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SIP 183 Session Progress Message

Status of this Memo

Abstract

   This document describes a proposed extension to SIP [1]. This exten-
   sion adds the 183 Session Progress response and a new header to indi-
   cate why a SDP message body is included in a 18x message.

   The introduction of the 183 informational response message would
   allow a called user agent to indicate to the calling user agent
   whether or not the calling user agent should apply local alerting for
   the session. The existing 180 Ringing message would indicate that
   the calling user agent has the option of providing local alerting
   (and generally should). The 183 Session Progress message would indi-
   cate that the calling user agent should not provide local alerting.
   In additionally, the calling user agent may be called on to establish
   a media session to be used by the called user agent to indicate the
   status of the session setup request as part of the indicated media
   stream. The indication of whether or not to play early media to the
   calling user would be controlled with a new Session header included
   in the 183 message.

1 Introduction

   There are instances, most notably dealing with SIP to PSTN interwork-
   ing, that necessitate that the SIP called User Agent (UA) be able to
   suppress local alerting by the SIP calling UA and to set up a prelim-
   inary media session from the called UA to the calling UA. This would
   allow the called UA to play back media prior to the full SIP session
   being set up. This media would be used to report on the status of
   the session setup request. It could also be used to play music while
   the session setup is attempted. This would be useful for find-me
   like services that involve attempting multiple locations for a single
   setup request.

   The only method in the current SIP specification that allows the
   called UA to playback media is to set up a full SIP session. In PSTN
   interworking situations (and likely in end-to-end SIP sessions) this
   will cause a billing relationship to be established between networks
   for the session. This causes a problem when the reason for setting
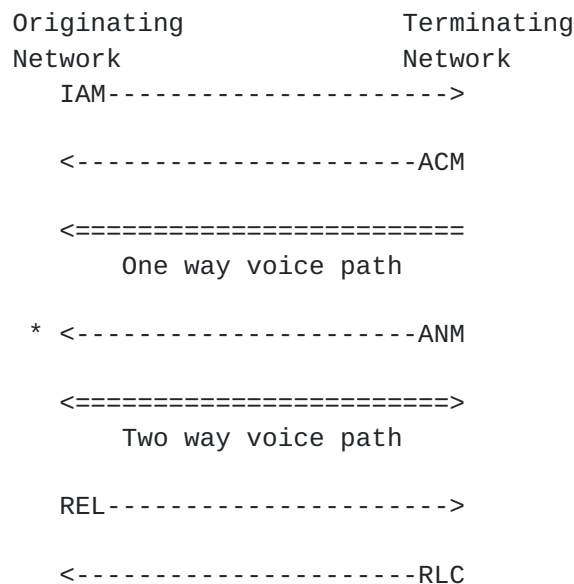   up the media session is to indicate a failure in the session setup.

   This document proposes an extension to the Session Initiation Proto-
   col (SIP) that introduces this capability.

2 PSTN Interworking Issues

   In the PSTN today there are times when a media (voice) path is set up
   from the called party to the calling party in order to play a treat-
   ment (a special tone or announcement). The treatment can range from
   alerting (ring back) to busy tones to announcements explaining why
   the call could not be set up. The participants in this call are not
   charged for the remote treatment portion of the call.

   This one way voice path is generally set up as part of the processing
   of the SS#7 ISUP ACM message.
   The following call flow illustrates call setup using SS7 ISUP in a
   PSTN network.

```
        Originating              Terminating
        Network                  Network
           IAM---------------------->

           <----------------------ACM

           <=======================
               One way voice path

         * <----------------------ANM

           <=======================>
               Two way voice path

           REL---------------------->

           <----------------------RLC
```

 * If the originating network is a Local Exchange Carrier and the termi-
    nating network is an Interexchange Carrier then the LEC will start
    charging for the call at this point in the call.

The following call flow illustrates the setup of a call that does not
result in a completed call but does involve a media path being set
up. In this case, the terminating network may be playing a busy sig-
nal or playing an announcement. The following are examples of
announcements that might be played in this scenario:

- The number you have dialed is no longer valid.

- The wireless subscriber you are calling is not currently reachable.

```
          Originating                   Terminating
          Network                       Network
             IAM---------------------->

             <---------------------ACM

             <========================
                 One way voice path

             REL---------------------->

             <---------------------RLC
```

2.1 PSTN to SIP Network Interworking Requirements

The following are a subset of the requirements for interworking
between a PSTN network and a SIP network.

When the SIP network is in the middle of two PSTN networks, it must
support the following:

- The ingress gateway into the SIP network shall have the ability to
determine, based on SIP signaling messages, when to send an ISUP ACM
message and when to send an ISUP ANM message.

- The SIP network shall have the ability to support fast setup. This
occurs when the terminating network does not send an ACM prior to
sending an ANM.

- The SIP network shall support the ability to cut through a voice

path from the terminating PSTN network to the originating PSTN net-
work without the interim SIP network incurring charges from the orig-
inating network.

The SIP network shall support the ability to place calls to a PSTN
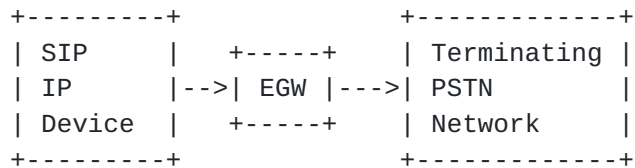network without the egress gateway knowing what type of device the
call was originated from. Thus, the egress gateway shall not need to
behave differently when the call originates from a PSTN network then
when the call originates from a native IP SIP device.

The following is an illustration of the two scenarios that must be
supported:
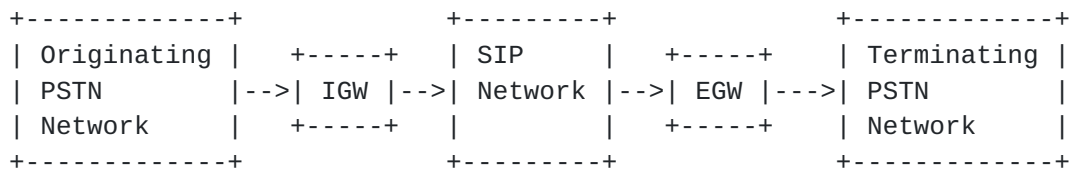
```
   +-------------+                 +---------+                 +-------------+
   | Originating |     +-----+     | SIP     |     +-----+     | Terminating |
   | PSTN        |-->| IGW |-->| Network |-->| EGW |--->| PSTN        |
   | Network     |     +-----+     |         |     +-----+     | Network     |
   +-------------+                 +---------+                 +-------------+

    IGW = Ingress Gateway
    EGW = Egress Gateway

                                   +---------+                 +-------------+
                                   | SIP     |     +-----+     | Terminating |
                                   | IP      |-->| EGW |--->| PSTN        |
                                   | Device  |     +-----+     | Network     |
                                   +---------+                 +-------------+
```

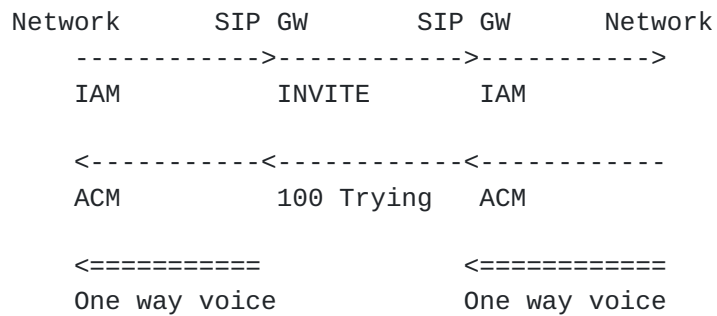## 2.2 Solution Options With Existing SIP Specification

The following sections show the results of investigating various
options for addressing the above requirements using existing SIP pro-
tocol capabilities. In each case, it is shown why the option either
cannot address the requirements or has short comings that can be bet-
ter addressed using the 183 Session Progress message.

2.2.1 100 Trying Mapped to ACM

The first option investigated involved mapping the 100 Trying message
to the ACM message.

The following call flow illustrates this option.

```
         Originating  Ingress      Egress     Terminating
```

```
        Network       SIP GW       SIP GW       Network
        ------------>------------>----------->
          IAM          INVITE        IAM

        <-----------<------------<------------
          ACM          100 Trying  ACM

        <===========              <============
        One way voice             One way voice
```

   The call flow breaks at this point for two reasons. First, at this
   point in the call flow a one way voice path is needed so that the
   terminating network can provide session setup status as part of the
   voice path. The 100 Trying does not cause a voice path cut-through
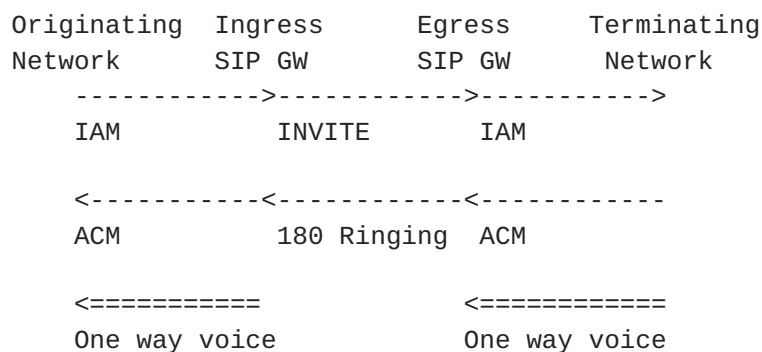   between the ingress and egress gateways. This potentially could be

   addressed by allowing the 100 Trying to carry SDP information to be
   used for carrying the preliminary session media. This option is
   explored in the context of the 180 Ringing message in section 3.3.

   The use of the 100 Trying also fails because a SIP Proxy Server sit-
   ting in the signaling path between the ingress gateway and the egress
   gateway might have generated the 100 Trying message. This would
   result in the ACM message being sent prior to the egress gateway
   receiving an ACM from the terminating network.

2.2.2 180 Ringing Mapped to ACM

   The second option investigated was to use a 180 Ringing message to
   trigger the ACM message at the ingress gateway.

   This option is illustrated in the following call flow:

```
        Originating  Ingress      Egress       Terminating
        Network       SIP GW       SIP GW        Network
        ------------>------------>----------->
          IAM          INVITE        IAM

        <-----------<------------<------------
          ACM          180 Ringing  ACM

        <===========              <============
        One way voice             One way voice
```
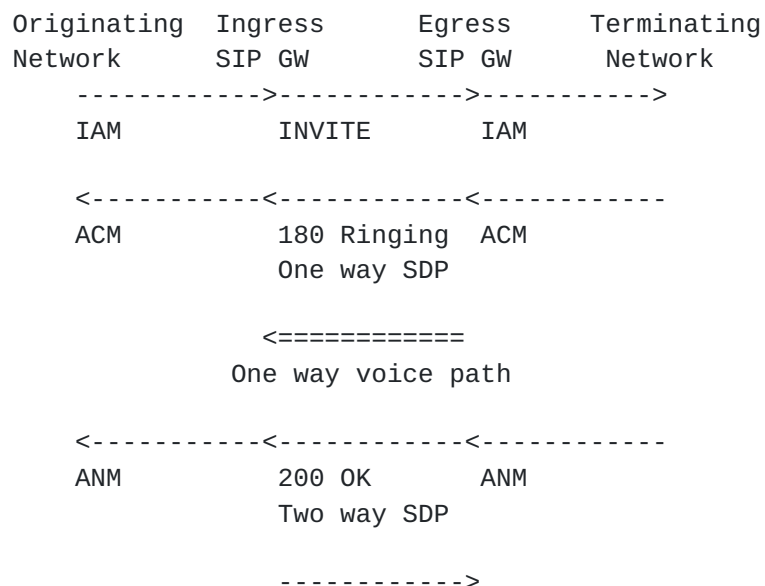
   This option breaks at this point because a media voice path cannot be
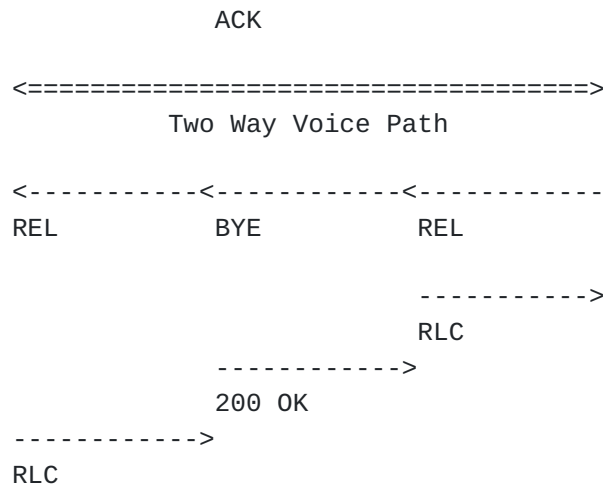   cut through at this point for the terminating PSTN network to report

on the session progress. This is due to the fact that the egress
gateway has not yet communicated its RTP information to the ingress
gateway. The next two options attempt to address this issue.

2.2.3 180 With SDP Mapped to ACM

The next option investigated involves using the presence of SDP in
the 180 Ringing message to indicate that session progress will be
communicated by the called user agent using the media stream. In
this case, absence of the SDP message body would indicate that local
alerting should take place.

The following call flow illustrates this option:

```
        Originating  Ingress      Egress     Terminating
        Network      SIP GW       SIP GW       Network
        ------------>------------>----------->
          IAM          INVITE       IAM


        <-----------<------------<------------
          ACM          180 Ringing  ACM
                       One way SDP


             <============
             One way voice path


        <-----------<------------<------------
          ANM          200 OK       ANM
                       Two way SDP


                       ------------>
```

```
                            ACK

            <=================================>
                     Two Way Voice Path

            <----------<------------<------------
            REL          BYE          REL

                            ----------->
                            RLC
                    ------------>
                    200 OK
            ------------>
            RLC
```
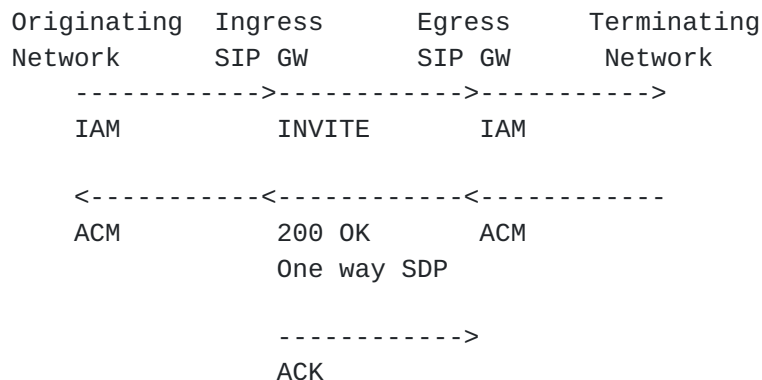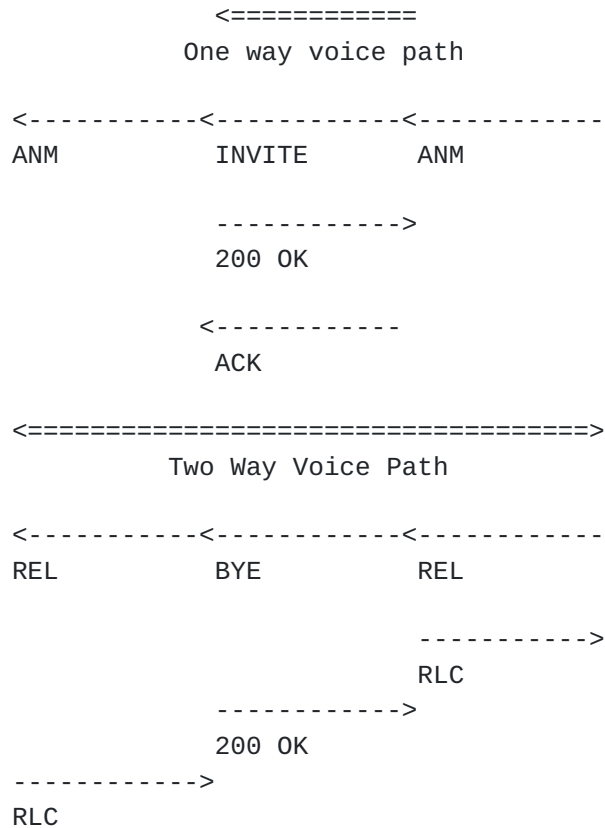
   Although this option looks promising on first review, it does not
   give the called user agent the ability to include SDP in the message
   and rely on the calling user agent (the ingress gateway in this sce-
   nario) to provide local alerting. As illustrated in [2] there are
   other reasons that SDP might be included in a 180 Ringing message.
   Thus the user requiring a coupling of SIP and QOS signaling, which
   requires inclusion of SDP in the 18x message, could not also request
   local alerting.

2.2.4 200 OK Mapped to ACM

   The final option investigated involves setting up a full media ses-
   sion in the SIP network prior to receiving the ANM from the terminat-
   ing PSTN network. This involves mapping the 200 OK to the ACM mes-
   sage at the ingress gateway and having the egress gateway send a re-
   INVITE upon receipt of the ANM. The ingress gateway would use the
   re-INVITE to trigger the ANM message.

   This option is illustrated in the following call flow:

```
        Originating  Ingress      Egress      Terminating
        Network      SIP GW       SIP GW        Network
        ------------>------------>----------->
          IAM          INVITE       IAM

        <----------<------------<------------
          ACM          200 OK       ACM
                       One way SDP

                       ------------>
                       ACK
```

```
                    <============
                    One way voice path


             <-----------<------------<------------
             ANM          INVITE       ANM


                    ------------>
                    200 OK


                    <-----------
                    ACK


             <================================>
                    Two Way Voice Path


             <-----------<------------<------------
             REL          BYE          REL


                              ----------->
                              RLC
                    ------------>
                    200 OK
             ------------>
             RLC
```
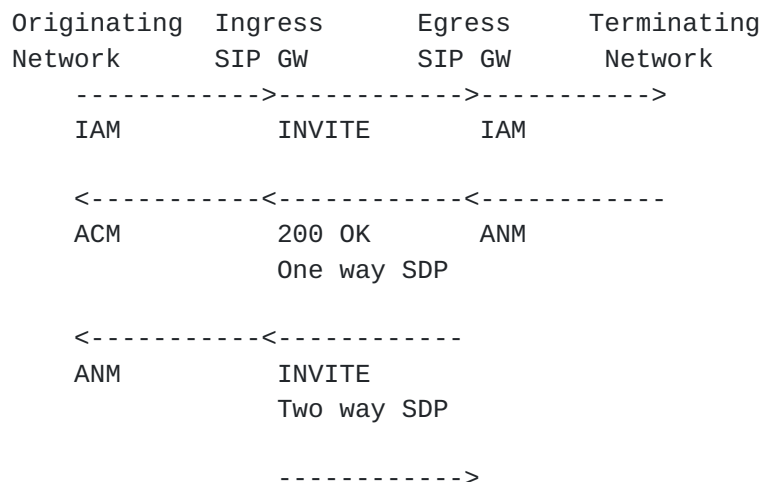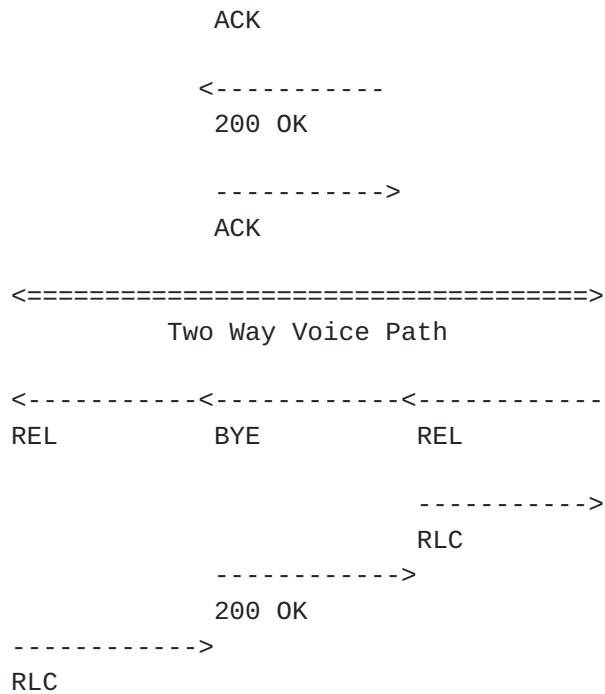
Although this will work in the above scenario, it introduces additional
messaging overhead. In addition, as illustrated in the following fast

answer call flow, it is at best awkward and may result in clipping off
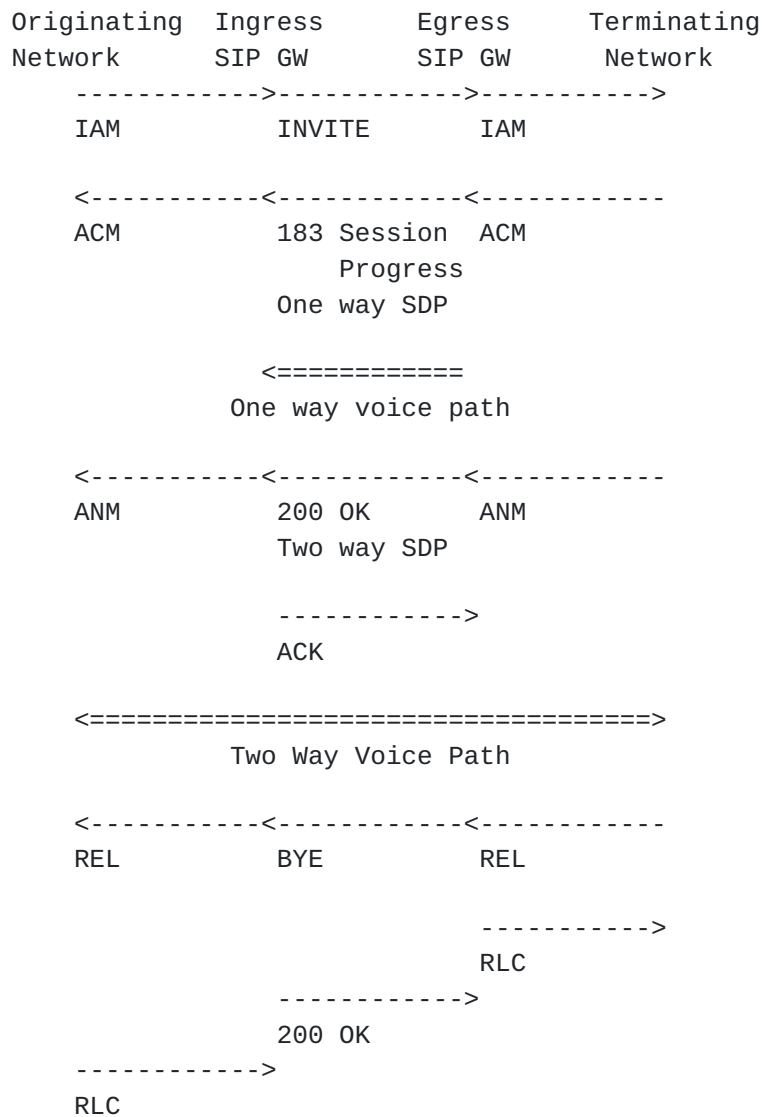of the beginning of the voice call.

```
         Originating  Ingress      Egress     Terminating
         Network      SIP GW       SIP GW      Network
             ------------>------------>----------->
             IAM          INVITE       IAM


             <-----------<------------<------------
             ACM          200 OK       ANM
                          One way SDP


             <-----------<------------
             ANM          INVITE
                          Two way SDP


                          ------------>
```

```
                     ACK

            <-----------
             200 OK

             ----------->
             ACK

      <================================>
              Two Way Voice Path

      <-----------<-----------<-----------
      REL          BYE          REL

                     ----------->
                     RLC
             ------------>
             200 OK
      ------------>
      RLC
```

2.3 **Proposed 183 Session Progress**

   The following session signaling flows show the proposed solution
   using the 183 Session Progress Message to map to the ISUP ACM message
   and how the 183 Session Progress message is used for when the call
   originates from a SIP IP Device.

2.3.1 **PSTN to SIP to PSTN Session Using 183 Session Progress**

   The following session signaling flow shows the use of the 183 Session
   Progress message for a session setup in a SIP based network when the
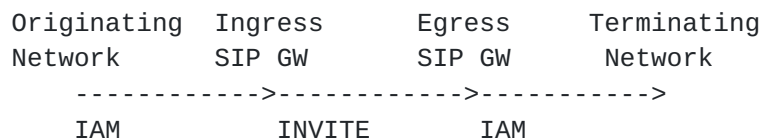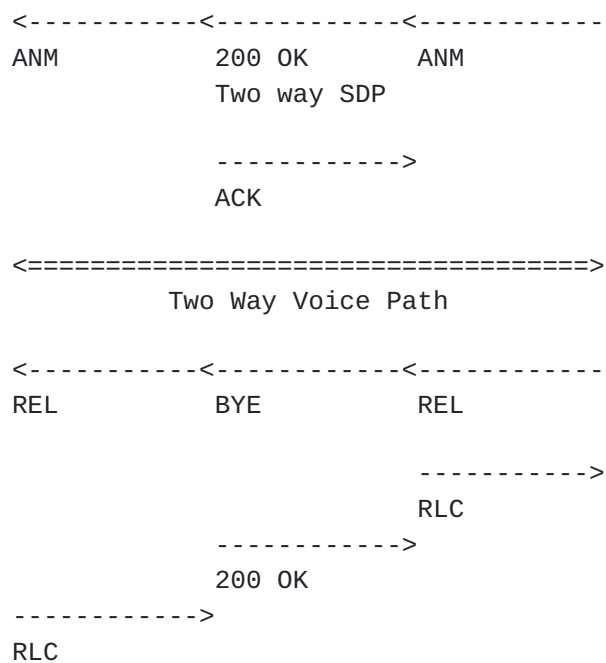   session will be between two PSTN networks.

```
         Originating  Ingress      Egress     Terminating
         Network      SIP GW       SIP GW      Network
             ------------>------------->----------->
             IAM           INVITE        IAM


             <-----------<------------<------------
             ACM           183 Session  ACM
                               Progress
                           One way SDP


                  <============
                  One way voice path


             <-----------<------------<------------
             ANM           200 OK        ANM
                           Two way SDP


                           ------------>
                           ACK


             <================================>
                  Two Way Voice Path


             <-----------<------------<------------
             REL           BYE          REL


                           ----------->
                           RLC
                  ------------>
                  200 OK
             ------------>
             RLC
```

2.3.2 PSTN Fast Answer

   The following session signaling flow shows the method for handling of
   the fast answer scenario. Note that in this case the 183 Session
   Progress message is not used, as the ANM is mapped directly to a 200

   OK. This meets the requirement that the SIP gateways must be able to
   differentiate between ACM and ANM messages.

```
         Originating  Ingress      Egress     Terminating
         Network      SIP GW       SIP GW      Network
             ------------>------------->----------->
             IAM           INVITE        IAM
```

```
           <------------<-------------<------------
           ANM           200 OK         ANM
                         Two way SDP


                         ------------>
                         ACK


           <===================================>
                     Two Way Voice Path


           <------------<-------------<------------
           REL           BYE            REL


                                   ----------->
                                   RLC
                         ------------>
                         200 OK
           ------------>
           RLC
```

2.3.3 SIP to PSTN Session Using 183 Session Progress

The following session signaling flow shows the use of the 183 Session
Progress message for a session setup in a SIP based network when the
session originates in the SIP network and terminates to a PSTN net-

work.

```
          Calling        Egress     Terminating
            UA           SIP GW        Network
            ------------>------------>
             INVITE        IAM

            <-----------
            100 Trying

            <-----------<------------
            183 Session  ACM
                 Progress
            One way SDP

            <======================
                 One way voice path

            <-----------<------------
            200 OK       ANM

            ------------>
            ACK

            <======================
                 Two Way Voice Path

            <-----------<------------
            BYE          REL

                         ------------>
                         RLC
            ------------>
            200 OK
```

** Session Description Message Bodies in 18x Responses**

The previous sections illustrated how the 183 response can be used to
communicate the need for the calling user agent to receive media
prior to receiving a final response from the called user agent.

There are other reasons for including SDP message bodies in an 18x
message.

One possible use is to use the SDP to establish security and/or QoS
relationship between the called and calling user agents, as

documented in [2].

As a result, it is necessary to communicate to the calling user agent
why the SDP was included in the 18x message. The following section
describes the Session header, which will be used for this purpose.

3.1 Session Header in 18x Message Bodies

The Session header will be used to communicate to the calling user
agent the reason for a SDP message body being included in an 18x mes-
sage. The valid values for the Session header are Media, QoS and
Security. Multiple of these values can be indicated.

A value of Media indicates that the SDP should be used for establish-
ing of an early media session. The early media session will gener-
ally be used to communicate the status of the session but could also
be used for other reasons. For instance, it could be used to play
music while the calling user is being alerted.

A value of QoS indicates that the SDP should be used for establishing
of a QoS relationship between the calling and called user agents.
This could involve the user agents requesting QoS resources using
RSVP or some other signaling mechanism.

A value of Security indicates that the SDP should be used for estab-
lishing of a security relationship between the calling and called
user agents. This could be an IPSEC based relationship, for example.

4 Format and Usage

The format of provisional responses with media session descriptions
is identical to that of 200-class responses to INVITE requests,
except as noted in section 7, "Reliability"; the body will contain a
session description (usually SDP; see ref [5]).

4.1. Temporary Media Establishment

Under most circumstances, provisional responses used to initiate tem-
porary media will contain SDP that is a subset of the media descrip-
tion presented in the INVITE message (as in normal 200 responses).

If the original INVITE message contains no media description, the
server will generate SDP representing the capabilities it requires
for media transmission and include it in the provisional response.
The client will include a final SDP in its acknowledgement of receipt
(see section 4, "Reliability").

In both cases, the media streams will be established after the mes-
sage confirming receipt of the provisional response has been sent
(from the client's perspective) or received (from the server's per-
spective).

   The designation of media capabilities in a provisional response has
   no implications on the capabilities of any subsequent temporary con-
   nections or the final connection. Each media stream is negotiated
   relative to the session description in the original INVITE request
   (or lack thereof).

4.2. Change of Temporary Media

   After a temporary media stream has been established, its parameters
   can be changed by sending further provisional responses that also
   contain session descriptions. Upon receipt of such a response, the
   client MUST immediately cease transmission of media relating to the
   old temporary stream. As before, the new temporary media stream is
   established after acknowledgement of the provisional response.

   Provisional responses which contain no session description SHOULD NOT
   have an effect on any currently established temporary media stream.

4.3. Discontinuation of Temporary Media

   Sending of temporary media MUST be discontinued upon the sending
   (from the server's perspective) or the receipt (from the client's
   perspective) of any INVITE final response.

   Sending a provisional response that contains a session description
   with all media stream port numbers set to zero can also discontinue a
   temporary media stream.

4.4. New Provisional 183 Status Code

   To allow for transmission of temporary media which does not corre-
   spond to the four provisional status codes defined in the SIP RFC
   (ref [1]), this protocol extension defines one additional response
   code of "183 Session Progress."

   The 183 Session Progress response can be used for any arbitrary in-
   band communication of call status. It SHOULD NOT, however, be used to
   convey ringing, forwarding, or call queueing situations.

   When applicable, the response text SHOULD include a text representa-
   tion of the information conveyed by the media stream. In the case of
   a recorded announcement, this text SHOULD be the text of the
   announcement. For a tone, this text SHOULD be either the name of the
   tone as defined in E.182 (ref [6]) (e.g. "Payphone Recognition Tone")
   or a description of the condition the tone is attempting to report
   (e.g. "The Called Party is a Payphone").

4.5 Reliability

Clients which understand this extension SHOULD also understand the
extension described in "Reliability of Provisional Responses in SIP"
(ref [3]) and indicate that they require reliable transmission of

provisional responses in Require: and Proxy-Require: headers.

4.6 Media Negotiation Failure for Temporary Media

If no acceptable media type is available in the client's INVITE
request session description, the server MAY return a "406 Not Accept-
able" message; the alternative is to forgo the transmission of provi-
sional media. While it is perhaps a more appropriate error code, "606
Not Acceptable" is not suggested, owing to its properties of termi-
nating any ongoing searches.

If the client finds the session description proposed by the server in
a provisional response unacceptable, its acknowledgement SHOULD con-
tain a session description with all media stream port numbers set to
zero. A server which receives such a message MAY respond with a "406
Not Acceptable" message; the alternative is to forgo the transmission
of provisional media.

4.7 Issues

There are situations when the called user agent (the UAS) requires
the support of the mechanisms defined in this document and does not
know through the INVITE message whether the calling user agent (the
UAC) supports the extension. There is currently no method for the
UAS to communicate to the UAC the required capabilities required to
properly setup the session.

This is a general problem with the SIP protocol that should be fixed
in an upcoming version. The mechanisms established for the SIP pro-
tocol will be used to indicate a need for this extension.

5 Proposed Extensions to the SIP Specification

The remainder of the document describes the proposed extensions to
the SIP specification. The section number indicates the section of
the SIP specification that requires modification. Thus section 5.M.N
would include proposed modifications to section M.N of the SIP speci-
fication.

Absence of a section indicates that no modifications are proposed for
that section.

5.5.1.1 Status Codes and Reason Phrases

The following is the updated Figure 5:

```
    Informational =  "100"  ; Trying
                  |   "180"  ; Ringing
                  |   "181"  ; Call is Being Forwarded
                  |   "182"  ; Queued
                  |   "183"  ; Session Progress
    Success       =  "200"  ; OK
```

Figure 5: Informational and Success codes

5.6 Header Field Definitions

The following needs to be added to Table 5

	where	enc.	e-e	ACK	BYE	CAN	INV	OPT	REG
Session	18x		e	-	-	-	-	-	-

5.6.43 Session

The session header is used in 18x response messages to communicate to the UAC the purpose of the session description message body included in response message.

The valid values are Media, QoS and Security.

A value of Media indicates that the SDP should be used for establish-ing of an early media session. The early media session will gener-ally be used to communicate the status of the session but could also be used for other reasons. For instance, it could be used to play music while the calling user is being alerted.

A value of QoS indicates that the SDP should be used for establishing of a QoS relationship between the calling and called user agents. This could involve the user agents requesting QoS resources using RSVP or some other signaling mechanism.

A value of Security indicates that the SDP should be used for estab-
lishing of a security relationship between the calling and called
user agents. This could be an IPSEC based relationship, for example.

See [2] for details on the handling of the QoS and Security values.

        Session = "Session" ":" 1#("Media" | "QoS" | "Security")

5.7 Status Code Definitions

5.7.1 Informational 1xx

5.7.1.2 180 Ringing

   The following text is proposed to be added to the description of the
   180 Ringing message:

   The calling UA should initiate local alerting (for instance, the
   playing of a ringing tone or other alerting mechanism) so as to indi-
   cate the progress of the session setup.

5.7.1.5 183 Session Progress

   The called UA may have the need to communicate session progress with-
   out the indication that the called user is being alerted.

   The calling UA should not apply local alerting upon receipt of the
   183 Session Progress Response.

   The 183 Session Progress may have been sent to communicate the status
   of the session setup attempt as part of a media stream. The called
   user agent will indicate this by including the Session header with a
   value of media.

   In this case, the calling UA shall establish a media session accord-
   ing to the contents of the session description contained in the 183
   message. The calling UA should not apply local alerting that would
   interfere with the media session information supplied by the called
   UA.

   The 183 message SHOULD include enough session description information
   to allow for a media session between the called UA and the calling
   UA.

   Although not strictly required for a one way voice path to be setup
   between the egress gateway and the ingress gateway, the SDP in the
   183 has the following benefits:

1. The list of audio (or video) codecs is reduced, so the calling
gateway need only expect a smaller set.

2. The 183 can contain security preconditions in the SDP (if they
were in the SDP in the INVITE), so that the calling gateway can per-
form appropriate authentication/encryption for each media stream from
each egress gateway.

3. If any kind of pre-call announcement requires two-way media (per-
haps some kind of speech recognition for credit card numbers, or even
DTMF too), the SDP in the 183 is needed.

5.8 SIP Message Body

5.8.1 Body Inclusion

The following is proposed rewording of paragraph 2 in Section 8.1 of
the SIP specification:

For response messages, the request method and the response status
code determine the type and interpretation of any message body. All
responses MAY include a body. Message bodies for 1xx responses con-
tain advisory information about the progress of the request. In
addition, message bodies for 1xx responses can contain session
descriptions. 2xx responses ...

5.10 Behavior of SIP Clients and Servers

5.10.1.2 Responses

The following is proposed text for inclusion in section 10.1.2 of the
SIP specification:

183 responses SHALL always be forwarded.

5.11 Behavior of SIP User Agents

5.11.6. Callee Needs Early Media

When the called UA receives and INVITE message that results in the
need to report on the status of the media setup through a media
stream, the called UA has the option to send a 183 message with a
session description to the calling UA.

5.11.7 Caller Receives 183 Response

When the calling UA receives a 183 response that contains a session

description and an indication that the session description is for
early media, it SHALL setup the associated media session and present
any media received from the called UA to the user.

5.13 Security Considerations

The security considerations for the 183 Session Progress message are
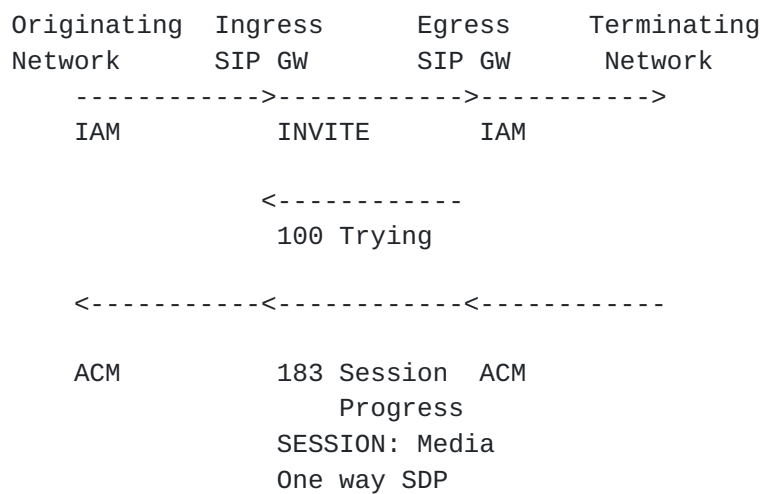the same as for SIP in general.

5.16 Examples

5.16.9 PSTN to PSTN Session Setup (SIP in the middle)

The following call flow illustrates the case where a call is origi-
nating from a PSTN network, transiting a SIP network and being deliv-
ered to a second PSTN network. In this case, the 183 message is used
to trigger the ACM message and results in a one way media session
being setup through the SIP network.

```
         Originating  Ingress       Egress     Terminating
         Network      SIP GW        SIP GW       Network
         ------------>------------->----------->
           IAM          INVITE        IAM


                      <------------
                        100 Trying


         <-----------<-------------<------------

           ACM          183 Session  ACM
                             Progress
                        SESSION: Media
                        One way SDP
```

```
            <================================
                    One way voice path


            <-----------<------------<------------
            ANM          200 OK        ANM


                         ------------>
                         ACK


            <================================>
                    Two Way Voice Path



            <-----------<------------<------------
            REL          BYE           REL


                         ------------>
                         RLC
                  ------------>
                  200 OK
            ------------>
            RLC
```

[5.16.10](5.16.10) SIP User Agent Session Setup to a PSTN Destination

```
      Calling        Egress     Terminating
        UA           SIP GW       Network
        ------------>------------>
          INVITE        IAM


        <-----------
        100 Trying


        <-----------<------------
        183 Session   ACM
            Progress
        Session: Media
        One way SDP


        <=======================
            One way voice path
```

```
            <-----------<------------
            200 OK        ANM

            ------------>
            ACK

            <=======================
                Two Way Voice Path

            <-----------<------------
            BYE           REL

                        ------------>
                        RLC
            ------------>
            200 OK
```

5.A Minimal Implementation

5.A.1 Client

   The following is a suggested addition to Appendix A.1 of the SIP
   specification:

   PSTN Interworking: If a client wishes to interwork properly with PSTN
   networks then it MUST support the 183 Session Progress message.

6 Further Examples

   Only the relevant headers have been included in the following exam-
   ples. Notably, the mandatory parameters Call-ID and CSeq are not

   shown.

6.1. Remote Ringtone, Followed by "Queued" Announcement

      Client to server:

            INVITE sip:+12145551212@bell.com SIP/2.0
            RAck: 0
            To: sip:+12145551212@bell.com
            From: sip:+15125559876@domain.com
            Require: org.ietf.sip.reliable-100
            Proxy-Require: org.ietf.sip.reliable-100
            Content-Type: application/sdp

```
            v=0
            o=- 929142225 929142225 IN IP4 vgw.domain.com
            c=IN IP4 vgw.domain.com
            M=audio 49152 RTP/AVP 0 1
            a=rtpmap:0 PCMU/8000
            a=rtpmap:1 1016/8000


        Server to client:

            SIP/2.0 180 Ringing
            RSeq: 1
            To: sip:+12145551212@bell.com
            From: sip:+15125559876@domain.com
            Session: Media
            Content-Type: application/sdp

            v=0
            o=- 929142942 929142942 IN IP4 media.bell.com
            c=IN IP4 media.bell.com
            M=audio 49180 RTP/AVP 0
            a=rtpmap:0 PCMU/8000


        Client to server:

            INVITE sip:+12145551212@bell.com SIP/2.0
            RAck: 1
            To: sip:+12145551212@bell.com
            From: sip:+15125559876@domain.com
            Require: org.ietf.sip.reliable-100
            Proxy-Require: org.ietf.sip.reliable-100



        [Remote ringing tone is played]
```

```
        Server to client:

            SIP/2.0 182 Call is queued; est. wait is 5 minutes
            RSeq: 2
            To: sip:+12145551212@bell.com
            From: sip:+15125559876@domain.com
            Session: Media
            Content-Type: application/sdp
```

```
        v=0
        o=- 929143057 929143057 IN IP4 media.bell.com
        c=IN IP4 media.bell.com
        M=audio 49180 RTP/AVP 1
        a=rtpmap:1 1016/8000

   [Ring tone is discontinued]



   Client to server:

        INVITE sip:+12145551212@bell.com SIP/2.0
        RAck: 2
        To: sip:+12145551212@bell.com
        From: sip:+15125559876@domain.com
        Require: org.ietf.sip.reliable-100
        Proxy-Require: org.ietf.sip.reliable-100

   ["Your call is queued" announcement is played, followed by hold
   music]



   Server to client:

        SIP/2.0 200 OK
        To: sip:+12145551212@bell.com
        From: sip:+15125559876@domain.com
        Content-Type: application/sdp

        v=0
        o=- 929143373 929143373 IN IP4 vgw.bell.com
        c=IN IP4 mg.bell.com
        M=audio 49199 RTP/AVP 1
        a=rtpmap:1 1016/8000

   [Hold music is discontinued]
```

```
   Client to server:

        ACK sip:+12145551212@bell.com SIP/2.0
        To: sip:+12145551212@bell.com
        From: sip:+15125559876@domain.com
```

[Final media stream is established]

6.2. Remote Announcement: "Call is being forwarded," local ring tone.

    Client to server:

        INVITE sip:+12145551212@bell.com SIP/2.0
        RAck: 0
        To: sip:+12145551212@bell.com
        From: sip:+15125559876@domain.com
        Require: org.ietf.sip.reliable-100
        Proxy-Require: org.ietf.sip.reliable-100
        Content-Type: application/sdp

        v=0
        o=- 929142225 929142225 IN IP4 vgw.domain.com
        c=IN IP4 vgw.domain.com
        M=audio 49152 RTP/AVP 0 1
        a=rtpmap:0 PCMU/8000
        a=rtpmap:1 1016/8000

    Server to client:

        SIP/2.0 180 Call is being forwarded
        RSeq: 1
        To: sip:+12145551212@bell.com
        From: sip:+15125559876@domain.com
        Session: Media
        Content-Type: application/sdp

        v=0
        o=- 929142942 929142942 IN IP4 media.bell.com
        c=IN IP4 media.bell.com
        M=audio 49180 RTP/AVP 0
        a=rtpmap:0 PCMU/8000

    Client to server:

```
     INVITE sip:+12145551212@bell.com SIP/2.0
     RAck: 1
     To: sip:+12145551212@bell.com
     From: sip:+15125559876@domain.com
     Require: org.ietf.sip.reliable-100
     Proxy-Require: org.ietf.sip.reliable-100
```

[Announcement plays: "Your call is being forwarded to a phone
outside the company's premises. Please wait."]

     Server to client:

```
     SIP/2.0 180 Ringing
     RSeq: 2
     To: sip:+12145551212@bell.com
     From: sip:+15125559876@domain.com
     Session: Media
     Content-Type: application/sdp

     v=0
     o=- 929143373 929143373 IN IP4 media.bell.com
     c=IN IP4 media.bell.com
     M=audio 0 RTP/AVP 0
```

[Media stream is discontinued. Local ring-tone is generated by
the client towards the PSTN user.]

     Client to server:

```
     INVITE sip:+12145551212@bell.com SIP/2.0
     RAck: 2
     To: sip:+12145551212@bell.com
     From: sip:+15125559876@domain.com
     Require: org.ietf.sip.reliable-100
     Proxy-Require: org.ietf.sip.reliable-100
```

    Server to client:

        SIP/2.0 200 OK
        To: sip:+12145551212@bell.com
        From: sip:+15125559876@domain.com
        Content-Type: application/sdp

        v=0
        o=- 929143373 929143373 IN IP4 vgw.bell.com
        c=IN IP4 mg.bell.com
        M=audio 49199 RTP/AVP 1
        a=rtpmap:1 1016/8000

    Client to server:

        ACK sip:+12145551212@bell.com SIP/2.0
        To: sip:+12145551212@bell.com
        From: sip:+15125559876@domain.com

    [Final media stream is established]

7 References

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