Control of Service Context using SIP Request-URI

Status of this Memo

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Abstract

This memo provides information for the Internet community. It describes a useful way to conceptualize the use of the standard SIP (Session Initiation Protocol) Request-URI (Uniform Resource Identifier) that the authors and many members of the SIP community think is suitable as a convention. It does not define any new protocol with respect to RFC 2543.

In a conventional telephony environment, extended service applications often use call state information, such as calling party, called party, reason for forward, etc, to infer application context. In a SIP/2.0 call, much of this information may be either non-existent or unreliable. This document proposes a mechanism to communicate context information to an application. Under this proposal, a client or proxy can communicate context through the use of a distinctive Request-URI. This document continues with examples of how this mechanism could be used in a voice mail application.
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1. Introduction

A communication service should make use of the information it has at hand when being accessed. For example, in most current voice mail implementations, a subscriber retrieving messages from his own desk does not have to reenter his voice mailbox number – the service assumes that the store being accessed is the one associated with the endpoint being used to access the service. Some services allow the user to validate this assumption using IVR techniques before prompting for a PIN.

This concept of context-awareness can be captured in a voice mail service implementing SIP as defined in RFC 2543[1], without modification, through the standard use of that protocol’s Request-URI. Furthermore, the concept is applicable to any SIP-based service where initial application state should be determined from context.

This concept is a usage convention of standard SIP as defined in RFC 2543[1] and does not modify or extend that protocol in any way.

2. Example Application

In this document, we use the example of voice mail to illustrate the technique. One motivation for applying this technique to this problem is allowing a proxy or location server to control the initial state of a voice service. For example, a voice client might register a contact list ending with the URL that would accept voice messages for the client.

2.1 Using URIs to Control Voice Mail Service Behavior

Many conventional voice mail systems use call state information, such as the calling party, called party, reason for forward, etc, to decide the initial application state. For example, it might play one outgoing message if the call reached voice mail because the called party did not answer and another if the line was busy. It decides whom the message is for based on the called party information. If the call originated from a subscriber’s phone number, it might authenticate the caller and then go directly to the message retrieval and account maintenance menu.

When a new subscriber is added to a system, a set of identities could be generated, each given a unique sip URI. The following tables show some of the identities that might be generated (it is not exhaustive). The example schemes show that the URIs could, but don’t necessarily have to, have mnemonic value.
In practical applications, it is important that an application does not apply semantic rules to the various URIs. Instead, it should allow any arbitrary string to be provisioned, and map the string to the desired behavior. The owner of the system may choose to provision mnemonic strings, but the application should not require it. In any large installation, the system owner is likely to have pre-existing rules for mnemonic URIs, and any attempt by an application to define its own rules may create a conflict. For our example, this means a voice mail system should allow an arbitrary mix of URLs from these schemes, or any other scheme that renders valid SIP URIs to be provisioned, rather than enforce one particular scheme.

<table>
<thead>
<tr>
<th>URI Identity</th>
<th>Example Scheme 1</th>
<th>Example Scheme 2</th>
<th>Example Scheme 3</th>
</tr>
</thead>
<tbody>
<tr>
<td>Deposit with</td>
<td>sip:<a href="mailto:sub-rjs-deposit@vm.wcom.com">sub-rjs-deposit@vm.wcom.com</a></td>
<td>sip:<a href="mailto:677283@vm.wcom.com">677283@vm.wcom.com</a></td>
<td>sip:<a href="mailto:rjs@vm.wcom.com">rjs@vm.wcom.com</a>;mode=deposit</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Deposit with</td>
<td>sip:sub-rjs-deposit-busy.vm.wcom.com</td>
<td>sip:<a href="mailto:677372@vm.wcom.com">677372@vm.wcom.com</a></td>
<td>sip:<a href="mailto:rjs@vm.wcom.com">rjs@vm.wcom.com</a>;mode=3991243</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Deposit with</td>
<td>sip:<a href="mailto:sub-rjs-deposit-sg@vm.wcom.com">sub-rjs-deposit-sg@vm.wcom.com</a></td>
<td>sip:<a href="mailto:677384@vm.wcom.com">677384@vm.wcom.com</a></td>
<td>sip:<a href="mailto:rjs@vm.wcom.com">rjs@vm.wcom.com</a>;mode=sg</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Retrieve - SIP</td>
<td>sip:<a href="mailto:sub-rjs-retrieve@vm.wcom.com">sub-rjs-retrieve@vm.wcom.com</a></td>
<td>sip:<a href="mailto:677405@vm.wcom.com">677405@vm.wcom.com</a></td>
<td>sip:<a href="mailto:rjs@vm.wcom.com">rjs@vm.wcom.com</a>;mode=retrieve</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Retrieve - prompt</td>
<td>sip:sub-rjs-retrieve-inpin.vm.wcom.com</td>
<td>sip:<a href="mailto:677415@vm.wcom.com">677415@vm.wcom.com</a></td>
<td>sip:<a href="mailto:rjs@vm.wcom.com">rjs@vm.wcom.com</a>;mode=inpin</td>
</tr>
</tbody>
</table>

When a service is first set up, identities such as the following could be created.

<table>
<thead>
<tr>
<th>URI Identity</th>
<th>Example Scheme 1</th>
<th>Example Scheme 2</th>
<th>Example Scheme 3</th>
</tr>
</thead>
<tbody>
<tr>
<td>Deposit -</td>
<td>sip:<a href="mailto:deposit@vm.wcom.com">deposit@vm.wcom.com</a></td>
<td>sip:<a href="mailto:670001@vm.wcom.com">670001@vm.wcom.com</a></td>
<td>sip:<a href="mailto:deposit@vm.wcom.com">deposit@vm.wcom.com</a></td>
</tr>
<tr>
<td>identify target</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>mailbox by To:</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
In addition to providing this set of URIs to the subscriber (to use as he sees fit), an integrated service provider could add these to the set of contacts in a find-me proxy. The proxy could then route calls to the appropriate URI based on the origin of the request, the subscriber’s preferences and current state.

3. Voice Mail Scenario Descriptions

In each of these scenarios, the PSTN gateway is configured to communicate only with a particular proxy-registrar.

3.1 Deposits

3.1.1 Direct Request to Deposit to a particular mailbox

3.1.1.1 SIP source

A SIP client that knew the URI for a particular deposit mailbox (sip:sub-rjs-deposit@vm.wcom.com) could place a direct invitation to the voicemail service, or through a protecting proxy. The proxy could restrict access to deposit identities with special greetings by authenticating the requester.

3.1.1.2 Arbitrary PSTN source

The gateway’s proxy would map a call from an unrecognized PSTN number to a number associated with a subscriber’s mailbox into an invite to the deposit with standard greeting URI (sip:sub-rjs-deposit@vm.wcom.com).
3.1.1.3 Recognized PSTN source

The gateway’s proxy would map a call from a recognized (exact or pattern match) PSTN number to a number associated with a subscriber’s mailbox into an invite to the appropriate special greeting URI (sip:sub-rjs-deposit-sg@vm.wcom.com). The gateway’s ability to identify the calling party (using calling party number) is trusted, so no further authentication of the requester is performed.

3.1.2 Direct Request to Deposit, mailbox to be determined

3.1.2.1 SIP source

A voice mail service or its protecting proxy could expose a generic deposit URL for use when a caller wished to go directly to voice mail. The service would likely play an IVR dialog to determine what message store to deposit a message into.

An application designer may be tempted to attempt to match the To: and From: headers on a call to infer information. However, this approach could cause complications when multiple proxy forwards occur in a call. For example, A calls B, who has all calls forwarded to C. C forwards the call to her voice mail service. If the voice mail service matches the To: header to determine the message store, it will get the information for B instead of C. But there is no reason to assume that C’s voice mail service has any knowledge of B.

3.1.2.2 PSTN source

The gateway’s proxy would map a call from an unrecognized PSTN number to the top level voice mail service access number to an invite to the Deposit - prompt for target mailbox in-band URI (sip:deposit-in@vm.wcom.com for example). Getting the call to the target mailbox would proceed as in the SIP source case.

3.1.2.3 Indirect Request to Deposit, due to find-me proxy decision

A find-me proxy could map an invitation to a subscriber (sip:rjs@wcom.com) to the appropriate voice mail service URI depending on the subscriber’s current state. The normal deposit URI could be chosen if the subscriber’s contact list has been otherwise exhausted with no answer. The busy-announcement URI would be chosen when a busy everywhere response is received from one of the contacts. A DND announcement URI could be selected if the subscriber had activated DND. Calls to sip:receptionist@wcom.com could be configured to roll to sip:deposit@wcom.com.
3.2 Retrievals

3.2.1 Request to Retrieve from a particular mailbox

3.2.1.1 Trusted SIP source

A request to retrieve the contents of a particular mailbox (sip:sub-rjs-retrieve@vm.wcom.com) coming from a trusted source could be honored without further authentication checks. A trusted source is one with which the voice mail service has secure communications, and to which it is willing to delegate authentication. This could be the service’s protecting proxy for example.

3.2.1.2 Authenticated SIP source

A service, or its protecting proxy, could choose to honor a retrieve request for a particular mailbox (sip:sub-rjs-retrieve@vm.wcom.com) based on SIP authentication. If SIP level authentication failed, the service or proxy could be configured to send the call to the in-band pin prompting URI (sip:sub-rjs-retrieve-inpin@vm.wcom.com).

3.2.1.3 Unauthenticated SIP source

A service, or its protecting proxy, receiving a retrieve request for a particular mailbox (sip:sub-rjs-retrieve@vm.wcom.com) with no other method of authenticating the requestor could send the request to the in-band pin prompting URI (sip:sub-rjs-retrieve-inpin@vm.wcom.com).

3.2.1.4 PSTN source

This scenario assumes that the service provider’s network has been configured such that a PSTN number could be dialed explicitly for retrieving messages from a particular mailbox. Such services currently exist, but are not common. In such a network, the gateway’s proxy would map the call to the in-band pin prompting URI (sip:sub-rjs-retrieve-inpin@vm.wcom.com).

3.2.2 Request to Retrieve, mailbox to be determined

3.2.2.1 SIP source

As in the Request to Deposit scenario, when a service receives a request for the top level retrieve URI it would most likely need to use in-band IVR techniques to determine the target mailbox and authenticate the caller.
3.2.2.2 Arbitrary PSTN source

This scenario assumes there is a single PSTN number that subscribers dial to access the voice mail service to retrieve messages. This is the most common access method provided by current voice mail services.

The gateway’s proxy would map a call to the top level PSTN number to the top level retrieve in-band prompting URI (sip:retrieve-in@vm.wcom.com). Once the system identifies the target mailbox, the call would be transferred to the appropriate in-band pin prompting URI (sip:sub-rjs-retrieve-inpin@vm.wcom.com).

3.2.2.3 Recognized PSTN source

This scenario also assumes there is a single PSTN number that subscribers dial to access the voice mail service to retrieve messages.

The gateway’s proxy would recognize the calling party number as a subscriber, and map the call to the subscriber’s in-band prompting URI (sip:sub-rjs-retrieve-inpin@vm.wcom.com).

4. Voice Mail Call Flow Examples

The following section describes some example call flows for a hypothetical voice mail service, with the host name of vm.wcom.com. All the call flows assume that a proxy protects the voice mail service and that a trust relationship exists between the voice mail service and the proxy.

4.1 Generic Scenario

4.1.1 Direct call to the voice mail system

User A calls the voice mail system directly. The voice mail system invokes the top-level menu, which might prompt the caller for an extension or the first few letters of a name.
Flow Id | Comments
--- | ---
INVITE F1 | INVITE sip:VoiceMail@wcom.com SIP/2.0
A->Proxy | Via: SIP/2.0/UDP here.com:5060
 | From: TheBigGuy <sip:UserA@here.com>
 | To: VoiceMail <sip:VoiceMail@wcom.com>
 | Call-Id: 12345600@here.com
 | CSeq: 1 INVITE
 | Contact: TheBigGuy <sip:UserA@here.com>
 | Proxy-Authorization:Digest username="UserA",
 | realm="MCI WorldCom SIP",
 | nonce="ea9c8e88df84f1cc4e341ae6cbe5a359", opaque="",
 | uri="sip:VoiceMail@wcom.com", response=<appropriately calculated hash goes here>
 | Content-Type: application/sdp
 | Content-Length: <appropriate value>
v=0
o=<UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/*Client for A prepares to receive data on port 49170
from the network. */

INVITE F2
Proxy->VM
INVITE sip:top@vm.wcom.com SIP/2.0
Via: SIP/2.0/UDP wcom.com:5060; branch=1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:VoiceMail@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=<UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying)
F3
Proxy->A)
SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing
F4
VM->Proxy
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP wcom.com:5060; branch=1
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
180 Ringing  SIP/2.0 180 Ringing
F5          Via: SIP/2.0/UDP here.com:5060
Proxy->A    From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

200 OK F6   SIP/2.0 200 OK
VM->Proxy   Via: SIP/2.0/UDP wcom.com:5060; branch=1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:VoiceMail@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: VoiceMailSystem <sip:top@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F7   SIP/2.0 200 OK
Proxy->A    Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:VoiceMail@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: VoiceMailSystem <sip:top@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
ACK F8 A->Proxy
Via: SIP/2.0/UDP here.com:5060
Route:<sip:top@vm.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F9 Proxy->VM
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>; tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and VM. VM system starts IVR dialog for top level menu */

/* User A Hangs Up with VM system. Alternatively, the VM system could initiate the BYE*/

BYE F10 A->Proxy
Via: SIP/2.0/UDP here.com:5060
Route:<sip: top@vm.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

BYE F11 Proxy->VM
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F12 VM->Proxy
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-ID: 12345600@here.com
4.2 Message Deposit Scenarios

4.2.1 Call to known subscriber forwarded on no answer

User A attempts to call User B, who does not answer. The call is forwarded to User B’s mailbox, and the voice mail system plays User B’s outgoing message for a ring-no-answer. The flow assumes that the URL of "sip:UserB-dep-fna@vm.wcom.com" maps to the desired behavior for depositing a message on a forward-no-answer.
User A | Proxy | User B | VM System
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE F1</td>
<td>INVITE F2</td>
<td>(100 Trying) F3</td>
<td>180 Ringing F4</td>
</tr>
<tr>
<td>----------</td>
<td>------------</td>
<td>&lt;---------------</td>
<td>&lt;---------------</td>
</tr>
<tr>
<td>180 Ringing F5</td>
<td>(Request Timeout)</td>
<td>Cancel F6</td>
<td>200 OK F7</td>
</tr>
<tr>
<td>&lt;---------------</td>
<td>&lt;---------------</td>
<td>&lt;---------------</td>
<td>&lt;---------------</td>
</tr>
<tr>
<td>200 OK F10</td>
<td>&lt;---------------</td>
<td>ACK F11</td>
<td>ACK F12</td>
</tr>
<tr>
<td>ACK F11</td>
<td>&lt;---------------</td>
<td>RTP Established Both Ways—Deposit Msg for B</td>
<td></td>
</tr>
<tr>
<td>&lt;---------------</td>
<td>&lt;---------------</td>
<td>&lt;---------------</td>
<td>&lt;---------------</td>
</tr>
<tr>
<td>OK F16</td>
<td>&lt;---------------</td>
<td>BYE F13</td>
<td>BYE F14</td>
</tr>
<tr>
<td>BYE F13</td>
<td>&lt;---------------</td>
<td>OK F15</td>
<td>&lt;---------------</td>
</tr>
<tr>
<td>&lt;---------------</td>
<td>&lt;---------------</td>
<td>&lt;---------------</td>
<td>&lt;---------------</td>
</tr>
</tbody>
</table>

**Flow Id** | **Comments**
--- | ---
INVITE F1 | INVITE sip:UserB@wcom.com SIP/2.0
A->Proxy | Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Proxy-Authorization:Digest username="UserA",
realm="MCI WorldCom SIP",
nonce="ea9c8e8bf84f1ce4341ae6cbe5a359", opaque="",
uri="sip:UserB@wcom.com", response=<appropriately
calculated hash goes here>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/*Client for A prepares to receive data on port 49170
from the network. */

INVITE F2  INVITE sip:UserB1@somewhere.wcom.com SIP/2.0
Proxy->B1
Via: SIP/2.0/UDP wcom.com:5060; branch=1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying  SIP/2.0 100 Trying
F3 Proxy->A)
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0
180 Ringing SIP/2.0 180 Ringing
F4 Via: SIP/2.0/UDP wcom.com:5060; branch=1
B1→Proxy Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

180 Ringing SIP/2.0 180 Ringing
F5 Via: SIP/2.0/UDP here.com:5060
Proxy→A From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

/* B1 rings for 9 seconds, this duration is a
configurable parameter in the Proxy Server. Proxy
sends Cancel and proceeds down the list of routes,
eventually hitting the voice mail URI for forward no
answer */

CANCEL F6 CANCEL sip:UserB1@wcom.com SIP/2.0
Proxy→B1 Via: SIP/2.0/UDP wcom.com:5060; branch=1
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

200 OK F7 SIP/2.0 200 OK
B1→Proxy Via: SIP/2.0/UDP wcom.com:5060; branch=1
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 CANCEL
Content-Length: 0

INVITE F8 INVITE sip:UserB-dep-fna@vm.wcom.com SIP/2.0
Proxy→VM Via: SIP/2.0/UDP wcom.com:5060;branch=2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o= UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F9 SIP/2.0 200 OK
VM->Proxy Via: SIP/2.0/UDP wcom.com:5060; branch=2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: TheLittleGuyVoiceMail <sip:UserB-dep-fna@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o= UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F10 SIP/2.0 200 OK
Proxy->A Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: TheLittleGuyVoiceMail <sip:UserB-dep-fna@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o= UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F11
A->Proxy
ACK sip:UserB@wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route:<sip: UserB-dep-fna@vm.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=123456
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F12
Proxy->VM
ACK sip:UserB-dep-fna@vm.wcom.com SIP/2.0
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=123456
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B2. VM system starts IVR dialog for message-deposit on no-answer for User B */

/* User A Hangs Up with VM system. Alternatively, the VM system could initiate the BYE*/

BYE F13
A->Proxy
BYE sip:UserB@wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route:<sip: UserB-dep-fna@vm.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=123456
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

BYE F14
Proxy->VM
BYE sip: UserB-dep-fna@vm.wcom.com SIP/2.0
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=123456
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0
4.2.2 Call to known subscriber forwarded on busy

User A attempts to call UserB, who is busy. The call is forwarded to UserB’s mailbox, and the voice mail system plays UserB’s outgoing message for a busy. This flow assumes that "sip:UserB-dep-fb@vm.wcom.com" maps to UserB’s mailbox and the behavior of "deposit message on busy."
INVITE F1
--------------> INVITE F2
(100 Trying) F3
<--------------
486 Busy Here F4
<--------------
ACK F5
<------------>
INVITE F6
-------------->
200 OK F8
<------------>
ACK F9
-------------> ACK F10
-------------->
RTP Established Both Ways—Deposit Msg for B
<---------------
BYE F11
-------------> BYE F12
-------------->
OK F14
<------------>

Flow Id   Comments
INVITE F1   INVITE sip:UserB@wcom.com SIP/2.0
A->Proxy   Via: SIP/2.0/UDP here.com:5060
            From: TheBigGuy <sip:UserA@here.com>
            To: TheLittleGuy <sip:UserB@wcom.com>
            Call-Id: 12345600@here.com
            CSeq: 1 INVITE
            Contact: TheBigGuy <sip:UserA@here.com>
            Proxy-Authorization:Digest username="UserA",
            realm="MCI WorldCom SIP",
            nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque="",
            url="sip:UserB@wcom.com", response=<appropriately
calculated hash goes here>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/*Client for A prepares to receive data on port 49170 from the network. */

INVITE F2     INVITE sip:UserB1@somewhere.wcom.com SIP/2.0
Proxy->B1     Via: SIP/2.0/UDP wcom.com:5060; branch=1
              Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying F3     SIP/2.0 100 Trying
Proxy->A)     Via: SIP/2.0/UDP here.com:5060
              From: TheBigGuy <sip:UserA@here.com>
              To: TheLittleGuy <sip:UserB@wcom.com>
              Call-Id: 12345600@here.com
              CSeq: 1 INVITE
              Content-Length: 0

486 Busy      SIP/2.0 486 Busy Here
Here F4       Via: SIP/2.0/UDP wcom.com:5060;branch=1
B1->Proxy     Via: SIP/2.0/UDP here.com:5060
              From: TheBigGuy <sip:UserA@here.com>
              To: TheLittleGuy <sip:UserB@wcom.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

ACK F5
Proxy->B
ACK sip: UserB@wcom.com SIP/2.0
Via: SIP/2.0/UDP wcom.com:5060; branch=1
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=123456
Call-ID: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

INVITE F6
Proxy->VM
INVITE sip:UserB-dep-fb@vm.wcom.com SIP/2.0
Via: SIP/2.0/UDP wcom.com:5060;branch=2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F7
VM->Proxy
SIP/2.0 200 OK
Via: SIP/2.0/UDP wcom.com:5060; branch=2
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Contact: TheLittleGuyVoiceMail <sip:UserB-dep-fb@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F8 SIP/2.0 200 OK
Proxy->A Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact TheLittleGuyVoiceMail <sip:UserB-dep-fb@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F9 ACK sip:UserB@wcom.com SIP/2.0
A->Proxy Via: SIP/2.0/UDP here.com:5060
Route:<sip:UserB-dep-fb@vm.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F10 ACK sip:UserB-dep-fb@vm.wcom.com SIP/2.0
Proxy->VM Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

/* RTP streams are established between A and B2. VM system starts IVR dialog for message-deposit on busy for UserB */
/* User A Hangs Up with VM system. Alternatively, the VM system could initiate the BYE*/

BYE F11  
A->Proxy
BYE sip:UserB@wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserB-dep-fnb@vm.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

BYE F12  
Proxy->VM
BYE sip: UserB-dep-fnb@vm.wcom.com SIP/2.0
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F13  
VM->Proxy
SIP/2.0 200 OK
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F14  
Proxy->A
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

4.2.3 Direct call to a subscriber’s mailbox

User A calls UserB’s mailbox directly. This flow assumes that "sip:UserB-dep@vm.wcom.com" maps to UserB’s mailbox and the behavior of "generic message deposit"
INVITE F1

(100 Trying) F3

200 OK F5

ACK F6

RTP Both Ways - Deposit Msg for B

BYE F8

200 OK F11

Flow Id Comments

INVITE F1 INVITE sip:UserB-VM@wcom.com SIP/2.0
A->Proxy Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Proxy-Authorization:Digest username="UserA",
realm="MCI WorldCom SIP",
nonce="ea9c8e88df84f1cecc4341ae6cbe5a359", opaque="",
uri="sip:UserB-VM@wcom.com", response=<appropriately
calculated hash goes here>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/*Client for A prepares to receive data on port 49170
from the network. */

INVITE F2
Proxy->B1

INVITE sip:UserB-dep@vm.wcom.com SIP/2.0
Via: SIP/2.0/UDP wcom.com:5060; branch=1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB-VM@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@vm.wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying)

F3
Proxy->A)

SIP/2.0 100 Trying
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

200 OK
F4
VM->Proxy

SIP/2.0 200 OK
Via: SIP/2.0/UDP wcom.com:5060; branch=1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB-VM@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheLittleGuyVoiceMail <sip:UserB-dep@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>
v=0
o=UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F5
Proxy->A
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:UserB-VM@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact TheLittleGuyVoiceMail <sip:UserB-dep@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F6
A->Proxy
Via: SIP/2.0/UDP here.com:5060
Route:<sip:UserB-dep@vm.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F7
Proxy->VM
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
/* RTP streams are established between A and VM. VM system starts IVR dialog for generic message-deposit for UserB */

/* User A Hangs Up with VM system. Alternatively, the VM system could initiate the BYE*/

BYE F8
A->Proxy
BYE sip:UserB-VM@wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route:<sip:UserB-dep@vm.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

BYE F9
Proxy->VM
BYE sip: UserB-dep@vm.wcom.com SIP/2.0
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F10
VM->Proxy
SIP/2.0 200 OK
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F11
Proxy->A
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuyVoiceMail <sip:UserB-VM@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0
4.3 Message Retrieval Scenarios

4.3.1 Call to retrieve messages believed to be from a known subscriber

Some user uses a SIP client on UserA’s desk to call the voice mail system to retrieve messages. The SIP client has authenticated itself to the proxy using credentials assigned to the device. The proxy can make a weak assumption that the caller is the device owner. The URI of "sip:UserA-retrieve@vm.wcom.com" maps to UserA’s mailbox and the behavior of "retrieve messages after prompting for and verifying PIN." The VM System trusts the proxy, and will not accept calls from an untrusted source. The proxy will not allow direct calls to UserA-retrieve@vm.wcom.com. The proxy will forward calls placed to VoiceMail@wcom.com to UserA-retrieve@vm.wcom.com only for calls placed from a client device assigned to UserA.

```
User A                      Proxy                      VM Service
INVITE F1
---------------------> INVITE F2
(100 Trying) F3         -------------------->
<----------------------  200 OK F4
200 OK F5
<----------------------  200 OK F10
                  RTP Both Ways - VM prompts for PIN
                  <-m-m-m-m-m-m-m-m-m-m-m-m-m-m-m-m-m-m-m-m-
BYE F8
---------------------> BYE F9
                  -------------------->
                  200 OK F10
200 OK F11
<----------------------
```
<table>
<thead>
<tr>
<th>Flow Id</th>
<th>Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>INVITE F1</td>
<td>A-&gt;Proxy</td>
</tr>
<tr>
<td>INVITE sip:<a href="mailto:VoiceMail@wcom.com">VoiceMail@wcom.com</a> SIP/2.0</td>
<td>Via: SIP/2.0/UDP here.com:5060</td>
</tr>
<tr>
<td>From: TheBigGuy <a href="">sip:UserA@here.com</a></td>
<td>To: VoiceMail <a href="">sip:VoiceMail@wcom.com</a></td>
</tr>
<tr>
<td>Call-Id: <a href="mailto:12345600@here.com">12345600@here.com</a></td>
<td>CSeq: 1 INVITE</td>
</tr>
<tr>
<td>Contact: TheBigGuy <a href="">sip:UserA@here.com</a></td>
<td>Proxy-Authorization:Digest username=&quot;UserAPhone&quot;, realm=&quot;MCI WorldCom SIP&quot;, nonce=&quot;ea9c8e8df84f1ce4341ae6cbe5a359&quot;, opaque=&quot;&quot;, uri=&quot;sip:<a href="mailto:VoiceMail@wcom.com">VoiceMail@wcom.com</a>&quot;, response=&lt;appropriately calculated hash goes here&gt;</td>
</tr>
<tr>
<td>Content-Type: application/sdp</td>
<td>Content-Length: &lt;appropriate value&gt;</td>
</tr>
</tbody>
</table>

```v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

/*Client for A prepares to receive data on port 49170 from the network. */
```

| INVITE F2 | Proxy->B1 |
| INVITE sip:UserA-retrieve@vm.wcom.com SIP/2.0 | Via: SIP/2.0/UDP wcom.com:5060; branch=1 |
| Via: SIP/2.0/UDP here.com:5060 | Record-Route: <sip:VoiceMail@wcom.com> |
| From: TheBigGuy <sip:UserA@here.com> | To: VoiceMail <sip:VoiceMail@wcom.com> |
| Call-Id: 12345600@here.com | CSeq: 1 INVITE |
| Contact: TheBigGuy <sip:UserA@here.com> | Content-Type: application/sdp |
| Content-Length: <appropriate value> |

```v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```
(100 Trying SIP/2.0 100 Trying
F3
Proxy->A) From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

200 OK F4 SIP/2.0 200 OK
VM->Proxy Via: SIP/2.0/UDP wcom.com:5060; branch=1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:VoiceMail@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: VoiceMailSystem <sip:UserA-
retrieve@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F5 SIP/2.0 200 OK
Proxy->A Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:VoiceMail@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact VoiceMailSystem <sip: UserA-
retrieve@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000
ACK F6  ACK sip:VoiceMail@wcom.com SIP/2.0  
A->Proxy  Via: SIP/2.0/UDP here.com:5060  
Route:<sip:UserA-retrieve@vm.wcom.com>  
From: TheBigGuy <sip:UserA@here.com>  
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678  
Call-Id: 12345600@here.com  
CSeq: 1 ACK  
Content-Length: 0  

ACK F7  ACK sip:UserA-retrieve@vm.wcom.com SIP/2.0  
Proxy->VM  Via: SIP/2.0/UDP wcom.com:5060  
Via: SIP/2.0/UDP here.com:5060  
From: TheBigGuy <sip:UserA@here.com>  
To: VoiceMail <sip:VoiceMail@wcom.com>; tag=3145678  
Call-Id: 12345600@here.com  
CSeq: 1 ACK  
Content-Length: 0  

/* RTP streams are established between A and VM. VM  
determines that the call is likely from UserA, and  
starts a message retrieval session, prompting for  
PIN*/  
/* User A Hangs Up with VM system. Alternatively, the  
VM system could initiate the BYE*/  

BYE F8  BYE sip: VoiceMail@wcom.com SIP/2.0  
A->Proxy  Via: SIP/2.0/UDP here.com:5060  
Route:<sip:UserA-retrieve@vm.wcom.com>  
From: TheBigGuy <sip:UserA@here.com>  
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678  
Call-Id: 12345600@here.com  
CSeq: 2 BYE  
Content-Length: 0  

BYE F9  BYE sip: UserA-retrieve@vm.wcom.com SIP/2.0  
Proxy->VM  Via: SIP/2.0/UDP wcom.com:5060  
Via: SIP/2.0/UDP here.com:5060  
From: TheBigGuy <sip:UserA@here.com>  
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678  
Call-Id: 12345600@here.com  
CSeq: 2 BYE  
Content-Length: 0  

200 OK F10  SIP/2.0 200 OK  
VM->Proxy  Via: SIP/2.0/UDP wcom.com:5060  
Via: SIP/2.0/UDP here.com:5060  
From: TheBigGuy <sip:UserA@here.com>
UserA to call the voice mail system to retrieve messages. Assumptions: The caller is authenticated using UserA’s credentials. "sip:UserA-retrieve-auth@vm.wcom.com" maps to UserA’s mailbox and the behavior of "retrieve messages." The voice mail service trusts the proxy not to forward any calls to that URI unless the call is authenticated to be from UserA.

Given these assumptions, The VM service may choose not require a PIN for calls to this URI.
INVITE F1
  ------------->
(100 Trying) F3  ------------->
<------------------
  200 OK F5  <------------------
<------------------
ACK F6  ------------->
  <------------------
RTP Both Ways - Deposit Msg for B
<------------------
BYE F8  ------------->
  <------------------
200 OK F10  <------------------
<------------------
200 OK F11

Flow Id  Comments

INVITE F1  INVITE sip:VoiceMail@wcom.com SIP/2.0
  A->Proxy  Via: SIP/2.0/UDP here.com:5060
           From: TheBigGuy <sip:UserA@here.com>
           To: VoiceMail <sip:VoiceMail@wcom.com>
           Call-Id: 12345600@here.com
           CSeq: 1 INVITE
           Contact: TheBigGuy <sip:UserA@here.com>
           Proxy-Authorization:Digest username="UserA",
           realm="MCI WorldCom SIP",
           nonce="ea9c8e88df84f1cec4341ae6cbe5a359", opaque="",
           uri="sip:VoiceMail@wcom.com", response=<appropriately
           calculated hash goes here>
           Content-Type: application/sdp
           Content-Length: <appropriate value>

           v=0
           o= UserA 2890844526 2890844526 IN IP4 client.here.com
           s=Session SDP
/*Client for A prepares to receive data on port 49170 from the network. */

INVITE F2
Proxy->B1
Via: SIP/2.0/UDP wcom.com:5060; branch=1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:VoiceMail@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: TheBigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserA 2890844526 2890844526 IN IP4 client.here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49170 RTP/AVP 0
a=rtpmap:0 PCMU/8000

(100 Trying)
F3
Proxy->A)
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Content-Length: 0

200 OK
F4
Proxy->VM
Via: SIP/2.0/UDP wcom.com:5060; branch=1
Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:VoiceMail@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact: VoiceMailSystem <sip:UserA-retrieve-auth@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>
v=0
o=UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

200 OK F5        SIP/2.0 200 OK
Proxy->A        Via: SIP/2.0/UDP here.com:5060
Record-Route: <sip:VoiceMail@wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 INVITE
Contact VoiceMailSystem <sip:UserA-retrieve-auth@vm.wcom.com>
Content-Type: application/sdp
Content-Length: <appropriate value>

v=0
o=UserB 2890844527 2890844527 IN IP4 vm.wcom.com
s=Session SDP
c=IN IP4 110.111.112.114
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

ACK F6        ACK sip:VoiceMail@wcom.com SIP/2.0
A->Proxy      Via: SIP/2.0/UDP here.com:5060
Route: <sip:UserA-retrieve-auth@vm.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0

ACK F7        ACK sip:UserA-retrieve-auth@vm.wcom.com SIP/2.0
Proxy->VM      Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>; tag=3145678
Call-Id: 12345600@here.com
CSeq: 1 ACK
Content-Length: 0
/* RTP streams are established between A and VM. VM determines that the call is likely from UserA, and starts a message retrieval session. Since the proxy has already authenticated the identity of UserA, the VM does not need to prompt for PIN. */

/* User A Hangs Up with VM system. Alternatively, the VM system could initiate the BYE*/

BYE F8
A->Proxy
BYE sip:VoiceMail@wcom.com SIP/2.0
Via: SIP/2.0/UDP here.com:5060
Route:<sip:UserA-retrieve-auth@vm.wcom.com>
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

BYE F9
Proxy->VM
BYE sip: UserA-retrieve-auth@vm.wcom.com SIP/2.0
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F10
VM->Proxy
SIP/2.0 200 OK
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0

200 OK F11
Proxy->A
SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: VoiceMail <sip:VoiceMail@wcom.com>;tag=3145678
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0
5. Security Considerations

This document discusses a usage of SIP/2.0 as defined by RFC 2543[1]. It introduces no additions, modifications, or restrictions to the protocol defined therein. Any implementation of the concepts in this document is subject to the issues discussed there.

6. Acknowledgments

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