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SIP 183 Session Progress Message

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Abstract

This document describes a proposed extension of the Session Initiation Protocol. This extension would add the 183 Session Progress response 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 indicate that the calling user agent should not provide local alerting and should 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.

1 Introduction

There are instances, most notably dealing with SIP to PSTN interworking, 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 preliminary 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 would be 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 Protocol (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 treatment (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 terminating 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 signal 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 network without the interim SIP network incurring charges from the originating 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     |
                         +---------+              +-------------+
```

3.0 Options With Existing SIP Specification

The following sections show the results of investigating various options for addressing the above requirements using existing SIP protocol capabilities. In each case, it is shown why the option either cannot address the requirements or has short comings that can be

better addressed using the 183 Session Progress message.

3.1 100 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 sitting in the signaling path between the ingress gateway and the egress gateway might have generated the 100 Trying message, causing the ACM message to be sent prior to the egress gateway receiving an ACM from the terminating network.

3.2 180 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.

3.3 180 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 scenario) 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.

3.4 200 OK Mapped to ACM

The final option investigated involves setting up a full media session in the SIP network prior to receiving the ANM from the terminating PSTN network. This involves mapping the 200 OK to the ACM message 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
```


4 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.

4.1 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
```

4.2 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
```

4.3 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 network.

```
        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

            ------------>------------>

            200 OK       RLC
```

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 specification.

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

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 indicate the progress of the session setup.

5.7.1.5 183 Session Progress

The called UA has the need to communicate the status of the session setup attempt as part of a media stream. The calling UA shall establish a media session according 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 perform appropriate authentication/encryption for each media stream from each egress gateway.

3. If any kind of pre-call announcement requires two-way media (perhaps 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 contain 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 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 originating from a PSTN network, transiting a SIP network and being delivered 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
                            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 SIP User Agent Session Setup to a PSTN Destination

```
        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

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

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 works then it MUST support the 183 Session Progress message.

6 References

[1] M. Handley, H. Schulzrinne, E. Schooler, and J. Rosenberg, SIP: Session Initiation Protocol", RFC 2543, March 1999.

[2] J. Rosenberg, H. Schulzrinne, S. Donovan, "Establishing QoS and Security Preconditions for SDP Sessions", draft-rosenberg-mmusic-sipqos-00.txt, To be published, Work in Progress.

[3] J. Rosenberg, H. Schulzrinne, "Reliability of Provisional Responses in SIP", draft-ietf-mmusic-sip-100rel-01.txt, May 21, 1999, Work in Progress.

[4] H. Schulzrinne, "RTP Profile for Audio and Video Conferences with Minimal Control ", RFC 1890, January, 1996.

6 Authors' Addresses

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