multiservice switching forum

IA for a SIP Profile between a UA and CA

MSF-IA-SIP.003-FINAL

Multiservice Switching Forum Implementation Agreement

Contribution Number: MSF2004.042
Document Filename: MSF-IA-SIP.003-FINAL
Working Group: Protocol & Control
Title: Implementation Agreement for a SIP Profile between a UA and CA
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Date: 28th September 2004

Abstract: This document describes a SIP profile between a UA and CA.

Keywords: SIP, GMI2004

The goal of the MSF is to promote multi-vendor interoperability as part of a drive to accelerate the deployment of next generation networks. To this end the MSF looks to adopt pragmatic solutions in order to maximize the chances for early deployment in real world networks.

To date the MSF has defined a number of detailed Implementation Agreements and detailed Test Plans for the signaling protocols between network components and is developing additional Implementation Agreements and Test Plans addressing some of the other technical issues such as QoS and Security to assist vendors and operators in deploying interoperable solutions.

In 2002, the MSF held a "Global MSF Interoperability 2002" (GMI 2002) event that tested interoperability between next generation network elements situated in Asia, Europe and North America. GMI 2002 validated the MSF release 1 architectural framework and Implementation Agreements by subjecting them to interoperability testing based on realistic network scenarios.

Following the success of GMI 2002 the MSF work program continues to address the key technical barriers to next generation network deployments. Global MSF Interoperability 2004 (GMI 2004) will demonstrate a deployable and operationally ready IP telephony network with Network Management, enhanced Quality-of-Service (QoS) and security features. GMI2004 will also demonstrate a service layer with application server, media server, and service broker functionality. This will enable the MSF to demonstrate a full end-to-end customer ready deployable network.

It is envisaged that GMI2004 will provide an industry showcase that will:

- Assist carriers achieve their goal: to deploy flexible, best of breed products.
- Assist vendors achieve their goal: to market products more cost effectively.
- Display the global interoperability of the MSF architecture as referenced in the Release 2 architecture document.
- Demonstrate a network scenario that can be managed to specific quality standards.

The MSF welcomes feedback and comment and would encourage interested parties to get involved in this work program. Information about the MSF and membership options can be found on the MSF website http://www.msforum.org/

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1 Multiservice Switching Forum

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2 Applicability and Scope

This Implementation Agreement covers the interface between a SIP User Agent & its Call Agent in preparation for GMI 2004.

Figure 1 shows the MSF architecture diagram for GMI 2004 and highlights the applicable interfaces for this IA.

The overall end to end Network Architecture for GMI 2004 is shown in figure 1 below.

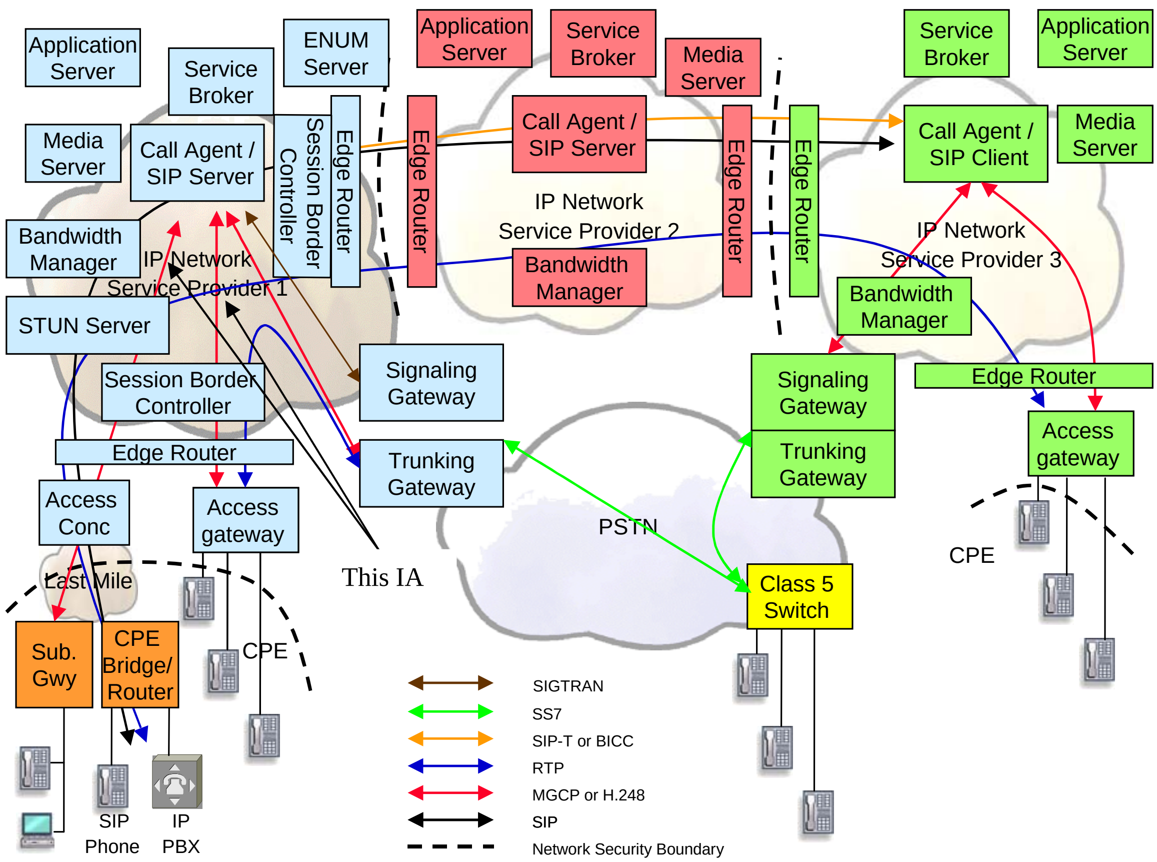

Figure 1: Architecture diagram indicating applicable interfaces for this IA in Service Provider 2

This profile is used to define SIP signalling between User Agents and Call Agents and ought to be read in conjunction with the MSF Core SIP Profile ([13]).

3 SIP Profile

3.1 Identifications

This profile shall be entitled “MSF SIP User Agent Profile”. The version number shall be 1.

3.2 Base specifications

A conforming entity shall implement the following specifications:

- IETF RFC 3261, "SIP: Session Initiation Protocol"
- IETF RFC 3262, "Reliability of Provisional Responses in SIP"
- IETF RFC 3311, “The SIP Update Method”
- IETF RFC 3264, “An Offer / Answer Model with SDP”
- IETF RFC 3265, “SIP Specific Event Notification”
- IETF RFC 3323, “A Privacy Mechanism for SIP”
- IETF RFC 3515, “The SIP REFER Method”
- IETF draft-ietf-sip-referredby-05.txt, “The SIP Referred-By Mechanism”

3.3 Optional specifications

A conforming entity shall optionally implement the following specifications:

- IETF draft draft-ietf-sip-session-timers-13.txt, “Session Timers in SIP”
- IETF RFC 3326, “The Reason Header field for SIP”
- IETF RFC 3312, “Integration of Resource Management & SIP”
- IETF draft-ietf-sipping-app-interaction-framework-02, “A Framework for Application Interaction in SIP”
- IETF draft-ietf-sipping-kpml-04, “ A SIP Event Package for Key Press Stimulus”
- IETF draft-ietf-sipping-mwi-04, “A Message Summary & Message Waiting Event Package for SIP”
- IETF draft-ietf-sip-resource-priority-03.txt “Resource Priority for SIP”

3.4 Configuration

A SIP User Agent shall be configured with the identity/address of its (Registrar) Call Agent

A CA/Registrar shall be configured with the identities of its dependent User Agents.

3.5 Transport

RFC 3261 [1] allow SIP requests to be sent using reliable or unreliable protocols: UDP, TCP, or SCTP.

The SIP Signaling Security IA ([14]) defines support of TLS to be mandatory for a Call Agent and optional for a User Agent. TLS requires support for TCP and optionally SCTP as a transport protocol.

Thus, for this profile, support of UDP shall be mandatory (for no TLS support) and support for TCP/SCTP shall be conditionally mandatory where TLS is mutually supported

3.6 RTP stream and SDP description

A conforming entity shall support unicast for RTP streams. Multicast may be supported.

SDP shall be exchanged according to the rules of RFC 3264 [7].

3.7 URL

A conforming entity shall support tel-URLs and SIP-URLs. SIPS URLs MAY also be supported.

Passwords in the userinfo field are not recommended and are to be ignored if present.

3.8 SIP Protocol

(1) Supported methods

A conforming entity shall support all the methods defined in the base specifications (section 3.2), and may support ones in the optional specifications (section 3.3).

Currently, INVITE, ACK, BYE, CANCEL, PRACK, REFER, SUBSCRIBE, NOTIFY, UPDATE & REGISTER are mandatory for a conforming entity.

(2) Message body in 1xx and 300 and greater responses

A conforming CA should support text/plain type and may support text/html.

(3) Compact Form

A conforming CA shall use only non-compact forms.

3.8.1 SIP Request Headers

All SIP requests shall support the following headers :- REQ-URI, TO, FROM, CALL-ID, CSEQ & CONTENT-LENGTH.

In addition, the following table describes the salient headers which are supported in the various requests:-

Method Name	**Mandatory Headers**	**Conditional Mandatory Headers**	**Optional Headers**
INVITE	Contact, Supported, Require, Allow (note),	Content-Type	Route, Session-Expires, Min-SE, Max-Forwards Resource Priority
ACK			Content-Type, Contact, Route, Allow, Max-Forwards Resource Priority
PRACK	Rack		Route, Content-Type, Contact, Supported, Resource Priority
CANCEL			Content-Type, Supported, Route, Max-Forwards

BYE			Content-Type, Supported, Route, Max-Forwards
REFER	Refer-To, Referred-By		Contact, Supported, Route, Max-Forwards
NOTIFY	Content-Type, Subscription-State, Event		Contact, Supported, Route, Max-Forwards
UPDATE	Content-Type		Route, Max-Forwards, Session-Expires Resource Priority
SUBSCRIBE	Contact, Expires, Event		Max-Frowards
REGISTER	Contact, Expires, Authorization		Max-Forwards

Note – Used to advertise support of certain Methods – e.g. UPDATE.

Conditional Mandatory headers are dependent on whether the message body is empty or not.

3.8.2 SIP Response Headers

All SIP 1XX responses support the following headers :- TO, FROM, VIA, CALL-ID, CSEQ & CONTENT-LENGTH. The following table details which additional headers are carried in specific 1XX responses :-

Header	**100**	**18X**
Content Type	N/A	O
Contact	O	M
Supported	O	O
RSeq	N/A	C
Require	O	O

Session Expires	O	O
Record-Route	N/A	O
Allow	O	O

Note :- M – Mandatory, O – Optional, C – Conditional (i.e. if SDP present, then M), N/A – Not applicable.

All 2XX responses other than 200 response are treated as for a 200 response for the corresponding transaction.

The 200 OK response always contains the following :- TO, FROM, VIA, CALL-ID, CSEQ, CONTACT & CONTENT-LENGTH. In addition, the following headers may be present :- CONTENT-TYPE (if SDP present), REQUIRE, SUPPORTED, RECORD-ROUTE, SESSION-EXPIRES, & ALLOW (to enable advertising of supported Methods).

Any SIP failure reason (3XX, 4XX, 5XX, 6XX) are allowed in the profile.

3XX responses will contain the following headers :- TO, FROM, VIA, CALL_ID, CSEQ, CONTENT-LENGTH & CONTACT. An entity MAY redirect the call on receipt of a 3XX response – else it is treated as a 4XX response and the call/session is released.

4XX/5XX/6XX responses will contain the following headers :- TO, FROM, VIA, CALL_ID, CSEQ, & CONTENT-LENGTH.

In general, the headers identified in this section are intended to be a minimum set and other headers may be present.

3.9 SDP

SDP shall be supported as defined in RFC 2327 [6] and RFC 3264 [7].

This profile shall support mid-call bearer change. It is recommended that this is performed via the UPDATE method.

3.9.1 SDP Usage

SDP usage shall conform to [4].

3.9.2 Basic Call Bearer Connection

For the initial / basic call bearer connection, a single block of SDP shall be exchanged as described in [7]. The typical sequence shall be:-

- An INVITE containing an originating side SDP Offer,
- A 18X message (typically 180 Alerting) containing the terminating side SDP Answer (which would result in a PRACK plus 200 OK due to use of 100rel),
- A 200 OK message containing the (repeated) terminating side SDP Answer.

Note that [7] mandates that the same SDP Time (t=) Line must appear in both blocks. In addition, there must be identical numbers of Media (m=) Lines in each SDP block.

3.9.3 Mid Call Bearer Change

Mid call bearer change is realized according to [7]. It is recommended that UPDATE be used ([12]) to convey the SDP change although re-INVITE shall also be supported.

[7] mandates that the same number of Media (m=) lines must be present. This raises an issue where a media stream is to be disconnected and thus must be disabled. There are a number of ways in which to disable a stream :-

- By setting the IP address to 0.0.0.0 (this is currently widely supported but is deprecated in [7]),
- By setting the ctream media attribute to inactive,
- By setting the port number to zero.

For this profile, to facilitate inter-operability, it is recommended that all three mechanisms are used to disable a media stream, e.g. compare the following SDP blocks :-

<u>Active</u>

v=0
s=-
t=0 0
o=- 0 0 IN IP4 128.96.63.25
c=IN IP4 128.96.63.25
m=audio 3456 RTP/AVP 0, 100
a=recvonly
a=rtpmap:100 G729D
a=ptime:20

<u>Inactive</u>
v=0
s=-
t=0 0
o=- 0 0 IN IP4 128.96.63.25
c=IN IP4 **0.0.0.0**
m=audio **0** RTP/AVP 0, 100
a=inactive
a=rtpmap:100 G729D
a=ptime:20

A new stream (i.e. bearer redirection) may be enabled via exchanging a new address and port in the Connection and Media Lines respectively – i.e. the contents of existing Media & Connection Lines can be altered as desired (e.g. to change address / port / media format / codec list).

3.10 Start of Day Processing

A Call Agent, on restart shall listen on its pre-configured port for receipt of signalling from its User Agents.

A User Agent, on restart shall listen on its pre-configured port for receipt of signalling from its Call Agent. Typically, a User Agent shall register with its Call Agent / Registrar.

A conformant entity shall run application level timers to terminate calls in the event of message loss / lack of response.

3.11 Reliable Provisional Responses

Due to the fact that SDP may be carried in provisional responses, support of RFC 3262 ([9]) is mandatory for this profile.

A confirming entity shall advertise its requirement of provisional reliable responses via a REQUIRE header containing the tag “100Rel”.

When sending a response in the range 101-199 (i.e. 100 is never sent reliably), a conformant entity shall include a REQUIRE header with tag “100rel”.

On receipt of a response in the range 101-199, a conformant entity shall check to see that a REQUIRE header is present with tag “100Rel”. If so, then a PRACK is generated.

See the appendix for a message flow example that uses 100rel.

3.12 SIP Session Timer

Support of session timer ([10]) is optional in this profile.

A confirming entity MAY advertise support of session timer via a SUPPORTED header containing the tag “timer”. In this case, the entity MAY also include a SESSION-EXPIRES header. A period of 3 minutes (180) is recommended. The rules governing which ends performs the refresh are as [10].

When the refresher timer expires, it is recommended for this profile that the UPDATE method ([12]) be used to perform the refresh.

4 Voice Codecs

This profile is applicable between a UA and a CA. The supported voice codecs are dependent on the capabilities of the SIP User Agent. This profile should not inhibit any media inter-working between the relevant endpoints. Codec negotiation is performed across this interface as described in [4].

5 Echo Control

Echo control is performed at the SIP User Agent.

6 Modem, Fax and TTY Support

Modem/Fax support is dependent on the capability of the SIP User Agent.

Such calls may start off as “voice” calls or else may be explicitly signalled as fax/modem at the start (e.g. advertisement of T38 in the SDP). In the former case, the User Agent would be responsible for detecting the fax/modem tone and this MAY result in a mid-call bearer change for the session in order to convey modified SDP blocks (e.g. additional media attribute lines such as *ecan* or *silencesupp*).

7 DTMF Digits and Telephony Tones

This is dependent on the capability of the user Agent.

For this profile, DTMF Digits may be transparently passed between the endpoints using RFC 2833 ([16]) or G711 or else signalled out of band via a SIP NOTIFY message (see [11] & [17]).

8 NAT & Firewall traversal

It is typical for there to be a NAT/FW traversal between an untrusted User Agent and a Call Agent.

When a NAT/FW is present, then suitable signalling and media PHs must be controlled and the SDP (conveyed in the SIP) appropriately modified. Such a FW device could be realized via a separate NAT/FW controlled by a discrete protocol such as the ETSI Gate Control Profile ([5]) or MSF Gate Control Profile ([19] or else via a SIP enabled ALG device. In all cases, the User Agent would be oblivious to the presence of the NAT/FW.

9 QOS Aspects

The overall solution for QoS is described in the MSF Quality of Service for next generation VoIP networks solution framework ([8]). This solution framework uses DiffServ to provide differential service to VoIP traffic and data traffic. VoIP traffic is marked using DSCP (Differentiated Services Code Points) in the IP header, it can then be given priority by routers in the IP network.

Where a NAT/FW is present between a User Agent and Call Agent, then the signalling/media packets (on entering the Operator's network) must be marked appropriately via DSCP. In general, Operators may wish to allocate the DSCP values within their network and therefore the DSCP actually used shall be configurable both for SIP signaling traffic and RTP/RTCP media.

In addition, this IA lists RFC3312 ([18]) as an optional specification. The intention of RFC3312 is to prevent the terminating user being alerted prior to ensuring that there are sufficient resources to handle the connection. RFC3312 shows the use of RSVP to reserve said resources. However, for this IA, there is no guarantee that the UA will support RFC3312 (in fact it's very likely that UAs will not support RFC3312). Moreover, if the UA does support RFC 3312, it is undesirable for RSVP signalling to be used to reserve said resources. Therefore, it is proposed that the CA shall set the RFC3312 parameters to inform the UA that QOS has been assured. If the UA doesn't support RFC 3312, then the additional media attributes are ignores. See the appendix for example flows. Note that there would typically be a SBC present between the CA and UA and thus the CA is able to perform its QOS checks/reservations based on knowledge of the SBC media address.

10 Security Aspects

Generic SIP security requirements are provided in [14].

SIP can be employed as an inter-domain or intra-domain signalling mechanism. In the former case, it may be subject to pre-existing trust relationships between different administrative domains. The level of security required is dependent on the level of trust between the respective entities. Examples of this are :-

- Authentication is based upon the known FQDN and potentially the IP address
- Use of HTTP digest to provide authentication between the CA and UA.
- Use of TLS and/or S/MIME.

For this IA, HTTP digest shall be mandatory whilst use of TLS and S/MIME shall be optional.

11 Redundancy & Resilience

The Call Agent must support redundancy. This is required to match the existing PSTN five-nine's level of reliability. In general, there are two ways to implement a redundant Call Agent :-

- Implement a primary and secondary Call Agent as separate components in the network
  - Primary and backup Call Agents have different domain names
- Implement a Call Agent with no single point of failure and full redundancy

To support the first scheme, User Agents should be provided with both Call Agent identities and should register with both. This is described in [13].

In the second scheme, the Call Agent has its own redundant components that share the same Domain name or IP address and the dependent User Agents will re-register to the new worker. The loss of the old worker would not be apparent to the User Agent (for UDP transport) but would require a new registration due to the loss of the TCP connection (for TCP transport).

User Agents are typically non redundant. Loss of a User Agent would become apparent to a Call Agent due to either loss of TCP connection (TCP transport) or application level timers expiring (UDP transport). Service would be resumed when the User Agent re-registers.

12 Management Information Model

12.1 SIP User Agent / Call Agent

The following items must be configurable on the User/Call Agent:-

- Fully Qualified Domain Name
- DHCP or fixed IP address (typically, CA would have a provisioned IP address and a User Agent would use DHCP)
- If DHCP is not used
  - IP address
  - Subnet mask
  - Default IP gateway
- UDP Port on which to receive SIP signaling (default 5060)
- DSCP (TOS) byte value for SIP signaling traffic (applicable to CA if FW present)
- DSCP (TOS) byte value for RTP voice traffic (applicable to CA if FW present)
- User name & password for HTTP digest authentication
- TLS Root Certificates (if TLS mutually supported)
- For the Call Agent
  - Domain Name of User Agent
- For the User Agent
  - Domain Name or IP address of Call Agent/Registrar
  - UDP Port to send SIP signalling to Call Agent/Registrar (default 5060)

13 References

[1]	IETF RFC 3261, "Session Initiation Protocol", June 2002.
[2]	Not Used
[3]	IETF RFC 3325, "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks" November 2002
[4]	MSF2003.059.03 Implementation Agreement for SDP Usage & Codec Negotiation for GMI2004, February 2004
[5]	TS 102 333 v1.1.2 ETSI Gate Control Profile, April 2004
[6]	IETF – RFC 2327 SDP: Session Description Protocol, April 1998
[7]	IETF RFC 3264 An Offer / Answer Model with SDP, June 2002
[8]	MSF2003.105.00 QOS for Next generation VoIP Networks Solution Framework, October 2003
[9]	IETF RFC 3262, "Reliability of Provisional Responses in SIP", June 2002
[10]	IETF draft draft-ietf-sip-session-timers-13.txt, Session Timers in SIP, January 2004.
[11]	IETF draft-ietf-sipping-app-interaction-framework-02, "A Framework for Application Interaction in SIP"
[12]	IETF RFC 3311, "The SIP UPDATE Method"
[13]	MSF2004.035.00 "IA for Core SIP IA for VoIP"
[14]	MSF2003.016.03 "SIP Signalling Security for GMI2004"
[15]	Not Used
[16]	IETF RFC 2833, "RTP Payload for Telephony Tones, DTMF Digits & Telephony Signals"
[17]	IETF draft-ietf-sipping-kpml-04 "A SIP Event Package for Key Press Stimulus"
[18]	IETF RFC 3312, "Resource Management in SIP"
[19]	MSF2004.032.01 "MSF Gate Control Profile using H248"

Appendix A SIP Call Flows

This appendix contains some example SIP call flows for :-

- Registration
- Basic Call Set-up and Tear-Down
- Early Audio (remote busy tone)
- Premature Call Termination (no answer)
- Call to busy subscriber
- Session Timer
- Unattended Transfer (use of REFER)
- Call Back When Free (Subscribe/Notify)
- RFC 3312 usage (A-side UA)
- RFC 3312 usage (B-side UA)

For further call flow examples, see :-

- IETF RFC 3665, "SIP Basic Call Flow Examples"
- IETF draft - draft-ietf-sip-service-examples-05.txt, " SIP Service Examples"

A.1 Registration

A User Agent (172.30.5.12) attempts to register with its Call Agent / Registrar (172.30.5.1). The initial registration request is rejected with a 401 response. The User Agent then repeats the registration with an Authorization header. The second registration attempt is successful.

```
[172.30.5.12]        [172.30.5.1]
   F1  |--- REGISTER -->|
   F2  |<-- 401 Unauth ----|
   F3  |--- REGISTER --->|
   F4 |<-----200 OK -------|
```

```
=================================================================
F1
172.30.1.12 (5060) ---- REGISTER ----> 172.30.1.5 (5060)

REGISTER sip:172.30.5.1;user=phone SIP/2.0
From: 9723542109 <sip:9723542109@172.30.5.12;user=phone>;tag=02ffb006-d93ddf4f
To: 9723542109 <sip:9723542109@172.30.5.12;user=phone>
Call-ID: 0eea65c0-630f598a-774af9dc@172.30.5.12
Via: SIP/2.0/UDP 172.30.5.12
CSeq: 439631084 REGISTER
Supported: timer, 100rel
Contact: 9723542109 <sip:9723542109@172.30.5.12;user=phone>
Content-Length: 0
-----------------------------------------------------------------

F2
172.30.1.5 (5060) ---- 401 Unauth ----> 172.30.1.12 (5060)

SIP/2.0 401 Unauthorized
From: 9723542109 <sip:9723542109@172.30.5.12;user=phone>;tag=02ffb006-d93ddf4f
To: 9723542109 <sip:9723542109@172.30.5.12;user=phone>
```

Call-ID: 0eea65c0-630f598a-774af9dc@172.30.5.12
CSeq: 439631084 REGISTER
WWW-Authenticate: Digest
realm="vocaldata.com",qop="auth",opaque="42e063c9b26f9b189c83cd5a596aa35a",nonce="6b9fe2f50fcad6eb134d952fe3a07160"
Content-Length: 0

F3
172.30.1.12 (5060) ---- REGISTER ----> 172.30.1.5 (5060)

REGISTER sip:172.30.5.1;user=phone SIP/2.0
From: 9723542109 <sip:9723542109@172.30.5.12;user=phone>;tag=02ffb006-d93ddf4f
To: 9723542109 <sip:9723542109@172.30.5.12;user=phone>
Call-ID: 0eea65c0-630f598a-774af9dc@172.30.5.12
Via: SIP/2.0/UDP 172.30.5.12
CSeq: 439631085 REGISTER
Supported: timer, 100rel
Authorization: Digest
realm="vocaldata.com",nonce="6b9fe2f50fcad6eb134d952fe3a07160",opaque="42e063c9b26f9b189c83cd5a596aa35a",qop=auth,username="9723542109",uri="sip:dummy",response="a3aa339fa168a357e2792d0a30e2f6b1",cnonce="4aafe4f95bfda97181fc90bf31bd981a",nc=00000001
Contact: 9723542109 <sip:9723542109@172.30.5.12;user=phone>
Content-Length: 0

F4
172.30.1.5 (5060) ---- 200 OK - ----> 172.30.1.12 (5060)

SIP/2.0 200 OK
From: 9723542109 <sip:9723542109@172.30.5.12;user=phone>;tag=02ffb006-d93ddf4f
To: 9723542109 <sip:9723542109@172.30.5.12;user=phone>
Call-ID: 0eea65c0-630f598a-774af9dc@172.30.5.12
CSeq: 439631085 REGISTER
Content-Length: 0

A.2 Basic Call

An endpoint from UAC (172.17.2.29) initiates a session by sending an INVITE to the UAS (172.30.1.5). UAS immediately replies with a '100 Trying'. After locating the destination (remote endpoint), the UAS rings the remote endpoint and at the same time alerts UAC with a '180 Ringing (SDP)' requires '100rel'. UAS is assumed to play the ring-back tone to the originating endpoint. UAC sends a provisional response acknowledge (PRACK) back to UAS.

A short time late the remote endpoint answers the call triggers UAS to send a '200 OK', which is acknowledged (ACK) by the UAC, and the call is connected.

At sometime later, originating endpoint terminates the call with a BYE, which is acknowledged by the UAS.

```
[172.17.2.29]        [172.30.1.5]
   F1  |----- INVITE ----->|
   F2  |<-- 100 Trying ----|
   F3  |<-- 180 Ringing ---|
   F4  |------ PRACK ----->|
   F5  |<---- 200 OK ------|
       UAC plays ring-back tone.

       Remote endpoint answers the call.
   F6  |<---- 200 OK ------|
   F7  |------- ACK ------>|
       Session begins. Call is connected.

   F8  |<------ BYE -------|
   F9  |----- 200 OK ----->|
       Session closes.
```

```
================================================================
F1
172.17.2.29 (5060) ---- INVITE ----> 172.30.1.5 (5060)

INVITE sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 207
Content-Type: application/sdp
Contact: sip:2143302105@172.17.2.29;user=phone
Session-Expires: 120
Require: 100rel
Allow: REFER, UPDATE, NOTIFY
Supported: timer

v=0
o=VocalData 7295 3647 IN IP4 172.17.2.29
s=VocalData SIP Call
c=IN IP4 172.17.2.31
t=0 0
```



```
m=audio 1000 RTP/AVP 0 4 18 101
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

-----------------------------------------------------------------
F2
172.30.1.5 (5060) ---- 100 Trying ----> 172.17.2.29 (5060)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 0

-----------------------------------------------------------------
F3
172.30.1.5 (5060) ---- 180 Ringing ----> 172.17.2.29 (5060)

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Require: 100rel
RSeq: 360
Allow: UPDATE, REFER, NOTIFY
Contact: <sip:9724411111@172.30.1.5:5060;user=phone>
Content-Type: application/sdp
Content-Length: 204

v=0
o=CiscoSystemsSIP-GW-UserAgent 1776 6232 IN IP4 172.30.1.5
s=SIP Call
c=IN IP4 172.30.1.5
t=0 0
m=audio 19374 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101

-----------------------------------------------------------------
F4
172.17.2.29 (5060) ---- PRACK ----> 172.30.1.5 (5060)

PRACK sip:9724411111@172.30.1.5:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
```


CSeq: 21479 PRACK
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone
RAck: 360 21478 INVITE

F5
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.29 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21479 PRACK
Content-Length: 0

F6
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.29 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Allow: UPDATE, REFER, NOTIFY
Contact: <sip:9724411111@172.30.1.5:5060;user=phone>
Content-Type: application/sdp
Content-Length: 204

v=0
o=CiscoSystemsSIP-GW-UserAgent 1776 6232 IN IP4 172.30.1.5
s=SIP Call
c=IN IP4 172.30.1.5
t=0 0
m=audio 19374 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101

F7
172.17.2.29 (5060) ---- ACK ----> 172.30.1.5 (5060)

ACK sip:9724411111@172.30.1.5:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 ACK
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone

F8
172.30.1.5 (51150) ---- BYE ----> 172.17.2.29 (5060)

BYE sip:2143302105@172.17.2.29:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Date: Sat, 01 Jan 2000 01:41:29 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
Max-Forwards: 6
CSeq: 21480 BYE
Content-Length: 0

F9
172.17.2.29 (5060) ---- 200 OK ----> 172.30.1.5 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21480 BYE
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone

A.3 Early Audio (remote busy tone)

In this case, busy tone is supplied from the remote end (via early media) to the originating endpoint. The session has never established, and is eventually terminated by a CANCEL.

```
 [172.17.2.29]        [172.30.1.5]
    F1  |----- INVITE ----->|
    F2  |<-- 100 Trying ----|
    F3  |<- 183 Session.. --|
```

UAS plays busy tone.

UAC cancels the session.

```
    F4  |------ PRACK ----->|
    F5  |<---- 200 OK ------|
    F6  |----- CANCEL ----->|
    F7  |<---- 200 OK ------|
    F8  |<- 487 Request.. --|
    F9  |------- ACK ------>|
==================================================================
F1
172.17.2.29 (5060) ---- INVITE ----> 172.30.1.5 (5060)

INVITE sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 2 <sip:2143302106@172.17.2.29;user=phone>;tag=C6D0.80E6
To: <sip:9724411111@172.30.1.5;user=phone>
Call-ID: 0800.20CE.A152.3C61.C6D0.80E6@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 205
Content-Type: application/sdp
Contact: sip:2143302106@172.17.2.29;user=phone
Session-Expires: 120
Require: 100Rel
Allow: UPDATE, REFER, NOTIFY
Supported: timer

v=0
o=VocalData 7387 3693 IN IP4 172.17.2.29
s=VocalData SIP Call
c=IN IP4 172.17.2.32
t=0 0
m=audio 1000 RTP/AVP 0 18 101
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

---------------------------------------------------------------
F2
172.30.1.5 (5060) ---- 100 Trying ----> 172.17.2.29 (5060)

SIP/2.0 100 Trying
```


Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 2 <sip:2143302106@172.17.2.29;user=phone>;tag=C6D0.80E6
To: <sip:9724411111@172.30.1.5;user=phone>
Date: Thu, 06 Jan 2000 04:03:06 GMT
Call-ID: 0800.20CE.A152.3C61.C6D0.80E6@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 0

F3
172.30.1.5 (5060) ---- 183 Session Progress ----> 172.17.2.29 (5060)

SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 2 <sip:2143302106@172.17.2.29;user=phone>;tag=C6D0.80E6
To: <sip:9724411111@172.30.1.5;user=phone>;tag=1A9E6A54-CB7
Date: Thu, 06 Jan 2000 04:03:06 GMT
Call-ID: 0800.20CE.A152.3C61.C6D0.80E6@172.17.2.29
CSeq: 21478 INVITE
Require: 100rel
Allow: UPDATE, REFER, NOTIFY
RSeq: 6732
Contact: <sip:9724411111@172.30.1.5:5060;user=phone>
Content-Type: application/sdp
Content-Length: 204

v=0
o=CiscoSystemsSIP-GW-UserAgent 9867 6054 IN IP4 172.30.1.5
s=SIP Call
c=IN IP4 172.30.1.5
t=0 0
m=audio 18042 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101

F4
172.17.2.29 (5060) ---- PRACK ----> 172.30.1.5 (5060)

PRACK sip:9724411111@172.30.1.5:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 2 <sip:2143302106@172.17.2.29;user=phone>;tag=C6D0.80E6
To: <sip:9724411111@172.30.1.5;user=phone>;tag=1A9E6A54-CB7
Call-ID: 0800.20CE.A152.3C61.C6D0.80E6@172.17.2.29
CSeq: 21479 PRACK
Content-Length: 0
Contact: sip:2143302106@172.17.2.29;user=phone
RAck: 6732 21478 INVITE

F5
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.29 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 2 <sip:2143302106@172.17.2.29;user=phone>;tag=C6D0.80E6


```
To: <sip:9724411111@172.30.1.5;user=phone>;tag=1A9E6A54-CB7
Date: Thu, 06 Jan 2000 04:03:06 GMT
Call-ID: 0800.20CE.A152.3C61.C6D0.80E6@172.17.2.29
CSeq: 21479 PRACK
Content-Length: 0

-----------------------------------------------------------------
F6
172.17.2.29 (5060) ---- CANCEL ----> 172.30.1.5 (5060)

CANCEL sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 2 <sip:2143302106@172.17.2.29;user=phone>;tag=C6D0.80E6
To: <sip:9724411111@172.30.1.5;user=phone>
Call-ID: 0800.20CE.A152.3C61.C6D0.80E6@172.17.2.29
CSeq: 21480 CANCEL
Content-Length: 0
Contact: sip:2143302106@172.17.2.29;user=phone

-----------------------------------------------------------------
F7
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.29 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 2 <sip:2143302106@172.17.2.29;user=phone>;tag=C6D0.80E6
To: <sip:9724411111@172.30.1.5;user=phone>
Date: Thu, 06 Jan 2000 04:03:08 GMT
Call-ID: 0800.20CE.A152.3C61.C6D0.80E6@172.17.2.29
Content-Length: 0
CSeq: 21480 CANCEL

-----------------------------------------------------------------
F8
172.30.1.5 (5060) ---- 487 Request Cancelled ----> 172.17.2.29 (5060)

SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 2 <sip:2143302106@172.17.2.29;user=phone>;tag=C6D0.80E6
To: <sip:9724411111@172.30.1.5;user=phone>;tag=1A9E6A54-CB7
Date: Thu, 06 Jan 2000 04:03:08 GMT
Call-ID: 0800.20CE.A152.3C61.C6D0.80E6@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 0

-----------------------------------------------------------------
F9
172.17.2.29 (5060) ---- ACK ----> 172.30.1.5 (5060)

ACK sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 2 <sip:2143302106@172.17.2.29;user=phone>;tag=C6D0.80E6
To: <sip:9724411111@172.30.1.5;user=phone>;tag=1A9E6A54-CB7
Call-ID: 0800.20CE.A152.3C61.C6D0.80E6@172.17.2.29
CSeq: 21478 ACK
Content-Length: 0
```

Contact: sip:2143302106@172.17.2.29;user=phone

A.4 Premature Call Termination (no answer)

The example in this section illustrates a case where the originating endpoint terminates (cancels) session before the session is completed.

```
[172.17.2.29]        [172.30.1.5]
   F1  |----- INVITE ----->|
   F2  |<-- 100 Trying ----|
   F3  |<-- 180 Ringing ---|
   F4  |------ PRACK ----->|
   F5  |<---- 200 OK ------|
       UAS  plays ring-back tone to the originating endpoint.
       Yet the remote endpoint has not answered.

       Originating terminates session request by a CANCEL request.
   F6  |----- CANCEL ----->|
   F7  |<---- 200 OK ------|
   F8  |<- 487 Request.. --|
   F9  |------- ACK ------>|
```

```
=================================================================
F1
172.17.2.29 (5060) ---- INVITE ----> 172.30.1.5 (5060)

INVITE sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0F00.809B
To: <sip:9724411111@172.30.1.5;user=phone>
Call-ID: 0800.20CE.A152.3C5B.0F00.809B@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 207
Content-Type: application/sdp
Contact: sip:2143302105@172.17.2.29;user=phone
Session-Expires: 120
Require: 100rel
Supported: timer

v=0
o=VocalData 7297 3648 IN IP4 172.17.2.29
s=VocalData SIP Call
c=IN IP4 172.17.2.31
t=0 0
m=audio 1000 RTP/AVP 0 4 18 101
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

-----------------------------------------------------------------
F2
172.30.1.5 (5060) ---- 100 Trying ----> 172.17.2.29 (5060)

SIP/2.0 100 Trying
```



```
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0F00.809B
To: <sip:9724411111@172.30.1.5;user=phone>
Date: Sat, 01 Jan 2000 01:45:09 GMT
Call-ID: 0800.20CE.A152.3C5B.0F00.809B@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 0

---------------------------------------------------------------
F3
172.30.1.5 (5060) ---- 180 Ringing ----> 172.17.2.29 (5060)

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0F00.809B
To: <sip:9724411111@172.30.1.5;user=phone>;tag=605240-25A8
Date: Sat, 01 Jan 2000 01:45:09 GMT
Call-ID: 0800.20CE.A152.3C5B.0F00.809B@172.17.2.29
CSeq: 21478 INVITE
Require: 100rel
RSeq: 5890
Contact: <sip:9724411111@172.30.1.5:5060;user=phone>
Content-Type: application/sdp
Content-Length: 204

v=0
o=CiscoSystemsSIP-GW-UserAgent 9867 6054 IN IP4 172.30.1.5
s=SIP Call
c=IN IP4 172.30.1.5
t=0 0
m=audio 18042 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101
---------------------------------------------------------------
F4
172.17.2.29 (5060) ---- PRACK ----> 172.30.1.5 (5060)

PRACK sip:9724411111@172.30.1.5:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0F00.809B
To: <sip:9724411111@172.30.1.5;user=phone>;tag=605240-25A8
Call-ID: 0800.20CE.A152.3C5B.0F00.809B@172.17.2.29
CSeq: 21479 PRACK
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone
RAck: 5890 21478 INVITE
---------------------------------------------------------------
F5
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.29 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0F00.809B
To: <sip:9724411111@172.30.1.5;user=phone>;tag=605240-25A8
Date: Sat, 01 Jan 2000 01:45:09 GMT
```


Call-ID: 0800.20CE.A152.3C5B.0F00.809B@172.17.2.29
CSeq: 21479 PRACK
Content-Length: 0

F6
172.17.2.29 (5060) ---- CANCEL ----> 172.30.1.5 (5060)

CANCEL sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0F00.809B
To: <sip:9724411111@172.30.1.5;user=phone>
Call-ID: 0800.20CE.A152.3C5B.0F00.809B@172.17.2.29
CSeq: 21480 CANCEL
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone

F7
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.29 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0F00.809B
To: <sip:9724411111@172.30.1.5;user=phone>
Date: Sat, 01 Jan 2000 01:45:13 GMT
Call-ID: 0800.20CE.A152.3C5B.0F00.809B@172.17.2.29
Content-Length: 0
CSeq: 21480 CANCEL

F8
172.30.1.5 (5060) ---- 487 Request Cancelled ----> 172.17.2.29 (5060)

SIP/2.0 487 Request Cancelled
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0F00.809B
To: <sip:9724411111@172.30.1.5;user=phone>;tag=605240-25A8
Date: Sat, 01 Jan 2000 01:45:13 GMT
Call-ID: 0800.20CE.A152.3C5B.0F00.809B@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 0

F9
172.17.2.29 (5060) ---- ACK ----> 172.30.1.5 (5060)

ACK sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0F00.809B
To: <sip:9724411111@172.30.1.5;user=phone>;tag=605240-25A8
Call-ID: 0800.20CE.A152.3C5B.0F00.809B@172.17.2.29
CSeq: 21478 ACK
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone

A.5 Call to busy subscriber

The User Agent (172.17.2.29) initiates a session by sending an INVITE to the Call Agent/UAS (172.30.1.5). UAS immediately replies with a ‘100 Trying’. The call is routed and terminates on a busy subscriber. The Call Agent returns a 486 (Busy) response.

```
[172.17.2.29]        [172.30.1.5]
   F1  |----- INVITE ----->|
   F2  |<-- 100 Trying ----|
   F3  |<-- 486 Busy--- ---|
   F4  |------- ACK ------>|

================================================================
F1
172.17.2.29 (5060) ---- INVITE ----> 172.30.1.5 (5060)

INVITE sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 207
Content-Type: application/sdp
Contact: sip:2143302105@172.17.2.29;user=phone
Session-Expires: 120
Require: 100rel
Allow: REFER, UPDATE, NOTIFY
Supported: timer

v=0
o=VocalData 7295 3647 IN IP4 172.17.2.29
s=VocalData SIP Call
c=IN IP4 172.17.2.31
t=0 0
m=audio 1000 RTP/AVP 0 4 18 101
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv

----------------------------------------------------------------
F2
172.30.1.5 (5060) ---- 100 Trying ----> 172.17.2.29 (5060)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 0

----------------------------------------------------------------
```


F3
172.30.1.5 (5060) ---- 486 Busy- ----> 172.17.2.29 (5060)

SIP/2.0 486 Busy Here
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 0

F4
172.17.2.29 (5060) ---- ACK ------> 172.30.1.5 (5060)

ACK sip:9724411111@172.30.1.5:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 ACK
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone

A.6 Session Timer

The following is a basic session timer. In the interest of brevity, all SDPs are intentionally left out of the SIP messages. This flow shows the UPDATE method being used to perform the refresh.

Example
1) UAC -> UAS
INVITE sip:9723543001@172.16.1.200;user=phone SIP/2.0
From: <sip:9723544001@172.16.1.189;user=phone>;tag=FC6F.803B
To: sip:9723543001@172.16.1.200;user=phone
Contact: sip:9723544001@172.16.1.189;user=phone
Call-ID: 8876.99B7.32B2.3B03.FC6F.803B@172.16.1.189
CSeq: 41 INVITE
Allow: UPDATE
Content-Length: 148
Content-Type: application/sdp
Supported: timer
Session-Expires: 240

2) UAC <- UAS
SIP/2.0 200 Ok
Via: SIP/2.0/UDP 172.16.1.189:5060
From: <sip:9723544001@172.16.1.189;user=phone>;tag=FC6F.803B
To: <sip:9723543001@172.16.1.200;user=phone>;tag=FC73.003D
Call-ID: 8876.99B7.32B2.3B03.FC6F.803B@172.16.1.189
CSeq: 41 INVITE
Allow: UPDATE
Content-Length: 144
Content-Type: application/sdp
Contact: sip:9723543001@172.16.1.200;user=phone
Require: timer
Session-Expires: 120;refresher=uac

In this example, the following actions are supposed;
UAC performs refreshes.
UAS wants a shorter timer (i.e. 120 instead of 240)
UAS starts session timer on sending of 200 OK
UAC starts session timer on receipt of 200 OK

3) UAC -> UAS
ACK

60 seconds later… UAC performs a refresh

4) UAC -> UAS
UPDATE sip:9723543001@172.16.1.200;user=phone SIP/2.0
From: sip:9723544001@172.16.1.189;user=phone;tag=FD7C.803C
To: sip:9723543001@172.16.1.200;user=phone;tag=FD80.003E
Contact: sip:9723544001@172.16.1.189;user=phone
Call-ID: 8876.99B7.32B2.3B03.FD7C.803C@172.16.1.189
CSeq: 26502 INVITE
Content-Length: 148
Content-Type: application/sdp
Supported: timer

Session-Expires: 120

5) UAC <- UAS
SIP/2.0 200 Ok
Via: SIP/2.0/UDP 172.16.1.189:5060
From: sip:9723544001@172.16.1.189;user=phone;tag=FD7C.803C
To: sip:9723543001@172.16.1.200;user=phone;tag=FD80.003E
Call-ID: 8876.99B7.32B2.3B03.FD7C.803C@172.16.1.189
CSeq: 26502 UPDATE
Content-Length: 144
Content-Type: application/sdp
Contact: sip:9723543001@172.16.1.200;user=phone
Require: timer
Session-Expires: 120

110 seconds later… The UAS did not receive a re-INVITE. Therefore, it considers the call terminated and sends a BYE:

6) UAC <- UAS
BYE sip:9723544001@172.16.1.189;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.16.1.200:5060
From: <sip:9723543001@172.16.1.200;user=phone>;tag=FE92.003F
To: <sip:9723544001@172.16.1.189;user=phone>;tag=FE8E.803B
Call-ID: 8876.99B7.32B2.3B03.FD7C.803C @172.16.1.189
Content-Length: 0
CSeq: 26962 BYE
Supported: timer

7) UAC -> UAS
SIP/2.0 200 Ok
Via: SIP/2.0/UDP 172.16.1.200:5060
From: <sip:9723543001@172.16.1.200;user=phone>;tag=FE92.003F
To: <sip:9723544001@172.16.1.189;user=phone>;tag=FE8E.803B
Call-ID: 8876.99B7.32B2.3B03.FD7C.803C @172.16.1.189
CSeq: 26962 BYE
Content-Length: 0
Contact: sip:9723544001@172.16.1.189;user=phone

A.7 Unattended Transfer

A basic call is set up as in appendix A.1.

At sometime later, originating endpoint terminates the call with a BYE, which is acknowledged by the UAS.

```
[172.17.2.29]        [172.30.1.5]
   F1  |----- INVITE ----->|
   F2  |<-- 100 Trying ----|
   F3  |<-- 180 Ringing ---|
   F4  |------ PRACK ----->|
   F5  |<---- 200 OK ------|
       UAC plays ring-back tone.

       Remote endpoint answers the call.
   F6  |<---- 200 OK ------|
   F7  |------- ACK ------>|
       Session begins. Call is connected.

      Call is transferred via use of REFER
   F8  |----- REFER ----->|
   F9  |<-- 202 Accepted--|
   F10 |<-- NOTIFY---- ---|
   F11|-------- BYE ------>|
   F12|<---- 200 OK ------|

       Final notification is sent
   F8  |<------ NOTIFY----|
   F9  |----- 200 OK ----->|
```

==
F1 – F7 as appendix A.1

F8
172.17.2.29 (5060) ---- REFER ----> 172.30.1.5 (5060)

REFER sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21480 REFER
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone
Refer-To: sip:9724412345@172.30.1.7;user=phone
Referred-By: sip:9724411111@172.30.1.5;user=phone

F9
172.17.2.29 (5060) ---- 202 Accepted ----> 172.30.1.5 (5060)

SIP/2.0 202 Accepted
Via: SIP/2.0/UDP 172.30.1.5:5060


```
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21480 REFER
Content-Length: 0
---------------------------------------------------------------
F10
172.30.1.5 (51150) ---- NOTIFY ----> 172.17.2.29 (5060)

NOTIFY sip:2143302105@172.17.2.29:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP  172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 101 NOTIFY
Event: refer
Subscription-State: active;expires=60
Content-Type: message/sipfrag
Content-Length: 20

SIP/2.0 100 Trying

---------------------------------------------------------------
F11
172.17.2.29 (5060) ---- 200 OK ----> 172.30.1.5 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 101 NOTIFY
Content-Length: 0
---------------------------------------------------------------
F12
172.17.2.29 (5060) ---- BYE ----> 172.30.1.5 (5060)

BYE sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21481 BYE
Content-Length: 0
---------------------------------------------------------------
F13
172.17.2.29 (5060) ---- 200 OK----- ----> 172.30.1.5 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21481 BYE
Content-Length: 0
---------------------------------------------------------------
```


F14
172.30.1.5 (51150) ---- NOTIFY ----> 172.17.2.29 (5060)

```
NOTIFY sip:2143302105@172.17.2.29:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP  172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 102 NOTIFY
Event: refer
Subscription-State: terminated;reason=noresource
Content-Type: message/sipfrag
Content-Length: 16

SIP/2.0 200 OK
```

F15
172.17.2.29 (5060) ---- 200 OK ----> 172.30.1.5 (5060)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 102 NOTIFY
Content-Length: 0
```

A.8 Call Back When Free (Subscribe/Notify)

In this flow, the initial call attempt fails due to busy. The originating UA then uses SUBSCRIBE/NOTIFY to be informed of when the terminating UA becomes free. At this point, the call is successfully terminated.

```
172.17.2.29]        [172.30.1.5]
   F1  |----- INVITE ----->|
   F2  |<-- 100 trying-- ----|
   F3  |<-- 486 Busy--- ----|
   F4  |--------ACK ------>|
```

UA uses SUBSCRIBE to monitor the far end

```
   F5  |--- SUBSCRIBE-->|
   F6  |<---- 200 OK--------|
   F7  |<-- NOTIFY---- ---|
   F8  |-----200 OK- ------>|
```

UA becomes free - final notification is sent

```
   F9 |<------ NOTIFY----|
   F10|----- 200 OK ----->|
```

Session is established

```
   F11 |----- INVITE ----->|
   F12|<-- 100 Trying ----|
   F13|<-- 180 Ringing ---|
   F14|------ PRACK ----->|
   F15|<---- 200 OK ------|
   F16|<---- 200 OK ------|
   F17|------- ACK ------>|
================================================================
```

F1-F4 As Appendix A.5.

```
---------------------------------------------------------------
F5
172.17.2.29 (5060) ---- SUBSCRIBE ----> 172.30.1.5 (5060)

SUBSCRIBE sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21479 SUBSCRIBE
Contact: sip:2143302105@172.17.2.29;user=phone
Event: dialog
Accept: application/dialog-info+xml
Content-Length: 0
---------------------------------------------------------------
F6
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.29 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
```

```
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21479 SUBSCRIBE
Contact: sip:2143302105@172.17.2.29;user=phone
Content-Length: 0
```

F7
172.30.1.5 (5060) ---- NOTIFY- ----> 172.17.2.29 (5060)

```
NOTIFY sip:2143302105@172.17.2.29:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP  172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
Max-Forwards: 70
CSeq: 1 NOTIFY
Event: dialog
Subscription-State: active;expires=3600
Content-Type: application/dialog-info+xml
Content-Length: xxx

<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
version="0" state="full" entity="sips:bob@biloxi.example.com">
<dialog id="562623442g3">
<state>confirmed</state>
</dialog>
</dialog-info>
```

F8
172.17.2.29 (5060) ---- 200 OK ----> 172.30.1.5 (5060)

```
SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 1 NOTIFY
Content-Length: 0
```

F9
172.30.1.5 (5060) ---- NOTIFY- ----> 172.17.2.29 (5060)

```
NOTIFY sip:2143302105@172.17.2.29:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP  172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
Max-Forwards: 70
CSeq: 2 NOTIFY
Event: dialog
Subscription-State: active;expires=3600
Content-Type: application/dialog-info+xml
Content-Length: xxx
```


<?xml version="1.0"?>
<dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info"
version="0" state="full" entity="sips:bob@biloxi.example.com">
<dialog id="562623442g3">
<state>terminated</state>
</dialog>
</dialog-info>

F10
172.17.2.29 (5060) ---- 200 OK ----> 172.30.1.5 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 2 NOTIFY
Content-Length: 0

F11 – F17 As appendix A.2.

A.9 RFC 3312 usage (A-side UA)

This flow illustrates usage of RFC 3312 between the CA and originating UA.

Even though NAT & SBC would be typically used in conjunction with RFC 3312, no such gate control messages (e.g. GCP-M or the ETSI gate control protocol) are shown in this flow.

The flow is similar to appendix A.1. However, in this case, there are additional RFC 3312 parameters. Additional contents are shown in bold to aid readability. In this flow, the UA is shown as supporting RFC3312. This is, in practice, unlikely and the CA could always include RFC3312 SDP attributes irrespective of the UA support.

Additional comments relating to the RFC 3312 usage are provided in-line *in italics* within the message flow. This flow should be read in conjunction with appendix A.10 which shows the corresponding flow at the B-side UA.

[172.17.2.29] [172.30.1.5]
F1 |----- INVITE ----->|
F2 |<-- 100 Trying ----|
F3 |<-- 183 Progress---|
F4 |------ PRACK ----->|
F5 |<---- 200 OK ------|
F6 |<-- 180 Ringing ---|

F7 |------ PRACK ----->|
F8 |<---- 200 OK ------|
UAC plays ring-back tone.

Remote endpoint answers the call.
F9 |<---- 200 OK ------|
F10|------- ACK ------>|
Session begins. Call is connected.

F11|<------ BYE -------|
F12|----- 200 OK ----->|
Session closes.

==
F1
172.17.2.29 (5060) ---- INVITE ----> 172.30.1.5 (5060)

INVITE sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Content-Length: **268**
Content-Type: application/sdp
Contact: sip:2143302105@172.17.2.29;user=phone
Session-Expires: 120
Require: 100rel
Allow: REFER, UPDATE, NOTIFY
Supported: timer

v=0
o=VocalData 7295 3647 IN IP4 172.17.2.29
s=VocalData SIP Call
c=IN IP4 172.17.2.31
t=0 0
m=audio 1000 RTP/AVP 0 4 18 101
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=curr:qos e2e none
a=des:qos mandatory e2e sendrecv

the UA is requesting QOS for the call but has not pre-allocated any resources.

F2
172.30.1.5 (5060) ---- 100 Trying ----> 172.17.2.29 (5060)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Content-Length: 0

F3
172.30.1.5 (5060) ---- 183 Progress ----> 172.17.2.29 (5060)

SIP/2.0 183 Progress
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone**>;tag=0E1D.8099**
To: <sip:9724411111@172.30.1.5;user=phone**>;tag=5CEE58-2187**
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Require: 100rel
RSeq: 360
Allow: UPDATE, REFER, NOTIFY
Contact: <sip:9724411111@172.30.1.5:5060;user=phone>
Content-Type: application/sdp
Content-Length: 261

v=0
o=CiscoSystemsSIP-GW-UserAgent 1776 6232 IN IP4 172.30.1.5
s=SIP Call
c=IN IP4 172.30.1.5
t=0 0
m=audio 19374 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101

a=curr: qos e2e sendrecv
a=des: qos mandatory e2e sendrecv

The CA is telling the UA that QOS has been satisfied. The UA awaits ringing.

F4
172.17.2.29 (5060) ---- PRACK ----> 172.30.1.5 (5060)

PRACK sip:9724411111@172.30.1.5:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21479 PRACK
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone
RAck: 360 21478 INVITE

F5
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.29 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21479 PRACK
Content-Length: 0

F6
172.30.1.5 (5060) ---- 180 Ringing ----> 172.17.2.29 (5060)

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Require: 100rel
RSeq: 361
Allow: UPDATE, REFER, NOTIFY
Contact: <sip:9724411111@172.30.1.5:5060;user=phone>
Content-Length: 0

F7
172.17.2.29 (5060) ---- PRACK ----> 172.30.1.5 (5060)

PRACK sip:9724411111@172.30.1.5:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099

To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21480 PRACK
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone
RAck: 361 21478 INVITE

F8
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.29 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21480 PRACK
Content-Length: 0

F9
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.29 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 INVITE
Allow: UPDATE, REFER, NOTIFY
Contact: <sip:9724411111@172.30.1.5:5060;user=phone>
Content-Type: application/sdp
Content-Length: **261**

v=0
o=CiscoSystemsSIP-GW-UserAgent 1776 6232 IN IP4 172.30.1.5
s=SIP Call
c=IN IP4 172.30.1.5
t=0 0
m=audio 19374 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101
a=curr:qos e2e sendrecv
a=des:qos mandatory e2e sendrecv

F10
172.17.2.29 (5060) ---- ACK ----> 172.30.1.5 (5060)

ACK sip:9724411111@172.30.1.5:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.17.2.29:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187


```
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21478 ACK
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone

----------------------------------------------------------------
F11
172.30.1.5 (51150) ---- BYE ----> 172.17.2.29 (5060)

BYE sip:2143302105@172.17.2.29:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP  172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Date: Sat, 01 Jan 2000 01:41:29 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
Max-Forwards: 6
CSeq: 21481 BYE
Content-Length: 0

----------------------------------------------------------------
F12
172.17.2.29 (5060) ---- 200 OK ----> 172.30.1.5 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.17.2.29
CSeq: 21481 BYE
Content-Length: 0
Contact: sip:2143302105@172.17.2.29;user=phone
----------------------------------------------------------------
```


A.10 RFC 3312 usage (B-side UA)

This flow illustrates usage of RFC 3312 at the terminating UA.

Even though NAT & SBC would be typically used in conjunction with RFC 3312, no such gate control messages (e.g. GCP-M or the ETSI gate control protocol) are shown in this flow.

The flow is similar to appendix A.1. However, in this case, there are additional RFC 33122 parameters. Additional contents are shown in bold to aid readability.

Additional comments relating to the RFC 3312 usage are provided in-line *in italics* within the message flow. This flow should be read in conjunction with appendix A.10 which shows the corresponding flow at the A-side UA.

[172.30.1.5] [172.17.2.30]
F1 |----- INVITE ----->|
F2 |<-- 100 Trying ----|
F3 |<-- 183 Progress---|
F4 |------ PRACK ----->|
F5 |<---- 200 OK ------|
F6 |<-- 180 Ringing ---|
F7 |------ PRACK ----->|
F8 |<---- 200 OK ------|
UAC plays ring-back tone.

Remote endpoint answers the call.
F9 |<---- 200 OK ------|
F10|------- ACK ------>|
Session begins. Call is connected.

F11|<------ BYE -------|
F12|----- 200 OK ----->|
Session closes.

==
F1
172.30.1.5 (5060) ---- INVITE ----> 172.17.2.30 (5060)

INVITE sip:9724411111@172.30.1.5;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.30.1.5:5060
From: IP30 Phone 1 <sip:2143302105@172.17.2.29;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.17.2.30;user=phone>
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21478 INVITE
Content-Length: **268**
Content-Type: application/sdp
Contact: sip:2143302105@172.30.1.5;user=phone
Session-Expires: 120
Require: 100rel
Allow: REFER, UPDATE, NOTIFY
Supported: timer

v=0
o=VocalData 7295 3647 IN IP4 172.30.1.5

s=VocalData SIP Call
c=IN IP4 172.30.1.5
t=0 0
m=audio 1000 RTP/AVP 0 4 18 101
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=curr:tp-qos e2e sendrecv
a=des:tp-qos mandatory e2e sendrecv

The CA tells the UA that QOS is assured.

F2
172.17.2.30 (5060) ---- 100 Trying ----> 172.30.1.5 (5060)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP 172.30.1.5:5060
From: IP30 Phone 1 <sip:2143302105@172.30.1.5;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.30.1.5;user=phone>
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21478 INVITE
Content-Length: 0

F3
172.17.2.30 (5060) ---- 183 Progress ----> 172.30.1.5 (5060)

SIP/2.0 183 Progress
Via: SIP/2.0/UDP 172.30.1.5:5060
From: IP30 Phone 1 <sip:2143302105@172.30.1.5;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.17.2.30;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21478 INVITE
Require: 100rel
RSeq: 360
Allow: UPDATE, REFER, NOTIFY
Contact: <sip:9724411111@172.30.1.5:5060;user=phone>
Content-Type: application/sdp
Content-Length: 261

v=0
o=CiscoSystemsSIP-GW-UserAgent 1776 6232 IN IP4 172.17.2.30
s=SIP Call
c=IN IP4 172.17.2.30
t=0 0
m=audio 19374 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101
a=curr:tp-qos e2e sendrecv
a=des:tp-qos mandatory e2e sendrecv

this SDP returns the reduced set of common codecs. Typically, the 3312 attributes would be missing.

F4
172.30.1.5 (5060) ---- PRACK ----> 172.17.2.30 (5060)

PRACK sip:9724411111@172.17.2.30:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.30.1.5:5060
From: IP30 Phone 1 <sip:2143302105@172.30.1.5;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.17.2.30;user=phone>;tag=5CEE58-2187
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21479 PRACK
Content-Length: 0
Contact: sip:2143302105@172.30.1.5;user=phone
RAck: 360 21478 INVITE

F5
172.17.2.30 (5060) ---- 200 OK ----> 172.30.1.5 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: IP30 Phone 1 <sip:2143302105@172.30.1.5;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.17.2.30;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21479 PRACK
Content-Length: 0

F6
172.17.2.30 (5060) ---- 180 Ringing ----> 172.30.1.5 (5060)

SIP/2.0 180 Ringing
Via: SIP/2.0/UDP 172.30.1.5:5060
From: IP30 Phone 1 <sip:2143302105@172.30.1.5;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.17.2.30;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21478 INVITE
Require: 100rel
RSeq: 361
Allow: UPDATE, REFER, NOTIFY
Contact: <sip:9724411111@172.30.1.5:5060;user=phone>
Content-Length: 0

F9
172.30.1.5 (5060) ---- PRACK ----> 172.17.2.30 (5060)

PRACK sip:9724411111@172.17.2.30:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.30.1.5:5060
From: IP30 Phone 1 <sip:2143302105@172.30.1.5;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.17.2.30;user=phone>;tag=5CEE58-2187
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21480 PRACK

Content-Length: 0
Contact: sip:2143302105@172.30.1.5;user=phone
RAck: 361 21478 INVITE

F10
172.17.2.30 (5060) ---- 200 OK ----> 172.30.1.5 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: IP30 Phone 1 <sip:2143302105@172.30.1.5;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.17.2.30;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21480 PRACK
Content-Length: 0

F11
172.17.2.30 (5060) ---- 200 OK ----> 172.30.1.5 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: IP30 Phone 1 <sip:2143302105@172.30.1.5;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.17.2.30;user=phone>;tag=5CEE58-2187
Date: Sat, 01 Jan 2000 01:41:27 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21478 INVITE
Allow: UPDATE, REFER, NOTIFY
Contact: <sip:9724411111@172.30.1.5:5060;user=phone>
Content-Type: application/sdp
Content-Length: **267**

v=0
o=CiscoSystemsSIP-GW-UserAgent 1776 6232 IN IP4 172.17.2.30
s=SIP Call
c=IN IP4 172.17.2.30
t=0 0
m=audio 19374 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101
a=curr:tp-qos e2e sendrecv
a=des:tp-qos mandatory e2e sendrecv

F12
172.30.1.5 (5060) ---- ACK ----> 172.17.2.30 (5060)

ACK sip:9724411111@172.17.2.30:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.30.1.5:5060
From: IP30 Phone 1 <sip:2143302105@172.30.1.5;user=phone>;tag=0E1D.8099
To: <sip:9724411111@172.17.2.30;user=phone>;tag=5CEE58-2187
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21478 ACK
Content-Length: 0

Contact: sip:2143302105@172.30.1.5;user=phone

F13
172.17.2.30 (51150) ---- BYE ----> 172.30.1.5 (5060)

BYE sip:2143302105@172.30.1.5:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.30;user=phone>;tag=0E1D.8099
Date: Sat, 01 Jan 2000 01:41:29 GMT
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
Max-Forwards: 6
CSeq: 21481 BYE
Content-Length: 0

F14
172.30.1.5 (5060) ---- 200 OK ----> 172.17.2.30 (5060)

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.30.1.5:5060
From: <sip:9724411111@172.30.1.5;user=phone>;tag=5CEE58-2187
To: IP30 Phone 1 <sip:2143302105@172.17.2.30;user=phone>;tag=0E1D.8099
Call-ID: 0800.20CE.A152.3C5B.0E1D.8099@172.30.1.5
CSeq: 21481 BYE
Content-Length: 0
Contact: sip:2143302105@172.30.1.5;user=phone

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