different parameters and more features. Some of the parameters have more possible values and provide more information about the status of the call.

The Circuit Query Message (CQM) and Circuit Query Response (CQR) are used in many ISUP variants. These messages have no analog in SIP, although receipt of a CQR may cause state reconciliation if the originating and destination switches have become desynchronized; as states are reconciled some calls may be terminated, which may cause SIP or ISUP messages to be sent (as described in Section 10).

However, differences in the message flows are more important. In ANSI [11] ISUP, the CON message MUST NOT be sent; an ANM is sent instead (when no ACM has been sent before the call is answered). In call forwarding situations, CPGs MAY be sent before the ACM is sent. SAMs MUST NOT be sent; 'en-bloc' signaling is always used. The ANSI Exit Message (EXM) SHOULD NOT result in any SIP signaling in gateways. ANSI also uses the Circuit Reservation Message (CRM) and Circuit Reservation Acknowledgment (CRA) as part of its interworking procedures - in the event that an MGC does receive a CRM, a CRA SHOULD be sent in return (in some implementations, transmissions of a CRA could conceivably be based on a resource reservation system); after a CRA is sent, the MGC SHOULD wait for a subsequent IAM and process it normally. Any further circuit reservation mechanism is outside the scope of this document.

Although receipt of a Confusion (CFN) message is an indication of a protocol error, corresponding SIP messages SHOULD NOT be sent on receipt of a CFN - the CFN should be handled with ISUP-specific procedures by the gateway (usually by retransmission of the packet to which the CFN responded). Only if ISUP procedures fails repeatedly should this cause a SIP error condition (and call failure) to arise.

In TTC ISUP CPGs MAY be sent before the ACM is sent. Messages such as a Charging Information Message (CHG) MAY be sent between ACM and ANM. 'En-bloc' signaling is always used and there is no T9 timer.

13.1 Guidelines for sending other ISUP messages

Some ISUP variants send more messages than the ones described in this document. Therefore, some guidelines are provided here with regard to transport and mapping of these ISUP message.

From the caller to the callee, other ISUP messages SHOULD be encapsulated (see [3]) inside INFO messages, even if the INVITE transaction is still not finished. Note that SIP does not ensure that INFO requests are delivered in order, and therefore in adverse network conditions an egress gateway might process INFOs out of order. This issue, however, does not represent an important problem since it is not likely to happen and its effects are negligible in most of the situations. The Information (INF) message and Information Response (INR) are examples of messages that should be encapsulated within an INFO. Gateway implementers might also consider building systems that wait for each INFO transaction to complete before initiating a new INFO transaction.

From the callee to the caller, if a message is received by a gateway before the call has been answered (i.e., ANM is received) it SHOULD be encapsulated in an INFO, provided that this will not be the first SIP message sent in the backwards direction (in which case it SHOULD be encapsulated in a provisional 1xx response). Similarly a message which is received on the originating side (probably in response to an INR) before a 200 OK has been received by the gateway should be carried within an INFO. In order for this mechanism to function properly in the forward direction, any necessary Contact or To-tag must have appeared in a previous provisional response or the message might not be correctly routed to its destination. As such all SIP-T gateways MUST send all provisional responses with a Contact header and any necessary tags in order to enable proper routing of new requests issued before a final response has been received. When the INVITE transaction is finished INFO requests SHOULD also be used in this direction.

14. Acronyms

ACK	Acknowledgment
ACM	Address Complete Message
ANM	Answer Message
ANSI	American National Standards Institute
BLA	Blocking ACK message
BLO	Blocking Message
CGB	Circuit Group Blocking Message
CGBA	Circuit Group Blocking ACK Message
CHG	Charging Information Message
CON	Connect Message
CPG	Call Progress Message
CUG	Closed User Group
GRA	Circuit Group Reset ACK Message
GRS	Circuit Group Reset Message
HLR	Home Location Register
IAM	Initial Address Message
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
ITU-T	International Telecommunication Union Telecommunication Standardization Sector
MG	Media Gateway
MGC	Media Gateway Controller
MTP	Message Transfer Part
REL	Release Message
RES	Resume Message
RLC	Release Complete Message
RTP	Real-time Transport Protocol
SCCP	Signaling Connection Control Part
SG	Signaling Gateway
SIP	Session Initiation Protocol
SS7	Signaling System No. 7
SUS	Suspend Message
TTC	Telecommunication Technology Committee
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
VoIP	Voice over IP

15. Security Considerations

The translation of ISUP parameters into SIP headers may introduce some privacy and security concerns above and beyond those that have been identified for other functions of SIP-T [9A]. Merely securing encapsulated ISUP, for example, would not provide adequate privacy for a user requesting presentation restriction if the Calling Party Number parameter is openly mapped to the From header. Section 12.2 shows how SIP Privacy [9B] should be used for this function. Since the scope of SIP-ISUP mapping has been restricted to only those parameters that will be translated into the headers and fields used to route SIP requests, gateways consequently reveal through translation the minimum possible amount of information.

A security analysis of ISUP is beyond the scope of this document. ISUP bridging across SIP is discussed more fully in [9A], but Section 7.2.1.1 discusses processing the translated ISUP values in relation to any embedded ISUP in a request arriving at PSTN gateway. Lack of ISUP security analysis may pose some risks if embedded ISUP is blindly interpreted. Accordingly, gateways SHOULD NOT blindly trust embedded ISUP unless the request was strongly authenticated [9A], and the sender is trusted, e.g., is another MGC that is authorized to use ISUP over SIP in bridge mode. When requests are received from arbitrary end points, gateways SHOULD filter any received ISUP. In particular, only known-safe commands and parameters should be accepted or passed through. Filtering by deleting believed-to-be dangerous entries does not work well.

In most respects, the information that is translated from ISUP to SIP has no special security requirements. In order for translated parameters to be used to route requests, they should be legible to intermediaries; end-to-end confidentiality of this data would be unnecessary and most likely detrimental. There are also numerous circumstances under which intermediaries can legitimately overwrite the values that have been provided by translation, and hence integrity over these headers is similarly not desirable.

There are some concerns however that arise from the other direction of mapping, the mapping of SIP headers to ISUP parameters, which are enumerated in the following paragraphs. When end users dial numbers in the PSTN today, their selections populate the telephone number portion of the Called Party Number parameter, as well as the digit portions of the Carrier Identification Code and Transit Network Selection parameters of an ISUP IAM. Similarly, the tel URL and its optional parameters in the Request-URI of a SIP, which can be created directly by end users of a SIP device, map to those parameters at a gateway. However, in the PSTN, policy can prevent the user from dialing certain (invalid or restricted) numbers, or selecting certain carrier identification codes. Thus, gateway operators MAY wish to use corresponding policies to restrict the use of certain tel URLs, or tel URL parameters, when authorizing a call.

The fields relevant to number portability, which include in ANSI ISUP the LRN portion of the Generic Address Parameter and the 'M' bit of the Forward Call Indicators, are used to route calls in the PSTN. Since these fields are rendered as tel URL parameters in the SIP-ISUP mapping, users can set the value of these fields arbitrarily. Consequently, an end-user could change the end office to which a call would be routed (though if LRN value were chosen at random, it is more likely that it would prevent the call from being delivered altogether). The PSTN is relatively resilient to calls that have been misrouted on account of local number portability, however. In some networks, a REL message with some sort of "misrouted ported number" cause code is sent in the backwards direction when such a condition arises. Alternatively, the PSTN switch to which a call was misrouted can forward the call along to the proper switch after making its own number portability query - this is an interim number portability practice that is still common in most segments of the PSTN that support portability. It is not anticipated that end users will typically set these SIP fields, and the risks associated with allowing an adventurous or malicious user to set the LRN do not seem to be grave, but they should be noted by network operators. The limited degree to which SIP signaling contributes to the interworking indicators of the Forward Call Indicators and Backward Call Indicator parameters incurs no foreseeable risks.

Some additional risks may result from the SIP response code to ISUP Cause Code parameter mapping. SIP user agents could conceivably respond to an INVITE from a gateway with any arbitrary SIP response code, and thus they can dictate (within the boundaries of the mappings supported by the gateway) the Q.850 cause code that will be sent by the gateway in the resulting REL message. Generally speaking, the manner in which a call is rejected is unlikely to provide any avenue for fraud or denial of service - to the best knowledge of the authors there is no cause code identified in this document that would signal that some call should not be billed, or that the network should take critical resources off-line. However, operators may want to scrutinize the set of cause codes that could be mapped from SIP response codes (listed in 7.2.6.1) to make sure that no undesirable network-specific behavior could result from operating a gateway supporting the recommended mappings. In some cases, operators MAY wish to implement gateway policies that use alternative mappings, perhaps selectively based on authorization data.

If the Request-URI and the To header field of a request received at a gateway differ, Section 7.2.1.1 recommends that the To header (if it is a telephone number) should map to the Original Called Number parameter, and the Request-URI to the Called Party Number parameter. However, the user can, at the outset of a request, select a To header field value that differs from the Request-URI; these two field values are not required to be the same. This essentially allows a user to set the ISUP Original Called Number parameter arbitrarily. Any applications that rely on the Original Called Number for settlement purposes could be affected by this mapping recommendation. It is anticipated that future SIP work in this space will arrive at a better general account of the re-targeting of SIP requests that may be applicable to the OCN mapping.

The arbitrary population of the From header of requests by SIP user agents has some well-understood security implications for devices that rely on the From header as an accurate representation of the identity of the originator. Any gateway that intends to use the From header to populate the called party's number parameter of an ISUP IAM message should authenticate the originator of the request and make sure that they are authorized to assert that calling number (or make use of some more secure method to ascertain the identity of the caller). Note that gateways, like all other SIP user agents, MUST support Digest authentication as described in [1].

There is another class of potential risk that is related to the cut-through of the backwards media path before the call is answered. Several practices described in this document recommend that a gateway signal an ACM when a called user agent returns a 18x provisional response code. At that time, backwards media will be cut through end-to-end in the ISUP network, and it is possible for the called user agent then to play arbitrary audio to the caller for an indefinite period of time before transmitting a final response (in the form of a 2xx or higher response code). There are conceivable respects in which this capability could be used illegitimately by the called user agent. It is also however a useful feature to allow progress tones and announcements to be played in the backwards direction in the 'ACM sent' state (so that the caller won't be billed for calls that don't actually complete but for which failure conditions must be rendered to the user as in-band audio). In fact, ISUP commonly uses this backwards cut-through capability in order to pass tones and announcements relating to the status of a call when an ISUP network interworks with legacy networks that are not capable of expressing Q.850 cause codes.

It is the contention of the authors that SIP introduces no risks with regard to backwards media that do not exist in Q.931-ISUP mapping, but gateways implementers MAY develop an optional mechanism (possibly something that could be configured by an operator) that would cut off such 'early media' on a brief timer - it is unlikely that more than 20 or 30 seconds of early media is necessary to convey status information about the call (see Section 7.2.6). A more conservative approach would be to never cut through backwards media in the gateway until a 2xx final response has been received, provided that the gateway implements some way of prevent clipping of the initial media associated with the call.

Unlike a traditional PSTN phone, a SIP user agent can launch multiple simultaneous requests in order to reach a particular resource. It would be trivial for a SIP user agent to launch 100 SIP requests at a 100 port gateway, thereby tying up all of its ports. A malicious user could choose to launch requests to telephone numbers that are known never to answer, which would saturate these resources indefinitely and potentially without incurring any charges. Gateways therefore MAY support policies that restrict the number of simultaneous requests originating from the same authenticated source, or similar mechanisms to address this possible denial-of-service attack.

16. IANA Considerations

This document introduces no new considerations for IANA.

17. Acknowledgments

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