Network Working Group   G. Camarillo
Request for Comments: 3398   Ericsson
Category: Standards Track   A. B. Roach
                                                          dynamicsoft
                                                         J. Peterson
                                                             NeuStar
                                                             L. Ong
                                                               Ciena
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           Integrated Services Digital Network (ISDN) User Part (ISUP)
                  to Session Initiation Protocol (SIP) Mapping

Status of this Memo

   This document specifies an Internet standards track protocol for the
   Internet community, and requests discussion and suggestions for
   improvements. Please refer to the current edition of the "Internet
   Official Protocol Standards" (STD 1) for the standardization state
   and status of this protocol. Distribution of this memo is unlimited.

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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)
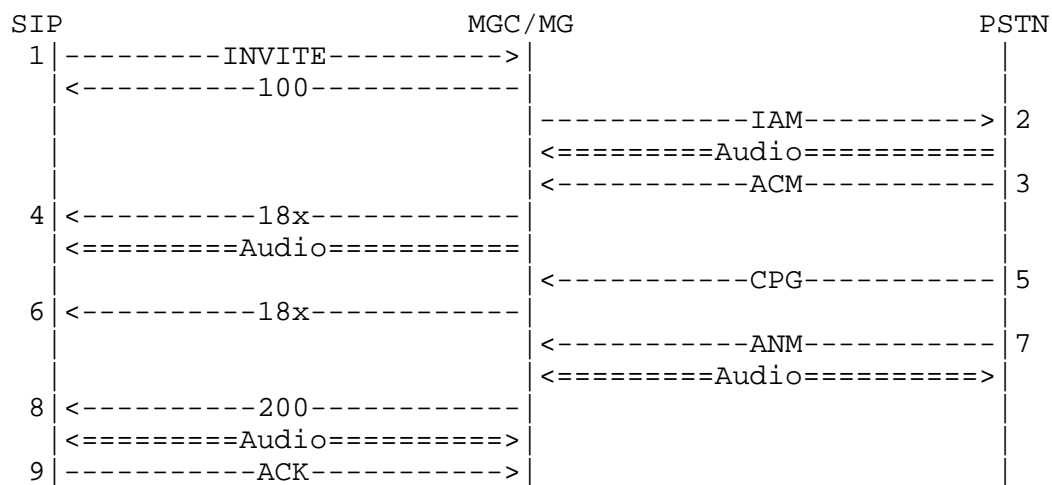
```
           SIP                       MGC/MG                       PSTN
           1|---------INVITE---------->|                           |
            |<----------100------------|                           |
            |                          |------------IAM----------->|2
            |                          |<========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|<----------200------------|                                |
      |<========Audio==========>|                                |
     5|----------ACK----------->|                                |
```
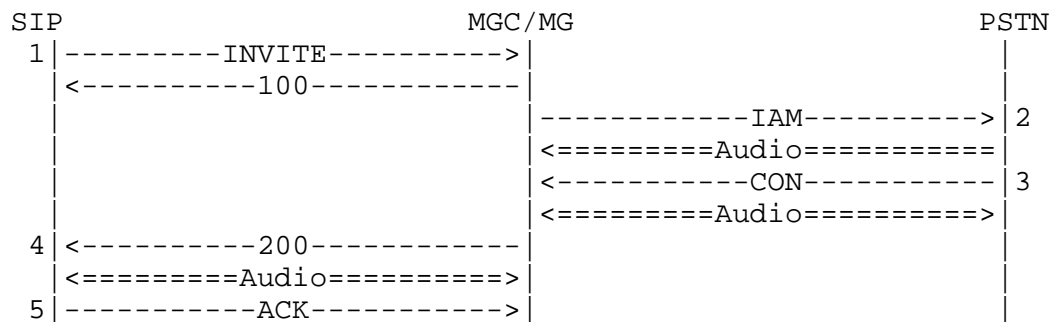
   Note that this flow is not supported in ANSI networks.

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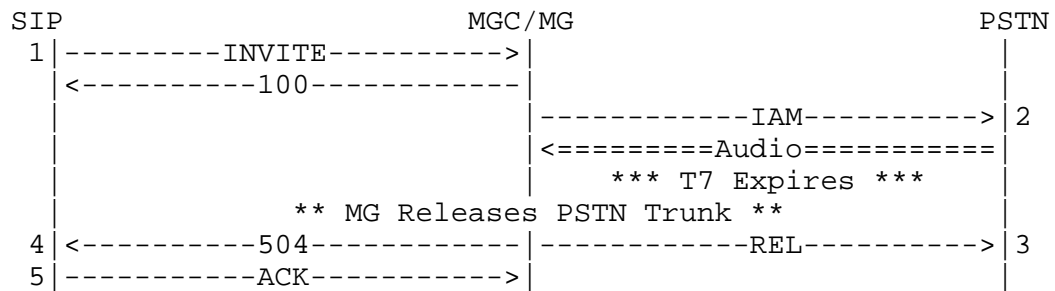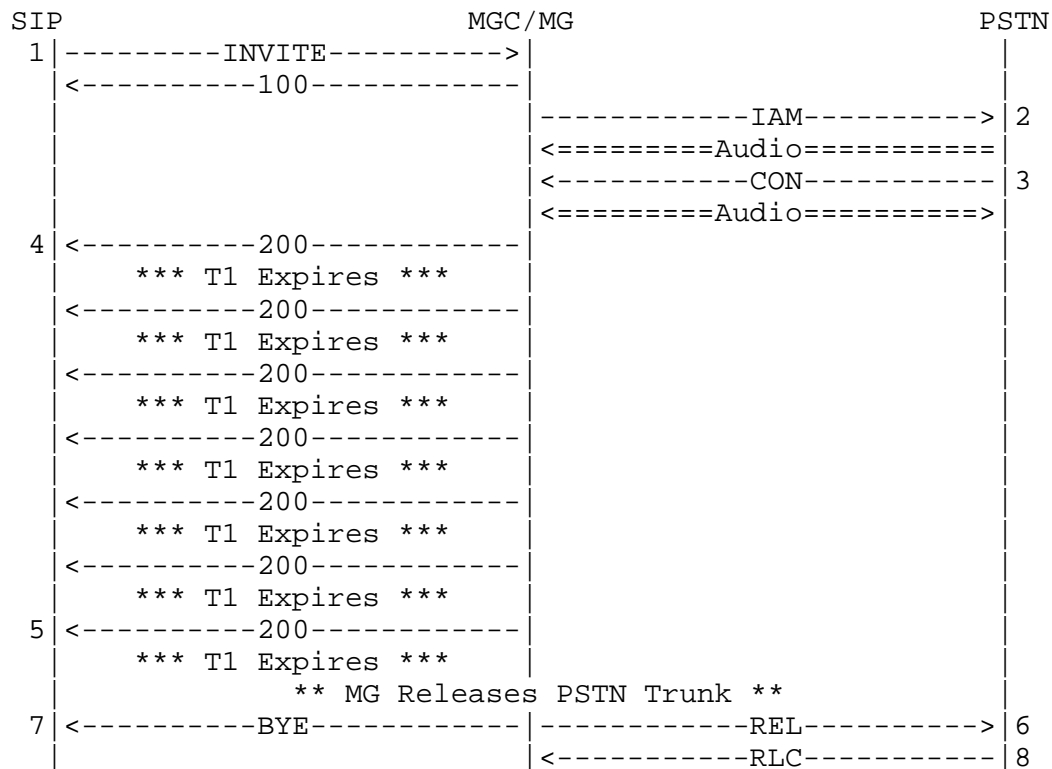
   1. When a SIP user wishes to begin a session with a PSTN user, the
       SIP node issues an INVITE request.

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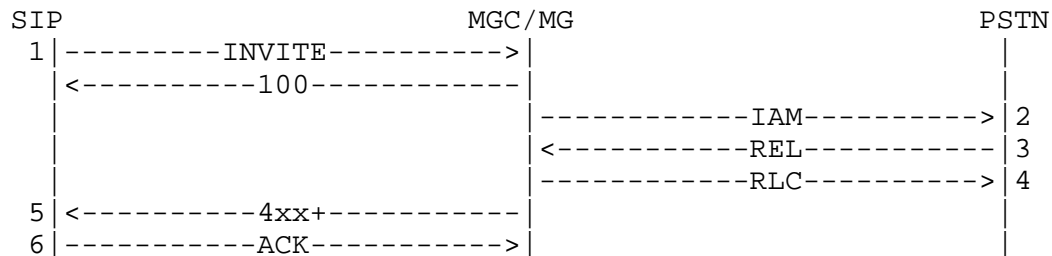
   1. When a SIP user wishes to begin a session with a PSTN user, the
       SIP node issues an INVITE request.

   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
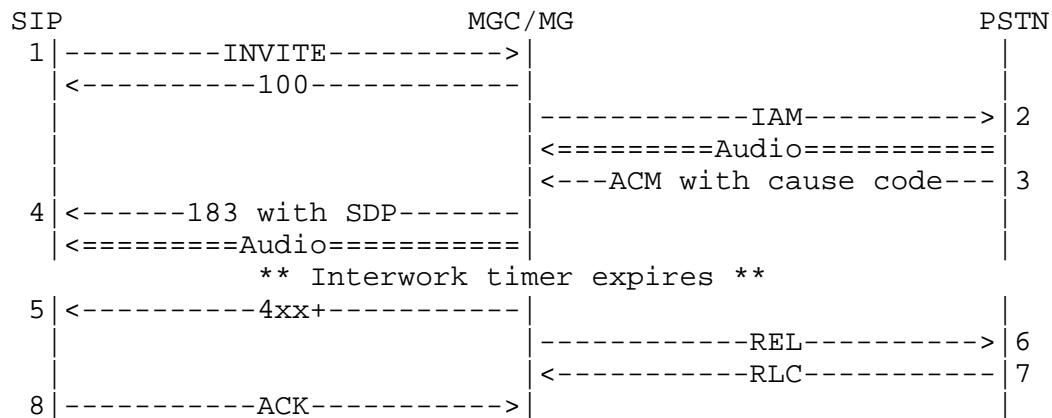
```
      SIP                        MGC/MG                        PSTN
       1|---------INVITE---------->|                            |
        |<----------100------------|                            |
        |                          |------------IAM---------->|2
        |                          |<========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----------->|                          |
```
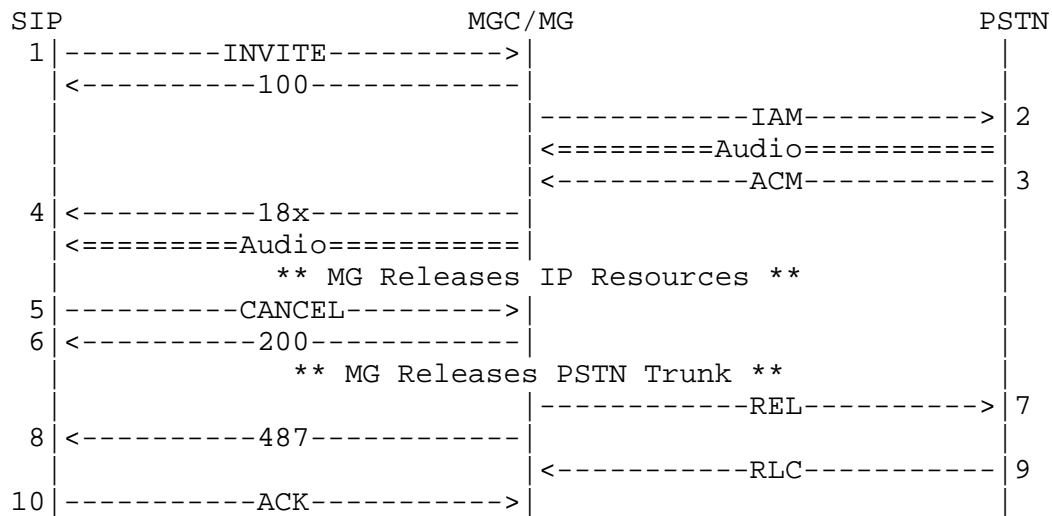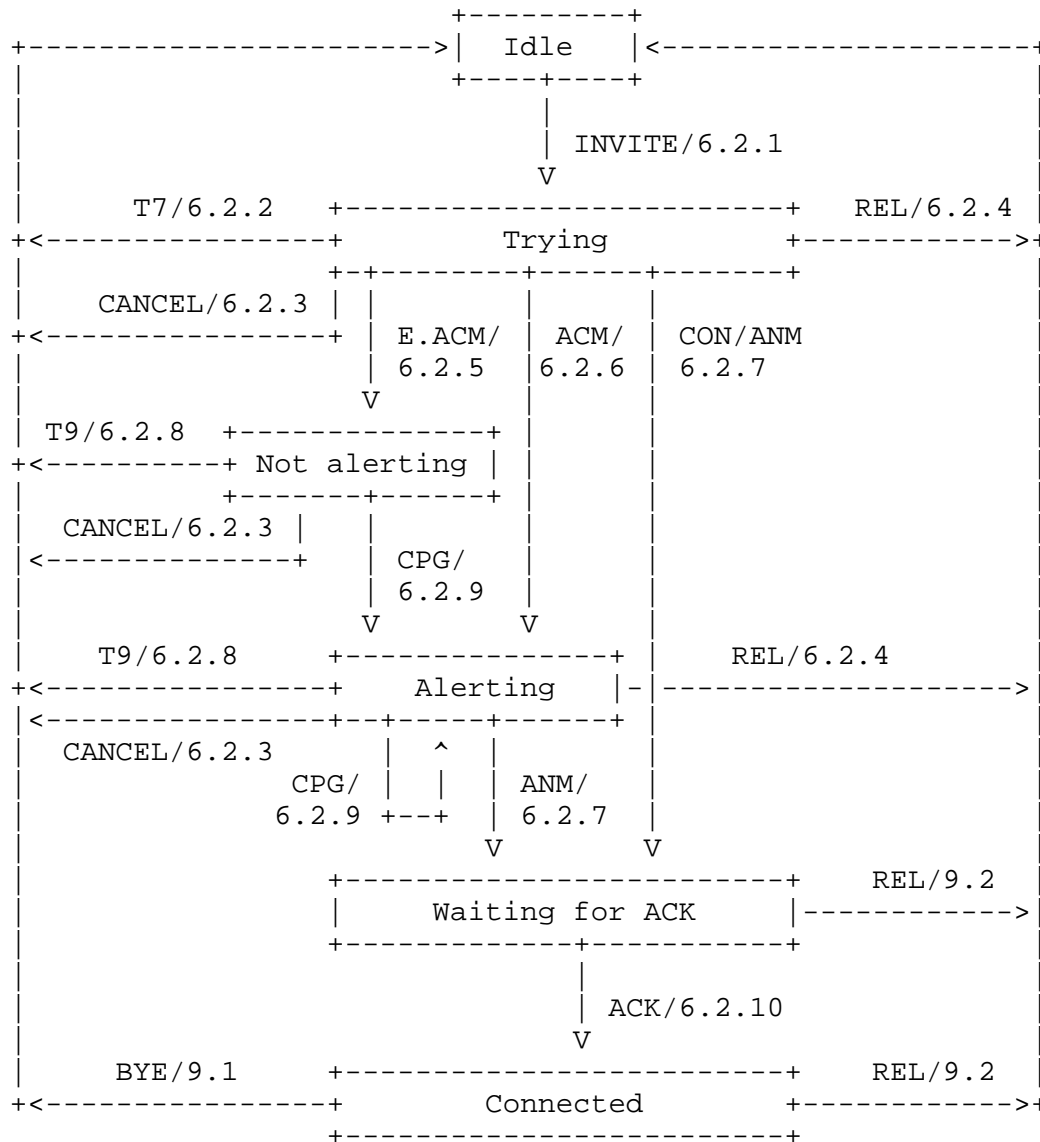
   1. When a SIP user wishes to begin a session with a PSTN user, the
       SIP node issues an INVITE request.

   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)
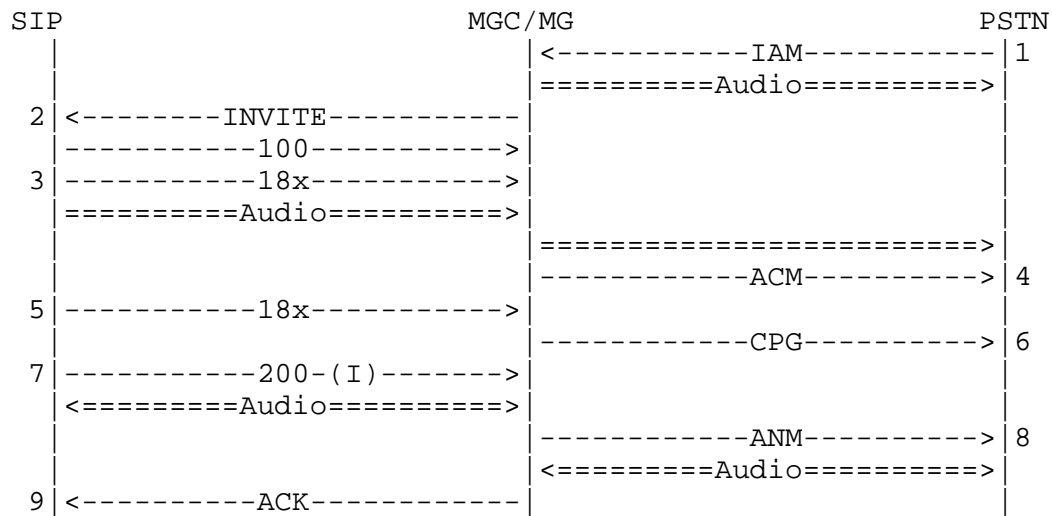
```
          SIP                      MGC/MG                        PSTN
           |                        |<-----------IAM-----------|1
           |                        |=========Audio=========>|
         2 |<--------INVITE-----------|                        |
           |-----------100----------->|                        |
         3 |-----------18x----------->|                        |
           |=========Audio=========>|                        |
           |                        |=========================>|
           |                        |-----------ACM---------->|4
         5 |-----------18x----------->|                        |
           |                        |-----------CPG---------->|6
         7 |-----------200-(I)------->|                        |
           |<========Audio=========>|                        |
           |                        |-----------ANM---------->|8
           |                        |<========Audio=========>|
         9 |<---------ACK-----------|                        |
```
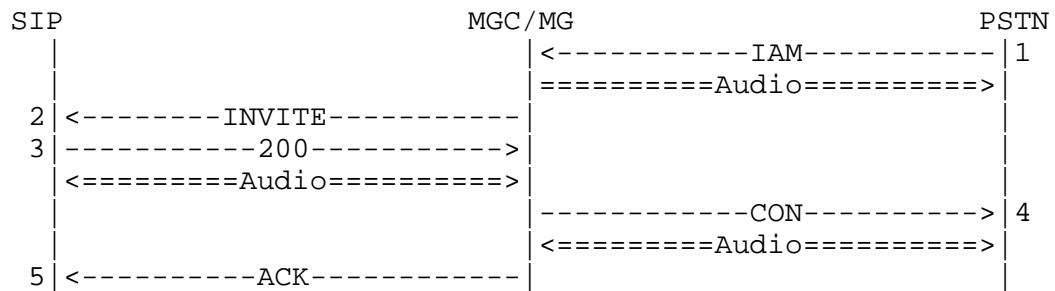
   1. When a PSTN user wishes to begin a session with a SIP user, the
       PSTN network generates an IAM message towards the gateway.

   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
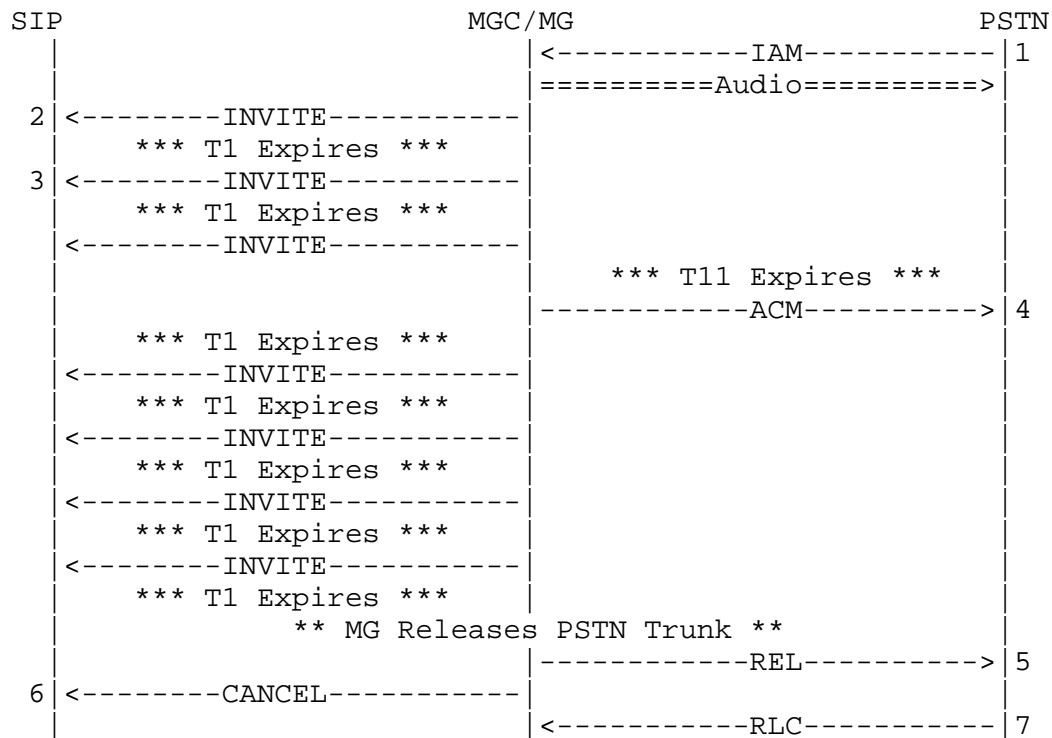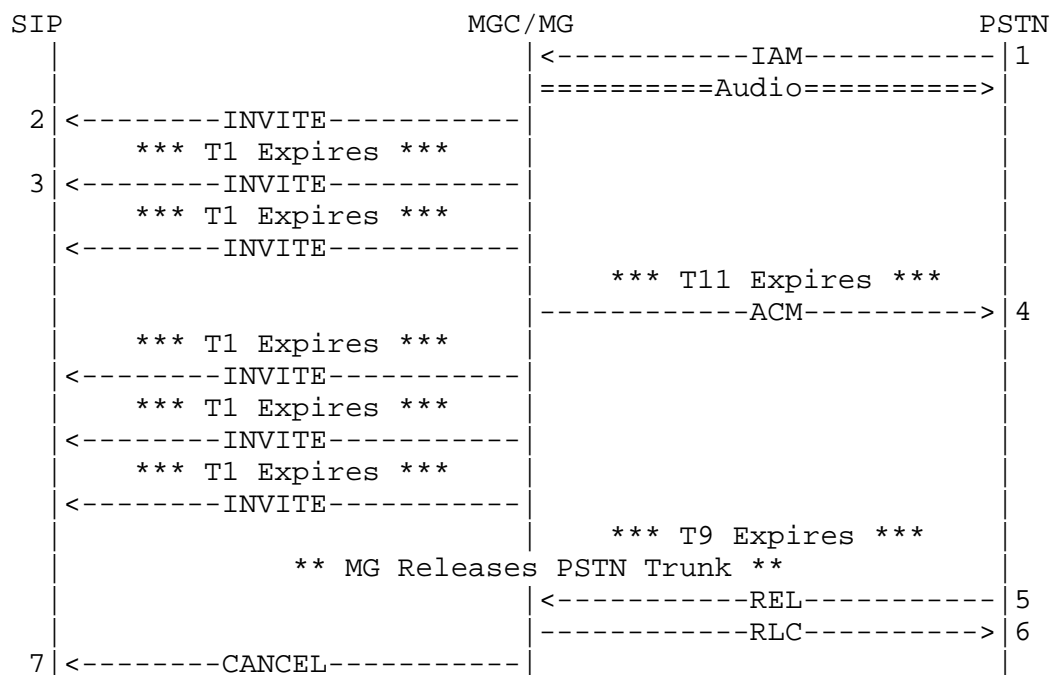   receipt of the INVITE final response.

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
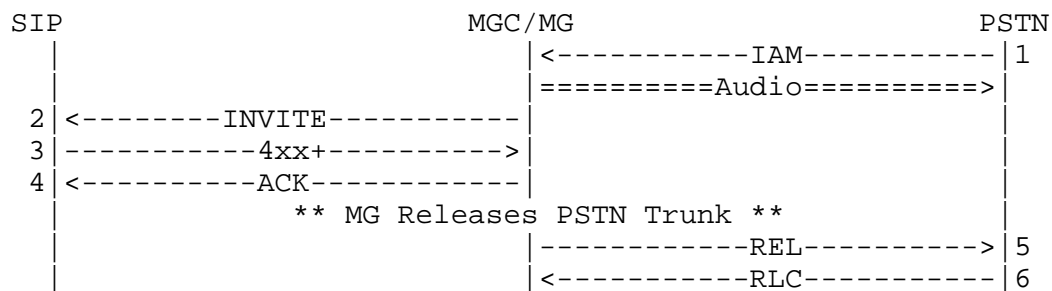   acknowledge.

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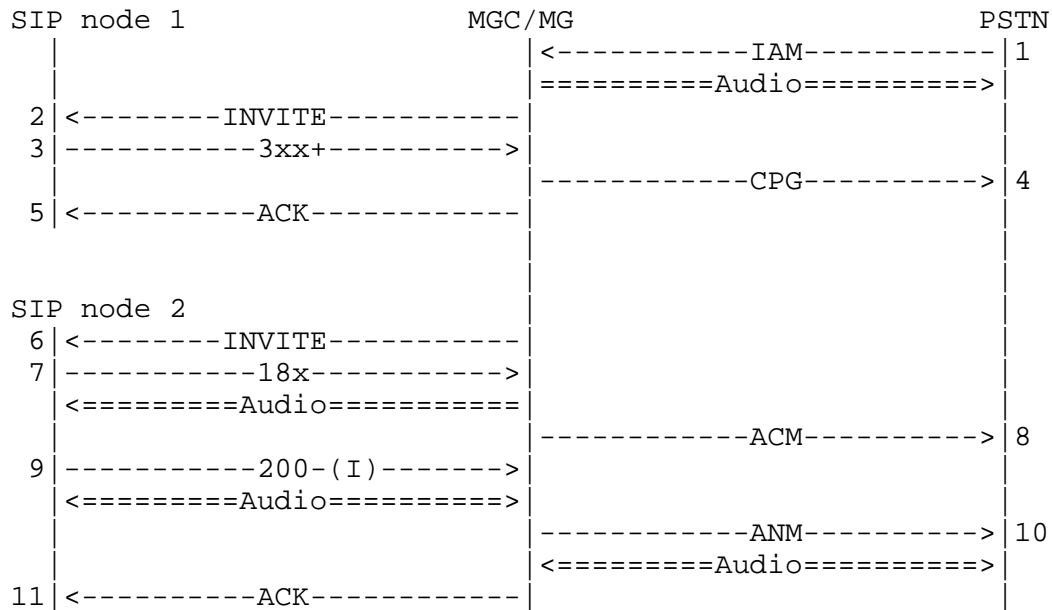
1. When a PSTN user wishes to begin a session with a SIP user, the
    PSTN network generates an IAM message towards the gateway.

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
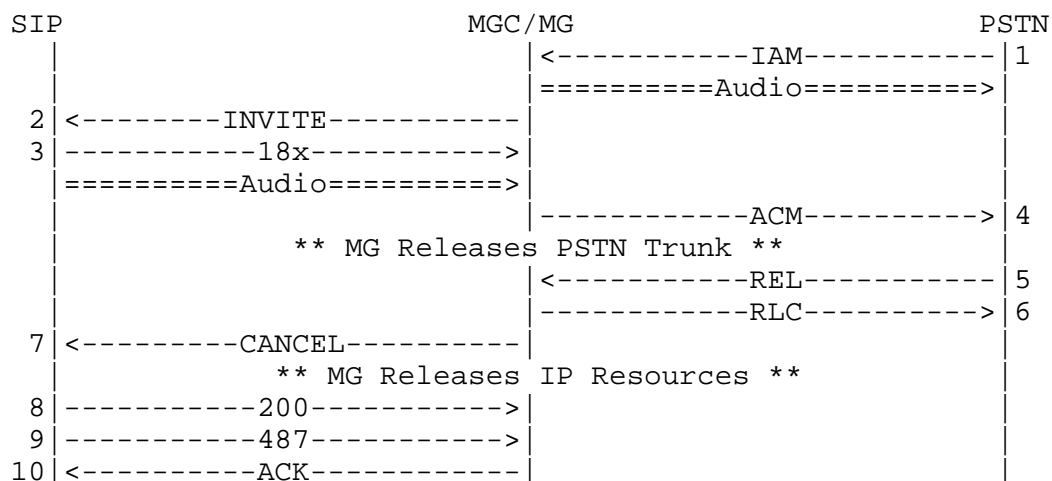   RLC message.

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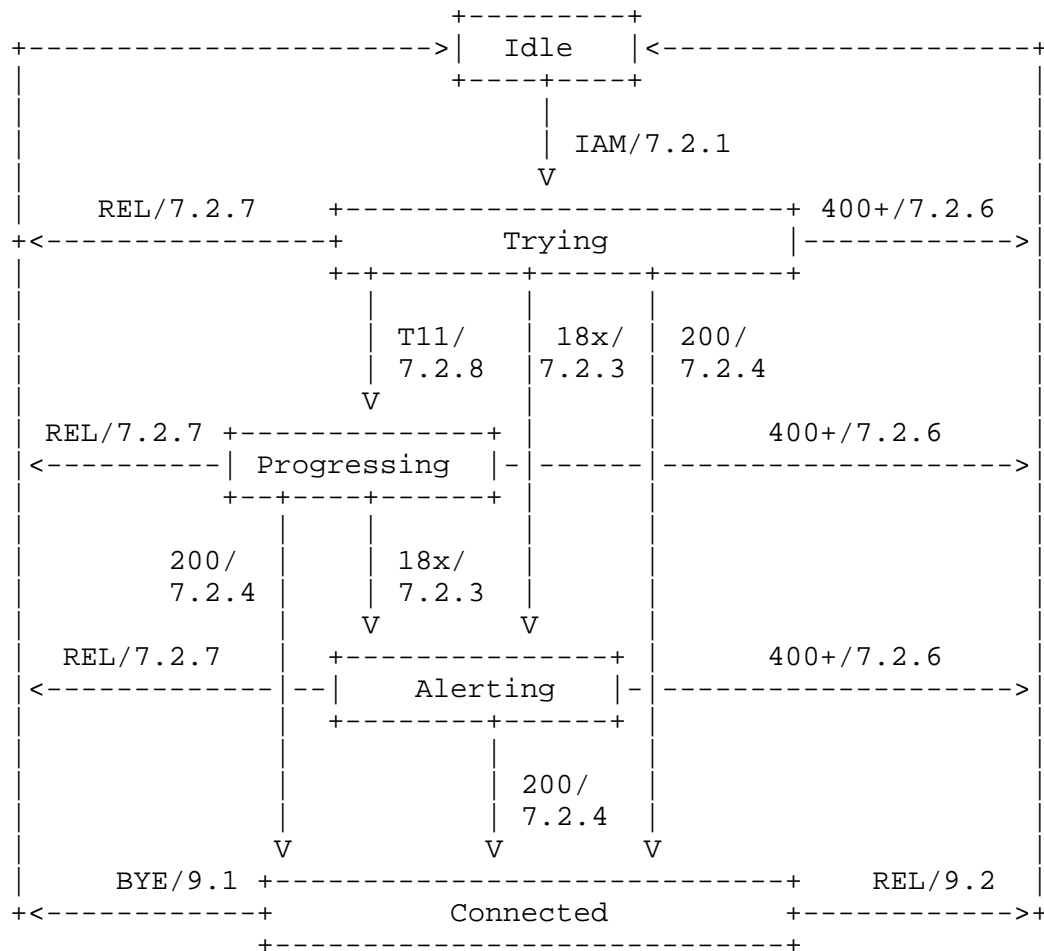
1. When a PSTN user wishes to begin a session with a SIP user, the
    PSTN network generates an IAM message towards the gateway.

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      REL/9.2        |
       |  BYE/9.1 +----------------------------+               |
      +<-----------+       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------------|                            |
```
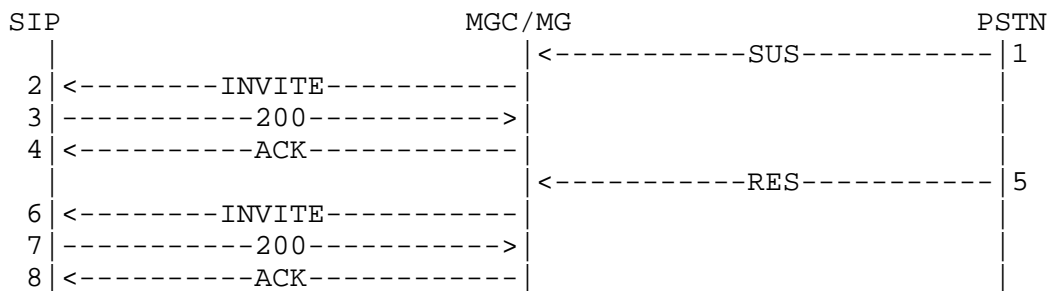
The handling of a network-initiated SUS immediately prior to call teardown is handled in Section 10.2.2.

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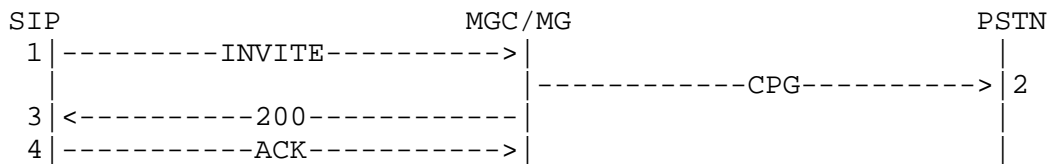
   When such a re-INVITE is received, the gateway SHOULD send a CPG in
   order to express that the call has been placed on hold. The CPG
   SHOULD contain a Generic Notification Indicator (or, in ANSI
   networks, a Notification Indicator) with a value of 'remote hold'.

   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
```
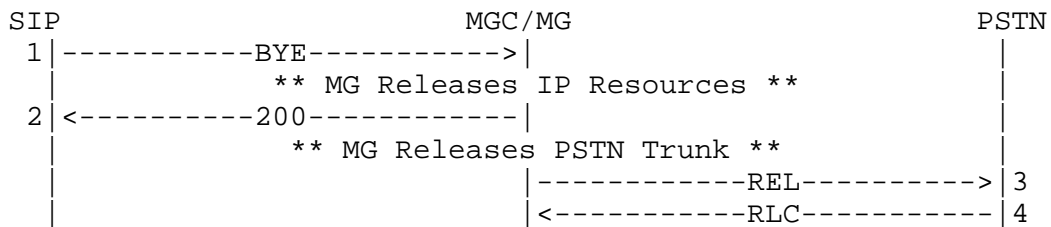
10.2 ISUP initiated release

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 **              |
         |                          |-----------RLC----------->|2
        3|<----------BYE------------|                              |
         |               ** MG Releases IP Resources **            |
        4|-----------200----------->|                              |
```
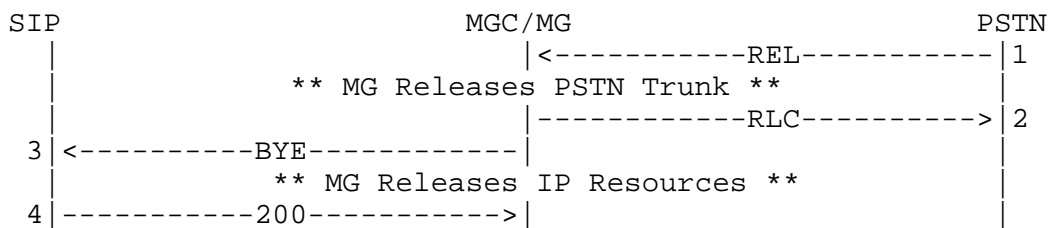
10.2.2 Callee hangs up (SUS)

   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----------->|                           |
```
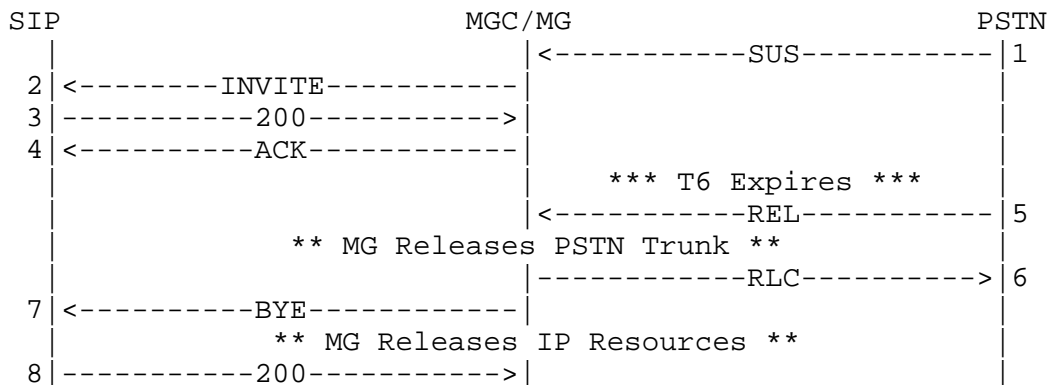
11. ISUP Maintenance Messages

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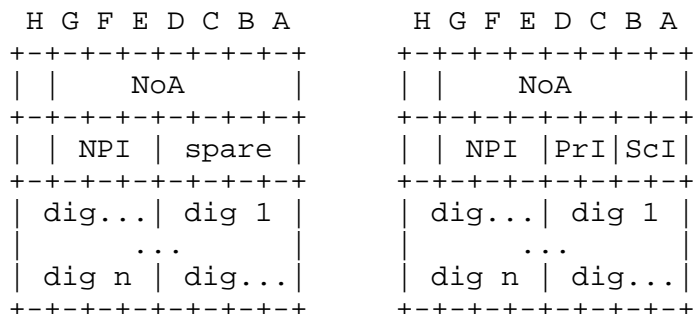
```
      H G F E D C B A           H G F E D C B A
     +-+-+-+-+-+-+-+-+          +-+-+-+-+-+-+-+-+
     | |    NoA      |          | |    NoA      |
     +-+-+-+-+-+-+-+-+          +-+-+-+-+-+-+-+-+
     | | NPI | spare |          | | NPI |PrI|ScI|
     +-+-+-+-+-+-+-+-+          +-+-+-+-+-+-+-+-+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

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Authors' Addresses

    Gonzalo Camarillo
    Ericsson
    Advanced Signalling Research Lab.
    FIN-02420 Jorvas
    Finland

    Phone: +358 9 299 3371
    URI: http://www.ericsson.com/
    EMail: Gonzalo.Camarillo@Ericsson.com

    Adam Roach
    dynamicsoft
    5100 Tennyson Parkway
    Suite 1200
    Plano, TX 75024
    USA

    URI: sip:adam@dynamicsoft.com
    EMail: adam@dynamicsoft.com

    Jon Peterson
    NeuStar, Inc.
    1800 Sutter St
    Suite 570
    Concord, CA 94520
    USA

    Phone: +1 925/363-8720
    EMail: jon.peterson@neustar.biz
    URI: http://www.neustar.biz/

    Lyndon Ong
    Ciena
    10480 Ridgeview Court
    Cupertino, CA 95014
    USA

    URI: http://www.ciena.com/
    EMail: lyOng@ciena.com

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