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SIP 183 Session Progress Message

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Abstract

This document describes a proposed extension to SIP [1]. This extension adds the 183 Session Progress response and a new header to indicate 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 indicate that the calling user agent should not provide local alerting. In addition, 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.

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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 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.

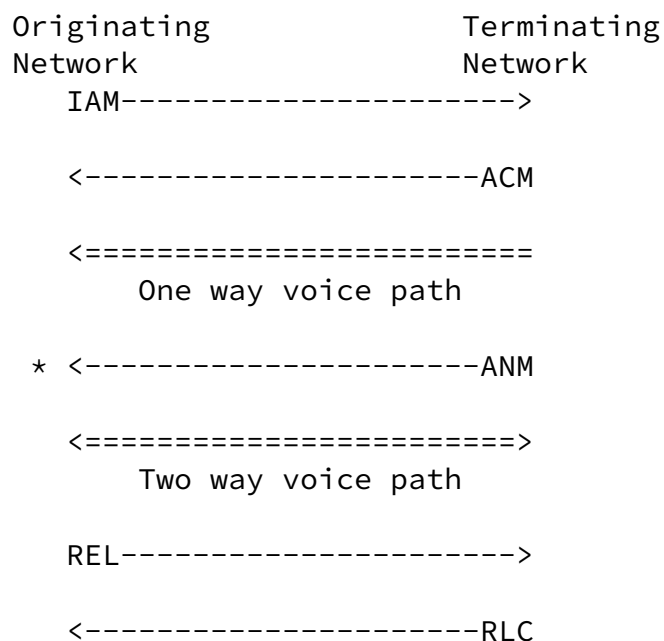
This document proposes an extension to the Session Initiation Protocol (SIP) that introduces this capability.

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.

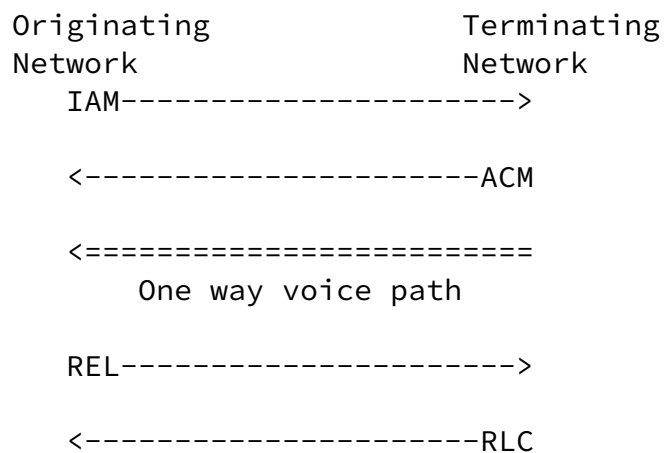
The following call flow illustrates call setup using SS7 ISUP in a PSTN network.



- * 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.



2.1 PSTN to SIP Network Interworking Requirements

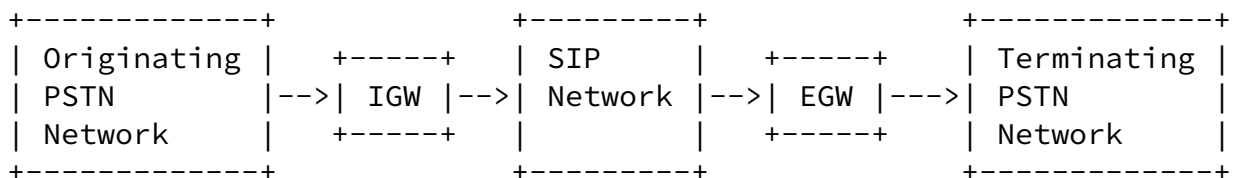
The following are a subset of the requirements for interworking between a PSTN network and a SIP network.

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

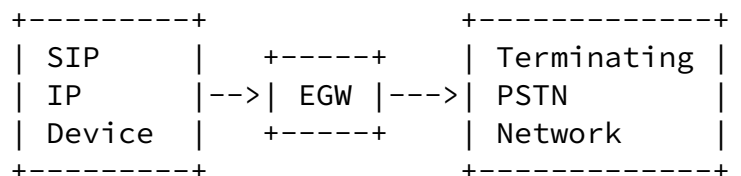
- The ingress gateway into the SIP network shall have the ability to determine, based on SIP signaling messages, when to send an ISUP ACM message and when to send an ISUP ANM message.
- The SIP network shall have the ability to support fast setup. This occurs when the terminating network does not send 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:



IGW = Ingress Gateway
EGW = Egress Gateway



2.2 Solution Options With Existing SIP Specification

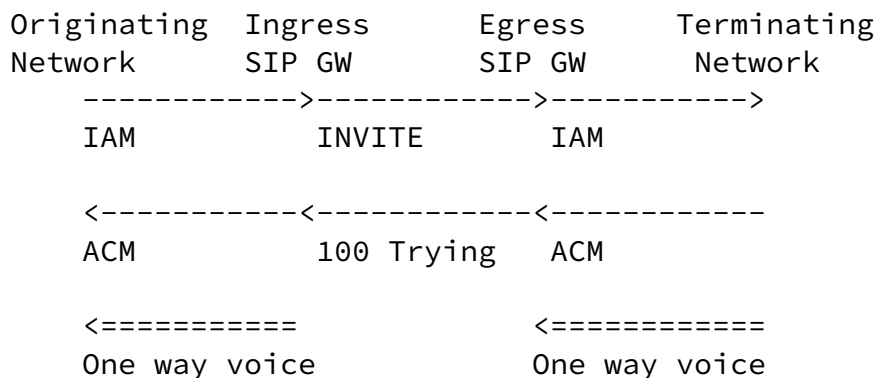
The following sections show the results of investigating various options for addressing the above requirements using existing SIP protocol capabilities. In each case, it is shown why the option either cannot address the requirements or has short comings that can be bet-

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.



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

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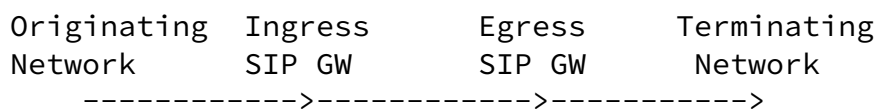
addressed by allowing the 100 Trying to carry SDP information to be used for carrying the preliminary session media. This option is explored in the context of the 180 Ringing message in [section 3.3](#).

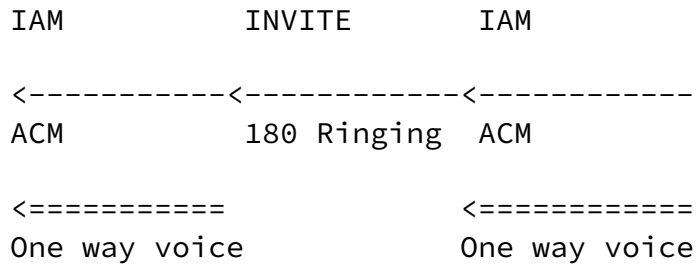
The use of the 100 Trying also fails because a SIP Proxy Server sitting in the signaling path between the ingress gateway and the egress gateway might have generated the 100 Trying message. This would result in the ACM message being sent prior to the egress gateway receiving an ACM from the terminating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



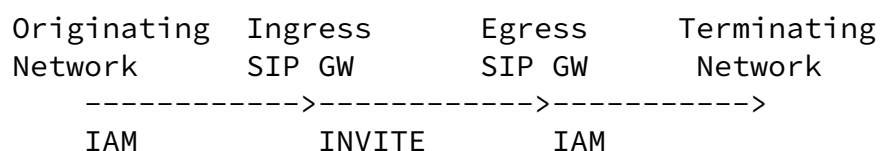


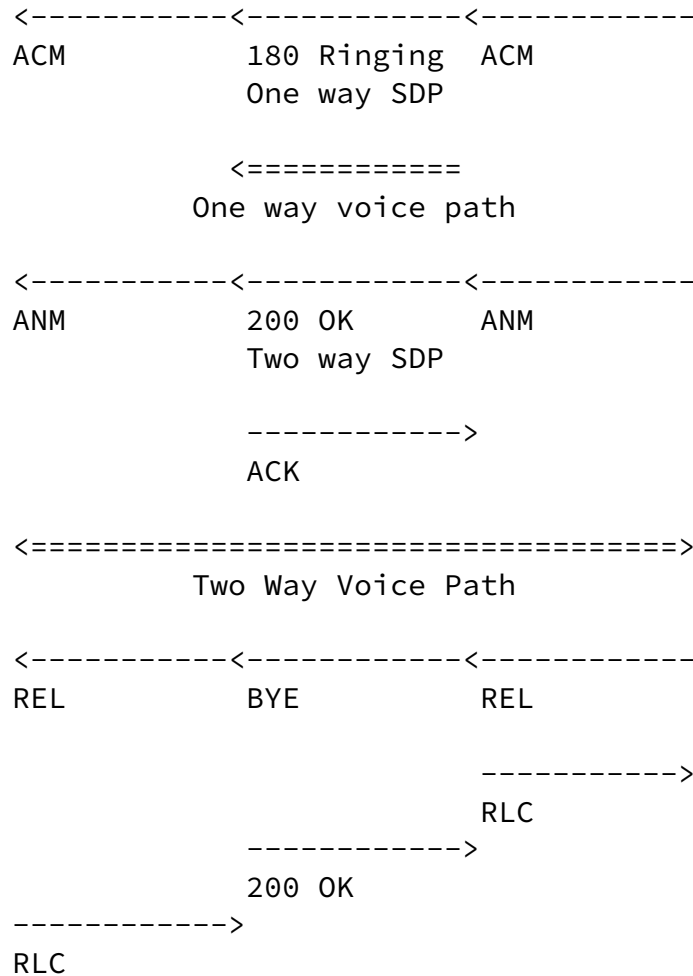
This option breaks at this point because a media voice path cannot be cut through at this point for the terminating PSTN network to report on the session progress. This is due to the fact that the egress gateway has not yet communicated its RTP information to the ingress gateway. The next two options 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:



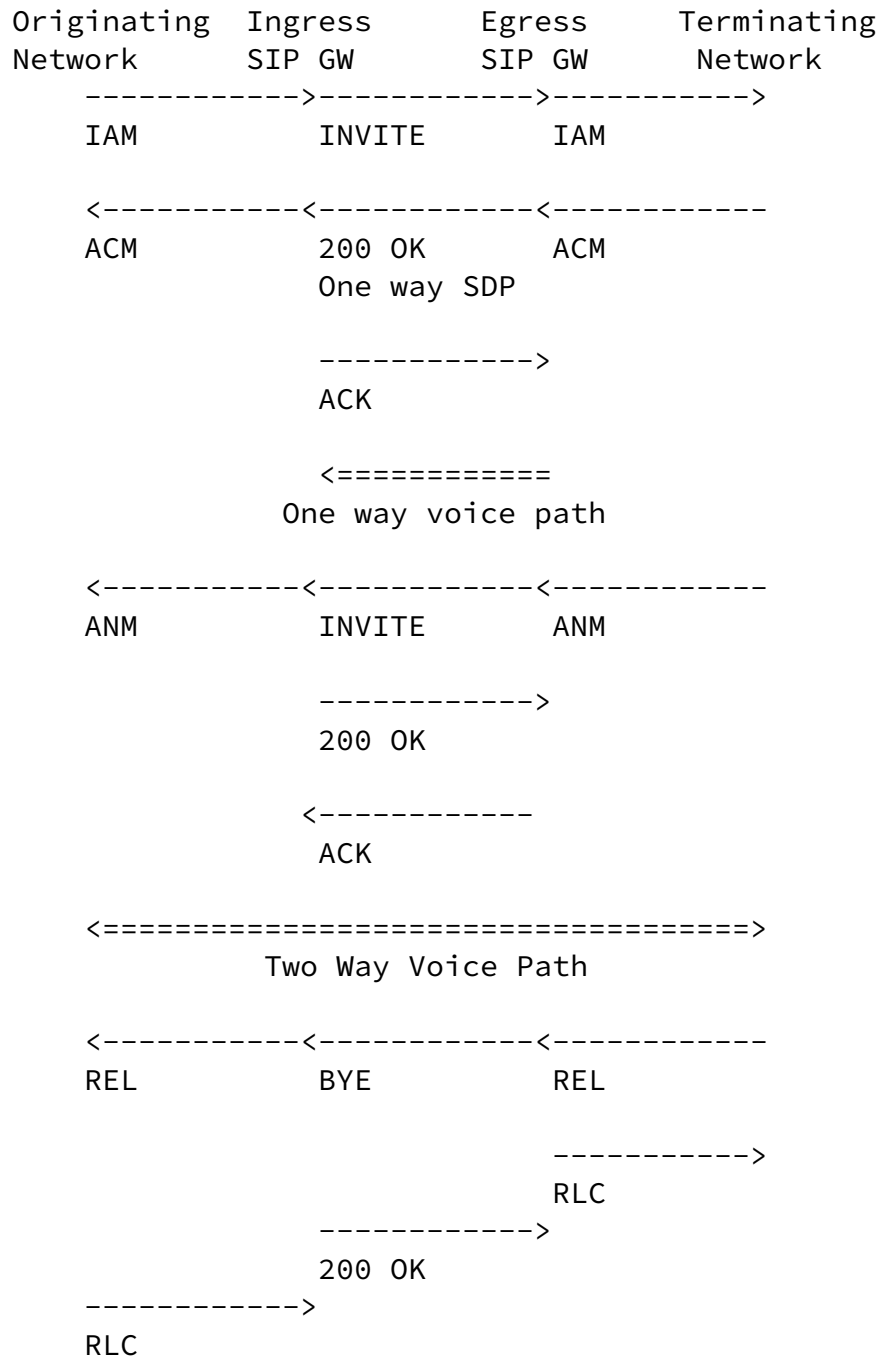


Although this option looks promising on first review, it does not give the called user agent the ability to include SDP in the message and rely on the calling user agent (the ingress gateway in this scenario) to provide local alerting. As illustrated in [2] there are other reasons that SDP might be included 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.

[2.2.4](#) 200 OK Mapped to ACM

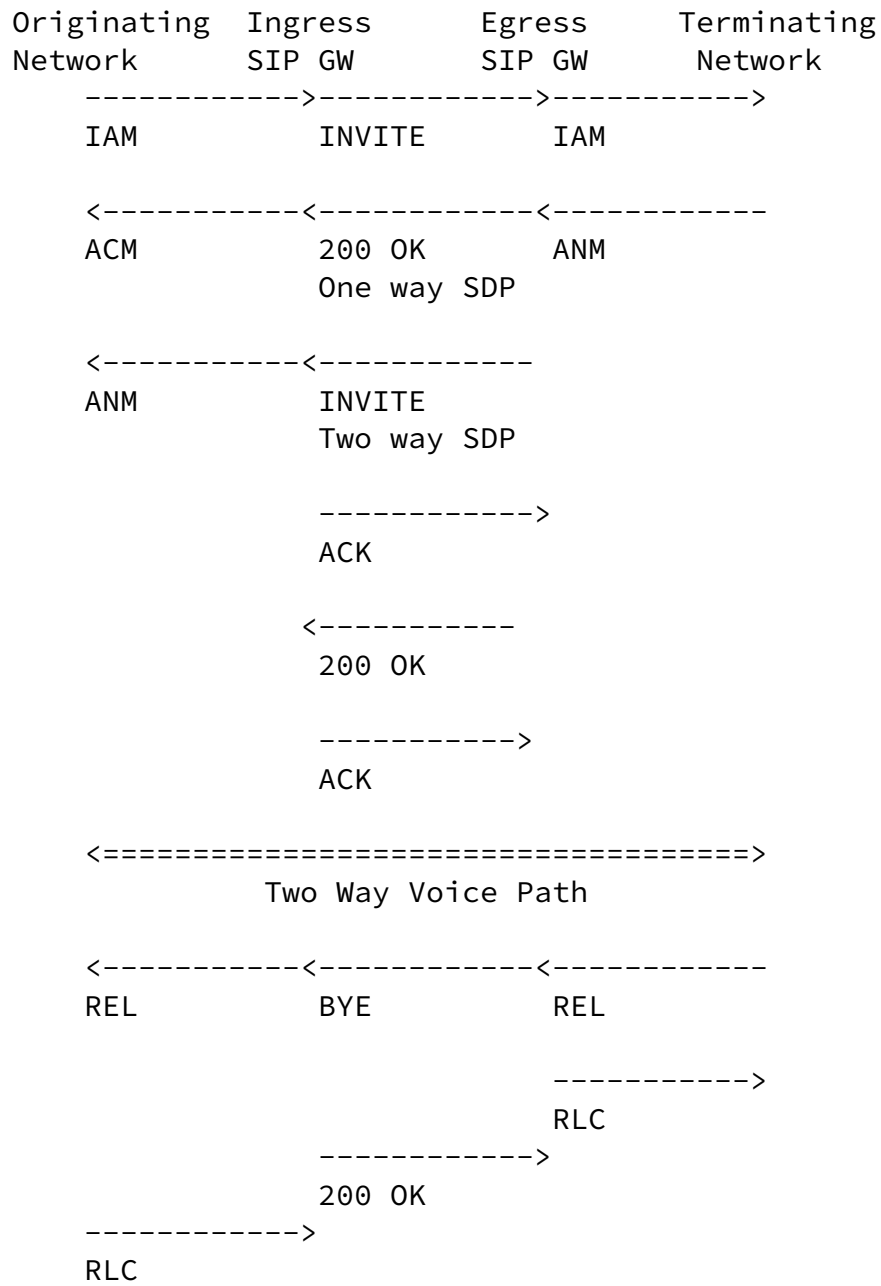
The final option investigated involves setting up a full media session in the SIP network prior to receiving the ANM from the terminating PSTN network. This involves mapping the 200 OK to the ACM message at the ingress gateway and having the egress gateway send a re-INVITE upon receipt of the ANM. The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

This option is illustrated in the following call flow:



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.

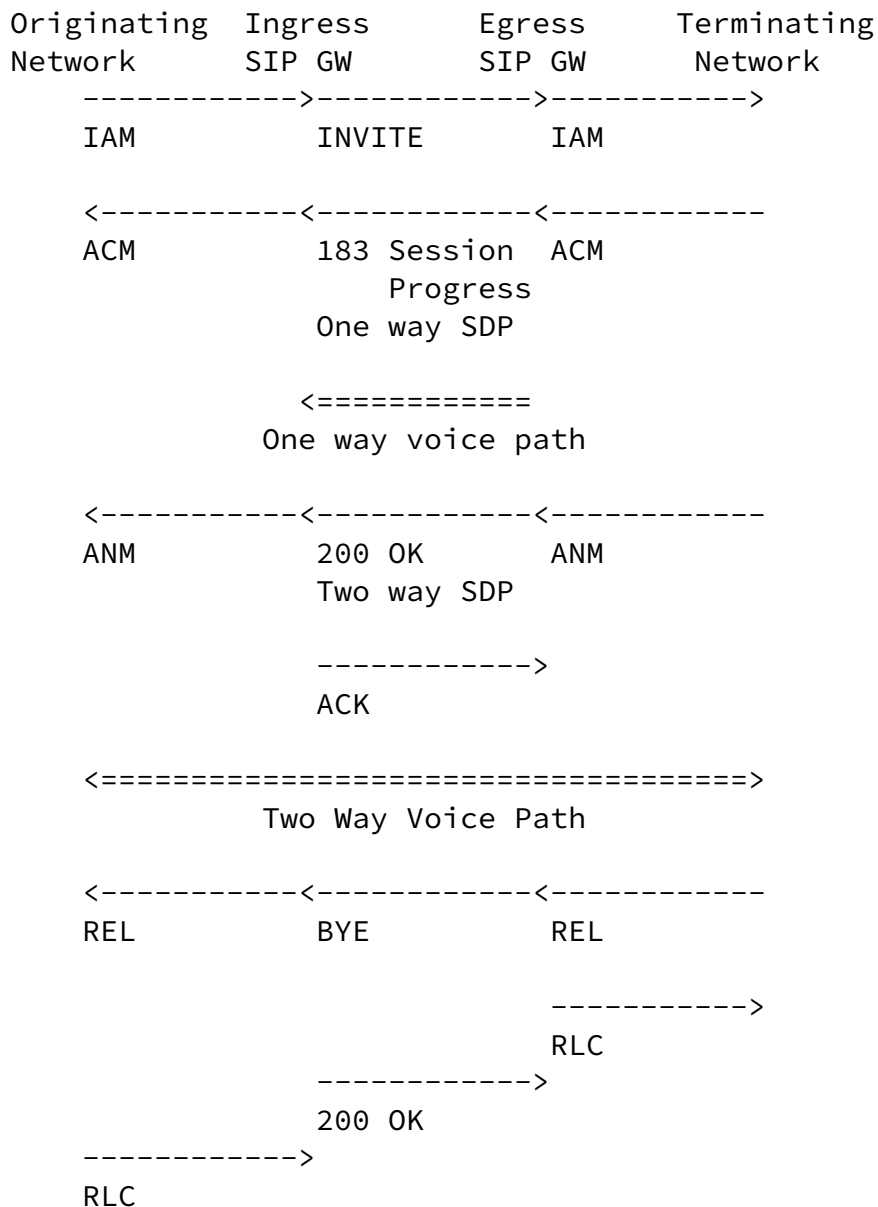


[2.3](#) Proposed 183 Session Progress

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.



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

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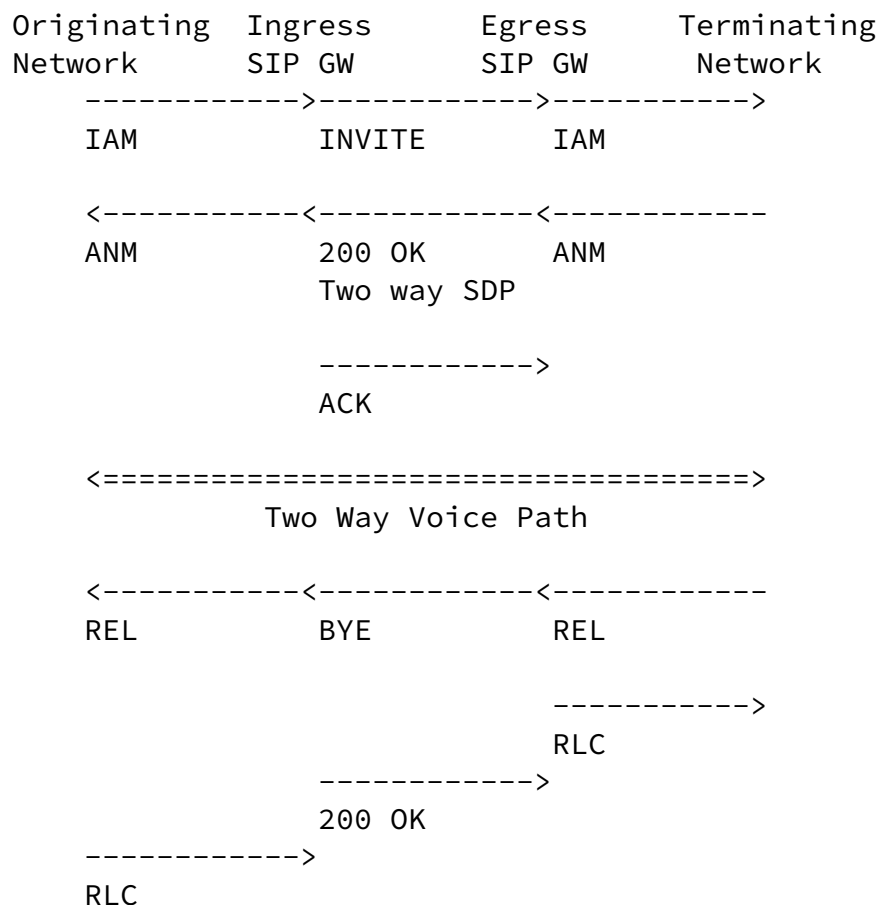
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OK. This meets the requirement that the SIP gateways must be able to differentiate between ACM and ANM messages.



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 network.

Calling UA	Egress SIP GW	Terminating 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
```

[3](#) 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

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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 message. 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 establishing of an early media session. The early media session will generally 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 establishing 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 temporary media will contain SDP that is a subset of the media description 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 message confirming receipt of the provisional response has been sent (from the client's perspective) or received (from the server's perspective).

The designation of media capabilities in a provisional response has no implications on the capabilities of any subsequent temporary connections 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 correspond 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 representation 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

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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 Acceptable" message; the alternative is to forgo the transmission of provisional media. While it is perhaps a more appropriate error code, "606 Not Acceptable" is not suggested, owing to its properties of terminating any ongoing searches.

If the client finds the session description proposed by the server in a provisional response unacceptable, its acknowledgement SHOULD contain 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 protocol 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 specification.

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:

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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 establishing of an early media session. The early media session will generally 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 establishing 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

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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 without 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 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 an 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.

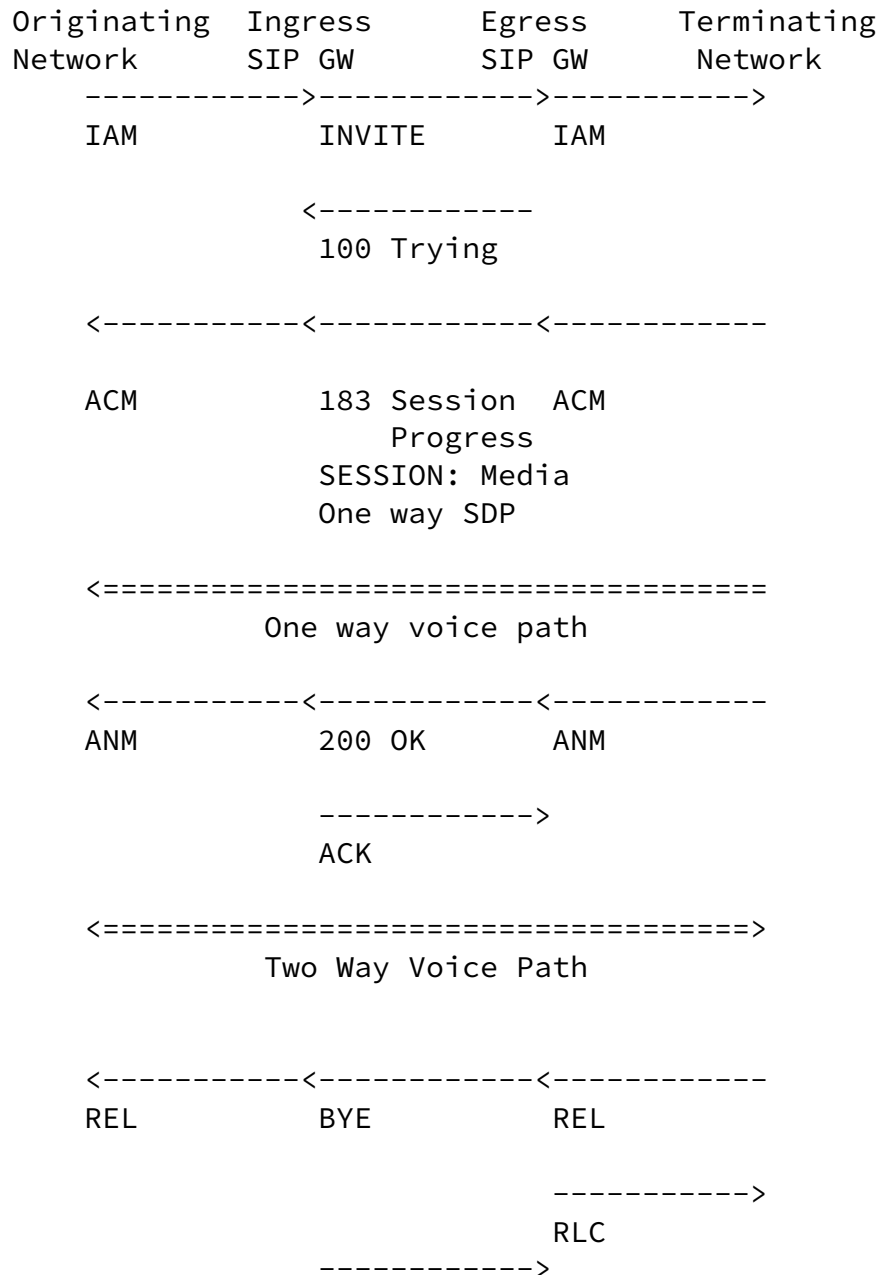
[5.13](#) Security Considerations

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 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.



```

                200 OK
----->
RLC

```

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

Calling UA	Egress SIP GW	Terminating 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 examples. Notably, the mandatory parameters Call-ID and CSeq are not

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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]

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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]

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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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