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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.

URI Identity	Example Scheme 1
	Example Scheme 2
	Example Scheme 3
Deposit with standard greeting	sip:sub-rjs-deposit@vm.wcom.com sip:677283@vm.wcom.com sip:rjs@vm.wcom.com;mode=deposit
Deposit with on phone greeting	sip:sub-rjs-deposit-busy.vm.wcom.com sip:677372@vm.wcom.com sip:rjs@vm.wcom.com;mode=3991243
Deposit with special greeting	sip:sub-rjs-deposit-sg@vm.wcom.com sip:677384@vm.wcom.com sip:rjs@vm.wcom.com;mode=sg
Retrieve - SIP authentication	sip:sub-rjs-retrieve@vm.wcom.com sip:677405@vm.wcom.com sip:rjs@vm.wcom.com;mode=retrieve
Retrieve - prompt for PIN in-band	sip:sub-rjs-retrieve-inpin.vm.wcom.com sip:677415@vm.wcom.com sip:rjs@vm.wcom.com;mode=inpin

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

URI Identity	Example Scheme 1
	Example Scheme 2
	Example Scheme 3
Deposit - identify target mailbox by To:	sip:deposit@vm.wcom.com sip:670001@vm.wcom.com sip:deposit@vm.wcom.com

Retrieve -	sip:retrieve@vm.wcom.com
identify target	sip:670002@vm.wcom.com
mailbox by SIP	sip:retrieve@vm.wcom.com
authentication	
Deposit - prompt	sip:deposit-in@vm.wcom.com
for target	sip:670003@vm.wcom.com
mailbox in-band	sip:deposit@vm.wcom.com;mode=inband
Retrieve - prompt	sip:retrieve-in@vm.wcom.com
for target	sip:670004@vm.wcom.com
mailbox and PIN	sip:retrieve@vm.wcom.com;mode=inband
in-band	

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)

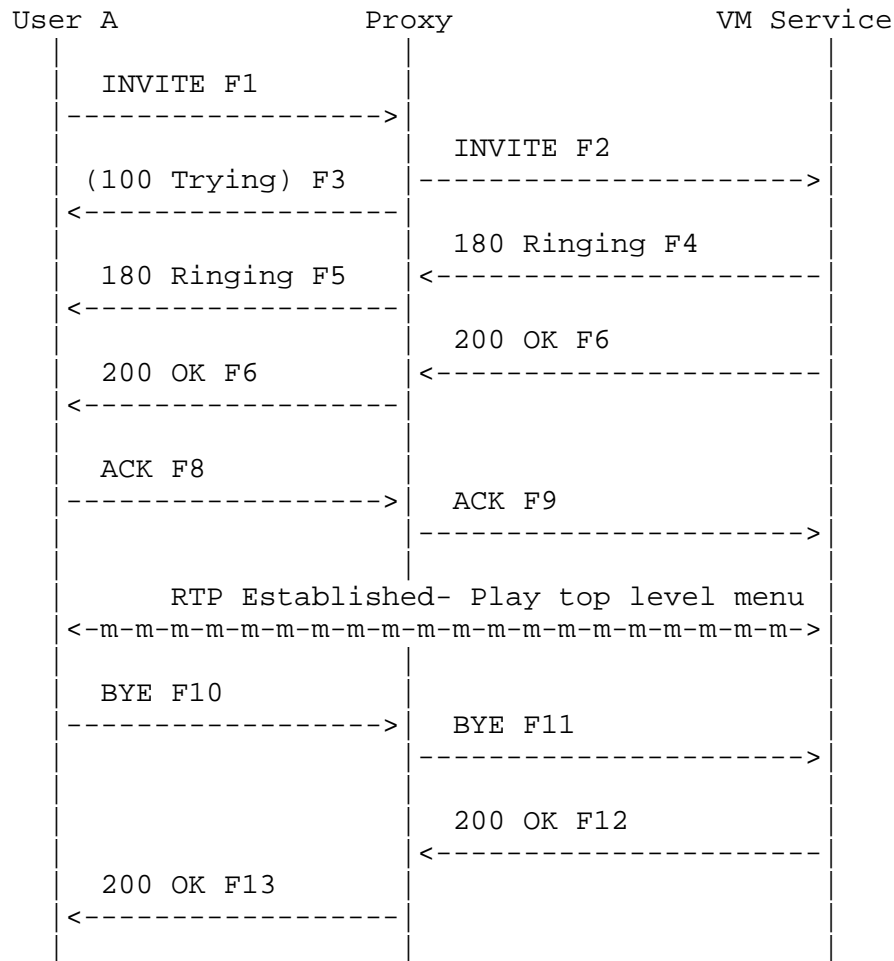
4. Voice Mail Call Flow Examples

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 A->Proxy	INVITE sip:VoiceMail@wcom.com SIP/2.0 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      INVITE sip:top@vm.wcom.com SIP/2.0
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>
                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             Via: SIP/2.0/UDP here.com:5060
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

```

```

180 Ringing    SIP/2.0 180 Ringing
F4             Via: SIP/2.0/UDP wcom.com:5060; branch=1
VM->Proxy      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 INVITE
                Content-Length: 0

```

180 Ringing
F5
Proxy->A
SIP/2.0 180 Ringing
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 INVITE
Content-Length: 0

200 OK F6
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: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
Proxy->A
SIP/2.0 200 OK
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

ACK sip:VoiceMail@wcom.com SIP/2.0
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

ACK sip:top@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: 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

BYE sip:VoiceMail@wcom.com SIP/2.0
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

BYE sip: top@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 F12
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

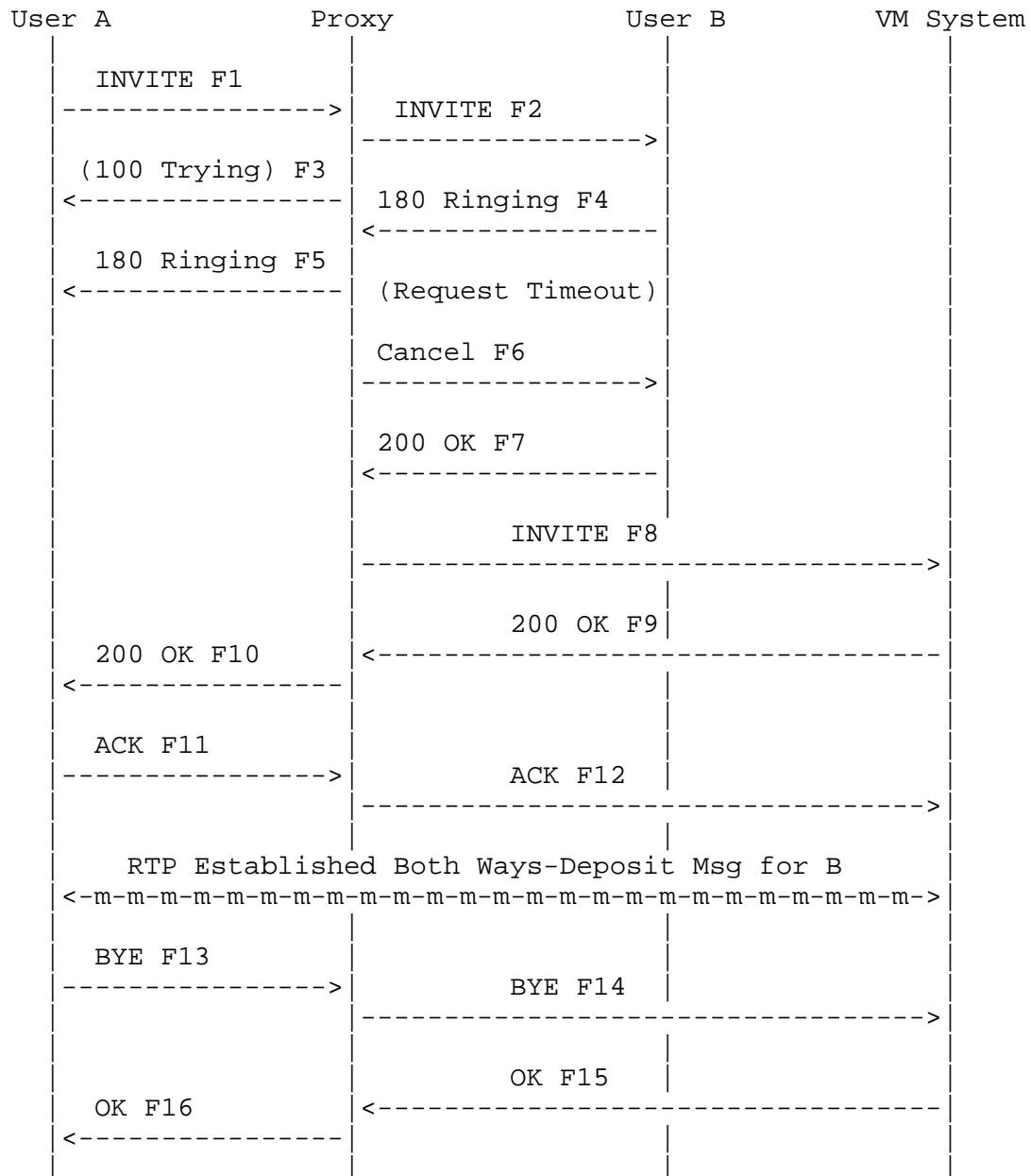
```
                CSeq: 2 BYE
                Content-Length: 0

200 OK F13      SIP/2.0 200 OK
Proxy->A        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
```

4.2 Message Deposit Scenarios

4.2.1 Call to known subscriber forwarded on no answer

User A attempts to call UserB, who does not answer. The call is forwarded to UserB's mailbox, and the voice mail system plays UserB'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.



Flow Id

Comments

INVITE F1
 A->Proxy
 INVITE sip:UserB@wcom.com SIP/2.0
 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="",
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
Proxy->B1

```
INVITE sip:UserB1@somewhere.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@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
Proxy->A)

```
SIP/2.0 100 Trying
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
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=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
Proxy->A

SIP/2.0 200 OK
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 UserB */
```

```
/* 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
```

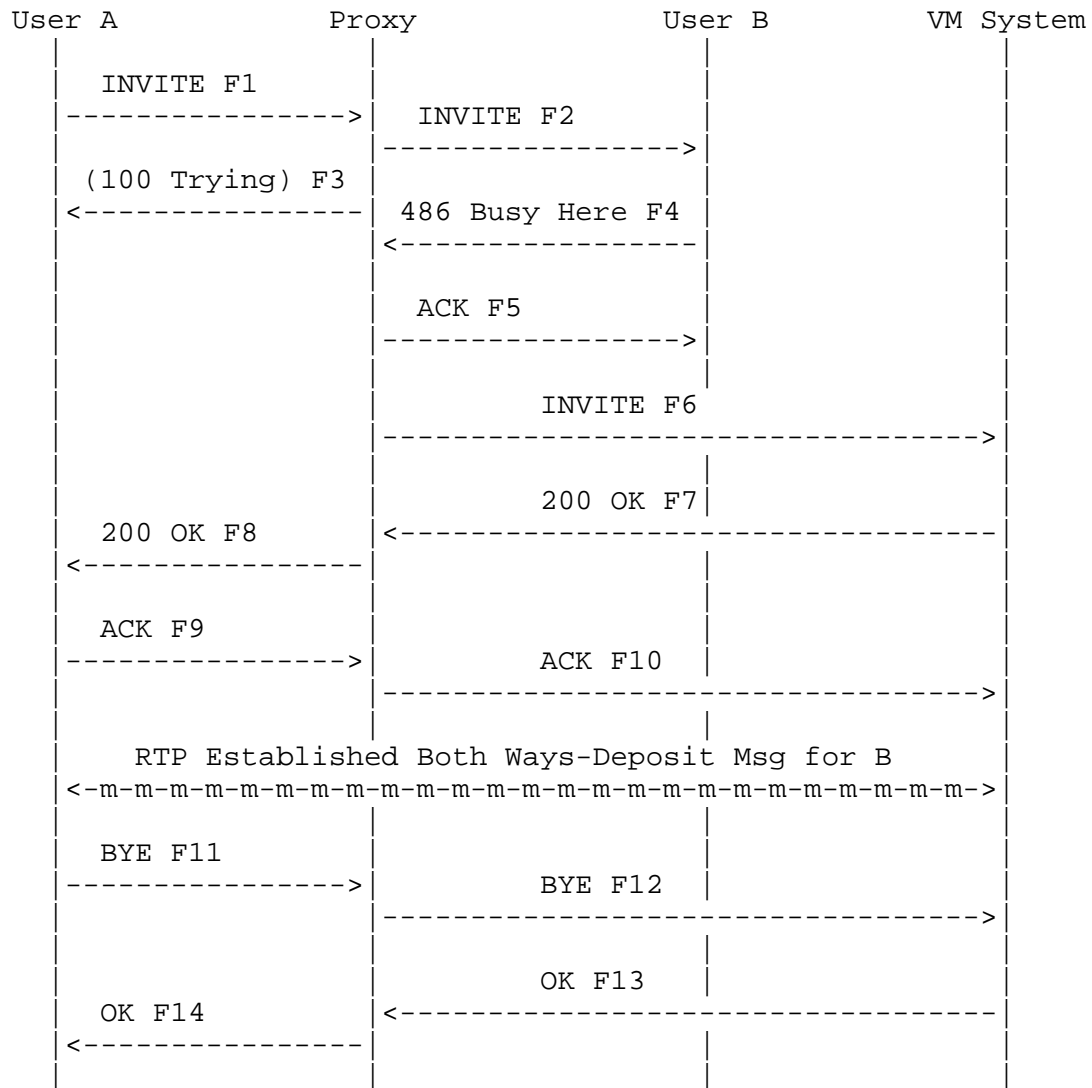
```
BYE F14
Proxy->VM
BYE sip: UserB-dep-fna@vm.wcom.com SIP/2.0
Via: SIP/2.0/UDP wcom.com:5060
Via: SIP/2.0/UDP here.com:5060
From: TheBigGuy <sip:UserA@here.com>
To: TheLittleGuy <sip:UserB@wcom.com>;tag=123456
Call-Id: 12345600@here.com
CSeq: 2 BYE
Content-Length: 0
```

```
200 OK F15      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>
                To: TheLittleGuy <sip:UserB@wcom.com>;tag=123456
                Call-Id: 12345600@here.com
                CSeq: 2 BYE
                Content-Length: 0

200 OK F16      SIP/2.0 200 OK
Proxy->A        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."



Flow Id

Comments

INVITE F1
A->Proxy

INVITE sip:UserB@wcom.com SIP/2.0
 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="",
 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 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
 Proxy->A)
 SIP/2.0 100 Trying
 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 Here F4
 B1->Proxy
 SIP/2.0 486 Busy Here
 Via: SIP/2.0/UDP wcom.com:5060;branch=1
 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: UserBl@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
Proxy->A
SIP/2.0 200 OK
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
A->Proxy
ACK sip:UserB@wcom.com SIP/2.0
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
Proxy->VM
ACK sip:UserB-dep-fb@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: 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 BYE sip:UserB@wcom.com SIP/2.0
A->Proxy 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

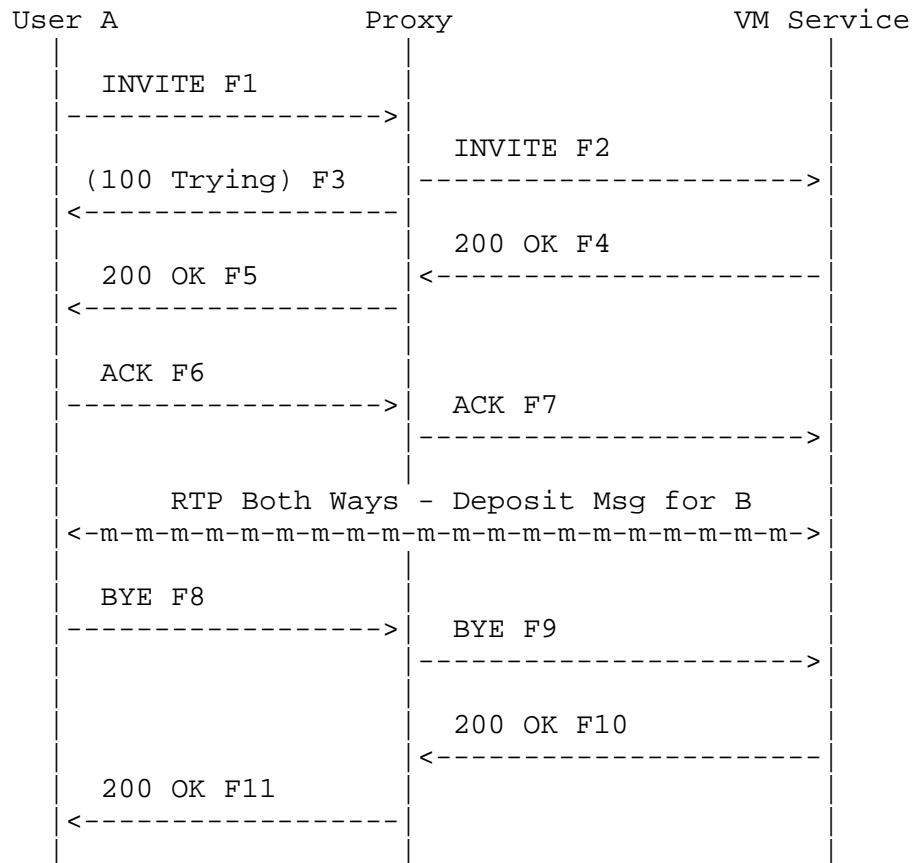
BYE F12 BYE sip: UserB-dep-fnb@vm.wcom.com SIP/2.0
Proxy->VM Via: SIP/2.0/UDP wcom.com:5060
 Via: SIP/2.0/UDP here.com:5060
 From: TheBigGuy <sip:UserA@here.com>
 To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
 Call-Id: 12345600@here.com
 CSeq: 2 BYE
 Content-Length: 0

200 OK F13 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>
 To: TheLittleGuy <sip:UserB@wcom.com>;tag=3145678
 Call-Id: 12345600@here.com
 CSeq: 2 BYE
 Content-Length: 0

200 OK F14 SIP/2.0 200 OK
Proxy->A 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"



Flow Id	Comments
INVITE F1 A->Proxy	INVITE sip:UserB-VM@vm.wcom.com SIP/2.0 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="ea9c8e88df84f1cec4341ae6cbe5a359", 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      INVITE sip:UserB-dep@vm.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-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   SIP/2.0 100 Trying
F3            Via: SIP/2.0/UDP here.com:5060
Proxy->A)      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      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: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 SIP/2.0 200 OK
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 ACK 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: 1 ACK
Content-Length: 0

ACK F7
Proxy->VM ACK 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: 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

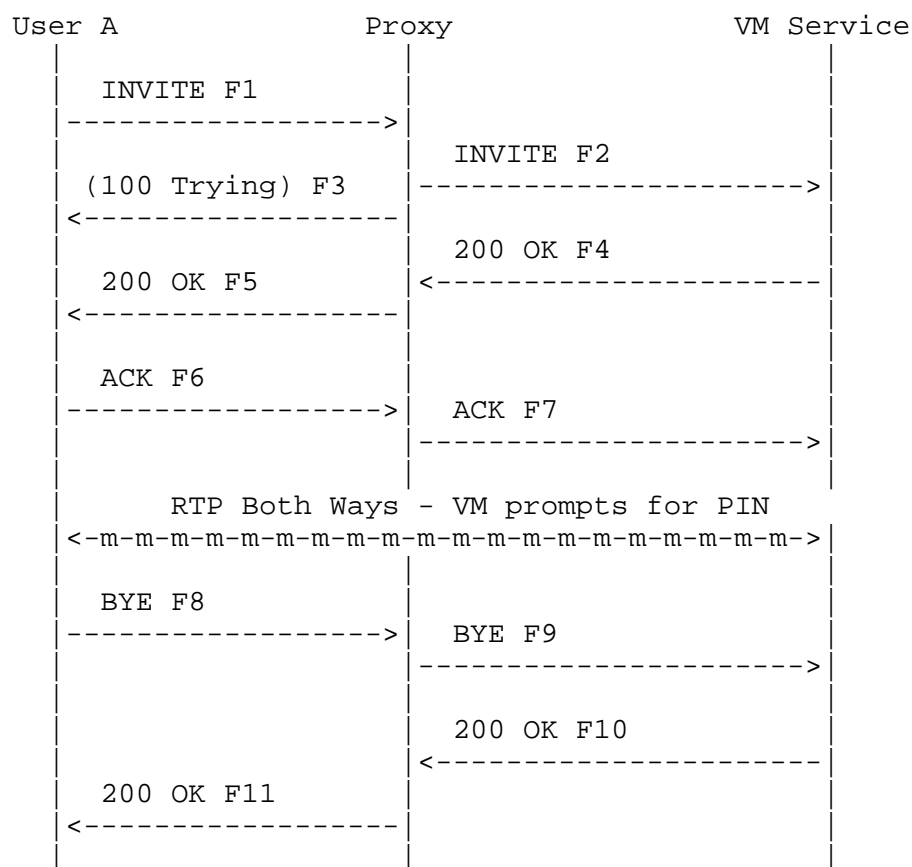
200 OK F11
Proxy->A

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.



Flow Id	Comments
INVITE F1 A->Proxy	<pre> INVITE sip:VoiceMail@wcom.com SIP/2.0 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="UserAPhone", 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 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. */ </pre>
INVITE F2 Proxy->B1	<pre> 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 </pre>

(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

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: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 Proxy->A) SIP/2.0 200 OK
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
A->Proxy

ACK sip:VoiceMail@wcom.com SIP/2.0
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
Proxy->VM

ACK sip:UserA-retrieve@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: 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
A->Proxy

BYE sip: VoiceMail@wcom.com SIP/2.0
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
Proxy->VM

BYE sip: UserA-retrieve@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      SIP/2.0 200 OK
Proxy->A       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
```

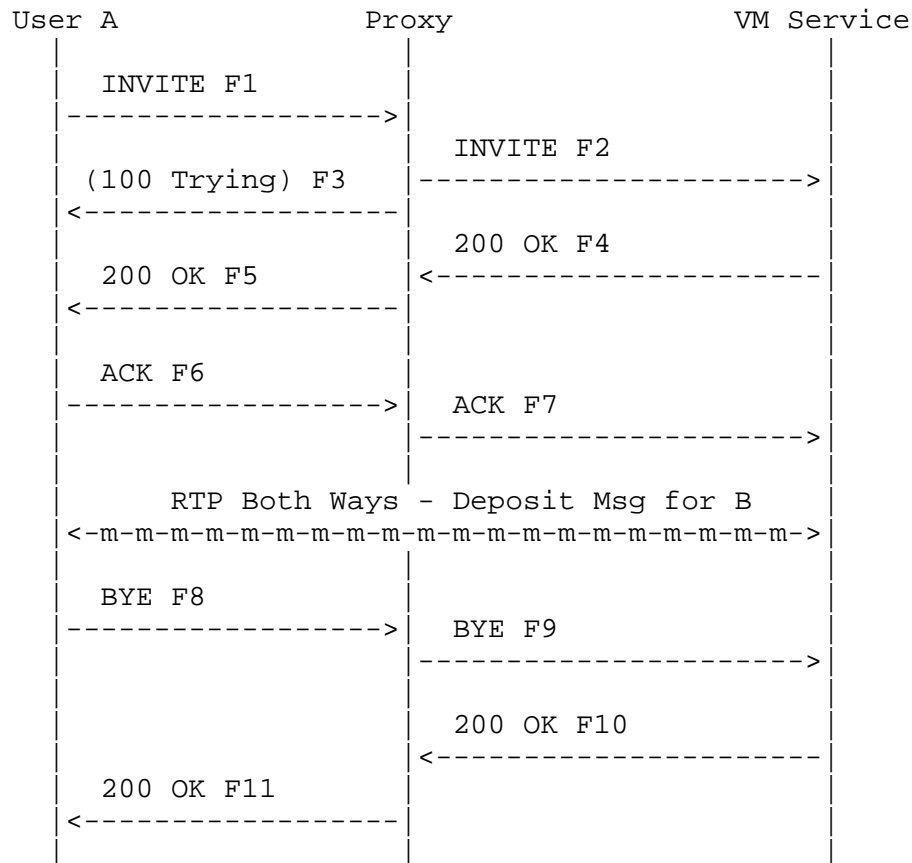
4.3.2 Call to retrieve messages from an authenticated subscriber

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.



Flow Id	Comments
INVITE F1 A->Proxy	INVITE sip:VoiceMail@wcom.com SIP/2.0 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

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:UserA-retrieve-auth@vm.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: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 Via: SIP/2.0/UDP here.com:5060
 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-
 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
Proxy->A SIP/2.0 200 OK
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
A->Proxy ACK 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: 1 ACK
Content-Length: 0

ACK F7
Proxy->VM ACK 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: 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

The authors would like to thank Chris Cunningham, Steve Donovan, Alan Johnston, Henry Sinnreich, Kevin Summers, John Truetken, and Dean Willis for their discussion of and contribution to this work.

References

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

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