

# VOCAL

Vovida Open Communication Application Library

System Administration Guide

Software Version 1.4.0

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**Guide Versions** The following table matches the software versions with the guide versions:

<b>Software Version</b>	<b>Guide Version</b>	<b>Date</b>	<b>Comments</b>
1.0.0			Internal Trials Only
1.1.0			Internal Trials Only
1.2.0	1.2	March 26, 2001	Open Release to Public
1.2.0	1.2 A	April 11, 2001	Copy edit errors corrected.
1.3.0	1.3	December 21, 2001	Support new open release to public
1.4.0	1.4	June 23, 2002	Changes to Call Flows and other minor editing.

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**Version** This manual is written to support VOCAL Version 1.4.0.

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**Support** The primary location for support, information and assistance for the VOCAL system is <http://www.vovida.org/>. This site contains other documentation, training materials, development tools, development resources and informational mailing lists.

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# Preface

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<b>Introduction</b>	This chapter is a general introduction to the System Administration manual, and provides information about the intentions and organization of the manual. It also provides information about additional resources available from <a href="http://www.vovida.org">http://www.vovida.org</a> .
<b>Objectives</b>	This guide provides Information about adding users and assigning features. Information for installing and provisioning a VOCAL system is provided in the <a href="#">Installation Guide</a> .
<b>Who Should Use This Book?</b>	This manual is intended for crafts person and system administrators who are familiar with either the Linux or Sun Solaris operating systems; and should also be familiar with Session Initiation Protocol (SIP) and the general concepts and principles of Voice over IP (VoIP) telephony networks.
<b>How is This Guide Organized?</b>	This guide is organized as follows:


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<b><i>Chapter</i></b>	<b><i>Title</i></b>	<b><i>Description</i></b>
<b><i>Chapter 1</i></b>	Setting Up Users	Describes how to add, view, edit and delete users from the VOCAL system.
<b><i>Chapter 2</i></b>	Network Management	Describes system monitoring and SNMP functions.
<b><i>Appendix A</i></b>	Features	Describes features supported by the VOCAL system.
<b><i>Appendix B</i></b>	Supported SIP Messages	Describes supported SIP request and response messages.

<i>Chapter</i>	<i>Title</i>	<i>Description</i>
<b>Appendix C</b>	Call Flows	Provides illustrations and descriptions of call flows for various call scenarios.

## Documentation Conventions

The following is a list of conventions used in this guide:

<i>Convention</i>	<i>Description</i>
<b>bold text</b>	Names of elements found on the GUI screen, including buttons, and selectable entities such as, servers and server groups.
< >	Text that appears between angle brackets describes variables such as, <group name>.
<code>courier font</code>	System responses and prompts either from the CLI or GUI.
<b>bold courier font</b>	Indicates information that you must enter.
■ <b>Note</b>	Highlights points of additional interest for the user.
 <b>Caution</b>	Be careful, this symbol highlights a potential for equipment damage or loss of data.

## Additional resources

### Publications

An [Installation Guide](#), which includes a system overview, installation instructions and information about provisioning the servers is also available from <http://www.vovida.org>.

### On Line Resources

Vovida.org (<http://www.vovida.org>) is a community web site dedicated to providing a forum for open source software used in datacom and telecom environment. This site was created to provide an environment where open source communications information and software can be easily located, accessed, retrieved and shared.

# Table of Contents

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## Preface

### Chapter 1.

#### Setting Up Users

Working With The GUI Environment .....	1-2
Adding, Viewing, Editing, and Deleting Users .....	1-9

### Chapter 2.

#### Network Management

SNMP Support .....	2-2
--------------------	-----

### Appendix A.

#### Features

Features .....	A-2
Core System Features .....	A-3
Set-Based Features .....	A-5

### Appendix B.

#### Supported SIP Messages

SIP Request Messages .....	B-2
SIP Response Messages .....	B-3

### Appendix C.

#### Call Flows

SIP Phone: Registration .....	C-3
SIP IP Phone to SIP IP Phone: Call Setup and Disconnect .....	C-8
SIP IP Phone to Analog Phone via Gateway .....	C-15
SIP Phone to Phone via Gateway: Called Party is Busy .....	C-24
SIP IP Phone to SIP IP Phone: Forward All Calls .....	C-32
Phone to SIP Phone via Gateway: Call Screening .....	C-41
SIP Phone to PSTN: Call Blocking .....	C-47
SIP IP Phone to SIP IP Phone: Call Return .....	C-52
User Agent to User Agent: Call Waiting .....	C-70
SIP IP Phone to SIP IP Phone: Forward to Voice Mail .....	C-85
User Agent to User Agent: Consulted Transfer .....	C-96
User Agent to User Agent: Blind Transfer .....	C-114
JTAPI .....	C-131
Ad Hoc Conference Call Between User Agents .....	C-149

## Table of Contents *(continued)*

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# Setting Up Users

This chapter describes how to add users to the system and how to maintain the user data base.

Topic	See Page
<b>Working With The GUI Environment.....</b>	<b>1-2</b>
Logging In.....	1-3
Overview of the User Configuration Screen.....	1-5
<b>Adding, Viewing, Editing, and Deleting Users.....</b>	<b>1-9</b>
Adding New Users .....	1-10
Adding Users: Administrator's Edit User Screen .....	1-11
Viewing Users: Individually.....	1-16
Viewing Users: Data Fields Descriptions .....	1-19
Viewing Users: All Users .....	1-24
Finding Users .....	1-26
Deleting Users .....	1-27
Editing Users: Administrator Controlled .....	1-28
Editing User: Show Alias .....	1-29
Editing User Features: User Controlled .....	1-30
<b>Editing User Feature: Edit User Screen .....</b>	<b>1-31</b>

## Working With The GUI Environment

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### Overview

This section describes:

- the login screen and how to log into the VOCAL system
- the user configuration screen and available the buttons, options boxes, and data fields.

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### Before you begin

The machine that was used to install the VOCAL system is known as the provisioning host. You can access the Provisioning Server from the provisioning host, or from any other PC that is connected to the network where the VOCAL system resides.

Before you can work with the Provisioning GUI, you must have the following:

- A web browser loaded on your machine. The browser can be any type that takes a Java Run-time Environment (JRE) plug-in version 1.3.1\_01.
- Access to the Internet to download the Java plug-in. All networking requirements are covered in the VOCAL Installation Guide.

You must also know the host name of your provisioning host and the system name that was entered during the software installation. This information is used to access the web page that contains the links to the GUI system utilities. This web address can be expressed as:

`http://<provisioning host name>/vocal/index.html`.

#### ■ Note

An example of this web address could be:  
`http://local_host/vocal/index.html`

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### Accessing the GUI

To access the Provisioning GUI, go to:

`http://<provisioning host name>/vocal/index.html`

and select **Provision System**.

The first Provisioning screen calls a Java plug-in. The plug-in is not shipped with the software, it must be downloaded from the Internet. Normally, this plug-in would have been downloaded to the provisioning host during the software installation process.

However, if you are accessing the system from another machine besides the provisioning host, a download prompt appears when the screen is first loaded. Accept the download file and load the plug-in onto your machine. Once the plug-in has been loaded the Provisioning Login screen appears as shown in [Figure 1-1 on page 1-3](#).

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## Logging In

**Introduction** The Provisioning Login screen provides access for Administrators to work with the users, and for Technicians to work with the servers.

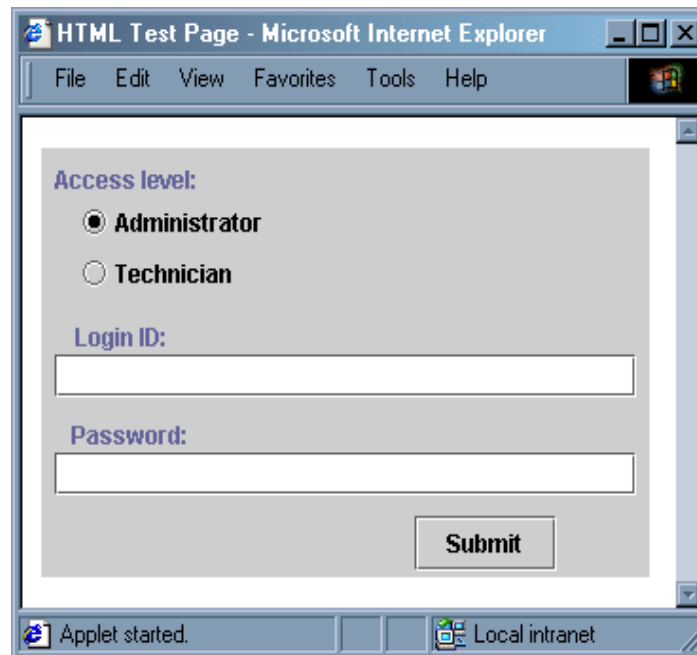
**Definition** The login screen is a java-enabled graphical user interface (GUI) that runs in a web browser. The browser can be any type that supports JRE 1.3.1\_01.

**Procedure** **To log in, follow these steps:**

**Table 1-1. Procedure—Logging In to the User Configuration**

Step	Description
1	Select <b>Administrator</b> . As shown in the figure below.
2	Type your user ID and password. The default user ID is vovida, password: vovida.
3	Click <b>Login</b> , or press <b>Enter</b> .

**Login Screen** Figure 1-1 shows the Login Screen.



**Figure 1-1. Login Screen**

**Items and Fields**

Table 1-2 describes the items found on the Login Screen.

**Table 1-2. Login Screen: Item and Field Description**

<i>Item</i>	<i>Description</i>
Access Level	<p><b>Administrator</b> As an Administrator, you can add, view, edit or delete user entries. In addition, you can setup feature subscriptions for users.</p> <p><b>Technician</b> As a Technician, you can edit the VOCAL server provisioning. For more information, see the VOCAL Installation Guide.</p>
Login ID	The default is <i>vovida</i> .
Password	The default is <i>vovida</i> .

**Password Administration**

There is a separate user interface for changing passwords and adding or removing accounts for administrators and technicians.

It is not setup as an applet, but you can run it stand alone. It is included in the psClient.jar. The main class is vocal.pw.AdminAcctManager.

**To maintain passwords, follow these instructions:**

**Table 1-3. Procedure for Maintaining Passwords**

<i>Step</i>	<i>Description</i>
<b>1</b>	<p>To run this user interface, type the following:</p> <pre>java -classpath /path/to/psClient.jar:path/to/xerces.jar vocal.pw.AdminAcctManager pServer_host pServer_port</pre> <p>A screen appears with a list of all the administrative accounts.</p>
<b>2</b>	<p>Use the popup menu to maintain the accounts.</p> <p><b>■ Note</b> It is possible to have an account that only accesses the administrator GUI, or only the technician GUI or both together.</p>
<b>3</b>	<p>To change the default password for a user, select the account in the GUI and click the <b>Change Password</b> button.</p>

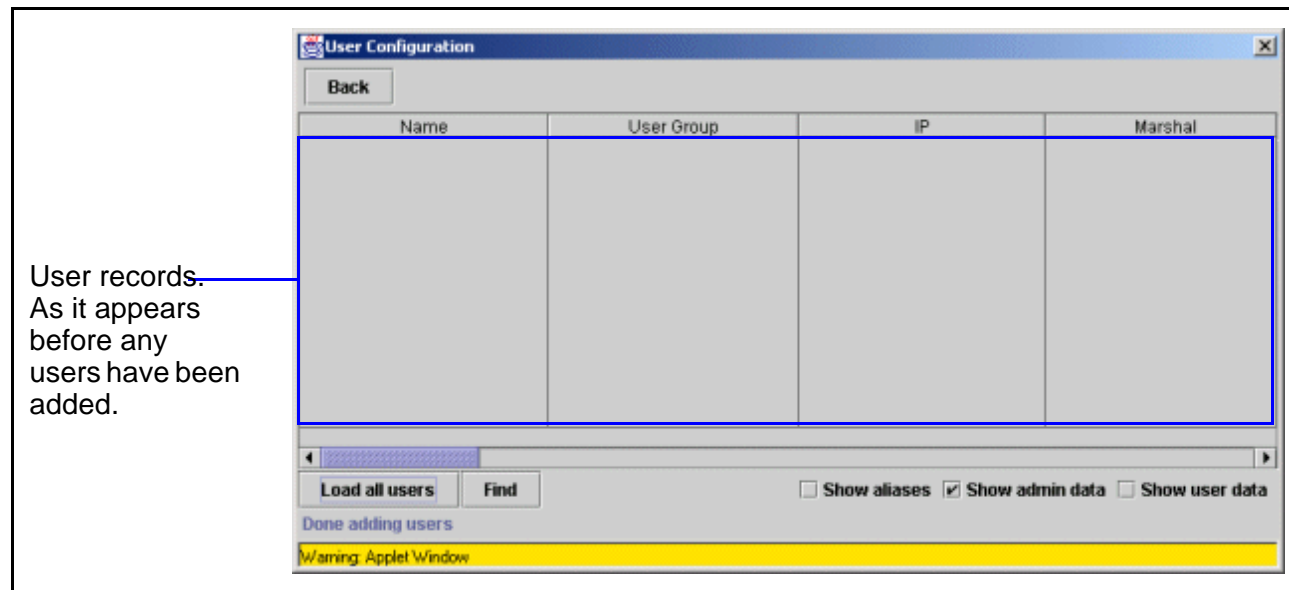
## Overview of the User Configuration Screen

### Introduction

This section describes the buttons, option boxes, and data fields on the User Configuration Screen.

### Screen Capture

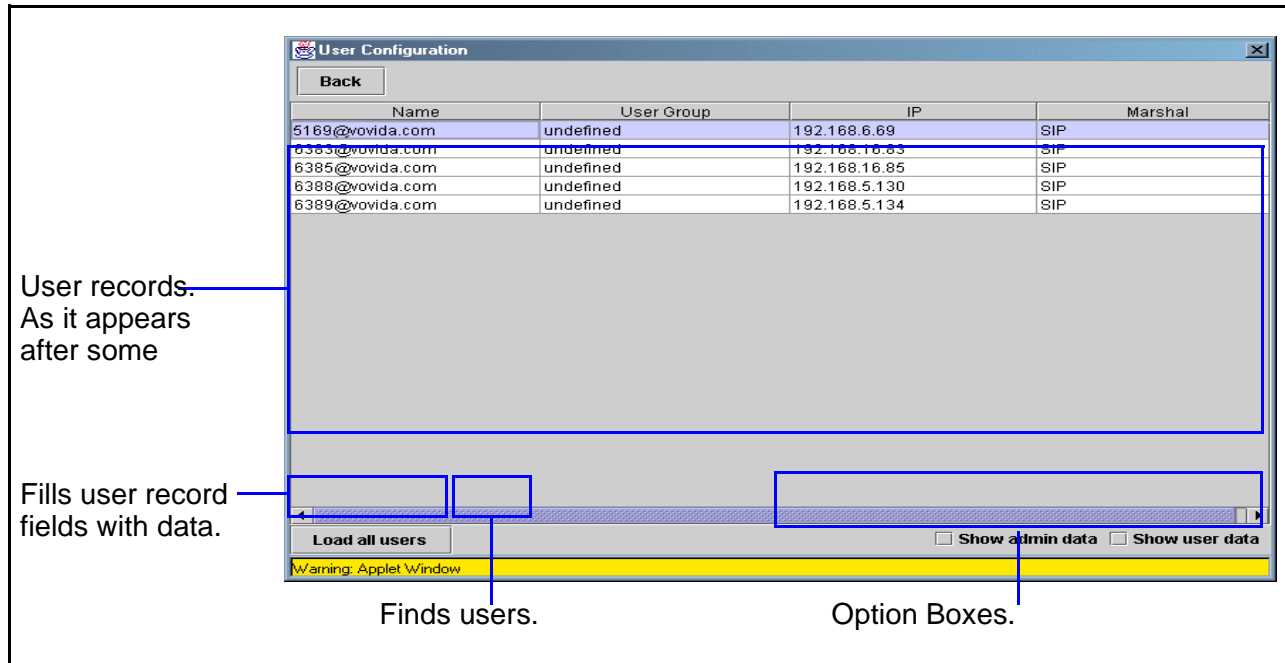
Figures 1-2 and 1-3 show the User Configuration screen as it appears when you login. Figure 1-2 shows what the screen looks like when you login for the first time and the user records are blank.



**Figure 1-2. User Configuration Screen: Before Data Entry**

**After Data Entry**

Figure 1-3 shows what the screen looks like after some users have been added. For more information about adding users, see [“Adding New Users” on page 1-10.](#)



**Figure 1-3. User Configuration Screen: After Data Entry**

**Buttons**

Table 1-4 describes buttons on the User Configuration screen.

**Table 1-4. User Configuration Screen: Buttons**

Button	Description
Back	Lets you return to the login screen.
Load all users	Lets you load information for all users. Clicking this button fills the user records fields with data. See <a href="#">Figure 1-8 on page 1-25</a> for more information.
Find	Lets you find users. See <a href="#">Figure 1-9 on page 1-26</a> for more information.

## Option Boxes

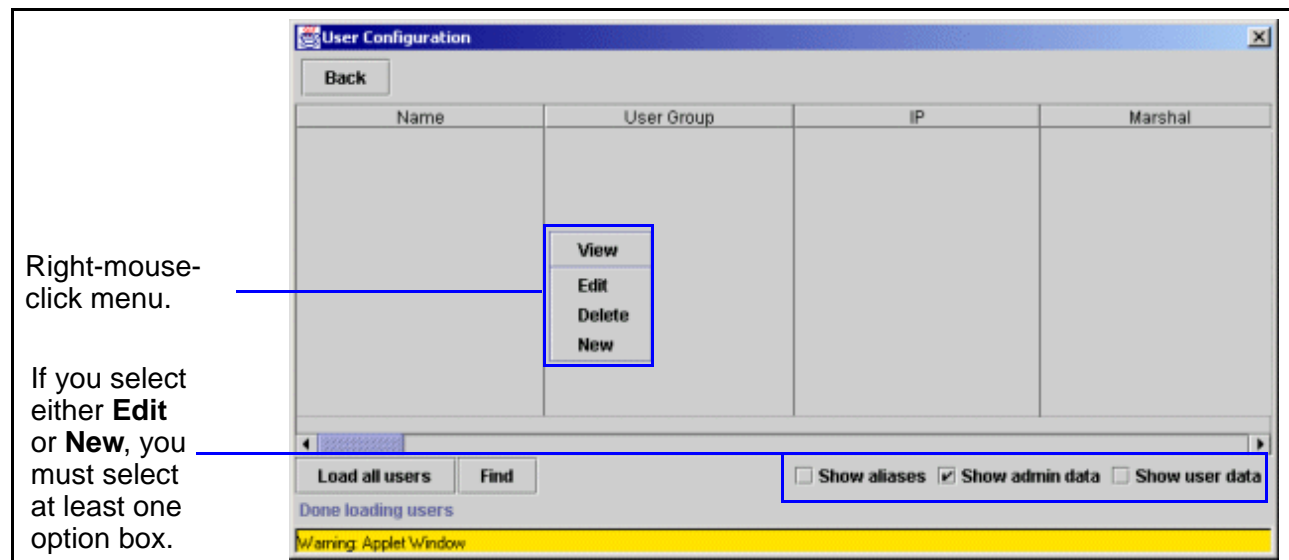
The option boxes filter the fields displayed on the User Configuration screen. If none of the boxes is selected, only the Name, User Group, IP and Marshal fields appear. If all of the boxes are selected, then all of the fields appear on the User Configuration screen. For a complete description of the data fields refer to the [“Viewing Users: Data Fields Descriptions”](#) section on page 1-19. Table 1-5 describes the option boxes.

**Table 1-5. User Configuration Screen: Option Boxes**

Option	Description
Show aliases	Displays the users that have aliases. Aliases are displayed in italics. For more information, see <a href="#">“Editing User: Show Alias”</a> on page 1-29.
Show admin data	Displays information and fields configured by the administrator. For more information, see <a href="#">“Adding Users: Administrator’s Edit User Screen”</a> on page 1-11.
Show user data	Displays information and fields configured by the user. For more information, see <a href="#">“Editing User Feature: Edit User Screen”</a> on page 1-31.

## Right-Mouse-Click Menu

The User Configuration screen has a hidden menu that you can bring up by right-mouse clicking the screen over the space reserved for the user records. Figure 1-4 shows this menu.



**Figure 1-4. User Configuration Screen: Right-Mouse-Click Menu**

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**Right-Mouse-Click  
Menu Options**

Table 1-6 shows the options available from the right-mouse-click menu.

**Table 1-6. User Configuration Screen: Right-Mouse-Click Menu Options**

<b>Option</b>	<b>Description</b>
<b>View</b>	View lets you view data field information in tabular format for one or more users. For more information, see <a href="#">Figure 1-6 on page 1-17</a> .
<b>Edit</b>	Edit lets you edit information for the user. Selecting edit will open the Edit user screen. For more information, see <a href="#">“Editing Users: Administrator Controlled” on page 1-28</a> .
<b>Delete</b>	Delete lets you delete the user. For more information, see <a href="#">“Deleting Users” on page 1-27</a> .
<b>New</b>	New lets you add a user. Selecting <b>New</b> opens a blank Edit user screen. For more information, see <a href="#">“Adding New Users” on page 1-10</a> .

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**Option Boxes**

If you select either **Edit** or **New**, you must select at least one option box as well. Depending on which option boxes (**show aliases**, **show admin data**, and/or **show user data**) are checked, different data fields or screens will appear when the right mouse options are used.

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## **Adding, Viewing, Editing, and Deleting Users**

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### **Introduction**

The "[Working With The GUI Environment](#)" section discussed the GUI buttons, option boxes and a right-mouse click menu that enables adding, viewing, editing and deleting users. This section provides information about using those GUI elements to perform tasks.

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## Adding New Users

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### Introduction


This section describes how to add new users.

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### Procedure: Adding a New User

To add a new user, follow these steps:

**Table 1-7. Adding New Users**

<b>Step</b>	<b>Action</b>
<b>1</b>	Select the <b>Show admin data</b> option box.
<b>2</b>	Right-mouse click and select <b>New</b> .
<b>3</b>	The Edit user screen appears.
<b>4</b>	Enter the user name in the <b>Name</b> field.  <b>Caution</b> You are only allowed to enter and modify the <b>Name</b> field when you add a new user entry. After adding the new user, you cannot modify the name. If you enter a name incorrectly, delete the user and then re-enter him or her as a new user.
<b>5</b>	Enter or select the other fields as required.
<b>6</b>	Select the <b>Add</b> button to save the new user entry.

---



## Adding Users: Administrator's Edit User Screen

**Edit User Screen** Figure 1-5 illustrates the edit user screen that appears when the **show administrator data option box** is checked.

The screenshot shows the 'Edit user' dialog box with the following fields and options:

- Name:** 6385
- Group:** San Jose
- Marshal**
  - Group:** UserAgentGroup
  - Authentication Type:** None
  - IP:** 192.168.16.85
  - Password:** password
  - Static Registration**
  - Terminating Contact:**
    - Host:** 000.000.000.000
    - Port:** yyyy
- Make Calls Using Jtapi**
  - Enabled** (JtapiGroup)
- Forward All Calls**
  - Enabled** (ForwardAllCallsGroup)
- Block Long Distance or 900/976 Outgoing Calls**
  - Enabled** (CallBlockingGroup)
  - Long Distance Blocked by Administrator**
  - 900 Calls Blocked by Administrator**
- Screen Incoming Calls**
  - Enabled** (CallScreeningGroup)
- Block Phone Number Display**
  - Enabled** (CallerIdBlockingGroup)
- Forward Busy or Unanswered Calls**
  - Enabled** (ForwardNoAnswerBusyGroup)
  - Failure Case:** unknown
- Return Calls**
  - Enabled** (CallReturnGroup)

Buttons: Change Password, Ok, Cancel

Warning: Applet Window

Features that can be enabled for the user by the administrator

**Figure 1-5. Edit User Screen: Show Admin Data**

**Name** Specify the name of the user in alphanumeric characters. A unique name must be specified for each user.

**Group**

This field is a text identifier to help you classify your users.

**Marshal**

**Group**

Allows you to select a User Agent Marshal server group from the pull down menu. The list of marshal server groups in the pull down menu corresponds to the marshal server groups provisioned under **servers/marshalServer/serverType UserAgent**. You can load balance users among different User Agent Marshal server groups.

**Authentication Type**

<i>Field</i>	<i>Description</i>
Authentication Type	Allows you to select the type of authentication to use: <ul style="list-style-type: none"> <li>• None—users are not authenticated by the User Agent Marshal server.</li> <li>• Access List—users are authenticated by comparing the user’s IP address with the address stored in the IP field.</li> <li>• Digest—users are authenticated by comparing their password to data stored in the Password field.</li> </ul>
IP	Specifies the IP address of user. This IP address is used to authenticate the user during registration. This is only used for access list authentication.
Password	Specifies the user’s password. This is only used for digest authentication.

**Static Registration**

<i>Field</i>	<i>Description</i>
Static Registration	With static registration, the user agent or gateway is reachable via a fixed host name or IP address. The user agent does not have to register with the Marshal server. Static registration is primary used for devices that do not support the SIP REGISTER message. Check the option box to enable static registration or uncheck the option box to disable static registration.
Terminating Contact: Host	Specify the IP address of user agent or gateway. You can only specify this field if the static registration option box is checked.
Terminating Contact: Port	Specify the port number of user agent or gateway. You can only specify this field if the static registration option box is checked.

---

**JTAPI**

Check the Enabled option box to enable the JTAPI feature. With this feature enabled the user can place calls using a JTAPI User Agent.

■ **Note**

You must load a Java application, such as the JTAPI User Agent application that is bundled with VOCAL, to work with this JTAPI feature.

---

**ForwardAllCalls**

**Option Box**

Check the Enabled option box to enable the ForwardAllCalls feature for the user.

**Pull Down Menu**

The pull down menu allows you to select a Feature server group from the pull down menu. The list of Feature server groups in the pull down menu corresponds to the Feature server groups provisioned under **servers/featureServer/serverType ForwardAllCalls**. The Feature Server groups enable user load balancing. A load balancing utility is not included in the current software release.

---

**Call Blocking**

**Option Box**

Check the Enabled option box to enable the Call Blocking feature for the user. This will allow:

- Long distance blocked by administrator
- Long distance blocked by the user (set by the user).
- 900 calls blocked by administrator
- 900 calls blocked by the user (set by the user).

■ **Note**

For version 1.4.0 of VOCAL, long-distance call blocking only works for calls originating from the North American Dialing Plan (NADP). Calls cannot be blocked if they originate from Europe, Asia or other locations that are not part of the NADP.

**Pull Down Menu**

The pull down menu allows you to select a Feature server group from the pull down menu. The list of Feature server groups in the pull down menu corresponds to the Feature server groups provisioned under **servers/featureServer/serverType CallBlocking**. You can load balance users among different Feature Server groups. A load balancing utility is not included in the current software release.

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## Call Screening

### Option Box

Check the Enabled option box to enable the Call Screening feature for the user.

### ■Note

For version 1.4.0 of VOCAL, phone numbers entered for call screening must include the area code, regardless if they are local or long-distance phone numbers. Call Processing Language does not provide a pattern matching method that differentiates seven digit (local) phone numbers from ten digit (long-distance) numbers.

### Pull Down Menu

The pull down menu allows you to select a Feature server group from the pull down menu. The list of Feature server groups in the pull down menu corresponds to the Feature server groups provisioned under **servers/featureServer/serverType CallScreening**. You can load balance users among different Feature Server groups. A load balancing utility is not included in version 1.4.0 of VOCAL.

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## CallerIDBlocking

### Option Box

Check the CallerIDBlocking option box to enable the CallerID Blocking feature for the user.

### Pull Down Menu

The pull down menu allows you to select a Feature server group from the pull down menu. The list of Feature server groups in the pull down menu corresponds to the Feature server groups provisioned under **servers/featureServer/serverType CallerIDBlocking**. You can load balance users among different Feature Server groups. A load balancing utility is not included in version 1.4.0 of VOCAL.

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## ForwardNoAnswer

### Option Box

Check the ForwardNoAnswer option box to enable the ForwardNoAnswer feature for the user.

### Pull Down Menu

The pull down menu allows you to select a Feature server group from the pull down menu. The list of Feature server groups in the pull down menu corresponds to the Feature server groups provisioned under **servers/featureServer/serverType ForwardNoAnswer**. You can load balance users among different Feature Server groups. A load balancing utility is not included in version 1.4.0 of VOCAL.

### Failure Case

Users can specify a number where calls are forwarded to when Call Forward No Answer or Call Forward Busy is turned on by the user. Refer to [“Editing User Feature: Edit User Screen” section on page 1-31](#) for information on setting the user feature.

The administrator can specify a failure case number or address. Incoming calls will be forwarded to this failure case number or address if the system receives a failure message instead of busy messages or a no answer signal.

**Call Return**

**Option Box**

Check the Call Return option box to enable the Call Return feature for the user.

**Pull Down Menu**

The pull down menu allows you to select a Feature server group from the pull down menu. The list of Feature server groups in the pull down menu corresponds to the Feature server groups provisioned under **servers/featureServer/serverType CallReturn**. You can load balance users among different Feature Server groups.

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**Change Password**

The Change Password button allows you the change the password for the user. This password is for the end user to gain access to his or her feature screen, as shown in [Figure 1-10 on page 1-31](#).

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## Viewing Users: Individually

### Introduction

This section describes how to view records for individual users. If you have thousands of users loaded into your system, you will find that it is faster to load the data for individual users, or small groups of users, as required, rather than loading the data all users every time you login as an Administrator.

Viewing individual records requires using the right-mouse-click menu, which is described below.

### Viewing User

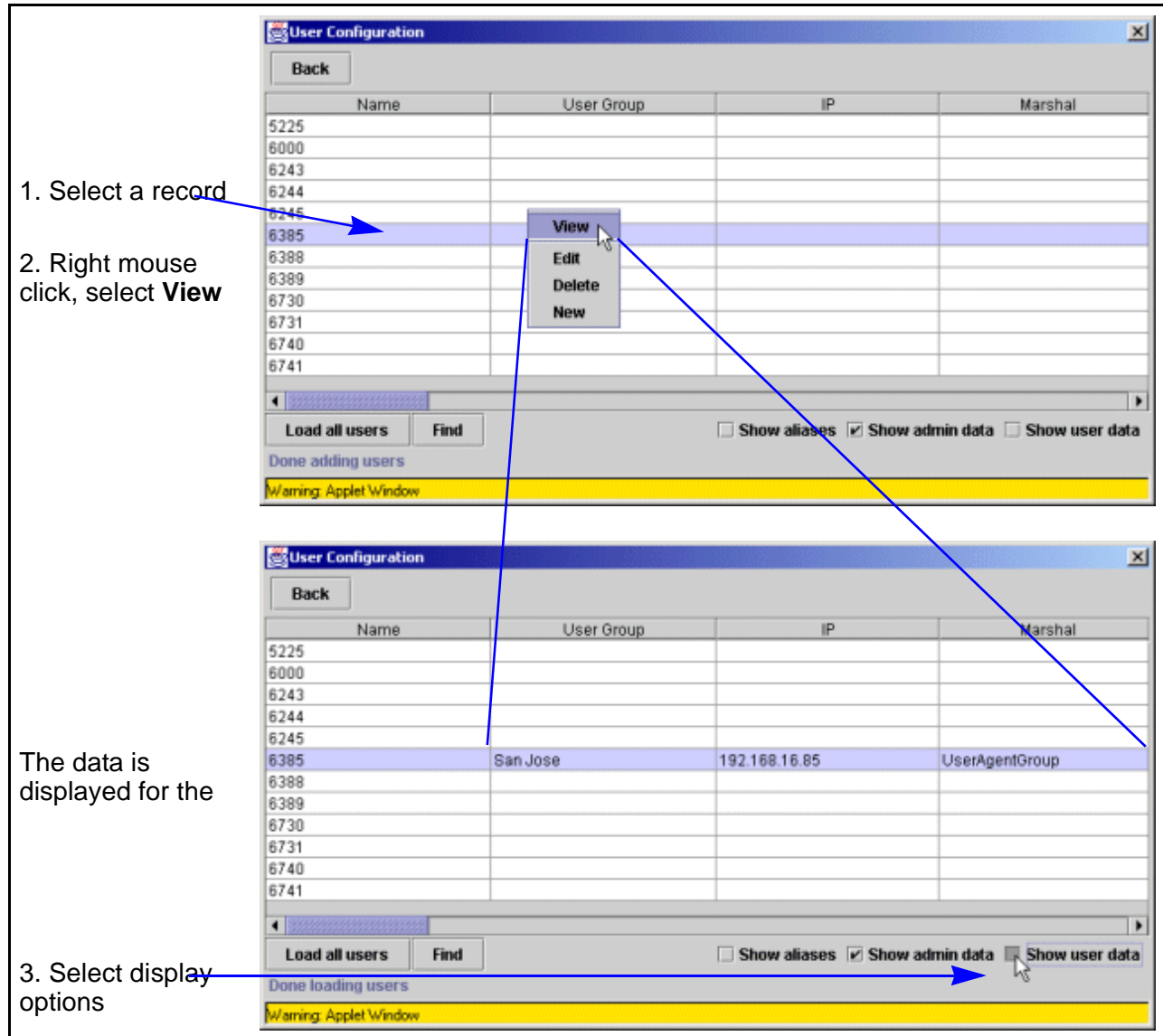
To view data fields for users, follow these steps:

**Table 1-8. Procedure for Viewing Users**

<b>Step</b>	<b>Action</b>
<b>1</b>	To select a user, click a table row. To select consecutive multiple users, hold down the shift key while clicking the table rows. To select non-consecutive multiple users, hold down the control key while clicking the table rows.
<b>2</b>	Using the right mouse click, select <b>View</b> .
<b>3</b>	Data fields with information will appear in the table. To sort the table by data type, check one or more of the option boxes: <ul style="list-style-type: none"><li>• <b>Show aliases</b>. User entries with aliases are shown in italics.</li><li>• <b>Show admin data</b>. Refer to <a href="#">Table 1-10 on page 1-19</a> for information on each admin data field.</li><li>• <b>Show user data</b>. Refer to <a href="#">Table 1-11 on page 1-22</a> for information on each user data field.</li></ul>

**Screen Capture:  
Viewing A Single  
User**

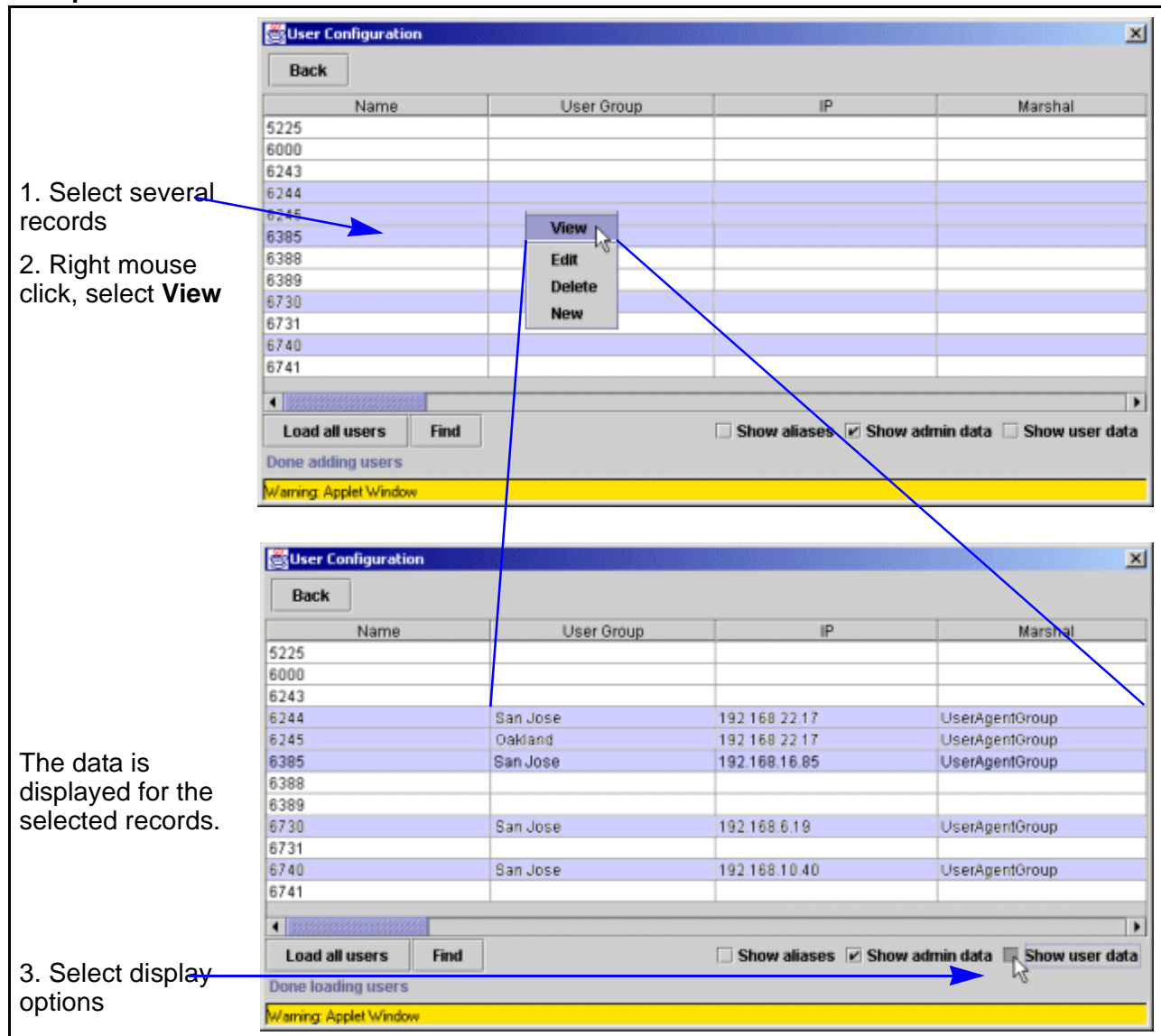
Figure 1-6 illustrates selecting the data for a single user.



**Figure 1-6. Displaying Data for a Single User**

**Screen Capture:  
Viewing Small  
Groups of Users**

Figure 1-7 illustrates selecting the data for a small group of users.



**Figure 1-7. Displaying Data for Small Groups of Users**

**Load all Users**

To view information for all users, select the **Load all users** button as shown in [Figure 1-8 on page 1-25](#).



## Viewing Users: Data Fields Descriptions

### Introduction

Different data fields appear in the user configuration screen depending on the option boxes selected.

### Default Data Fields

When none of the option boxes are checked, the user configuration screen displays these default data fields:

**Table 1-9. Default Data Field**

<i>Field</i>	<i>Description</i>
Name	Specifies the unique name of the user.
User Group	Specifies the group name that the user is grouped in. An administrator can classify users by specifying a common user group name.
IP	Optional: Specifies the IP address of the user. If Access List is selected for authentication type, the IP address is used to validate the user. If None or Digest is selected for authentication type, the IP address is not used.
Marshal	Specifies the name of the Marshal server group.

### Admin Data Field

When the Show admin data option box is checked, these data field appear in addition to the default data fields.

**Table 1-10. Admin Data Fields**

<i>Field</i>	<i>Description</i>
Forward All Calls Enabled	Indicates whether the Forward All Calls Enabled feature is enabled for the user: <ul style="list-style-type: none"> <li>• Deselected: indicates that this feature is disabled for the user.</li> <li>• Selected: indicates that this feature is enabled for the user.</li> </ul>
Forward All Group	Indicates the name of the ForwardAll Feature server group.
Forward Busy/No Ans. Enabled	Indicates whether the Forward Busy/No Answer feature is enabled for the user: <ul style="list-style-type: none"> <li>• Deselected: indicates that this feature is disabled for the user.</li> <li>• Selected: indicates that this feature is enabled for the user.</li> </ul>

**Table 1-10. Admin Data Fields (Continued)**

<b>Field</b>	<b>Description</b>
Forward Busy/No Ans. Group	Indicates the name of the ForwardBusyNoAnswer Feature server group.
Failure Case	Indicates the number or address to forward calls to there is a problem with contacting the destination called party.
Call Screen Enabled	Indicates whether the Call Screen feature is enabled for the user: <ul style="list-style-type: none"> <li>• Deselected: indicates that this feature is disabled for the user.</li> <li>• Selected: indicates that this feature is enabled for the user.</li> </ul>
Call Screen Group	Indicates the name of the Call Screen Feature server group.
Call Block Enabled	Indicates whether the Call Block feature is enabled for the user: <ul style="list-style-type: none"> <li>• Deselected: indicates that this feature is disabled for the user.</li> <li>• Selected: indicates that this feature is enabled for the user.</li> </ul>
Call Block Group	Indicates the name of the Call Block Feature server group.
Long Distance Admin Block	Indicates whether the Long Distance blocking feature is enabled by the administrator: <ul style="list-style-type: none"> <li>• Deselected: indicates that this feature is disabled.</li> <li>• Selected: indicates that this feature is enabled.</li> </ul>
900 # Admin Block	Indicates whether the 900 number blocking feature is enabled by the administrator: <ul style="list-style-type: none"> <li>• Deselected: indicates that this feature is disabled.</li> <li>• Selected: indicates that this feature is enabled.</li> </ul>
JTAPI Enabled	Indicates whether JTAPI feature is enabled: <ul style="list-style-type: none"> <li>• Deselected: indicates that this feature is disabled.</li> <li>• Selected: indicates that this feature is enabled.</li> </ul>
JTAPI Group	Indicates the name of the JTAPI Feature server group.

**Table 1-10. Admin Data Fields (Continued)**

<b>Field</b>	<b>Description</b>
Call Return Enabled	Indicates whether the Call Return feature is enabled for the user: <ul style="list-style-type: none"> <li>• Deselected: indicates that this feature is disabled for the user.</li> <li>• Selected: indicates that this feature is enabled for the user.</li> </ul>
Call Return Group	Indicates the name of the CallReturn Feature server group.
Caller ID Block Enabled	Indicates whether the Call ID Block feature is enabled for the user: <ul style="list-style-type: none"> <li>• Deselected: indicates that this feature is disabled for the user.</li> <li>• Selected: indicates that this feature is enabled for the user.</li> </ul>
Caller ID Group	Indicates the name of the CallerID Feature server group.
Authentication Type	Indicates the authentication type used to authenticate the user: <ul style="list-style-type: none"> <li>• None</li> <li>• Access List The IP address is required.</li> <li>• Digest A password is required.</li> </ul>
Password	Indicates the user's password when digest authentication is used.
Static Reg Enabled	Indicates whether static registration is enabled.
Terminating Host	Indicates the IP address of the terminating host when static registration is used.
Terminating Port	Indicates the port number on the termination host when static registration is used.

**User Data Field**

When the Show user data option box is checked, these data field appear in addition to the default data fields:

**Table 1-11. User Data Fields**

<b>Field</b>	<b>Description</b>
Forward All Set	Indicates whether Forward All feature is set by the user: <ul style="list-style-type: none"> <li>• OFF: indicates that call forwarding off.</li> <li>• ON: indicates that call forwarding is on and all calls are forward to a number specified by the user.</li> </ul>
Forward All To	Indicates the address or number where all calls are redirected when Forward All feature is set to ON or enabled.
Forward Busy Set	Indicates whether Forward Busy feature is set by the user: <ul style="list-style-type: none"> <li>• Deselected: indicates that call forwarding when busy is not on.</li> <li>• Selected: indicates that call forwarding busy is on and all calls are forward to a number specified by the user when the user is busy.</li> </ul>
Forward Busy To	Indicates the address or number where all calls are redirected when Forward Busy Set is enabled.
Forward No Ans. Set	Indicates whether Forward No Answer feature is set by the user: <ul style="list-style-type: none"> <li>• Deselected: indicates that call forwarding with no answer is not on.</li> <li>• Selected: indicates that call forwarding no answer is on and all calls are forward to a number specified by the user when the user does not answer.</li> </ul>
Forward No Ans To	Indicates the address or number where calls are redirected when the user does not answer the call and Forward No Ans. Set is enabled.
Long Distance User Block	Indicates whether Long Distance User Block feature is set by the user: <ul style="list-style-type: none"> <li>• Deselected: indicates that long distance block is not on.</li> <li>• Selected: indicates that long distance block is on and that the user cannot make long distance calls.</li> </ul>

**Table 1-11. User Data Fields (Continued)**

<b>Field</b>	<b>Description</b>
900 # User Block	<p>Indicates whether 900 Number Block feature is set by the user:</p> <ul style="list-style-type: none"> <li>• Deselected: indicates that 900 Number block is not enabled by the user. 900 numbers are not blocked and user can dial 900 numbers.</li> <li>• Selected: indicates that 900 Number block is enabled by the user. 900 numbers are blocked and users cannot dial 900 numbers.</li> </ul>
JTAPI Set	<p>Indicates whether the JTAPI feature is set by the user:</p> <ul style="list-style-type: none"> <li>• Deselected: indicates that JTAPI feature is not on.</li> <li>• Selected: indicates that JTAPI feature is on and that the user can make calls from a JTAPI User Agent.</li> </ul> <p>■ <b>Note</b> You must load a Java-based application, such as a JTAPI User Agent, to work with this JTAPI feature.</p>
Caller ID Blocking Set	<p>Indicates whether the Caller ID Blocking feature is set by the user:</p> <ul style="list-style-type: none"> <li>• Deselected: indicates that Caller ID feature is not on.</li> <li>• Selected: indicates the Caller ID feature is on.</li> </ul>

## Viewing Users: All Users

---

### Introduction

This section explains how to use the **Load all users** button and the option boxes to view user data.

---

### Overview

For situations where you need to compare the data between users, you can click the **Load all users** button. This button activates a program that reads a flat file on the Provisioning Server, containing all user data, and displays the data in the GUI.

---

### Procedure: Viewing Data

**To load and view the data, follow these steps:**

**Table 1-12. Procedure for Viewing User Data on the User Configuration Screen**

<b>Step</b>	<b>Description</b>
<b>1</b>	Click <b>Load all users</b> . The user records are filled with data.
<b>2</b>	Select one or any combination of the following options: <ul style="list-style-type: none"><li>• Show aliases</li><li>• Show admin data</li><li>• Show user data</li></ul>
<b>3</b>	Use the horizontal scroll bar to view the data.

Load All Users

Figure 1-8 shows the use of the **Load all users** button.

1. Click **Load all users**

The data is displayed.

2. Select options

3. Use the horizontal scrollbar to see the other fields.

Name	User Group	IP	Marshal
5225			
6000			
6243			
6244			
6245			
6385			
6388			
6389			
6730			
6731			
6740			
6741			

Name	User Group	IP	Marshal
5225	San Jose	192.168.22.17	UserAgentGroup
6000	San Jose	192.168.5.31	UserAgentGroup
6243	Oakland	192.168.22.17	UserAgentGroup
6244	San Jose	192.168.22.17	UserAgentGroup
6245	Oakland	192.168.22.17	UserAgentGroup
6385	San Jose	192.168.16.85	UserAgentGroup
6388	San Francisco	192.168.5.130	UserAgentGroup
6389	San Francisco	192.168.5.134	UserAgentGroup
6730	San Jose	192.168.6.19	UserAgentGroup
6731	Oakland	192.168.6.22	UserAgentGroup
6740	San Jose	192.168.10.40	UserAgentGroup
6741	San Francisco	192.168.10.41	UserAgentGroup

Figure 1-8. User Configuration Screen: Loading All Users

## Finding Users

### Introduction

You can highlight any of the users by clicking their record with the mouse. If you have thousands of users, the Find User utility will make your search easier.

### Procedure: Finding Users

The **Find** button activates a program that automatically searches the Name column for the first match of your criteria as you type it in. For example, if you type a 6, the first name that starts with 6 will be highlighted. If you type 63, the first name that starts with 63 will be highlighted. You can continue typing in the user name until the desired name appears on the screen.

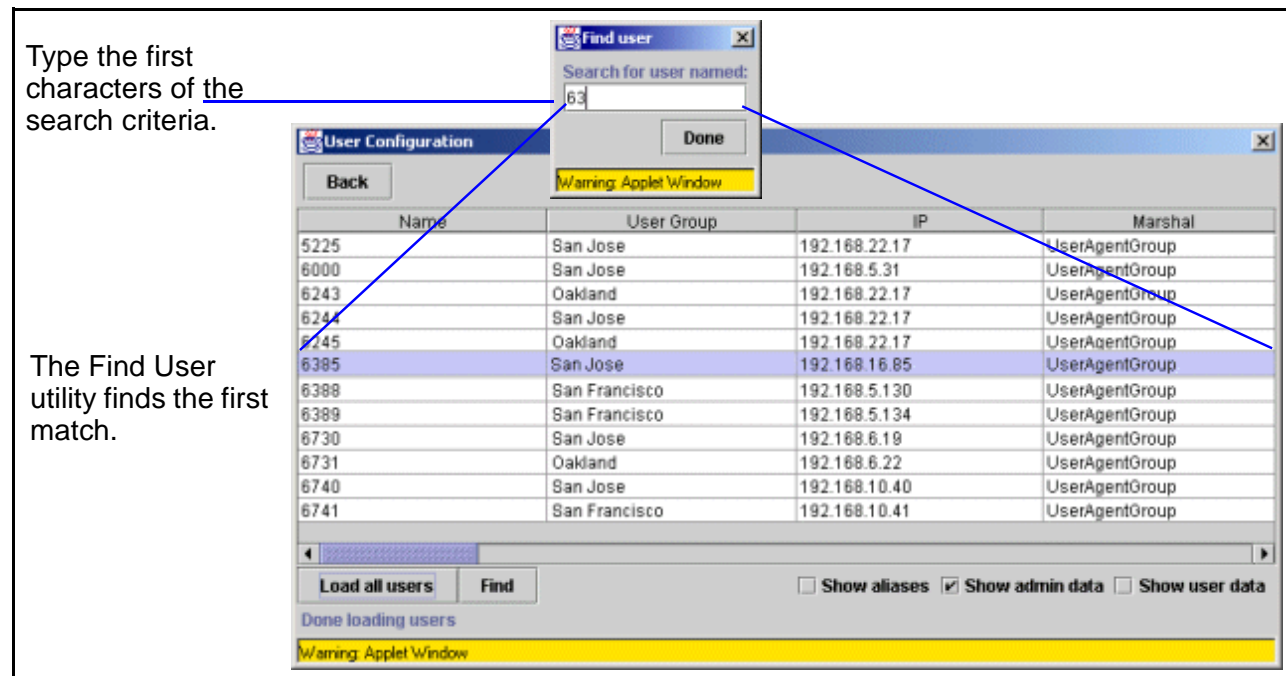
To find users, follow these steps:

**Table 1-13. Procedure—Finding Users**

Step	Description
1	Click <b>Find</b> . The Find Users dialog box appears.
2	Type the first characters of the name. The first instance of each character will be automatically selected in the list.
3	When you are finished searching, click <b>Done</b> .

### Screen Capture

Figure 1-9 shows the use of the Find User utility.



**Figure 1-9. User Configuration Screen: Finding Users**



## Deleting Users

---

### Deleting User

To delete a user or multiple users, follow these steps:

**Table 1-14. Procedure for Deleting Users**

<b>Step</b>	<b>Action</b>
<b>1</b>	To select a user, left mouse click on a row in the table. To select multiple users, hold down the shift key while left mouse clicking the rows in the table.
<b>2</b>	Using the right mouse click, select <b>Delete</b> .

---

## Editing Users: Administrator Controlled

---

**Introduction** This section describes how to edit users.

---

**Procedure: Editing a User** To edit a user, follow these steps:

**Table 1-15. Procedure for Editing User**

<b>Step</b>	<b>Action</b>
<b>1</b>	Select the <b>Show admin data</b> option box.
<b>2</b>	Use the Left mouse click to select a user entry.
<b>3</b>	Right mouse click and select <b>Edit</b> . The Edit user screen appears.
<b>4</b>	Edit the fields and option boxes as required.
<b>5</b>	Select the OK button to save the entry.

---

**Procedure: Editing Multiple Users** To edit multiple users, follow these steps:

**Table 1-16. Procedure for Editing Multiple Users**

<b>Step</b>	<b>Action</b>
<b>1</b>	Select the <b>Show admin data</b> option box.
<b>2</b>	To select multiple users, hold down the shift key while left mouse clicking the rows in the table.
<b>3</b>	Right mouse click and select <b>Edit</b> . The Edit user screen appears.
<b>4</b>	Edit the fields and option boxes as required.
<b>5</b>	Select the OK button to save.

**■Note**

When you edit multiple users, only one edit user screen will appear. All selected users will have the same field and features settings.

---

## Editing User: Show Alias

---

### Introduction

The aliases names associated with each users can be displayed using the show alias option box. User names with aliases appear in italics.

---

### What's an Alias?

An alias is another address or phone number by which a user can be reached. A telephone call directed to the alias will terminate at the user's telephone. For example, in an office, a user's complete phone number might be 555-6024, but to provide for simple on-site dialing, 6024 can be set up as an alias to substitute for the full phone number.

---

### Selecting Show Alias

If you select **Show Aliases**, then you must also select **Show Admin Data** or **Show User Data** as well. Selecting **Show Aliases** by itself is not allowed and returns an error.

---

## Editing User Features: User Controlled

### Introduction

The VOCAL system provides a web page for users to maintain some of their features. These features are call User Controlled Features and they include:

- JTAPI
- Forward all calls
- Call blocking
- Call screening
- Caller ID blocking
- Forward unanswered
- Forward busy

### Procedure: Editing a User

To edit a user, follow these steps:

**Table 1-17. Procedure for Editing User**

<b>Step</b>	<b>Action</b>
1	Select the <b>Show user data</b> option box.
2	Use the Left mouse click to select a user entry.
3	Right mouse click and select <b>Edit</b> . An Edit user screen appears.
4	Edit the fields and option boxes as required.
5	Select the OK button to save the entry.

### Procedure: Editing Multiple User

To edit multiple users, follow these steps:

■ **Note**

When you edit multiple users, only one edit user screen will appear. All selected users will have the same field and option box settings.

**Table 1-18. Procedure for Editing Multiple Users**

<b>Step</b>	<b>Action</b>
1	Select the <b>Show user data</b> option box.
2	To select multiple users, hold down the shift key while clicking the table rows.
3	Right-mouse click over the user records, and select <b>Edit</b> . The Edit screen appears.
4	Edit the fields and option boxes as required.
5	Select the OK button to save the entry.

## Editing User Feature: Edit User Screen

### Show User Data View

Figure 1-10 illustrates the edit user screen that appears when **Show user data** option box checked and the edit right mouse option is used. This screen displays features that can be enabled by the user. This screen is provided for the administrator to view the user's settings. If required, the administrator can modify the user's setting.

The screenshot shows a dialog box titled "Edit user" with a close button (X) in the top right corner. The dialog is divided into several sections:

- Aliases:** An empty text area.
- Make Calls Using Jtapi:** A checkbox labeled "Set" which is unchecked.
- Forward All Calls:** A checkbox labeled "Set" which is unchecked. To its right is a text field labeled "Forward To:" containing the text "none".
- Block Long Distance or 900/976 Outgoing Calls:** Two checkboxes: "Long Distance Blocked by User" and "900 Calls Blocked by User", both of which are unchecked.
- Screen Incoming Calls:** A table with two columns: "Name" and "Number".
 

Name	Number
John Smith	6740
- Block Phone Number Display:** A checkbox labeled "Set" which is unchecked.
- Forward Busy or Unanswered Calls:** Two checkboxes: "Forward Unanswered" and "Forward Busy", both of which are unchecked. To their right are two text fields labeled "Forward To:" containing the text "7000".

At the bottom of the dialog, there are three buttons: "Change Password", "Ok", and "Cancel". A yellow warning bar at the very bottom of the window reads "Warning: Applet Window".

Figure 1-10. Edit User Screen: Show User Data

---

### Aliases

This field displays aliases associated with this user.

#### To add aliases for the user:

- 1) Right mouse click over the Alias area and select add.
- 2) Type the alias name for the user.

To remove aliases, right mouse click the alias name and select remove.

---

### JTAPI

The user can use a JTAPI dial pad to place calls if the JTAPI set option box is checked.

#### ■Note

If the JTAPI option box is greyed out, this means that the Administrator has not enabled the JTAPI feature for the user.

---

### Forward All Calls

The user can set Forward All Calls to redirect all incoming calls to a specific number.

#### To turn on this feature:

- 1) Check the **Set** option box.
- 2) Enter the user ID in the **Forward to** text box, for example, 7000.

#### ■Note

If the Set option box and the Forward to text box are greyed out, this means that the Administrator has not enabled the Forward All Calls feature for the user.

---

### Call Blocking

The user can turn on call blocking for:

- Long distance numbers—this prevents the user from dialing long distance numbers beginning with 1.
- 900 numbers—this prevents the user from dialing 900 numbers.

#### ■Note

If the Long distance blocked by user and 900 calls blocked by user option boxes are greyed out, this means that the Administrator has not enabled these features for the user.

For version 1.4.0 of VOCAL, long-distance call blocking only works for calls originating from the North American Numbering Plan (NANP). Calls cannot be blocked if they originate from Europe, Asia or other locations that are not part of the NANP.

## **Call Screening**

The user can screen a call by name and number.

### **To add numbers to screen:**

- 1) Right mouse click near the name and number box. Select Add.
- 2) A Screen Calls From dialog box appears. Enter the name and number to screen. Click OK.  
The format is the user ID, for example, 7000.
- 3) If you enter “6” in the number field, then all numbers beginning with 6 will be screened. Entering a specific phone number, such as, “408-555-1000” will screen only that phone number.

To remove a screened number, right mouse click near the name and number box and select Remove.

### **■Note**

If the call screening table is greyed out, this means that the Administrator has not enabled this feature for the user.

For version 1.4.0 of VOCAL, phone numbers entered for call screening must include the area code, regardless if they are local or long-distance phone numbers. Call Processing Language does not provide a pattern matching method that differentiates seven digit (local) phone numbers from ten digit (long-distance) numbers.

---

## **Caller ID Blocking**

The user can set Caller ID blocking to prevent the caller’s number or address from being delivered and displayed to the called party.

### **■Note**

If the Caller ID blocking option box is greyed out, this means that the Administrator has not enabled these features for the user.

---

## **Forward No Answer Busy**

The user can set Forward No Answer Busy to forward all incoming calls to another number if:

- The user is busy—Forward Busy.
- The user if does not answer the call—Forward No Answer.

### **To turn on these features:**

- 1) Check the Set option box.
  - 2) Enter the number in the Forward to text box, for example, 7000.
-

## *Adding, Viewing, Editing, and Deleting Users*



# 2

## Network Management

This chapter describes network management and statistics for the VOCAL system.

<b>Topic</b>	<b>See Page</b>
<b>SNMP Support</b> .....	<b>2-2</b>
MIBs .....	2-3
VOCAL SNMP GUI .....	2-5
VOCAL SNMP GUI Screen Elements .....	2-6

## SNMP Support

---

**Overview**

VOCAL supports Simple Network Management Protocol (SNMP) monitoring from:

- the VOCAL SNMP GUI - this supports monitoring of VOCAL server status.
- a third party SNMP network manager.

---

**SNMP Version**

VOCAL supports SNMP version 2 (RFCs 1441 to 1452).

---

## MIBs

### Introduction

---

In a TCP/IP-based network, each device maintains a set of variables describing its state. In Simple Network Management Protocol (SNMP), these variables are known as objects, but these objects do not hold the same meaning as those within an object-oriented programming architecture. SNMP objects contain information about their state without any methods, other than the ability to read and write their values. A collection of SNMP objects are known as a Management Information Base (MIB).

As networks are built, they normally use several MIBs to describe the various structures within the overall architecture. It is more common for developers to use public MIBs for the non-proprietary components of their system and to build their own for the proprietary components.

### Public MIBs

---

The VOCAL system supports the following public MIBs:

#### **RFC 1213 - MIB II**

#### **Network Services Monitoring MIB**

VOCAL supports a subset of the network services monitoring MIBs defined in RFC 2788. This includes the minimum identifiers of applications for use in other MIBs as indexes.

For more information refer to this file in the Linux directory: `/usr/local/vocal/proxies/netMgmt/NETWORK-SERVICES-MIB.txt`.

#### **SIP MIBs**

VOCAL supports the initial version of Session Initiation Protocol (SIP) MIB dated July 2000 (`draft-ietf-sip.mib-01.txt`). The MIB is temporarily located under `enterprise.vovida.vovida` Temporary until the MIB is assigned a permanent location by IANA. For more information refer to `/usr/local/vocal/proxies/netMgmt` directory for these SIP MIBs:

- SIP-COMMON-MIB.txt
- SIP-MIB-SMI.txt
- SIP-REGISTRAR-MIB.txt
- SIP-SERVER-MIB.txt
- SIP-TC.txt
- SIP-UA-MIB.txt

#### **UCD Enterprise MIB**

VOCAL supports and implements the UCD Enterprise MIBs. For additional information, refer to this web site:

<http://net-snmp.sourceforge.net>

**VOCAL Enterprise  
MIB**

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For more information refer to the /usr/local/vocal/proxies/netMgmt directory:

- VOVIDA-LOCAL-GRP-MIB.txt
  - VOVIDA-NOTIFICATIONS-MIB.txt
  - VOVIDA-SERVERGRP-MIB.txt
  - VOVIDA-SOFTSWITCHSTATS-MIB.txt
  - VOVIDA-SUBSCRIBERSTATS-MIB.txt
-

## VOCAL SNMP GUI

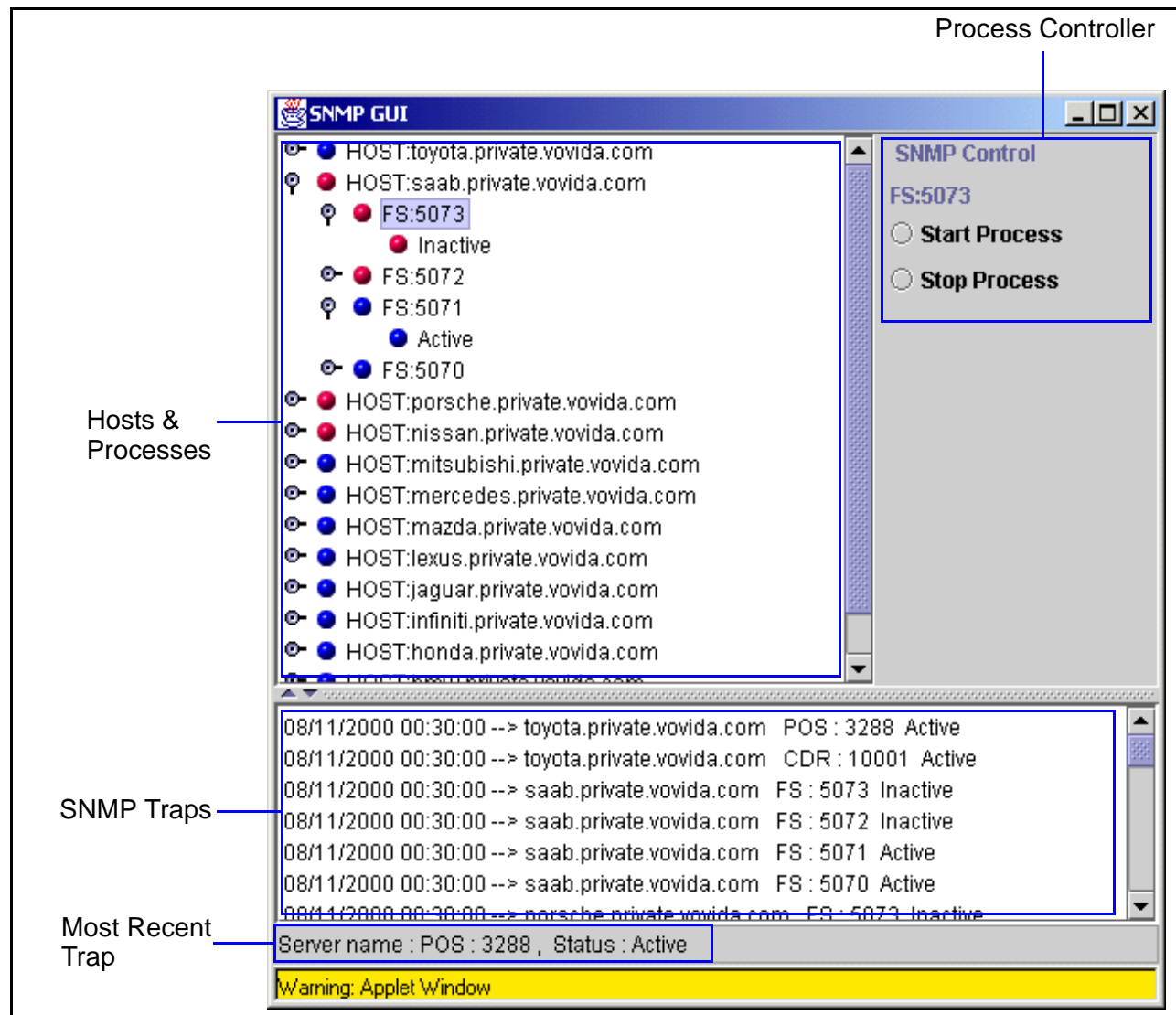
### Server Status Monitoring

Each VOCAL system server sends (via multicast) heartbeat packets to its peers at a predefined interval. The Heartbeat Server monitors the exchange of heartbeat packets between VOCAL servers and sends server status trap messages to the network management system. The network management system displays server status on the VOCAL SNMP GUI.

### VOCAL SNMP GUI

Figure 2-1 illustrates an example of the VOCAL SNMP GUI. Access the VOCAL SNMP GUI from the URL:

<http://<hostname of the provisioning server>/<system name>/index.html>



**Figure 2-1. VOCAL SNMP GUI Screen**

## VOCAL SNMP GUI Screen Elements

---

<b>Hosts &amp; Processes</b>	This frame displays the host server and indicates whether they are active (blue) or inactive (red). If a host server contains several processes, it will display a red ball if one or more of the processes is inactive.
<b>SNMP Controller</b>	<p>The process controller allows you to start or stop the SNMP control process.</p> <p><b>To start or stop the process controller, follow these steps:</b></p> <ol style="list-style-type: none"><li>1) From the Host View, select a hostname and select a process.</li><li>2) Select <b>Start Process</b> to begin the SNMP control or <b>Stop Process</b> to stop the SNMP Process.</li></ol> <p>■ <b>Note</b></p> <p>The process or function running on each host server is listed under each hostname. When the process is selected, the process abbreviation and port number appear above the <b>Start Process</b> and <b>Stop Process</b> buttons. Processes or functions are abbreviated as:</p> <ul style="list-style-type: none"><li>• FS - Feature Server</li><li>• MS - Marshal Server</li><li>• POS - Policy Server</li><li>• RS -Redirect Server</li><li>• CDR - Call Detail Record Server</li><li>• PS - Provisioning Server</li></ul> <p>■ <b>Note</b></p> <p>There is no warning prompt for starting or stopping processes. If you select a process, and click either <b>Start Process</b> or <b>Stop Process</b>, the instruction is sent immediately to the host.</p>
<b>SNMP Traps</b>	The SNMP Traps screen displays SNMP traps from the system with timestamps. Traps are used to report unexpected events, such as a server crashing. Trap is a conventional term that comes from the early developers of SNMP.
<b>Most Recent SNMP Trap</b>	Displays the server name and status of the most recent trap.

---



# Features

This chapter describes features supported by the VOCAL system.

<b>Topic</b>	<b>See Page</b>
<b>Features</b> .....	<b>A-2</b>
<b>Core System Features</b> .....	<b>A-3</b>
<b>Set-Based Features</b> .....	<b>A-5</b>

## Features

---

### Introduction

This section describes the types of feature supported by the VOCAL system.

---

### Overview

The VOCAL system supports two types of features—core system features and set-based features.

#### **Core System Features**

Core system features are those that involve rerouting calls under certain conditions, such as the called party is busy, or the phone number being called is forbidden.

#### **Set-Based Features**

Set-based features are those that require functions to built into the phone sets to interpret information coming from the SIP messages. For example, SIP messages contain Caller ID information, but the phone cannot make use of it unless it has a Caller ID Display.



## Core System Features

### Types of Core System Features

There are two types of system features—calling features and called features. The calling features are assigned to the call originator. The called features are assigned to the calling destination.

<i>Calling Features</i>	<i>Called Features</i>
Call Blocking	Forward All Calls
Caller ID Blocking	Forward Busy
	Forward No Answer
	Call Return
	Call Screening
	Meet-Me Conferencing
	Voice Mail

### Call Blocking

The administrator or the user can turn on call blocking for:

- Long distance numbers—this prevents the user from dialling long distance numbered beginning with 1.
- 900 numbers—this prevents the user from dialling 900 numbers.

#### ■ Note

For version 1.4.0 of VOCAL, activating call blocking for long distance numbers unblocks 1-8xx phone numbers. This includes toll-free as well as regular 8xx area codes such as 801 and 802.

Also, for version 1.4.0 of VOCAL, long-distance call blocking only works for calls originating from the North American Numbering Plan (NANP). Calls cannot be blocked if they originate from Europe, Asia or other locations that are not part of the NANP. For more information about the NANP, see [www.nanpa.com](http://www.nanpa.com).

### Caller ID Blocking

The Caller ID blocking feature can be enabled for the user. With Caller ID blocking, the user's number and name will not be delivered or displayed on the called party's display equipment or application.

### Forward All Calls

The Forward All Calls feature redirects all incoming calls to a specified, alternate number or voice messaging system.

### Forward No Answer and Forward Busy

Forward No Answer Busy features allows redirecting of all incoming calls to another number if:

- The user is busy—Forward Busy.
- The user if does not answer the call after a specified number of rings-Forward No Answer.

## Core System Features

For example, Forward No Answer Busy feature can be used to redirect incoming calls to the voice messaging system or alternate number if the user is busy or does not answer.

---

### Call Return

Call return allows the user to call back the last caller. The user can either dial \*69 to dial up the last caller or select from a list of callers on his or her phone.

---

### Call Screening

Call screening allows the user to block calls from a list of numbers. For example, when an screened number calls the user, the caller will receive a busy signal. When call screening is enabled the user's phone set will not ring for a screened number.

#### ■ Note

For version 1.4.0 of VOCAL, phone numbers entered for call screening must include the area code, regardless if they are local or long-distance phone numbers. Call Processing Language does not provide a pattern matching method that differentiates seven digit (local) phone numbers from ten digit (long-distance) numbers.

---

### Meet-Me Conferencing

A conference bridge is provided with a well-known phone number. Users agree to call the well-known phone number at the same time, and are linked up together in the same audio path. The linking is a function of the conference bridge.

---

### Unified Messaging

The VOCAL system provides a unified messaging system whereby voicemail messages are converted into wave files and sent to the users attached to e-mail messages.

---

## Set-Based Features

<b>Definition</b>	<p>Set based features are features that a user can enable from a phone set. These features are an example of how SIP-based networks are able to transfer much of its intelligence to its end-points.</p> <p>Many SIP IP phone sets have a variety of “smart” features. The VOCAL system supports the following set-based features:</p> <ul style="list-style-type: none"> <li>• Call Transfer</li> <li>• Calling Number Delivery</li> <li>• Calling Name Delivery</li> <li>• Caller ID</li> <li>• Call Waiting</li> <li>• AdHoc Conferencing</li> </ul>
<b>Transfer</b>	<p>The Transfer key on a phone set allows the user to direct a call to another number or person. There are two types of transfers:</p> <ul style="list-style-type: none"> <li>• Consult Transfer -With consult transfer you consult the other party before transferring the call. To consult transfer a caller, you would press the transfer button to place the caller on hold. You will hear a dial tone and can dial the number you want to transfer the call to. Wait until the other party answers the call and inform them you are transferring the call. Press the transfer button on the phone set to complete the transfer.</li> <li>• Blind Transfer—With blind transfer you transfer the caller to another party without announcing the caller to the called party. To blind transfer a caller, you would press transfer to place the caller on hold. You will hear a dial tone and can dial the number you want to transfer the call to. Pressing transfer after dialling the number will immediately transfer the call.</li> </ul> <p>■ <b>Note</b> Some analog phones, which are attached to IP gateways, use a “flashback” key in place of a “transfer” key.</p>
<b>Calling Number Delivery</b>	<p>Calling Number Delivery, or commonly called Call Display Number, provides the called party with the number of the caller, the date and time of call.</p>
<b>Calling Name Delivery</b>	<p>Calling Name Delivery, or commonly called Call Display Name, provides the called party with the name of the caller, the date and time of call.</p>
<b>Caller ID</b>	<p>Caller ID requires display equipment or application. The name and/or number of the caller are displayed on the called party’s display equipment or application.</p>
<b>Call Waiting</b>	<p>Call waiting allows a user to receive an incoming call while the user is on another call. The user places the first call on hold while answering the second call.</p>

## *Set-Based Features*

### **AdHoc Conferencing**

---

The Conference key on a phone set allows the user to set up a conference call with a number of people. To set up a conference call:

- Call the first person. Press the conference button to place the first caller on hold.
  - Call the second person. Press the conference button to add the second caller to the call.
  - Repeat until all callers are added to the call.
-

# B

## Supported SIP Messages

Topic	See Page
SIP Request Messages .....	B-2
SIP Response Messages .....	B-3

## SIP Request Messages

**Supported SIP  
Request Messages**

The VOCAL system supports these SIP request messages:

***Table B-1. SIP Request Messages Descriptions***

<b><i>SIP Request Messages</i></b>	<b><i>Descriptions</i></b>
INVITE	Indicates that the user or service is being invited to participate in a session.
ACK	Confirms that the client has received a final response to an INVITE request.
BYE	Indicate the user wishes to terminate the call.
CANCEL	Cancels a pending request but does not affect a completed request.
REGISTER	Register the address listed in the To header field with a SIP server.
OPTIONS	Queries the capability of the servers.
INFO	Allows for the carrying of the session related control information that is generated during a session.

## SIP Response Messages

---

### **SIP Response Messages Category**

The VOCAL system supports all SIP response messages:

- 1xx Responses - Information Responses
  - 2xx Responses - Successful Responses
  - 3xx Responses - Redirection Responses
  - 4xx Responses - Request Failures Responses
  - 5xx Responses - Server Failure Responses
  - 6xx Responses - Global Failure Responses
- 

### **For More Information**

Refer to the SIP RFC 2543 for a list of the status codes and their reason codes:

<http://www.ietf.org/rfc/rfc2543.txt>

---

### **1xx and 2xx Responses**

1xx SIP response message are informational responses:

- 100 Trying
  - 180 Ringing
  - 181 Call Is Being Forwarded
  - 182 Queued
  - 183 Session Progress
  - 200 OK
- 

### **3xx Responses**

3xx SIP response message are redirection responses:

- 300 Multiple Choices
  - 301 Moved Permanently
  - 302 Moved Temporarily
  - 303 See Other
  - 305 Use Proxy
  - 380 Alternative Service
- 

### **4xx Responses**

4xx SIP response message are client error responses:

- 400 Bad Request
- 401 Unauthorized
- 402 Payment Required
- 403 Forbidden
- 404 Not Found
- 405 Method Not Allowed
- 406 Not Acceptable
- 407 Proxy Authentication Required
- 408 Request Timeout

## *SIP Response Messages*

- 409 Conflict
- 410 Gone
- 411 Length Required
- 413 Request Entity Too Large
- 414 Request-URI Too Large
- 415 Unsupported Media Type
- 420 Bad Extension
- 480 Temporarily not available
- 481 Call Leg/Transaction Does Not Exist
- 482 Loop Detected
- 483 Too Many Hops
- 484 Address Incomplete
- 485 Ambiguous
- 486 Busy Here

---

### **5xx Responses**

5xx SIP response message are server error responses:

- 500 Internal Server Error
- 501 Not Implemented
- 502 Bad Gateway
- 503 Service Unavailable
- 504 Gateway Time-out
- 505 SIP Version not supported

---

### **6xx Responses**

6xx SIP response message are global failure responses:

- 600 Busy Everywhere
  - 603 Decline
  - 604 Does not exist anywhere
  - 606 Not Acceptable
-





# Call Flows

This chapter provides call flows diagram and IP trace logs for several call scenarios.

<b>Topic</b>	<b>See Page</b>
<b>SIP Phone: Registration</b> .....	<b>C-3</b>
Registration: Access List Authentication .....	C-4
Registration: Digest Authentication .....	C-6
<b>SIP IP Phone to SIP IP Phone: Call Setup and Disconnect</b> .....	<b>C-8</b>
<b>SIP IP Phone to Analog Phone via Gateway</b> .....	<b>C-15</b>
<b>SIP Phone to Phone via Gateway: Called Party is Busy</b> .....	<b>C-24</b>
<b>SIP IP Phone to SIP IP Phone: Forward All Calls</b> .....	<b>C-32</b>
<b>Phone to SIP Phone via Gateway: Call Screening</b> .....	<b>C-41</b>
<b>SIP Phone to PSTN: Call Blocking</b> .....	<b>C-47</b>
<b>SIP IP Phone to SIP IP Phone: Call Return</b> .....	<b>C-52</b>
<b>User Agent to User Agent: Call Waiting</b> .....	<b>C-70</b>
<b>SIP IP Phone to SIP IP Phone: Forward to Voice Mail</b> .....	<b>C-85</b>

**Topic** *(continued)* **See Page**

---

**User Agent to User Agent: Consulted Transfer..... C-96**

**User Agent to User Agent: Blind Transfer..... C-114**

**JTAPI..... C-131**

**Ad Hoc Conference Call Between User Agents..... C-149**

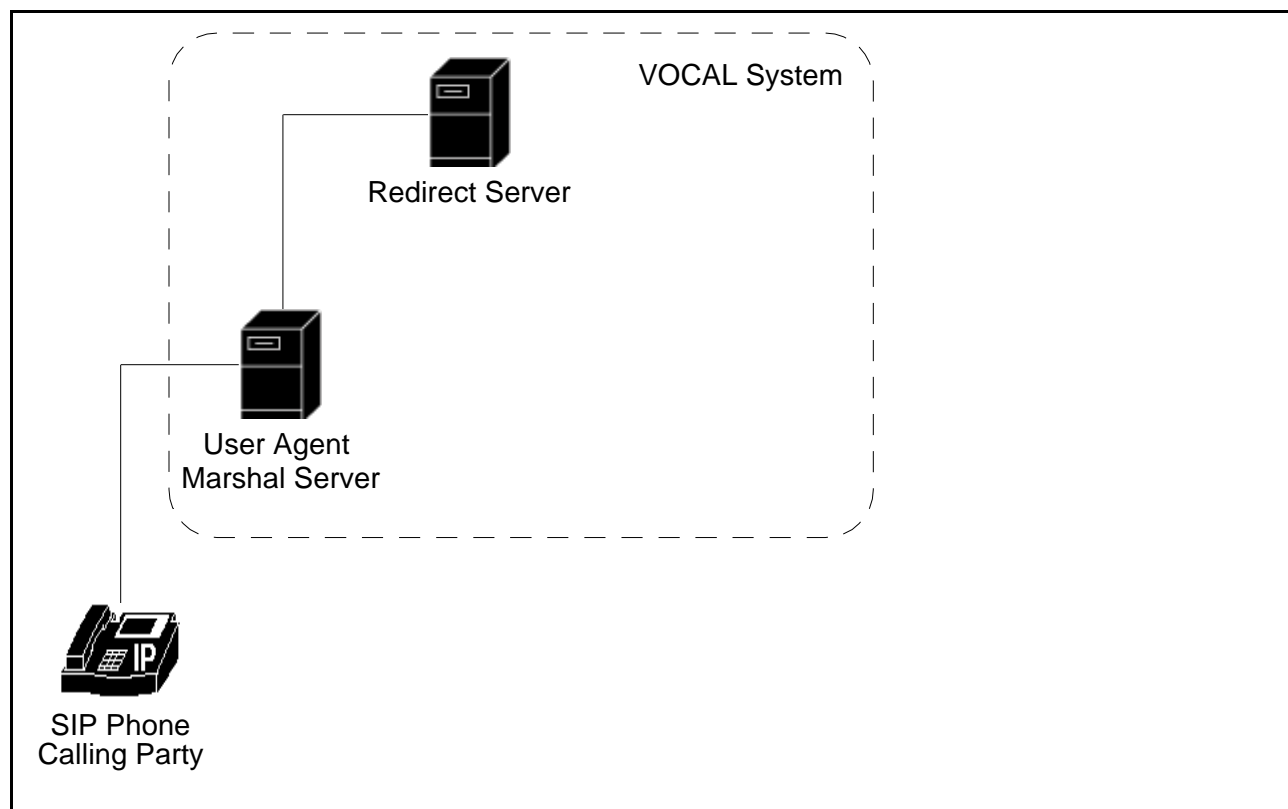
## SIP Phone: Registration

**Call Scenario** Figure C-1 illustrates a SIP phone registering with the Marshal server.

**Authentication Methods** There are three registration methods, no authentication, access list authentication or digest authentication. Table C-1 shows the authentication criteria used by each method.

**Table C-1. Authentication Methods and Criteria**

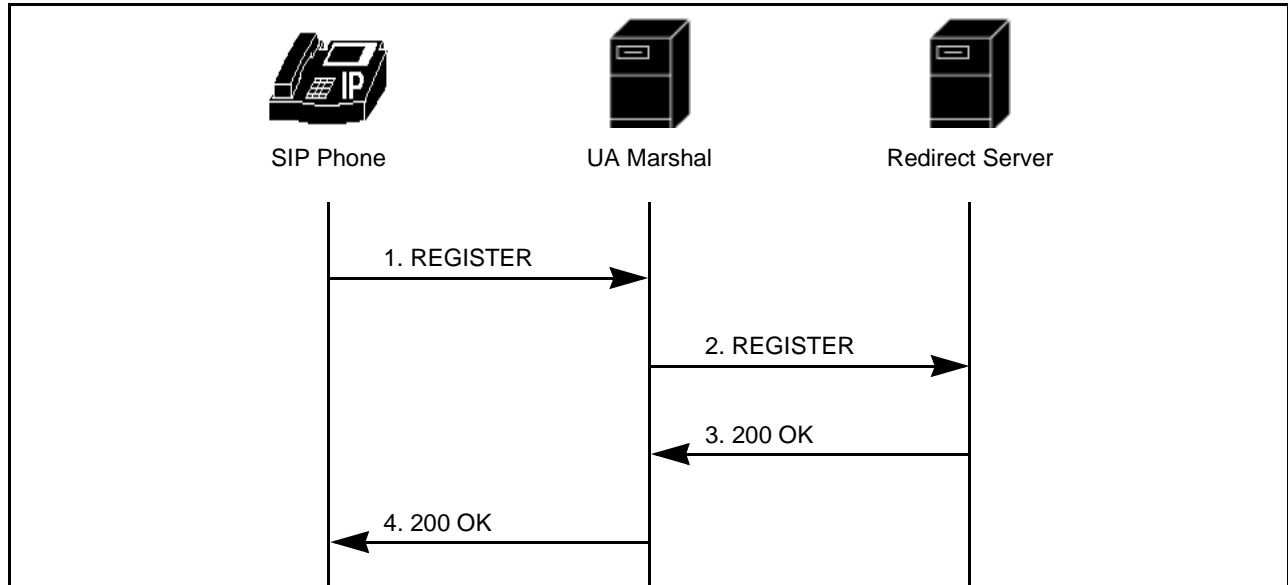
<i>Method</i>	<i>Authentication Criteria</i>
No Authentication	None
Access List	User's IP Address
Digest	User's name and password



**Figure C-1. Registration**

## Registration: Access List Authentication

**Call Flow Diagram** Figure C-2 shows a SIP IP phone registering with the Redirect server. The User Agent Marshal server is using the Access List authentication method.



**Figure C-2. Call Flow Diagram: SIP Phone Registration**

**Call Trace** The following trace shows a SIP IP phone registering with the Redirect server. The User Agent Marshal server is using the Access List authentication method.

```

-----
SIP Headers
-----
sip-req: REGISTER sip:192.168.26.180 SIP/2.0 [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: sip:6711@192.168.26.180
Header: To: sip:6711@192.168.26.180
Header: Call-ID: c2943000-1e262-14ae-2e323931@192.168.26.10
Header: CSeq: 100 REGISTER
Header: Contact: <sip:6711@192.168.26.10:5060>
Header: Expires: 3600
Header: Content-Length: 0
-----

```

```

-----
SIP Headers
-----
sip-req: REGISTER sip:@192.168.26.200:5060 SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.180:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c2943000-1e262-14ae-2e323931@192.168.26.10
Header: CSeq: 100 REGISTER
Header: Proxy-Authorization: Basic 123
Header: Expires: 3600
Header: Contact: <sip:6711@192.168.26.180:5060>
Header: Contact: <sip:6711@192.168.26.10:5060>
Header: Content-Length: 0
-----

```

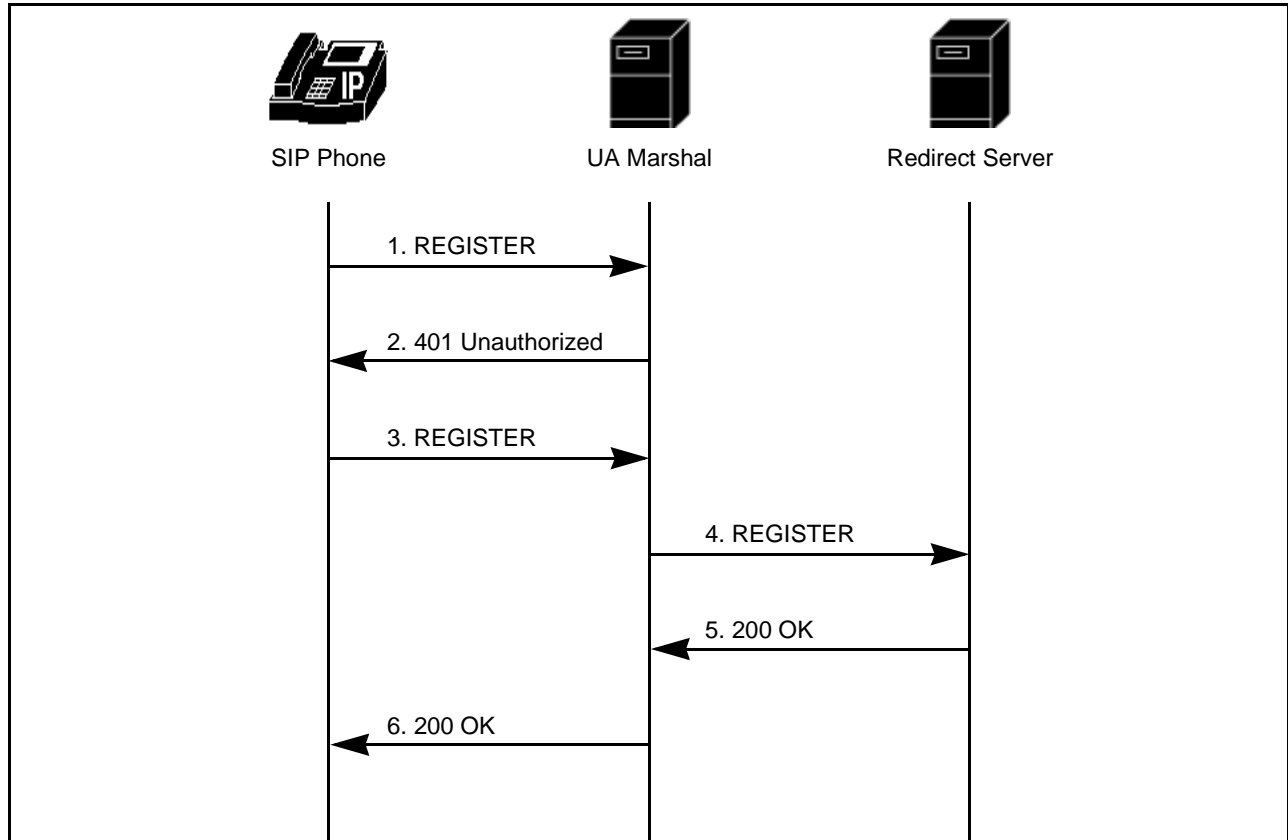
```
-----  
SIP Headers  
-----  
sip-res: SIP/2.0 200 OK [192.168.26.200:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.180:5060>  
Header: To: <sip:6711@192.168.26.180:5060>  
Header: Call-ID: c2943000-1e262-14ae-2e323931@192.168.26.10  
Header: CSeq: 100 REGISTER  
Header: Contact: <sip:6711@192.168.26.180:5060>  
Header: Contact: <sip:6711@192.168.26.10:5060>  
Header: Expires: 3600  
Header: Content-Length: 0  
-----
```

```
-----  
SIP Headers  
-----  
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.26.10:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.180:5060>  
Header: To: <sip:6711@192.168.26.180:5060>  
Header: Call-ID: c2943000-1e262-14ae-2e323931@192.168.26.10  
Header: CSeq: 100 REGISTER  
Header: Contact: <sip:6711@192.168.26.10:5060>  
Header: Expires: 3600  
Header: Content-Length: 0  
-----
```

---

## Registration: Digest Authentication

**Call Flow Diagram** Figure C-3 shows a SIP IP phone registering with the Redirect server. The User Agent Marshal server is using the Digest authentication method.



**Figure C-3. Call Flow Diagram: SIP IP Phone Registration — Digest Authentication**

**Call Trace** The following call trace shows a SIP IP phone registering with the Redirect server. The User Agent Marshal server is using the Digest authentication method.

```
-----
SIP Headers
-----
sip-req: REGISTER sip:192.168.26.180 SIP/2.0 [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: sip:6711@192.168.26.180
Header: To: sip:6711@192.168.26.180
Header: Call-ID: c2943000-1e262-513-2e323931@192.168.26.10
Header: CSeq: 100 REGISTER
Header: Contact: <sip:6711@192.168.26.10:5060>
Header: Expires: 3600
Header: Content-Length: 0
-----
SIP Headers
-----
sip-res: SIP/2.0 401 Unauthorized [192.168.26.180:5060->192.168.26.10:5060]
```

```
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.180:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c2943000-1e262-513-2e323931@192.168.26.10
Header: CSeq: 100 REGISTER
Header: WWW-Authenticate: Digest
realm=vovida.com,algorithm=MD5,nonce=966645751
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: REGISTER sip:192.168.26.180 SIP/2.0 [192.168.26.10:50373-
>192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: sip:6711@192.168.26.180
Header: To: sip:6711@192.168.26.180
Header: Call-ID: c2943000-1e262-513-2e323931@192.168.26.10
Header: CSeq: 101 REGISTER
Header: Authorization: Digest
username="6711",realm="vovida.com",uri="sip:192.168.26.180",response="fee2efef60a99b4576c
0437947959deb",nonce="966645751",algorithm=MD5
Header: Contact: <sip:6711@192.168.26.10:5060>
Header: Expires: 3600
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: REGISTER sip:@192.168.26.200:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.26.200:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.180:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c2943000-1e262-513-2e323931@192.168.26.10
Header: CSeq: 101 REGISTER
Header: Authorization: Basic VovidaClassXswitch
Header: Expires: 3600
Header: Contact: <sip:6711@192.168.26.180:5060>
Header: Contact: <sip:6711@192.168.26.10:5060>
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

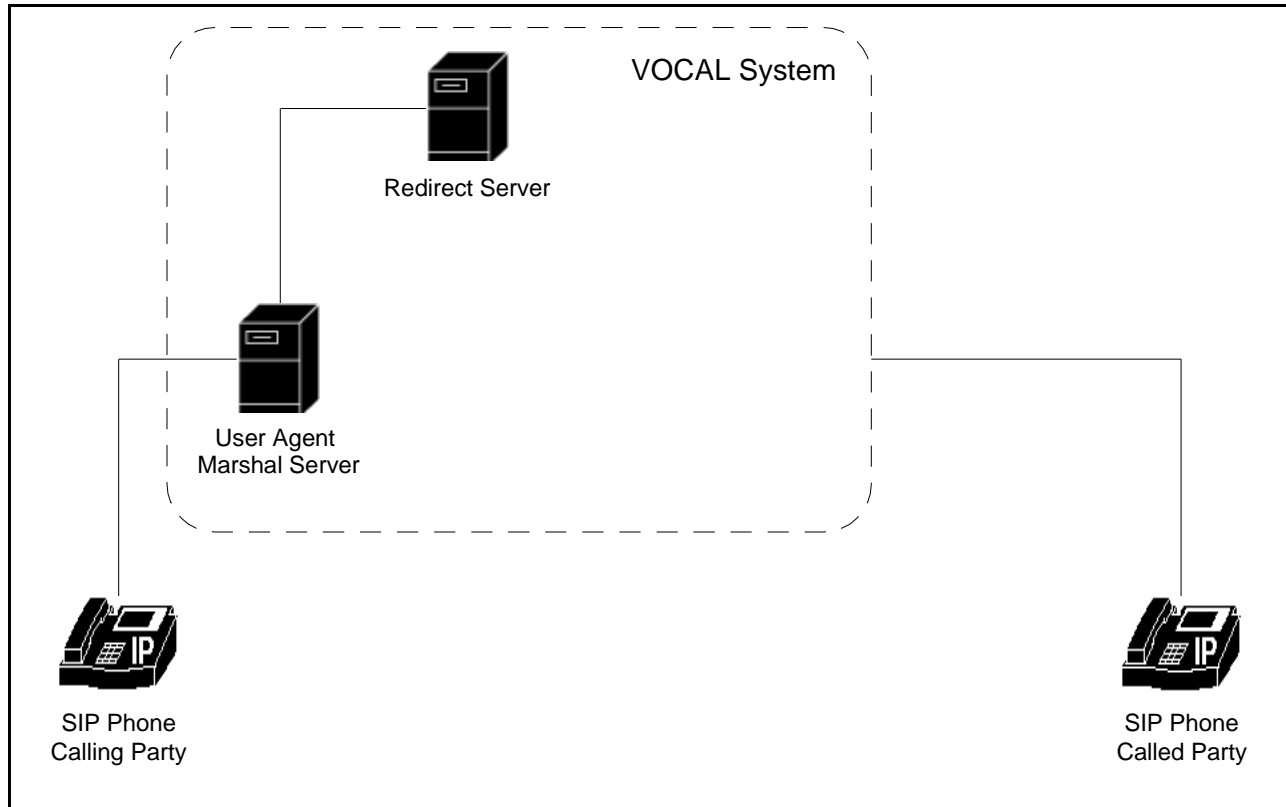
```
sip-res: SIP/2.0 200 OK [192.168.26.200:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.180:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c2943000-1e262-513-2e323931@192.168.26.10
Header: CSeq: 101 REGISTER
Header: Contact: <sip:6711@192.168.26.180:5060>
Header: Contact: <sip:6711@192.168.26.10:5060>
Header: Expires: 3600
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.26.10:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.180:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c2943000-1e262-513-2e323931@192.168.26.10
Header: CSeq: 101 REGISTER
Header: Contact: <sip:6711@192.168.26.10:5060>
Header: Expires: 3600
Header: Content-Length: 0
```

## SIP IP Phone to SIP IP Phone: Call Setup and Disconnect

**Call Scenario** Figure C-4 illustrates a call between two, on-network SIP IP phones.

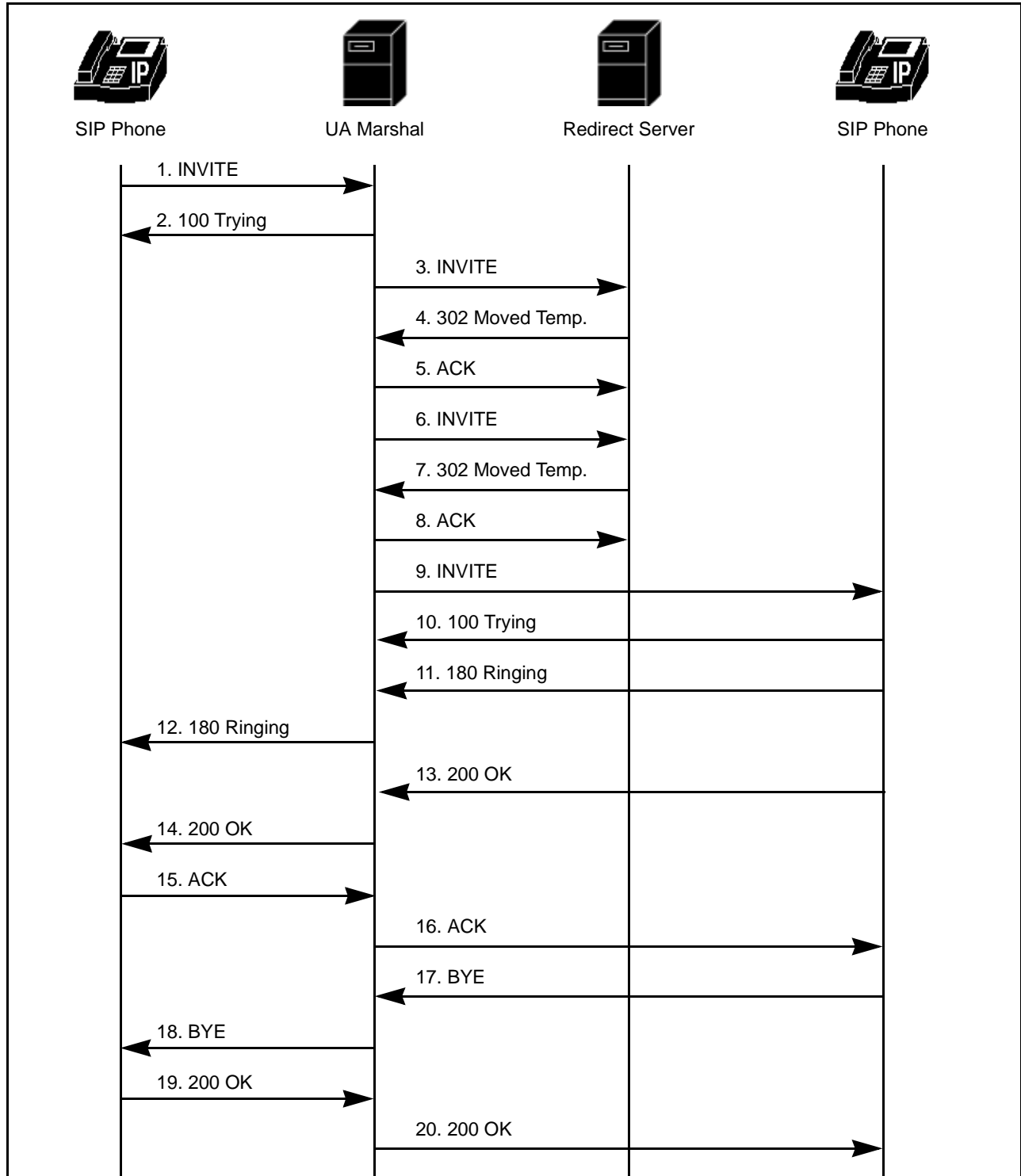


**Figure C-4. SIP Phone to SIP Phone**

**Call Flow Diagrams** Figure C-5 shows a successful call setup between two, on-network SIP IP phones. In this example, the called party terminates the call.



*SIP IP Phone to SIP IP Phone: Call Setup and Disconnect*



**Figure C-5. Call Flow Diagram: SIP Phone to SIP Phone**

### Call Trace

The following call trace shows a successful call setup between two, on-network IP phones. In this example, the called party terminates the call.

```
-----
                          SIP Headers
-----
sip-reg:    INVITE sip:5120@192.168.36.180 SIP/2.0      [192.168.6.21:50623->192.168.36.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.6.21:5060
Header:    From: sip:5121@192.168.6.21
Header:    To: <sip:5120@192.168.36.180>
Header:    Call-ID: c2943000-e0563-2alce-2e323931@192.168.6.21
Header:    CSeq: 100 INVITE
Header:    Expires: 180
Header:    User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:    Accept: application/sdp
Header:    Contact: sip:5121@192.168.6.21:5060
Header:    Content-Type: application/sdp
Header:    Content-Length: 219
-----
                          SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 16264 18299 IN IP4 192.168.6.21
Header:    s=SIP Call
Header:    c=IN IP4 192.168.6.21
Header:    t=0 0
Header:    m=audio 25282 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000
Header:    a=fmtp:101 0-11
-----
                          SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.36.180:5060->192.168.6.21:5060]
Header:    Via: SIP/2.0/UDP 192.168.6.21:5060
Header:    From: <sip:5121@192.168.6.21:5060>
Header:    To: <sip:5120@192.168.36.180:5060>
Header:    Call-ID: c2943000-e0563-2alce-2e323931@192.168.6.21
Header:    CSeq: 100 INVITE
Header:    Content-Length: 0
-----
                          SIP Headers
-----
sip-reg:    INVITE sip:5120@192.168.36.200:5060;user=phone SIP/2.0 [192.168.36.180:5060->192.168.36.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.36.180:5060;branch=1
Header:    Via: SIP/2.0/UDP 192.168.6.21:5060
Header:    From: <sip:5121@192.168.6.21:5060>
Header:    To: <sip:5120@192.168.36.180:5060>
Header:    Call-ID: c2943000-e0563-2alce-2e323931@192.168.6.21
Header:    CSeq: 100 INVITE
Header:    Proxy-Authorization: Basic VovidaClassXSwitch
Header:    Expires: 180
Header:    Record-Route: <sip:5120@192.168.36.180:5060;maddr=192.168.36.180>
Header:    Contact: <sip:5121@192.168.6.21:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 219
-----
                          SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 16264 18299 IN IP4 192.168.6.21
Header:    s=SIP Call
Header:    c=IN IP4 192.168.6.21
Header:    t=0 0
Header:    m=audio 25282 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000
```

## SIP IP Phone to SIP IP Phone: Call Setup and Disconnect

Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 302 Moved Temporarily [192.168.36.200:5060->192.168.36.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.6.21:5060  
Header: From: <sip:5121@192.168.6.21:5060>  
Header: To: <sip:5120@192.168.36.180:5060>  
Header: Call-ID: c2943000-e0563-2alce-2e323931@192.168.6.21  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:5120@192.168.36.180:5060>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: ACK sip:5120@192.168.36.200:5060;user=phone SIP/2.0 [192.168.36.180:5060->192.168.36.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=1  
Header: From: <sip:5121@192.168.6.21:5060>  
Header: To: <sip:5120@192.168.36.180:5060>  
Header: Call-ID: c2943000-e0563-2alce-2e323931@192.168.6.21  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:5120@192.168.36.200:5060;user=phone SIP/2.0 [192.168.36.180:5060->192.168.36.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=3  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.6.21:5060  
Header: From: <sip:5121@192.168.6.21:5060>  
Header: To: <sip:5120@192.168.36.180:5060>  
Header: Call-ID: c2943000-e0563-2alce-2e323931@192.168.6.21  
Header: CSeq: 100 INVITE  
Header: Expires: 180  
Header: Record-Route: <sip:5120@192.168.36.180:5060;maddr=192.168.36.180>, <sip:5120@192.168.36.180:5060;maddr=192.168.36.180>  
Header: Contact: <sip:5121@192.168.6.21:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 219

-----  
SDP Headers

Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 16264 18299 IN IP4 192.168.6.21  
Header: s=SIP Call  
Header: c=IN IP4 192.168.6.21  
Header: t=0 0  
Header: m=audio 25282 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 302 Moved Temporarily [192.168.36.200:5060->192.168.36.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=3  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.6.21:5060  
Header: From: <sip:5121@192.168.6.21:5060>  
Header: To: <sip:5120@192.168.36.180:5060>  
Header: Call-ID: c2943000-e0563-2alce-2e323931@192.168.6.21  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:5120@192.168.6.20:5060>  
Header: Content-Length: 0

-----  
SIP Headers

## SIP IP Phone to SIP IP Phone: Call Setup and Disconnect

```
-----  
sip-req:   ACK sip:5120@192.168.36.200:5060;user=phone SIP/2.0 [192.168.36.180:5060-  
>192.168.36.200:5060]
```

```
Header:   Via: SIP/2.0/UDP 192.168.36.180:5060;branch=3  
Header:   From: <sip:5121@192.168.6.21:5060>  
Header:   To: <sip:5120@192.168.36.180:5060>  
Header:   Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21  
Header:   CSeq: 100 ACK  
Header:   Content-Length: 0  
-----
```

### SIP Headers

```
-----  
sip-req:   INVITE sip:5120@192.168.6.20:5060 SIP/2.0 [192.168.36.180:5060-  
>192.168.6.20:5060]
```

```
Header:   Via: SIP/2.0/UDP 192.168.36.180:5060;branch=4  
Header:   Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2  
Header:   Via: SIP/2.0/UDP 192.168.6.21:5060  
Header:   From: <sip:5121@192.168.6.21:5060>  
Header:   To: <sip:5120@192.168.36.180:5060>  
Header:   Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21  
Header:   CSeq: 100 INVITE  
Header:   Expires: 180  
Header:   Record-Route:  
<sip:5120@192.168.36.180:5060;maddr=192.168.36.180>,<sip:5120@192.168.36.180:5060;maddr=1  
92.168.36.180>  
Header:   Contact: <sip:5121@192.168.6.21:5060>  
Header:   Content-Type: application/sdp  
Header:   Content-Length: 219  
-----
```

### SDP Headers

```
-----  
Header:   v=0  
Header:   o=CiscoSystemsSIP-IPPhone-UserAgent 16264 18299 IN IP4 192.168.6.21  
Header:   s=SIP Call  
Header:   c=IN IP4 192.168.6.21  
Header:   t=0 0  
Header:   m=audio 25282 RTP/AVP 0 101  
Header:   a=rtpmap:0 pcmu/8000  
Header:   a=rtpmap:101 telephone-event/8000  
Header:   a=fmtp:101 0-11  
-----
```

### SIP Headers

```
-----  
sip-res:   SIP/2.0 100 Trying [192.168.6.20:50753->192.168.36.180:5060]  
Header:   Via: SIP/2.0/UDP 192.168.36.180:5060;branch=4,SIP/2.0/UDP  
192.168.36.180:5060;branch=2,SIP/2.0/UDP 192.168.6.21:5060  
Header:   From: <sip:5121@192.168.6.21:5060>  
Header:   To: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0  
Header:   Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21  
Header:   Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header:   CSeq: 100 INVITE  
Header:   Content-Length: 0  
-----
```

### SIP Headers

```
-----  
sip-res:   SIP/2.0 180 Ringing [192.168.6.20:50753->192.168.36.180:5060]  
Header:   Via: SIP/2.0/UDP 192.168.36.180:5060;branch=4,SIP/2.0/UDP  
192.168.36.180:5060;branch=2,SIP/2.0/UDP 192.168.6.21:5060  
Header:   From: <sip:5121@192.168.6.21:5060>  
Header:   To: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0  
Header:   Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21  
Header:   Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header:   CSeq: 100 INVITE  
Header:   Content-Length: 0  
-----
```

### SIP Headers

```
-----  
sip-res:   SIP/2.0 180 Ringing [192.168.36.180:5060->192.168.6.21:5060]  
Header:   Via: SIP/2.0/UDP 192.168.6.21:5060  
Header:   From: <sip:5121@192.168.6.21:5060>  
-----
```

## SIP IP Phone to SIP IP Phone: Call Setup and Disconnect

```
Header:      To: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0
Header:      Call-ID: c2943000-e0563-2alce-2e323931@192.168.6.21
Header:      CSeq: 100 INVITE
Header:      Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:      Content-Length: 0
-----
SIP Headers
-----
sip-res:     SIP/2.0 200 OK [192.168.6.20:50753->192.168.36.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.36.180:5060;branch=4,SIP/2.0/UDP
192.168.36.180:5060;branch=2,SIP/2.0/UDP 192.168.6.21:5060
Header:      From: <sip:5121@192.168.6.21:5060>
Header:      To: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0
Header:      Call-ID: c2943000-e0563-2alce-2e323931@192.168.6.21
Header:      Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:      Contact: sip:5120@192.168.6.20:5060
Header:      Record-Route:
<sip:5120@192.168.36.180:5060;maddr=192.168.36.180>,<sip:5120@192.168.36.180:5060;maddr=1
92.168.36.180>
Header:      CSeq: 100 INVITE
Header:      Content-Type: application/sdp
Header:      Content-Length: 218
-----
SDP Headers
-----
Header:      v=0
Header:      o=CiscoSystemsSIP-IPPhone-UserAgent 13045 2886 IN IP4 192.168.6.20
Header:      s=SIP Call
Header:      c=IN IP4 192.168.6.20
Header:      t=0 0
Header:      m=audio 30658 RTP/AVP 0 101
Header:      a=rtpmap:0 pcmu/8000
Header:      a=rtpmap:101 telephone-event/8000
Header:      a=fmtp:101 0-11
-----
SIP Headers
-----
sip-res:     SIP/2.0 200 OK [192.168.36.180:5060->192.168.6.21:5060]
Header:      Via: SIP/2.0/UDP 192.168.6.21:5060
Header:      From: <sip:5121@192.168.6.21:5060>
Header:      To: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0
Header:      Call-ID: c2943000-e0563-2alce-2e323931@192.168.6.21
Header:      CSeq: 100 INVITE
Header:      Contact: <sip:5120@192.168.6.20:5060>
Header:      Record-Route:
<sip:5120@192.168.36.180:5060;maddr=192.168.36.180>,<sip:5120@192.168.36.180:5060;maddr=1
92.168.36.180>
Header:      Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:      Content-Type: application/sdp
Header:      Content-Length: 218
-----
SDP Headers
-----
Header:      v=0
Header:      o=CiscoSystemsSIP-IPPhone-UserAgent 13045 2886 IN IP4 192.168.6.20
Header:      s=SIP Call
Header:      c=IN IP4 192.168.6.20
Header:      t=0 0
Header:      m=audio 30658 RTP/AVP 0 101
Header:      a=rtpmap:0 pcmu/8000
Header:      a=rtpmap:101 telephone-event/8000
Header:      a=fmtp:101 0-11
-----
SIP Headers
-----
sip-req:     ACK sip:5120@192.168.36.180:5060 SIP/2.0      [192.168.6.21:50623-
>192.168.36.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.6.21:5060
Header:      From: <sip:5121@192.168.6.21:5060>
Header:      To: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0
```

## SIP IP Phone to SIP IP Phone: Call Setup and Disconnect

Header: Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21  
Header: Route: <sip:5120@192.168.36.180:5060;maddr=192.168.36.180>,  
<sip:5120@192.168.6.20:5060>  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: ACK sip:5120@192.168.6.20:5060 SIP/2.0 [192.168.36.180:5060-  
>192.168.6.20:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.6.21:5060  
Header: From: <sip:5121@192.168.6.21:5060>  
Header: To: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0  
Header: Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: BYE sip:5120@192.168.36.180:5060 SIP/2.0 [192.168.6.20:50753-  
>192.168.36.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.6.20:5060  
Header: From: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0  
Header: To: <sip:5121@192.168.6.21:5060>  
Header: Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21  
Header: User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: CSeq: 101 BYE  
Header: Route: <sip:5120@192.168.36.180:5060;maddr=192.168.36.180>,  
<sip:5121@192.168.6.21:5060>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: BYE sip:5121@192.168.6.21:5060 SIP/2.0 [192.168.36.180:5060-  
>192.168.6.21:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.6.20:5060  
Header: From: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0  
Header: To: <sip:5121@192.168.6.21:5060>  
Header: Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21  
Header: CSeq: 101 BYE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.6.21:50623->192.168.36.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=4,SIP/2.0/UDP  
192.168.36.180:5060;branch=2,SIP/2.0/UDP 192.168.6.20:5060  
Header: From: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0  
Header: To: <sip:5121@192.168.6.21:5060>  
Header: Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: CSeq: 101 BYE  
Header: Content-Length: 0

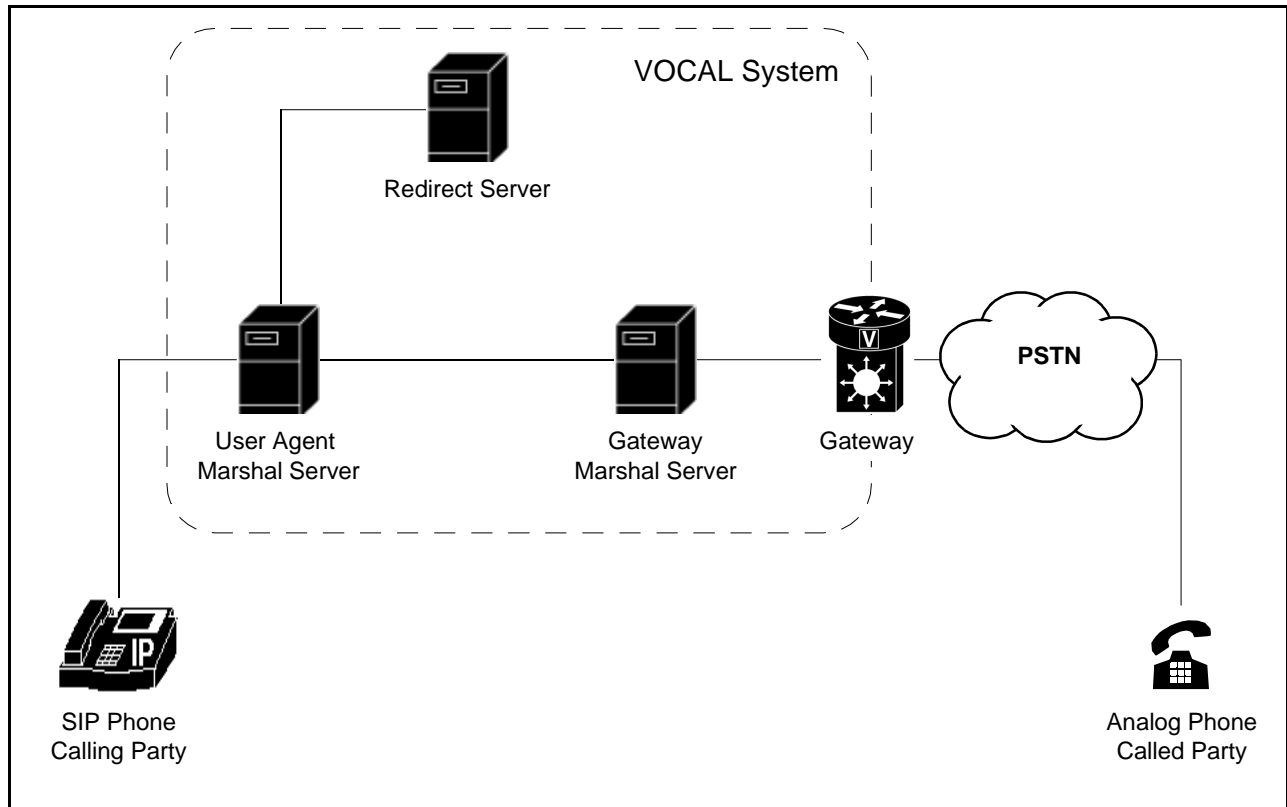
-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.36.180:5060->192.168.6.20:5060]  
Header: Via: SIP/2.0/UDP 192.168.6.20:5060  
Header: From: <sip:5120@192.168.36.180:5060>;tag=c29430002e0620-0  
Header: To: <sip:5121@192.168.6.21:5060>  
Header: Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21  
Header: CSeq: 101 BYE  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: Content-Length: 0

## SIP IP Phone to Analog Phone via Gateway

**Call Scenario**

Figure C-6 illustrates a SIP phone to analog phone call made over an IP network via a gateway.

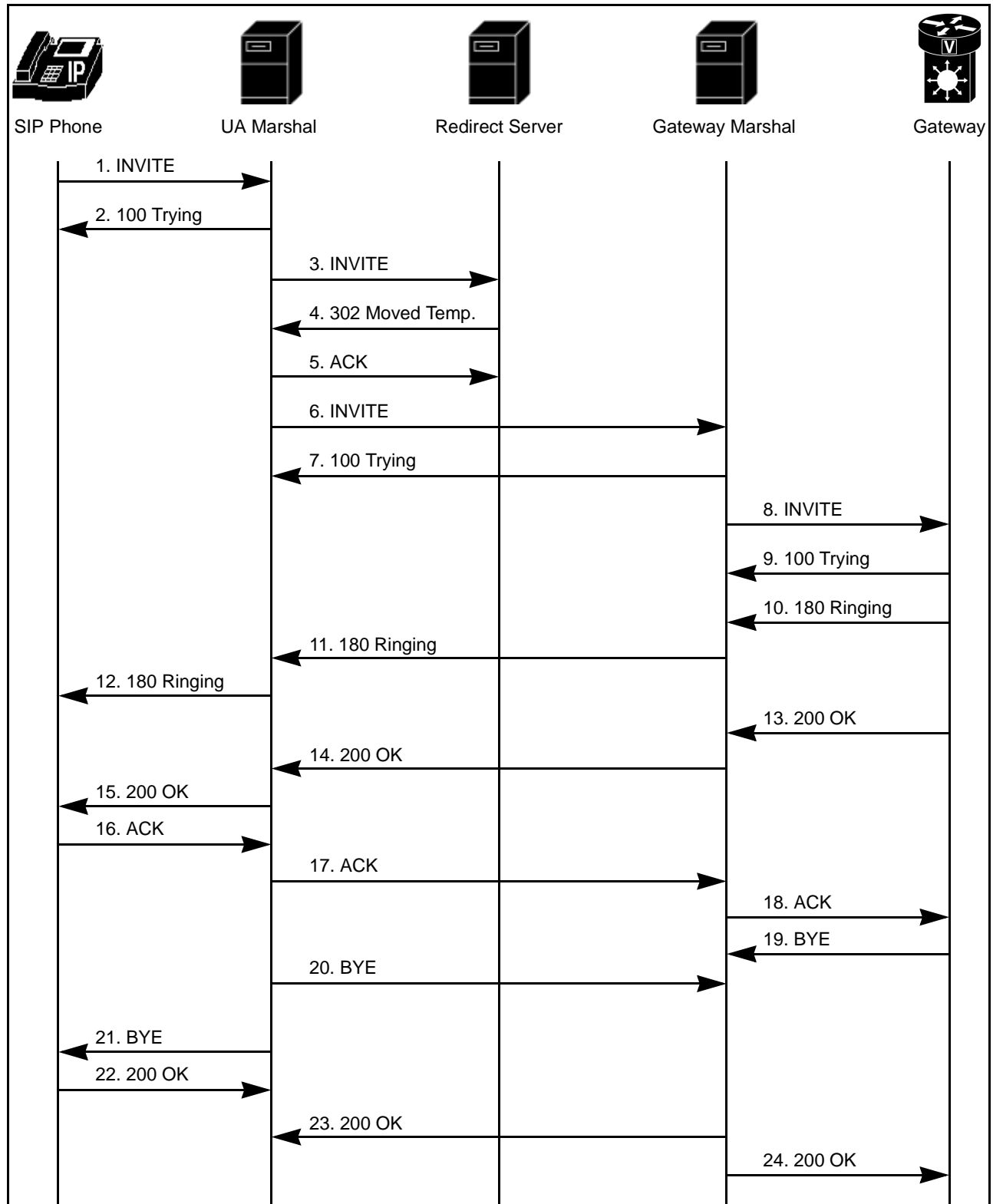


**Figure C-6. SIP IP Phone to Analog Phone via Gateway**

**Call Flow Diagram**

Figures C-7 and C-7 show a call originating from an on-network SIP phone and being routed through a gateway to the PSTN.

## SIP IP Phone to Analog Phone via Gateway



**Figure C-7. Call Flow Diagram: SIP IP Phone to SIP IP Phone via SIP Gateway**



Call Trace

The following trace shows a call originating from an on-network SIP phone and being routed through a gateway to the PSTN.

```

-----
                        SIP Headers
-----
sip-reg:    INVITE sip:93831073@192.168.36.180 SIP/2.0    [192.168.6.20:50753->192.168.36.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.6.20:5060
Header:    From: sip:5120@192.168.6.20
Header:    To: <sip:93831073@192.168.36.180>
Header:    Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header:    CSeq: 100 INVITE
Header:    Expires: 180
Header:    User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:    Accept: application/sdp
Header:    Contact: sip:5120@192.168.6.20:5060
Header:    Content-Type: application/sdp
Header:    Content-Length: 218
-----
                        SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 21012 9466 IN IP4 192.168.6.20
Header:    s=SIP Call
Header:    c=IN IP4 192.168.6.20
Header:    t=0 0
Header:    m=audio 25776 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000
Header:    a=fmtp:101 0-11
-----
                        SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.36.180:5060->192.168.6.20:5060]
Header:    Via: SIP/2.0/UDP 192.168.6.20:5060
Header:    From: <sip:5120@192.168.6.20:5060>
Header:    To: <sip:93831073@192.168.36.180:5060>
Header:    Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header:    CSeq: 100 INVITE
Header:    Content-Length: 0
-----
                        SIP Headers
-----
sip-reg:    INVITE sip:93831073@192.168.36.200:5060;user=phone SIP/2.0
[192.168.36.180:5060->192.168.36.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.36.180:5060;branch=1
Header:    Via: SIP/2.0/UDP 192.168.6.20:5060
Header:    From: <sip:5120@192.168.6.20:5060>
Header:    To: <sip:93831073@192.168.36.180:5060>
Header:    Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header:    CSeq: 100 INVITE
Header:    Proxy-Authorization: Basic VovidaClassXSwitch
Header:    Expires: 180
Header:    Record-Route:
<sip:93831073@192.168.36.180:5060;maddr=192.168.36.180>
Header:    Contact: <sip:5120@192.168.6.20:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 218
-----
                        SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 21012 9466 IN IP4 192.168.6.20
Header:    s=SIP Call
Header:    c=IN IP4 192.168.6.20
Header:    t=0 0
Header:    m=audio 25776 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000

```

## SIP IP Phone to Analog Phone via Gateway

Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers  
-----

sip-res: SIP/2.0 302 Moved Temporarily [192.168.36.200:5060->192.168.36.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.6.20:5060  
Header: From: <sip:5120@192.168.6.20:5060>  
Header: To: <sip:93831073@192.168.36.180:5060>  
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:93831073@192.168.36.110:5060;user=phone>  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: ACK sip:93831073@192.168.36.200:5060;user=phone SIP/2.0 [192.168.36.180:5060->192.168.36.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=1  
Header: From: <sip:5120@192.168.6.20:5060>  
Header: To: <sip:93831073@192.168.36.180:5060>  
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: INVITE sip:93831073@192.168.36.110:5060;user=phone SIP/2.0 [192.168.36.180:5060->192.168.36.110:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.6.20:5060  
Header: From: <sip:5120@192.168.6.20:5060>  
Header: To: <sip:93831073@192.168.36.180:5060>  
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20  
Header: CSeq: 100 INVITE  
Header: Proxy-Authorization: Basic VovidaClassXSwitch  
Header: Expires: 180  
Header: Record-Route:  
<sip:93831073@192.168.36.180:5060;maddr=192.168.36.180>  
Header: Contact: <sip:5120@192.168.6.20:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 218

-----  
SDP Headers  
-----

Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 21012 9466 IN IP4 192.168.6.20  
Header: s=SIP Call  
Header: c=IN IP4 192.168.6.20  
Header: t=0 0  
Header: m=audio 25776 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers  
-----

sip-res: SIP/2.0 100 Trying [192.168.36.110:5060->192.168.36.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.6.20:5060  
Header: From: <sip:5120@192.168.6.20:5060>  
Header: To: <sip:93831073@192.168.36.180:5060>  
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20  
Header: CSeq: 100 INVITE  
Header: Content-Length: 0

-----  
SIP Headers  
-----

## SIP IP Phone to Analog Phone via Gateway

```
sip-req: INVITE sip:93831073@192.168.16.210:5060;user=phone SIP/2.0
[192.168.36.110:5060->192.168.16.210:5060]
Header: Via: SIP/2.0/UDP 192.168.36.110:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.6.20:5060
Header: From: <sip:5120@192.168.6.20:5060>
Header: To: <sip:93831073@192.168.36.180:5060>
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header: CSeq: 100 INVITE
Header: Expires: 180
Header: Record-Route:
<sip:93831073@192.168.36.110:5060;maddr=192.168.36.110>,<sip:93831073@192.168.36.180:5060
;maddr=192.168.36.180>
Header: Contact: <sip:5120@192.168.6.20:5060>
Header: Content-Type: application/sdp
Header: Content-Length: 218
```

### SDP Headers

```
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 21012 9466 IN IP4 192.168.6.20
Header: s=SIP Call
Header: c=IN IP4 192.168.6.20
Header: t=0 0
Header: m=audio 25776 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

### SIP Headers

```
sip-res: SIP/2.0 100 Trying [192.168.16.210:5060->192.168.36.110:5060]
Header: Via: SIP/2.0/UDP 192.168.36.110:5060;branch=4,SIP/2.0/UDP
192.168.36.180:5060;branch=2,SIP/2.0/UDP 192.168.6.20:5060
Header: From: <sip:5120@192.168.6.20:5060>
Header: To: <sip:93831073@192.168.36.180:5060>
Header: Date:
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header: Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header: CSeq: 100 INVITE
Header: Content-Length: 0
```

### SIP Headers

```
sip-res: SIP/2.0 180 Ringing [192.168.16.210:5060->192.168.36.110:5060]
Header: Via: SIP/2.0/UDP 192.168.36.110:5060;branch=4,SIP/2.0/UDP
192.168.36.180:5060;branch=2,SIP/2.0/UDP 192.168.6.20:5060
Header: From: <sip:5120@192.168.6.20:5060>
Header: To: <sip:93831073@192.168.36.180:5060>
Header: Date:
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header: Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header: CSeq: 100 INVITE
Header: Content-Type: application/sdp
Header: Content-Length: 138
```

### SDP Headers

```
Header: v=0
Header: o=CiscoSystemsSIP-GW-UserAgent 3721 2232 IN IP4 192.168.16.210
Header: s=SIP Call
Header: c=IN IP4 192.168.16.210
Header: t=0 0
Header: m=audio 20334 RTP/AVP 0
```

### SIP Headers

```
sip-res: SIP/2.0 180 Ringing [192.168.36.110:5060-
>192.168.36.180:5060]
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2
```

## SIP IP Phone to Analog Phone via Gateway

```
Header: Via: SIP/2.0/UDP 192.168.6.20:5060
Header: From: <sip:5120@192.168.6.20:5060>
Header: To: <sip:93831073@192.168.36.180:5060>
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header: CSeq: 100 INVITE
Header: Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header: Content-Type: application/sdp
Header: Content-Length: 138
-----
SDP Headers
-----
Header: v=0
Header: o=CiscoSystemsSIP-GW-UserAgent 3721 2232 IN IP4 192.168.16.210
Header: s=SIP Call
Header: c=IN IP4 192.168.16.210
Header: t=0 0
Header: m=audio 20334 RTP/AVP 0
-----
SIP Headers
-----
sip-res: SIP/2.0 180 Ringing [192.168.36.180:5060->192.168.6.20:5060]
Header: Via: SIP/2.0/UDP 192.168.6.20:5060
Header: From: <sip:5120@192.168.6.20:5060>
Header: To: <sip:93831073@192.168.36.180:5060>
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header: CSeq: 100 INVITE
Header: Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header: Content-Type: application/sdp
Header: Content-Length: 138
-----
SDP Headers
-----
Header: v=0
Header: o=CiscoSystemsSIP-GW-UserAgent 3721 2232 IN IP4 192.168.16.210
Header: s=SIP Call
Header: c=IN IP4 192.168.16.210
Header: t=0 0
Header: m=audio 20334 RTP/AVP 0
-----
SIP Headers
-----
sip-res: SIP/2.0 200 OK [192.168.16.210:5060->192.168.36.110:5060]
Header: Via: SIP/2.0/UDP 192.168.36.110:5060;branch=4,SIP/2.0/UDP
192.168.36.180:5060;branch=2,SIP/2.0/UDP 192.168.6.20:5060
Header: From: <sip:5120@192.168.6.20:5060>
Header: To: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50
Header: Date:
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header: Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header: Contact: <sip:93831073@192.168.16.210:5060;user=phone>
Header: Record-Route:
<sip:93831073@192.168.36.110:5060;maddr=192.168.36.110>,<sip:93831073@192.168.36.180:5060
;maddr=192.168.36.180>
Header: CSeq: 100 INVITE
Header: Content-Type: application/sdp
Header: Content-Length: 138
-----
SDP Headers
-----
Header: v=0
Header: o=CiscoSystemsSIP-GW-UserAgent 3721 2232 IN IP4 192.168.16.210
Header: s=SIP Call
Header: c=IN IP4 192.168.16.210
Header: t=0 0
Header: m=audio 20334 RTP/AVP 0
-----
SIP Headers
-----
sip-res: SIP/2.0 200 OK [192.168.36.110:5060-
>192.168.36.180:5060]
```

## SIP IP Phone to Analog Phone via Gateway

```
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.6.20:5060
Header: From: <sip:5120@192.168.6.20:5060>
Header: To: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header: CSeq: 100 INVITE
Header: Contact: <sip:93831073@192.168.16.210:5060;user=phone>
Header: Record-Route:
<sip:93831073@192.168.36.110:5060;maddr=192.168.36.110>,<sip:93831073@192.168.36.180:5060
;maddr=192.168.36.180>
Header: Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header: Content-Type: application/sdp
Header: Content-Length: 138
```

### ----- SDP Headers

```
Header: v=0
Header: o=CiscoSystemsSIP-GW-UserAgent 3721 2232 IN IP4 192.168.16.210
Header: s=SIP Call
Header: c=IN IP4 192.168.16.210
Header: t=0 0
Header: m=audio 20334 RTP/AVP 0
```

### ----- SIP Headers

```
sip-res: SIP/2.0 200 OK [192.168.36.180:5060->192.168.6.20:5060]
Header: Via: SIP/2.0/UDP 192.168.6.20:5060
Header: From: <sip:5120@192.168.6.20:5060>
Header: To: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header: CSeq: 100 INVITE
Header: Contact: <sip:93831073@192.168.16.210:5060;user=phone>
Header: Record-Route:
<sip:93831073@192.168.36.110:5060;maddr=192.168.36.110>,<sip:93831073@192.168.36.180:5060
;maddr=192.168.36.180>
Header: Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header: Content-Type: application/sdp
Header: Content-Length: 138
```

### ----- SDP Headers

```
Header: v=0
Header: o=CiscoSystemsSIP-GW-UserAgent 3721 2232 IN IP4 192.168.16.210
Header: s=SIP Call
Header: c=IN IP4 192.168.16.210
Header: t=0 0
Header: m=audio 20334 RTP/AVP 0
```

### ----- SIP Headers

```
sip-req: ACK sip:93831073@192.168.36.180:5060 SIP/2.0 [192.168.6.20:50753-
>192.168.36.180:5060]
Header: Via: SIP/2.0/UDP 192.168.6.20:5060
Header: From: <sip:5120@192.168.6.20:5060>
Header: To: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header: Route: <sip:93831073@192.168.36.110:5060;maddr=192.168.36.110>,<
sip:93831073@192.168.16.210:5060;user=phone>
Header: CSeq: 100 ACK
Header: Content-Length: 0
```

### ----- SIP Headers

```
sip-req: ACK sip:93831073@192.168.36.110:5060;maddr=192.168.36.110 SIP/2.0
[192.168.36.180:5060->192.168.36.110:5060]
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.6.20:5060
Header: From: <sip:5120@192.168.6.20:5060>
Header: To: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
```

## SIP IP Phone to Analog Phone via Gateway

Header: CSeq: 100 ACK  
Header: Route: <sip:93831073@192.168.16.210:5060>  
Header: Proxy-Authorization: Basic VovidaClassXSwitch  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: ACK sip:93831073@192.168.16.210:5060 SIP/2.0 [192.168.36.110:5060->192.168.16.210:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.110:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.6.20:5060  
Header: From: <sip:5120@192.168.6.20:5060>  
Header: To: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50  
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: BYE sip:93831073@192.168.36.110:5060 SIP/2.0 [192.168.16.210:50110->192.168.36.110:5060]  
Header: Via: SIP/2.0/UDP 192.168.16.210:50110  
Header: From: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50  
Header: To: <sip:5120@192.168.6.20:5060>  
Header: Date:  
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20  
Header: User-Agent: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled  
Header: Max-Forwards: 6  
Header: Route: <sip:93831073@192.168.36.180:5060;maddr=192.168.36.180>,  
<sip:5120@192.168.6.20:5060>  
Header: Timestamp: 947137048  
Header: CSeq: 101 BYE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: BYE sip:93831073@192.168.36.180:5060;maddr=192.168.36.180 SIP/2.0 [192.168.36.110:5060->192.168.36.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.110:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.16.210:50110  
Header: From: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50  
Header: To: <sip:5120@192.168.6.20:5060>  
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20  
Header: CSeq: 101 BYE  
Header: Timestamp: 947137048  
Header: Max-Forwards: 5  
Header: Proxy-Authorization: Basic VovidaClassXSwitch  
Header: Route: <sip:5120@192.168.6.20:5060>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: BYE sip:5120@192.168.6.20:5060 SIP/2.0 [192.168.36.180:5060->192.168.6.20:5060]  
Header: Via: SIP/2.0/UDP 192.168.36.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.36.110:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.16.210:50110  
Header: From: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50  
Header: To: <sip:5120@192.168.6.20:5060>  
Header: Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20  
Header: CSeq: 101 BYE  
Header: Timestamp: 947137048  
Header: Max-Forwards: 5  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.6.20:50753->192.168.36.180:5060]

## SIP IP Phone to Analog Phone via Gateway

```
Header:          Via: SIP/2.0/UDP 192.168.36.180:5060;branch=4,SIP/2.0/UDP
192.168.36.110:5060;branch=2,SIP/2.0/UDP 192.168.16.210:50110
Header:          From: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50
Header:          To: <sip:5120@192.168.6.20:5060>
Header:          Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header:          Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:          CSeq: 101 BYE
Header:          Content-Length: 0
```

---

### SIP Headers

```
-----
sip-res:        SIP/2.0 200 OK [192.168.36.180:5060->192.168.36.110:5060]
Header:         Via: SIP/2.0/UDP 192.168.36.110:5060;branch=2
Header:         Via: SIP/2.0/UDP 192.168.16.210:50110
Header:         From: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50
Header:         To: <sip:5120@192.168.6.20:5060>
Header:         Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header:         CSeq: 101 BYE
Header:         Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:         Content-Length: 0
```

---

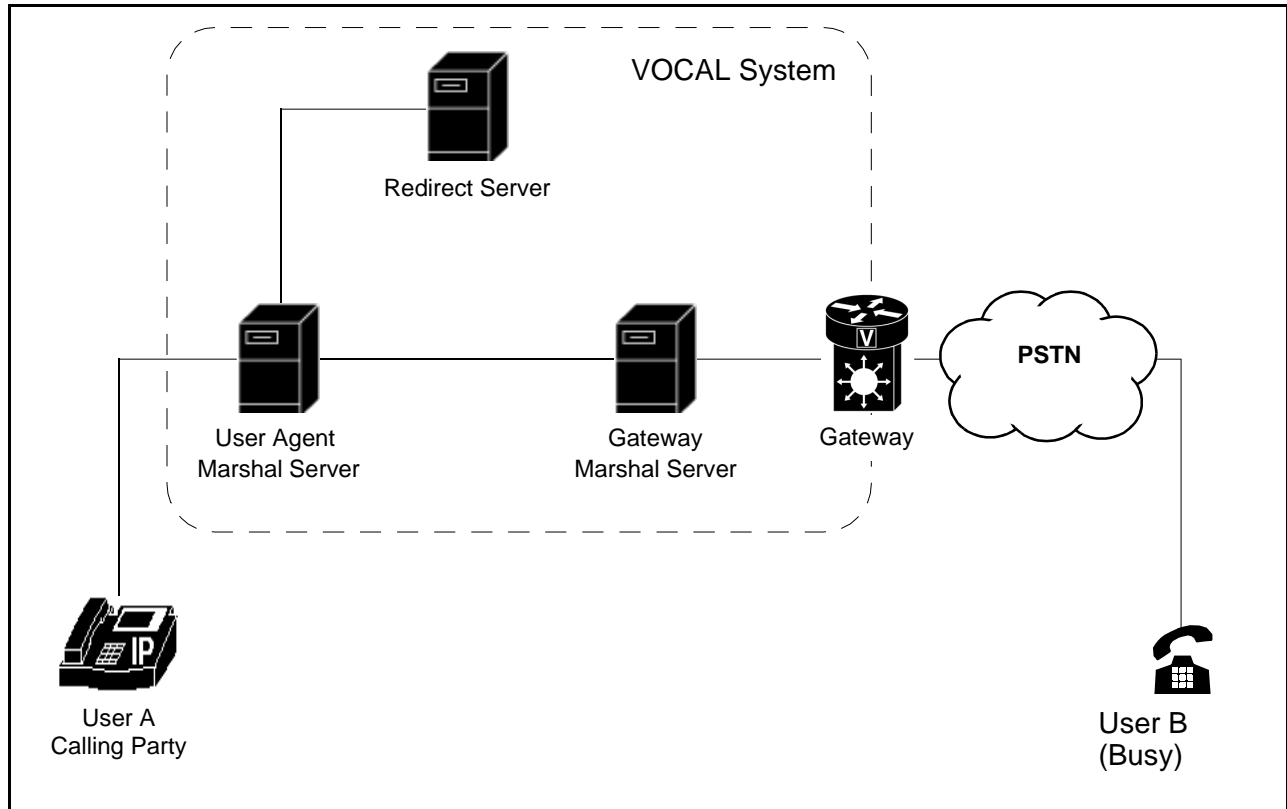
### SIP Headers

```
-----
sip-res:        SIP/2.0 200 OK [192.168.36.110:5060->192.168.16.210:50110]
Header:         Via: SIP/2.0/UDP 192.168.16.210:50110
Header:         From: <sip:93831073@192.168.36.180:5060>;tag=1AF49448-1D50
Header:         To: <sip:5120@192.168.6.20:5060>
Header:         Call-ID: c2943000-23e062-2e278-2e323931@192.168.6.20
Header:         CSeq: 101 BYE
Header:         Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:         Content-Length: 0
-----
```

---

## SIP Phone to Phone via Gateway: Called Party is Busy

**Call Scenario** Figure C-8 illustrates User A initiating a call to User B while User B is busy.

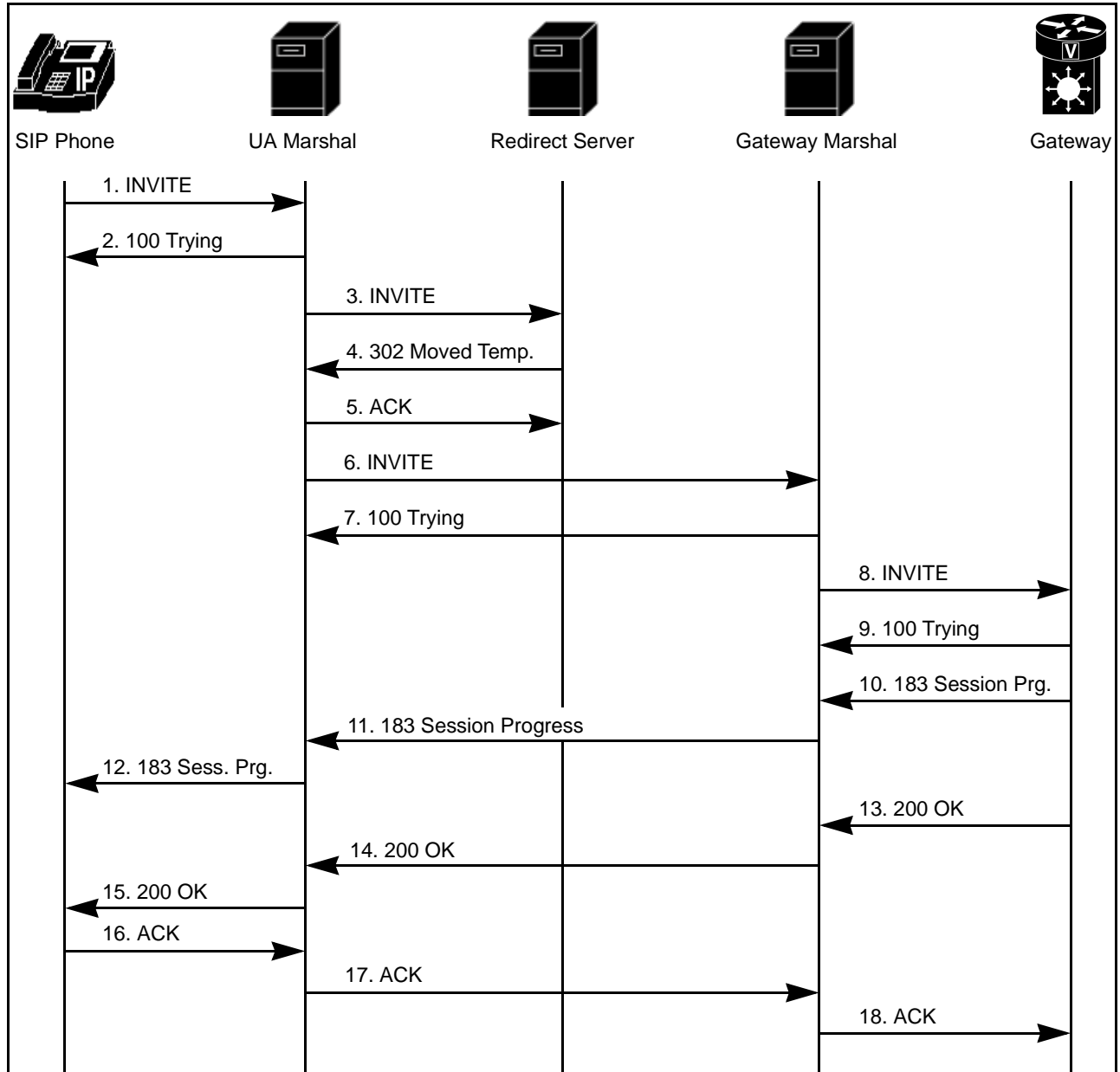


**Figure C-8. SIP Phone to Phone: Called Party is Busy**

**Call Flow Diagram** Figure C-9 shows a call originating from an on-network SIP phone, being routed through a gateway to the PSTN, and returning a busy signal.



*SIP Phone to Phone via Gateway: Called Party is Busy*



**Figure C-9. Call Flow Diagram: SIP Phone to Phone: Called Party is Busy**

## SIP Phone to Phone via Gateway: Called Party is Busy

### Call Trace

The following call trace shows a call originating from an on-network SIP phone, being routed through a gateway to the PSTN, and returning a busy signal.

```
-----
                          SIP Headers
-----
sip-reg:    INVITE sip:93831069@192.168.26.180 SIP/2.0    [192.168.26.10:50373-
>192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.10:5060
Header:    From: sip:6711@192.168.26.10
Header:    To: <sip:93831069@192.168.26.180>
Header:    Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:    CSeq: 100 INVITE
Header:    Expires: 180
Header:    User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:    Accept: application/sdp
Header:    Contact: sip:6711@192.168.26.10:5060
Header:    Content-Type: application/sdp
Header:    Content-Length: 221
-----
                          SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 16302 19778 IN IP4
192.168.26.10
Header:    s=SIP Call
Header:    c=IN IP4 192.168.26.10
Header:    t=0 0
Header:    m=audio 26268 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000
Header:    a=fmtp:101 0-11
-----
                          SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.26.180:5060->192.168.26.10:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.10:5060
Header:    From: <sip:6711@192.168.26.10:5060>
Header:    To: <sip:93831069@192.168.26.180:5060>
Header:    Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:    CSeq: 100 INVITE
Header:    Content-Length: 0
-----
                          SIP Headers
-----
sip-reg:    INVITE sip:93831069@192.168.26.200:5060;user=phone SIP/2.0
[192.168.26.180:5060->192.168.26.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1
Header:    Via: SIP/2.0/UDP 192.168.26.10:5060
Header:    From: <sip:6711@192.168.26.10:5060>
Header:    To: <sip:93831069@192.168.26.180:5060>
Header:    Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:    CSeq: 100 INVITE
Header:    Proxy-Authorization: Basic 123
Header:    Expires: 180
Header:    Record-Route:
<sip:93831069@192.168.26.180:5060;maddr=192.168.26.180>
Header:    Contact: <sip:6711@192.168.26.10:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 221
-----
                          SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 16302 19778 IN IP4
192.168.26.10
Header:    s=SIP Call
```

## SIP Phone to Phone via Gateway: Called Party is Busy

Header: c=IN IP4 192.168.26.10  
Header: t=0 0  
Header: m=audio 26268 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:93831069@192.168.26.180:5060>  
Header: Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:93831069@192.168.26.110:5060;user=phone>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: ACK sip:93831069@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:93831069@192.168.26.180:5060>  
Header: Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:93831069@192.168.26.110:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.110:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:93831069@192.168.26.180:5060>  
Header: Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Proxy-Authorization: Basic 123  
Header: Expires: 180  
Header: Record-Route:  
<sip:93831069@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:6711@192.168.26.10:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 221

-----  
SDP Headers

-----  
Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 16302 19778 IN IP4 192.168.26.10  
Header: s=SIP Call  
Header: c=IN IP4 192.168.26.10  
Header: t=0 0  
Header: m=audio 26268 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.26.110:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:93831069@192.168.26.180:5060>  
Header: Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE

## SIP Phone to Phone via Gateway: Called Party is Busy

```
Header:                               Content-Length: 0
-----
SIP Headers
-----
sip-req:    INVITE sip:93831069@192.168.16.210:5060;user=phone SIP/2.0
[192.168.26.110:5060->192.168.16.210:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.110:5060;branch=4
Header:     Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:     Via: SIP/2.0/UDP 192.168.26.10:5060
Header:     From: <sip:6711@192.168.26.10:5060>
Header:     To: <sip:93831069@192.168.26.180:5060>
Header:     Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:     CSeq: 100 INVITE
Header:     Expires: 180
Header:     Record-Route:
<sip:93831069@192.168.26.110:5060;maddr=192.168.26.110>,<sip:93831069@192.168.26.180:5060
;maddr=192.168.26.180>
Header:     Contact: <sip:6711@192.168.26.10:5060>
Header:     Content-Type: application/sdp
Header:     Content-Length: 221
-----
SDP Headers
-----
Header:     v=0
Header:     o=CiscoSystemsSIP-IPPhone-UserAgent 16302 19778 IN IP4
192.168.26.10
Header:     s=SIP Call
Header:     c=IN IP4 192.168.26.10
Header:     t=0 0
Header:     m=audio 26268 RTP/AVP 0 101
Header:     a=rtpmap:0 pcmu/8000
Header:     a=rtpmap:101 telephone-event/8000
Header:     a=fmtp:101 0-11
-----
SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.16.210:5060->192.168.26.110:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.110:5060;branch=4,SIP/2.0/UDP
192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.10:5060
Header:     From: <sip:6711@192.168.26.10:5060>
Header:     To: <sip:93831069@192.168.26.180:5060>
Header:     Date:
Header:     Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:     Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header:     CSeq: 100 INVITE
Header:     Content-Length: 0
-----
SIP Headers
-----
sip-res:    SIP/2.0 183 Session Progress [192.168.16.210:5060->192.168.26.110:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.110:5060;branch=4,SIP/2.0/UDP
192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.10:5060
Header:     From: <sip:6711@192.168.26.10:5060>
Header:     To: <sip:93831069@192.168.26.180:5060>
Header:     Date:
Header:     Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:     Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header:     CSeq: 100 INVITE
Header:     Content-Type: application/sdp
Header:     Session: Media
Header:     Content-Length: 138
-----
SDP Headers
-----
Header:     v=0
Header:     o=CiscoSystemsSIP-GW-UserAgent 3804 4963 IN IP4 192.168.16.210
Header:     s=SIP Call
Header:     c=IN IP4 192.168.16.210
Header:     t=0 0
Header:     m=audio 20334 RTP/AVP 0
```

## SIP Phone to Phone via Gateway: Called Party is Busy

```
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 183 Session Progress [192.168.26.110:5060->192.168.26.180:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:     Via: SIP/2.0/UDP 192.168.26.10:5060
Header:     From: <sip:6711@192.168.26.10:5060>
Header:     To: <sip:93831069@192.168.26.180:5060>
Header:     Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:     CSeq: 100 INVITE
Header:     Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header:     Content-Type: application/sdp
Header:     Content-Length: 138
-----
                                SDP Headers
-----
Header:     v=0
Header:     o=CiscoSystemsSIP-GW-UserAgent 3804 4963 IN IP4 192.168.16.210
Header:     s=SIP Call
Header:     c=IN IP4 192.168.16.210
Header:     t=0 0
Header:     m=audio 20334 RTP/AVP 0
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 183 Session Progress [192.168.26.180:5060->192.168.26.10:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.10:5060
Header:     From: <sip:6711@192.168.26.10:5060>
Header:     To: <sip:93831069@192.168.26.180:5060>
Header:     Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:     CSeq: 100 INVITE
Header:     Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header:     Content-Type: application/sdp
Header:     Content-Length: 138
-----
                                SDP Headers
-----
Header:     v=0
Header:     o=CiscoSystemsSIP-GW-UserAgent 3804 4963 IN IP4 192.168.16.210
Header:     s=SIP Call
Header:     c=IN IP4 192.168.16.210
Header:     t=0 0
Header:     m=audio 20334 RTP/AVP 0
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.16.210:5060->192.168.26.110:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.110:5060;branch=4,SIP/2.0/UDP
192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.10:5060
Header:     From: <sip:6711@192.168.26.10:5060>
Header:     To: <sip:93831069@192.168.26.180:5060>;tag=25A5AD44-1FE9
Header:     Date:
Header:     Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:     Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header:     Contact: <sip:93831069@192.168.16.210:5060;user=phone>
Header:     Record-Route:
<sip:93831069@192.168.26.110:5060;maddr=192.168.26.110>,<sip:93831069@192.168.26.180:5060
;maddr=192.168.26.180>
Header:     CSeq: 100 INVITE
Header:     Content-Type: application/sdp
Header:     Content-Length: 138
-----
                                SDP Headers
-----
Header:     v=0
Header:     o=CiscoSystemsSIP-GW-UserAgent 3804 4963 IN IP4 192.168.16.210
Header:     s=SIP Call
Header:     c=IN IP4 192.168.16.210
Header:     t=0 0
Header:     m=audio 20334 RTP/AVP 0
```

## SIP Phone to Phone via Gateway: Called Party is Busy

```
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.26.110:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.26.10:5060
Header:    From: <sip:6711@192.168.26.10:5060>
Header:    To: <sip:93831069@192.168.26.180:5060>;tag=25A5AD44-1FE9
Header:    Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:    CSeq: 100 INVITE
Header:    Contact: <sip:93831069@192.168.16.210:5060;user=phone>
Header:    Record-Route:
<sip:93831069@192.168.26.110:5060;maddr=192.168.26.110>,<sip:93831069@192.168.26.180:5060
;maddr=192.168.26.180>
Header:    Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header:    Content-Type: application/sdp
Header:    Content-Length: 138
-----
                                SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-GW-UserAgent 3804 4963 IN IP4 192.168.16.210
Header:    s=SIP Call
Header:    c=IN IP4 192.168.16.210
Header:    t=0 0
Header:    m=audio 20334 RTP/AVP 0
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.26.180:5060->192.168.26.10:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.10:5060
Header:    From: <sip:6711@192.168.26.10:5060>
Header:    To: <sip:93831069@192.168.26.180:5060>;tag=25A5AD44-1FE9
Header:    Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:    CSeq: 100 INVITE
Header:    Contact: <sip:93831069@192.168.16.210:5060;user=phone>
Header:    Record-Route:
<sip:93831069@192.168.26.110:5060;maddr=192.168.26.110>,<sip:93831069@192.168.26.180:5060
;maddr=192.168.26.180>
Header:    Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header:    Content-Type: application/sdp
Header:    Content-Length: 138
-----
                                SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-GW-UserAgent 3804 4963 IN IP4 192.168.16.210
Header:    s=SIP Call
Header:    c=IN IP4 192.168.16.210
Header:    t=0 0
Header:    m=audio 20334 RTP/AVP 0
-----
                                SIP Headers
-----
sip-req:    ACK sip:93831069@192.168.26.180:5060 SIP/2.0 [192.168.26.10:50373-
>192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.10:5060
Header:    From: <sip:6711@192.168.26.10:5060>
Header:    To: <sip:93831069@192.168.26.180:5060>;tag=25A5AD44-1FE9
Header:    Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header:    Route: <sip:93831069@192.168.26.110:5060;maddr=192.168.26.110>,<
sip:93831069@192.168.16.210:5060;user=phone>
Header:    CSeq: 100 ACK
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    ACK sip:93831069@192.168.26.110:5060;maddr=192.168.26.110 SIP/2.0
[192.168.26.180:5060->192.168.26.110:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
```

## SIP Phone to Phone via Gateway: Called Party is Busy

```
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:93831069@192.168.26.180:5060>;tag=25A5AD44-1FE9
Header: Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header: CSeq: 100 ACK
Header: Route: <sip:93831069@192.168.16.210:5060>
Header: Proxy-Authorization: Basic 123
Header: Content-Length: 0
```

---

### SIP Headers

```
-----
sip-req: ACK sip:93831069@192.168.16.210:5060 SIP/2.0 [192.168.26.110:5060-
>192.168.16.210:5060]
Header: Via: SIP/2.0/UDP 192.168.26.110:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:93831069@192.168.26.180:5060>;tag=25A5AD44-1FE9
Header: Call-ID: c2943000-3e262-e4dc-2e323931@192.168.26.10
Header: CSeq: 100 ACK
Header: Content-Length: 0
-----
```

---

## SIP IP Phone to SIP IP Phone: Forward All Calls

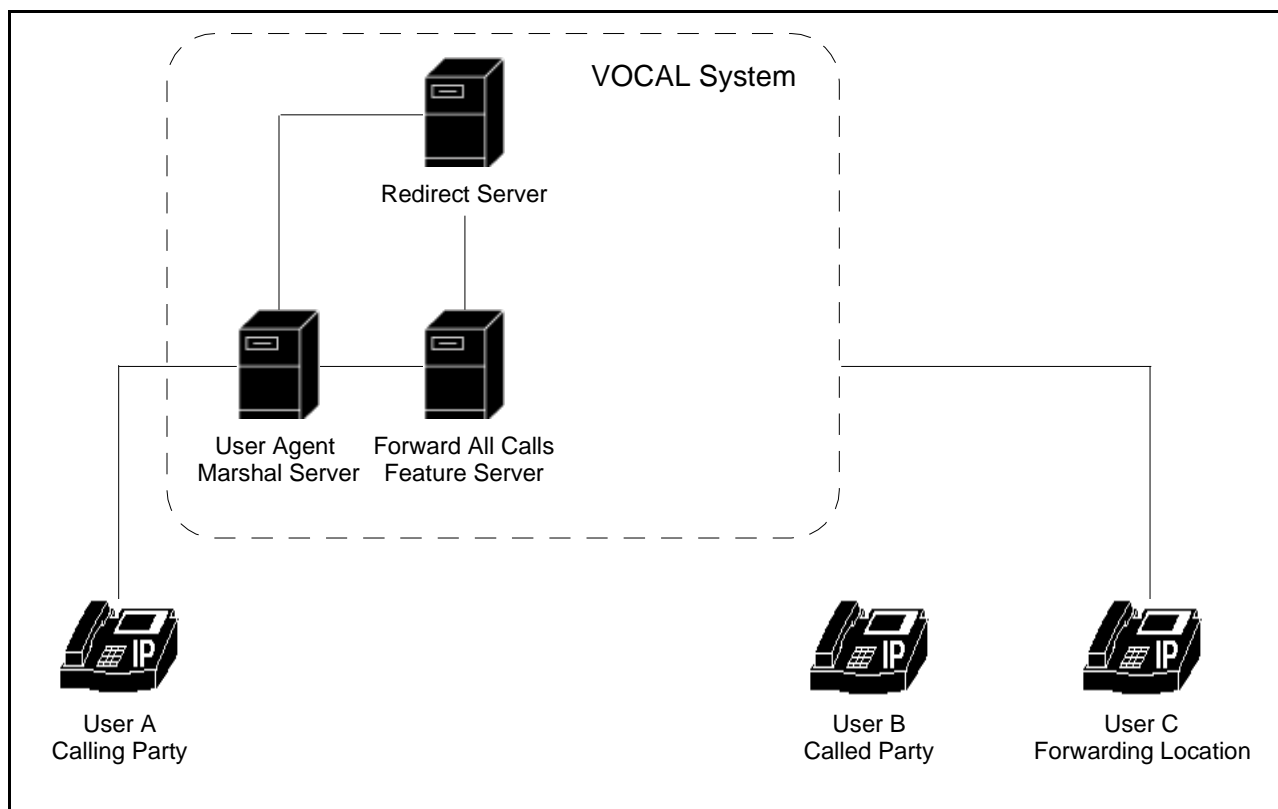
### Call Scenario

Figure C-10 illustrates the following call scenario:

- User A initiates a call to User B
- User B has call forwarding enabled
- The call is forwarded to User C

#### ■Note

In this example, all SIP phones are connected to the same Marshal server.



**Figure C-10. SIP IP Phone to SIP IP Phone: Forward All Calls**

### Call Flow Diagrams

Figure C-11 shows a call originating from an on-network SIP IP phone being forwarded to a call forward destination.



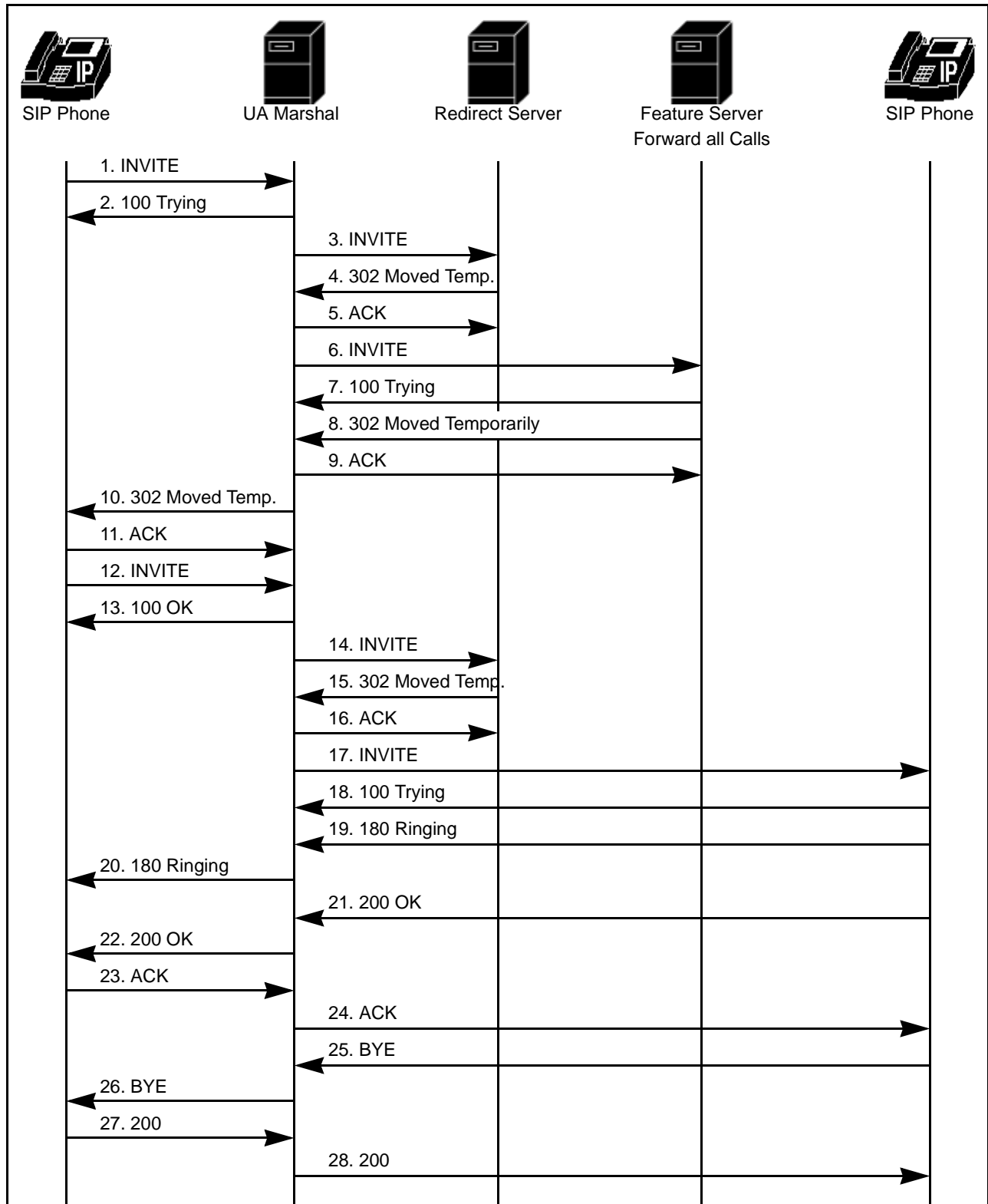


Figure C-11. Call Flow Diagram: SIP IP Phone to SIP IP Phone: Forward All Calls — Diagram 1

### Call Trace

The following call trace shows a call originating from an on-network SIP IP phone being forwarded to a call forward destination.

```
-----
                          SIP Headers
-----
sip-req:    INVITE sip:6715@192.168.26.180 SIP/2.0      [192.168.26.10:50373-
>192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.10:5060
Header:    From: sip:6711@192.168.26.10
Header:    To: <sip:6715@192.168.26.180>
Header:    Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10
Header:    CSeq: 100 INVITE
Header:    Expires: 180
Header:    User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:    Accept: application/sdp
Header:    Contact: sip:6711@192.168.26.10:5060
Header:    Content-Type: application/sdp
Header:    Content-Length: 220
-----
                          SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 6321 26207 IN IP4 192.168.26.10
Header:    s=SIP Call
Header:    c=IN IP4 192.168.26.10
Header:    t=0 0
Header:    m=audio 30224 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000
Header:    a=fmtp:101 0-11
-----
                          SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.26.180:5060->192.168.26.10:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.10:5060
Header:    From: <sip:6711@192.168.26.10:5060>
Header:    To: <sip:6715@192.168.26.180:5060>
Header:    Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10
Header:    CSeq: 100 INVITE
Header:    Content-Length: 0
-----
                          SIP Headers
-----
sip-req:    INVITE sip:6715@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1
Header:    Via: SIP/2.0/UDP 192.168.26.10:5060
Header:    From: <sip:6711@192.168.26.10:5060>
Header:    To: <sip:6715@192.168.26.180:5060>
Header:    Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10
Header:    CSeq: 100 INVITE
Header:    Proxy-Authorization: Basic 123
Header:    Expires: 180
Header:    Record-Route: <sip:6715@192.168.26.180:5060;maddr=192.168.26.180>
Header:    Contact: <sip:6711@192.168.26.10:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 220
-----
                          SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 6321 26207 IN IP4 192.168.26.10
Header:    s=SIP Call
Header:    c=IN IP4 192.168.26.10
Header:    t=0 0
Header:    m=audio 30224 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
```

## SIP IP Phone to SIP IP Phone: Forward All Calls

Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers  
-----

sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6715@192.168.26.180:5060>  
Header: Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:6715@192.168.26.220:6071;user=phone>  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: ACK sip:6715@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6715@192.168.26.180:5060>  
Header: Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: INVITE sip:6715@192.168.26.220:6071;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.220:6071]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6715@192.168.26.180:5060>  
Header: Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Proxy-Authorization: Basic 123  
Header: Expires: 180  
Header: Record-Route: <sip:6715@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:6711@192.168.26.10:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 220

-----  
SDP Headers  
-----

Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 6321 26207 IN IP4 192.168.26.10  
Header: s=SIP Call  
Header: c=IN IP4 192.168.26.10  
Header: t=0 0  
Header: m=audio 30224 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers  
-----

sip-res: SIP/2.0 100 Trying [192.168.26.220:6071->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6715@192.168.26.180:5060>  
Header: Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.220:6071->192.168.26.180:5060]

## SIP IP Phone to SIP IP Phone: Forward All Calls

```
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6715@192.168.26.180:5060>
Header: Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10
Header: CSeq: 100 INVITE
Header: Contact: <sip:6716@192.168.26.180:5060>
Header: Content-Length: 0
Header: CC-Redirect: <sip:6716@192.168.26.180:5060>;redir-
reason=unconditional;redir-counter=0;redir-limit=99
-----
SIP Headers
-----
sip-req: ACK sip:6715@192.168.26.220:6071;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.220:6071]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6715@192.168.26.180:5060>
Header: Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10
Header: CSeq: 100 ACK
Header: Content-Length: 0
-----
SIP Headers
-----
sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.180:5060->192.168.26.10:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6715@192.168.26.180:5060>
Header: Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10
Header: CSeq: 100 INVITE
Header: Contact: <sip:6716@192.168.26.180:5060>
Header: Content-Length: 0
Header: CC-Redirect: <sip:6716@192.168.26.180:5060>;redir-
reason=unconditional;redir-counter=0;redir-limit=99
-----
SIP Headers
-----
sip-req: ACK sip:6715@192.168.26.180 SIP/2.0 [192.168.26.10:50373-
>192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: sip:6711@192.168.26.10
Header: To: <sip:6715@192.168.26.180>
Header: Call-ID: c2943000-ce262-1b5c2-2e323931@192.168.26.10
Header: CSeq: 100 ACK
Header: Content-Length: 0
-----
SIP Headers
-----
sip-req: INVITE sip:6716@192.168.26.180:5060 SIP/2.0 [192.168.26.10:50373-
>192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: sip:6711@192.168.26.10
Header: To: sip:6716@192.168.26.180:5060
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: CSeq: 101 INVITE
Header: Expires: 180
Header: User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Accept: application/sdp
Header: Contact: sip:6711@192.168.26.10:5060
Header: Content-Type: application/sdp
Header: Content-Length: 221
-----
SDP Headers
-----
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 20090 22228 IN IP4
192.168.26.10
Header: s=SIP Call
Header: c=IN IP4 192.168.26.10
Header: t=0 0
```

## SIP IP Phone to SIP IP Phone: Forward All Calls

Header: m=audio 30224 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers  
-----

sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.26.10:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6716@192.168.26.180:5060>  
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10  
Header: CSeq: 101 INVITE  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: INVITE sip:6716@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6716@192.168.26.180:5060>  
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10  
Header: CSeq: 101 INVITE  
Header: Proxy-Authorization: Basic 123  
Header: Expires: 180  
Header: Record-Route: <sip:6716@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:6711@192.168.26.10:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 221

-----  
SDP Headers  
-----

Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 20090 22228 IN IP4  
192.168.26.10  
Header: s=SIP Call  
Header: c=IN IP4 192.168.26.10  
Header: t=0 0  
Header: m=audio 30224 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers  
-----

sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6716@192.168.26.180:5060>  
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10  
Header: CSeq: 101 INVITE  
Header: Contact: <sip:6716@192.168.26.180:5060>  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: ACK sip:6716@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6716@192.168.26.180:5060>  
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10  
Header: CSeq: 101 ACK  
Header: Content-Length: 0

-----  
SIP Headers  
-----

## SIP IP Phone to SIP IP Phone: Forward All Calls

```
sip-req: INVITE sip:6716@192.168.26.12:5060 SIP/2.0 [192.168.26.180:5060->192.168.26.12:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6716@192.168.26.180:5060>
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: CSeq: 101 INVITE
Header: Expires: 180
Header: Record-Route:
<sip:6716@192.168.26.180:5060;maddr=192.168.26.180>,<sip:6716@192.168.26.180:5060;maddr=192.168.26.180>
Header: Contact: <sip:6711@192.168.26.10:5060>
Header: Content-Type: application/sdp
Header: Content-Length: 221
```

### SDP Headers

```
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 20090 22228 IN IP4
192.168.26.10
Header: s=SIP Call
Header: c=IN IP4 192.168.26.10
Header: t=0 0
Header: m=audio 30224 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

### SIP Headers

```
sip-res: SIP/2.0 100 Trying [192.168.26.12:50383->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: CSeq: 101 INVITE
Header: Content-Length: 0
```

### SIP Headers

```
sip-res: SIP/2.0 180 Ringing [192.168.26.12:50383->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: CSeq: 101 INVITE
Header: Content-Length: 0
```

### SIP Headers

```
sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.26.10:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: CSeq: 101 INVITE
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Content-Length: 0
```

### SIP Headers

```
sip-res: SIP/2.0 200 OK [192.168.26.12:50383->192.168.26.180:5060]
```

## SIP IP Phone to SIP IP Phone: Forward All Calls

```
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Contact: sip:6716@192.168.26.12:5060
Header: Record-Route:
<sip:6716@192.168.26.180:5060;maddr=192.168.26.180>,<sip:6716@192.168.26.180:5060;maddr=1
92.168.26.180>
Header: CSeq: 101 INVITE
Header: Content-Type: application/sdp
Header: Content-Length: 218
```

### ----- SDP Headers

```
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 1817 287 IN IP4 192.168.26.12
Header: s=SIP Call
Header: c=IN IP4 192.168.26.12
Header: t=0 0
Header: m=audio 30642 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

### ----- SIP Headers

```
-----  
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.26.10:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: CSeq: 101 INVITE
Header: Contact: <sip:6716@192.168.26.12:5060>
Header: Record-Route:
<sip:6716@192.168.26.180:5060;maddr=192.168.26.180>,<sip:6716@192.168.26.180:5060;maddr=1
92.168.26.180>
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Content-Type: application/sdp
Header: Content-Length: 218
```

### ----- SDP Headers

```
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 1817 287 IN IP4 192.168.26.12
Header: s=SIP Call
Header: c=IN IP4 192.168.26.12
Header: t=0 0
Header: m=audio 30642 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

### ----- SIP Headers

```
-----  
sip-req: ACK sip:6716@192.168.26.180:5060 SIP/2.0 [192.168.26.10:50373-
>192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: Route: <sip:6716@192.168.26.180:5060;maddr=192.168.26.180>,<
sip:6716@192.168.26.12:5060>
Header: CSeq: 101 ACK
Header: Content-Length: 0
```

### ----- SIP Headers

## SIP IP Phone to SIP IP Phone: Forward All Calls

```
sip-req: ACK sip:6716@192.168.26.12:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.26.12:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: CSeq: 101 ACK
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: BYE sip:6716@192.168.26.180:5060 SIP/2.0 [192.168.26.12:50383-
>192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.12:5060
Header: From: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: To: <sip:6711@192.168.26.10:5060>
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: CSeq: 102 BYE
Header: Route: <sip:6716@192.168.26.180:5060;maddr=192.168.26.180>,
<sip:6711@192.168.26.10:5060>
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: BYE sip:6711@192.168.26.10:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.26.10:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.12:5060
Header: From: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: To: <sip:6711@192.168.26.10:5060>
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: CSeq: 102 BYE
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res: SIP/2.0 200 OK [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.12:5060
Header: From: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: To: <sip:6711@192.168.26.10:5060>
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: CSeq: 102 BYE
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.26.12:5060]
Header: Via: SIP/2.0/UDP 192.168.26.12:5060
Header: From: <sip:6716@192.168.26.180:5060>;tag=c2943000312610-0
Header: To: <sip:6711@192.168.26.10:5060>
Header: Call-ID: c2943000-de262-1b626-2e323931@192.168.26.10
Header: CSeq: 102 BYE
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Content-Length: 0
```

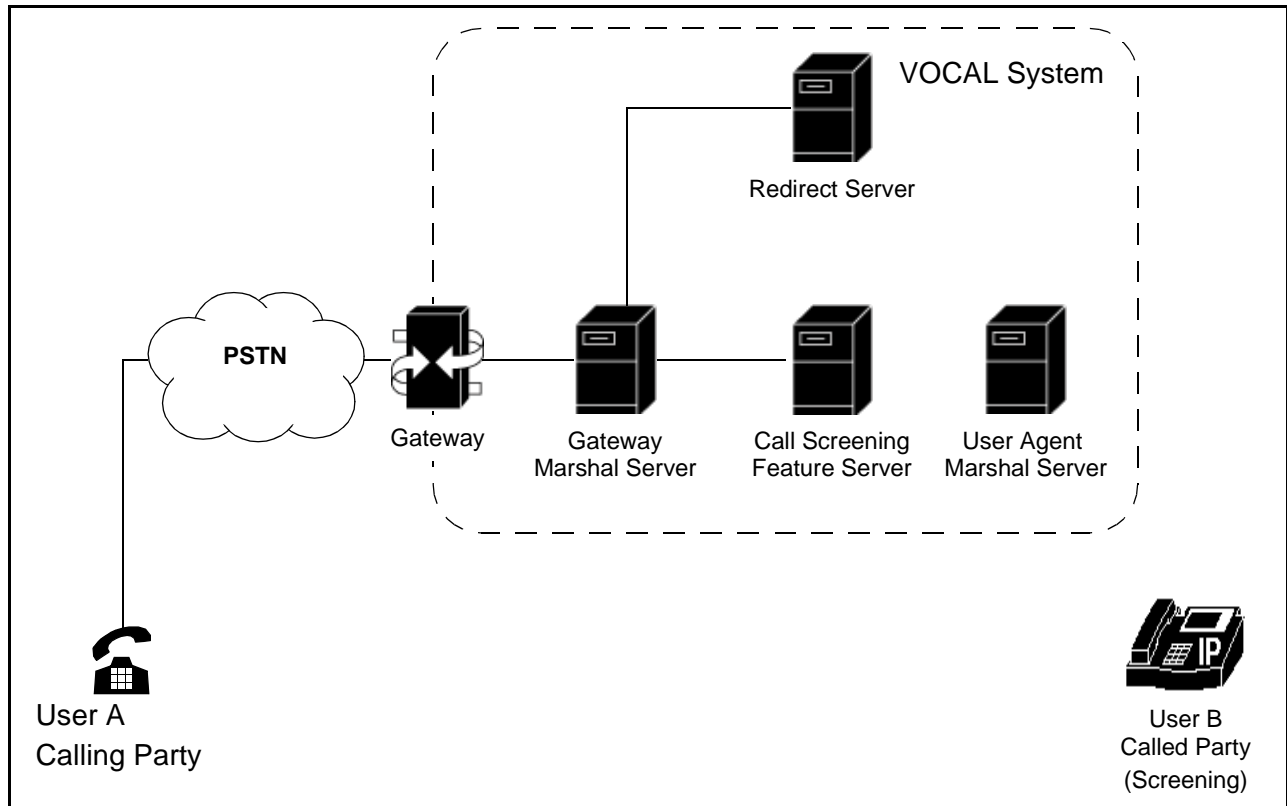


## Phone to SIP Phone via Gateway: Call Screening

### Call Scenario

Figure C-12 illustrates the following call scenario:

- User A initiates a call to User B
- User B has call screening enabled
- The feature server screens the call and returns a forbidden call message back to the gateway.

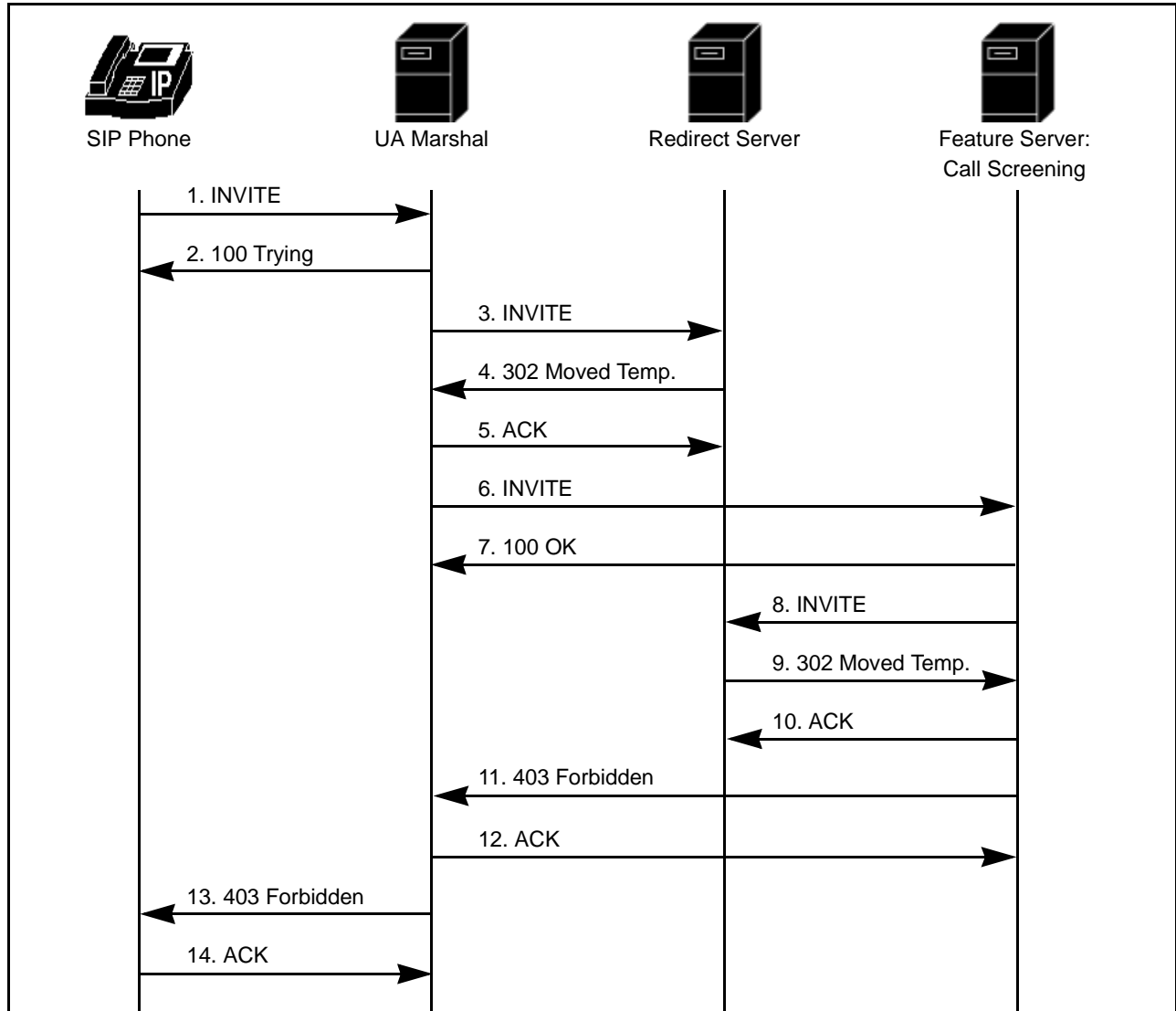


**Figure C-12. SIP IP Phone to Phone: Call Screening**

### Call Flow Diagram

Figure C-13 shows a call, originating from an on-network SIP IP phone, being screened by the feature server.

*Phone to SIP Phone via Gateway: Call Screening*



**Figure C-13. Call Flow Diagram: Call Screening**

**Call Trace**

The following call trace shows a call, originating from an on-network SIP IP phone, being screened by the feature server.

```

-----
                        SIP Headers
-----
sip-reg:    INVITE sip:6711@192.168.26.180 SIP/2.0      [192.168.26.11:50783-
>192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.11:5060
Header:    From: sip:6715@192.168.26.11
Header:    To: <sip:6711@192.168.26.180>
Header:    Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header:    CSeq: 100 INVITE
Header:    Expires: 180
Header:    User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:    Accept: application/sdp
Header:    Contact: sip:6715@192.168.26.11:5060
Header:    Content-Type: application/sdp
Header:    Content-Length: 220
-----
                        SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 5311 17807 IN IP4 192.168.26.11
Header:    s=SIP Call
Header:    c=IN IP4 192.168.26.11
Header:    t=0 0
Header:    m=audio 23070 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000
Header:    a=fmtp:101 0-11
-----
                        SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.26.180:5060->192.168.26.11:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.11:5060
Header:    From: <sip:6715@192.168.26.11:5060>
Header:    To: <sip:6711@192.168.26.180:5060>
Header:    Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header:    CSeq: 100 INVITE
Header:    Content-Length: 0
-----
                        SIP Headers
-----
sip-reg:    INVITE sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1
Header:    Via: SIP/2.0/UDP 192.168.26.11:5060
Header:    From: <sip:6715@192.168.26.11:5060>
Header:    To: <sip:6711@192.168.26.180:5060>
Header:    Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header:    CSeq: 100 INVITE
Header:    Proxy-Authorization: Basic 123
Header:    Expires: 180
Header:    Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header:    Contact: <sip:6715@192.168.26.11:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 220
-----
                        SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 5311 17807 IN IP4 192.168.26.11
Header:    s=SIP Call
Header:    c=IN IP4 192.168.26.11
Header:    t=0 0
Header:    m=audio 23070 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000

```

## Phone to SIP Phone via Gateway: Call Screening

```
Header:          a=fmtp:101 0-11
-----
SIP Headers
-----
sip-res:        SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]
Header:         Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1
Header:         Via: SIP/2.0/UDP 192.168.26.11:5060
Header:         From: <sip:6715@192.168.26.11:5060>
Header:         To: <sip:6711@192.168.26.180:5060>
Header:         Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header:         CSeq: 100 INVITE
Header:         Contact: <sip:6711@192.168.26.220:6074;user=phone>
Header:         Content-Length: 0
-----
SIP Headers
-----
sip-req:        ACK sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.200:5060]
Header:         Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1
Header:         From: <sip:6715@192.168.26.11:5060>
Header:         To: <sip:6711@192.168.26.180:5060>
Header:         Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header:         CSeq: 100 ACK
Header:         Content-Length: 0
-----
SIP Headers
-----
sip-req:        INVITE sip:6711@192.168.26.220:6074;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.220:6074]
Header:         Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:         Via: SIP/2.0/UDP 192.168.26.11:5060
Header:         From: <sip:6715@192.168.26.11:5060>
Header:         To: <sip:6711@192.168.26.180:5060>
Header:         Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header:         CSeq: 100 INVITE
Header:         Proxy-Authorization: Basic 123
Header:         Expires: 180
Header:         Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header:         Contact: <sip:6715@192.168.26.11:5060>
Header:         Content-Type: application/sdp
Header:         Content-Length: 220
-----
SDP Headers
-----
Header:         v=0
Header:         o=CiscoSystemsSIP-IPPhone-UserAgent 5311 17807 IN IP4 192.168.26.11
Header:         s=SIP Call
Header:         c=IN IP4 192.168.26.11
Header:         t=0 0
Header:         m=audio 23070 RTP/AVP 0 101
Header:         a=rtpmap:0 pcmu/8000
Header:         a=rtpmap:101 telephone-event/8000
Header:         a=fmtp:101 0-11
-----
SIP Headers
-----
sip-res:        SIP/2.0 100 Trying [192.168.26.220:6074->192.168.26.180:5060]
Header:         Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:         Via: SIP/2.0/UDP 192.168.26.11:5060
Header:         From: <sip:6715@192.168.26.11:5060>
Header:         To: <sip:6711@192.168.26.180:5060>
Header:         Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header:         CSeq: 100 INVITE
Header:         Content-Length: 0
-----
SIP Headers
-----
sip-req:        INVITE sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.220:6074-
>192.168.26.200:5060]
Header:         Via: SIP/2.0/UDP 192.168.26.220:6074;branch=101
```

## Phone to SIP Phone via Gateway: Call Screening

```
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.11:5060
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header: CSeq: 100 INVITE
Header: Proxy-Authorization: Basic 123
Header: Expires: 180
Header: Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header: Contact: <sip:6715@192.168.26.11:5060>
Header: Content-Type: application/sdp
Header: Content-Length: 220
```

### SDP Headers

```
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 5311 17807 IN IP4 192.168.26.11
Header: s=SIP Call
Header: c=IN IP4 192.168.26.11
Header: t=0 0
Header: m=audio 23070 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

### SIP Headers

```
-----
sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.220:6074]
Header: Via: SIP/2.0/UDP 192.168.26.220:6074;branch=101
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.11:5060
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header: CSeq: 100 INVITE
Header: Contact: <sip:6711@192.168.26.220:6073;user=phone>
Header: Content-Length: 0
```

### SIP Headers

```
-----
sip-req: ACK sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.220:6074-
>192.168.26.200:5060]
Header: Via: SIP/2.0/UDP 192.168.26.220:6074;branch=101
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header: CSeq: 100 ACK
Header: Content-Length: 0
```

### SIP Headers

```
-----
sip-res: SIP/2.0 403 Forbidden [192.168.26.220:6074-
>192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.11:5060
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header: CSeq: 100 INVITE
Header: Content-Length: 0
```

### SIP Headers

```
-----
sip-req: ACK sip:6711@192.168.26.220:6074;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.220:6074]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header: CSeq: 100 ACK
```

## Phone to SIP Phone via Gateway: Call Screening

```
Header:                               Content-Length: 0
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 403 Forbidden [192.168.26.180:5060->192.168.26.11:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.11:5060
Header:     From: <sip:6715@192.168.26.11:5060>
Header:     To: <sip:6711@192.168.26.180:5060>
Header:     Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header:     CSeq: 100 INVITE
Header:     Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    ACK sip:6711@192.168.26.180 SIP/2.0 [192.168.26.11:50783-
>192.168.26.180:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.11:5060
Header:     From: sip:6715@192.168.26.11
Header:     To: <sip:6711@192.168.26.180:5060>
Header:     Call-ID: c3943000-6978b-2995c-2e323931@192.168.26.11
Header:     CSeq: 100 ACK
Header:     Content-Length: 0
-----
```

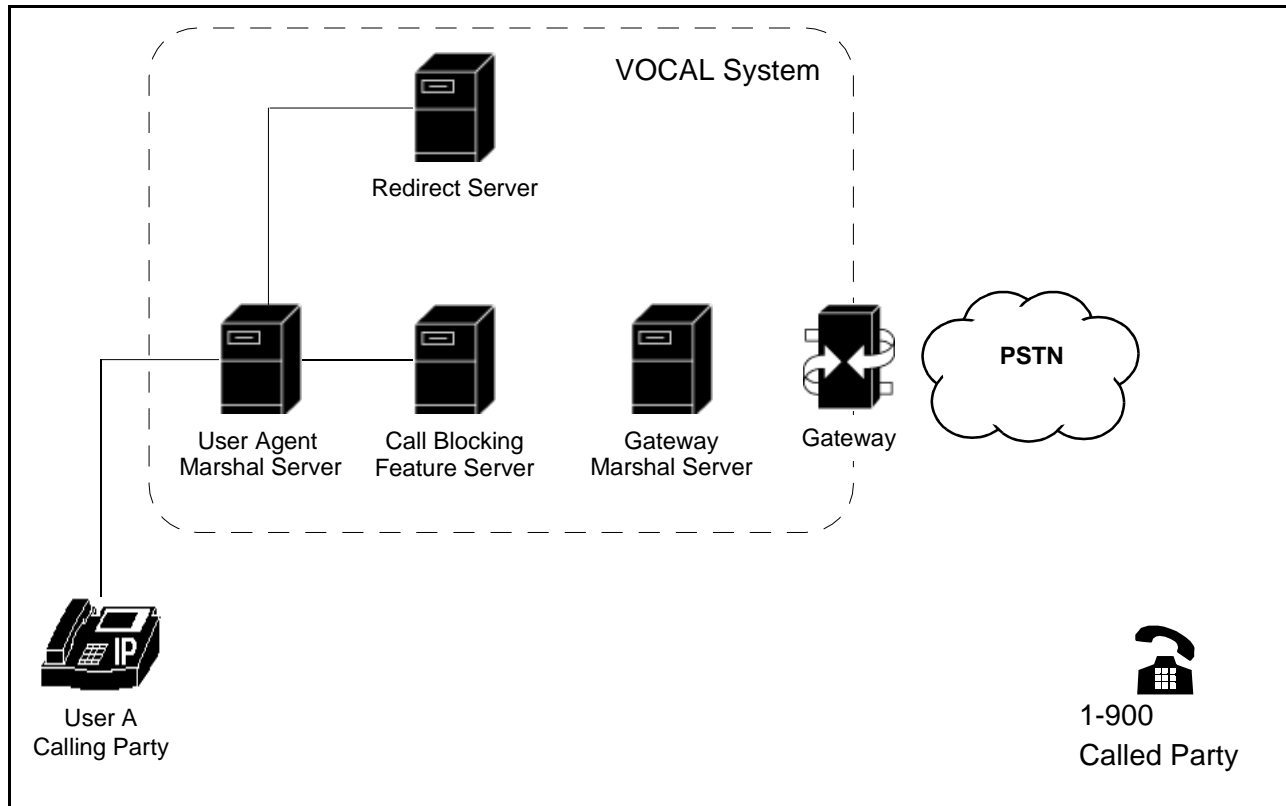
---

## SIP Phone to PSTN: Call Blocking

**Call Scenario**

Figure C-12 illustrates the following call scenario:

- User A initiates a long distance or 1-900 number call
- The VOCAL System blocks the call



**Figure C-14. SIP IP Phone to Phone: Call Blocking**

**Call Flow Diagram**

Figure C-15 shows a call, originating from an on-network SIP IP phone, being blocked by the feature server.

## SIP Phone to PSTN: Call Blocking

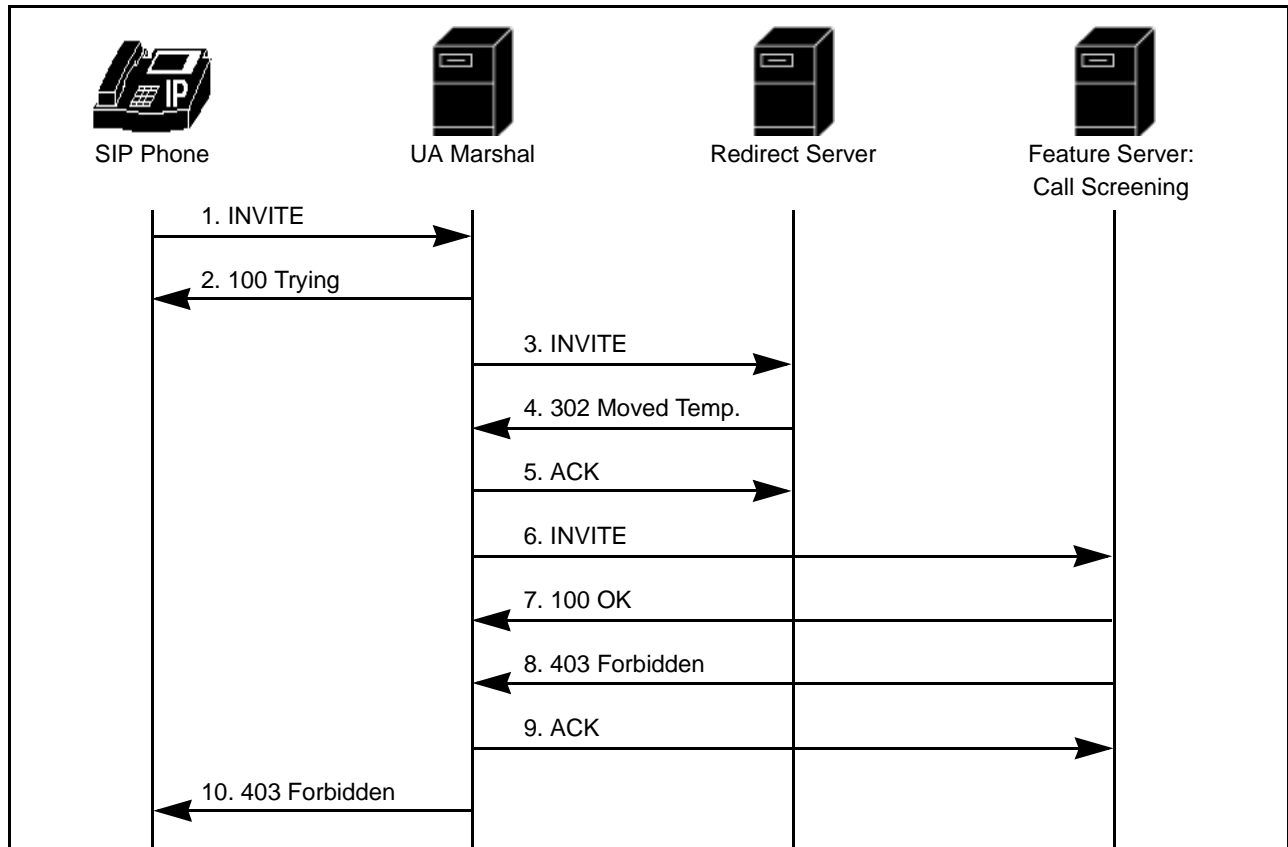


Figure C-15. Call Flow Diagram: SIP IP Phone to PSTN: Call Blocking



## Call Trace

The following call trace shows a call, originating from an on-network SiP IP phone, being blocked by the feature server.

```

-----
                        SIP Headers
-----
sip-reg:    INVITE sip:914083831059@192.168.26.180 SIP/2.0 [192.168.26.12:50420-
>192.168.26.180:6060]
Header:    Via: SIP/2.0/UDP 192.168.26.12:5060
Header:    From: sip:6715@192.168.26.12
Header:    To: <sip:914083831059@192.168.26.180>
Header:    Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12
Header:    CSeq: 100 INVITE
Header:    Expires: 180
Header:    User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:    Accept: application/sdp
Header:    Contact: sip:6715@192.168.26.12:5060
Header:    Content-Type: application/sdp
Header:    Content-Length: 220
-----
                        SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 14800 9904 IN IP4 192.168.26.12
Header:    s=SIP Call
Header:    c=IN IP4 192.168.26.12
Header:    t=0 0
Header:    m=audio 24246 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000
Header:    a=fmtp:101 0-11
-----
                        SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.26.180:6060->192.168.26.12:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.12:5060
Header:    From: <sip:6715@192.168.26.12:5060>
Header:    To: <sip:914083831059@192.168.26.180:5060>
Header:    Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12
Header:    CSeq: 100 INVITE
Header:    Content-Length: 0
-----
                        SIP Headers
-----
sip-reg:    INVITE sip:914083831059@192.168.26.200:6060;user=phone SIP/2.0
[192.168.26.180:6060->192.168.26.200:6060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:6060;branch=1
Header:    Via: SIP/2.0/UDP 192.168.26.12:5060
Header:    From: <sip:6715@192.168.26.12:5060>
Header:    To: <sip:914083831059@192.168.26.180:5060>
Header:    Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12
Header:    CSeq: 100 INVITE
Header:    Proxy-Authorization: Basic VovidaClassXSwitch
Header:    Expires: 180
Header:    Record-Route:
<sip:914083831059@192.168.26.180:6060;maddr=192.168.26.180>
Header:    Contact: <sip:6715@192.168.26.12:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 220
-----
                        SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 14800 9904 IN IP4 192.168.26.12
Header:    s=SIP Call
Header:    c=IN IP4 192.168.26.12
Header:    t=0 0
Header:    m=audio 24246 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000

```

## SIP Phone to PSTN: Call Blocking

Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers  
-----

sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:6060->192.168.26.180:6060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:6060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.26.12:5060  
Header: From: <sip:6715@192.168.26.12:5060>  
Header: To: <sip:914083831059@192.168.26.180:5060>  
Header: Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:914083831059@192.168.26.220:6072;user=phone>  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: ACK sip:914083831059@192.168.26.200:6060;user=phone SIP/2.0 [192.168.26.180:6060->192.168.26.200:6060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:6060;branch=1  
Header: From: <sip:6715@192.168.26.12:5060>  
Header: To: <sip:914083831059@192.168.26.180:5060>  
Header: Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: INVITE sip:914083831059@192.168.26.220:6072;user=phone SIP/2.0 [192.168.26.180:6060->192.168.26.220:6072]  
Header: Via: SIP/2.0/UDP 192.168.26.180:6060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.26.12:5060  
Header: From: <sip:6715@192.168.26.12:5060>  
Header: To: <sip:914083831059@192.168.26.180:5060>  
Header: Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12  
Header: CSeq: 100 INVITE  
Header: Proxy-Authorization: Basic VovidaClassXSwitch  
Header: Expires: 180  
Header: Record-Route: <sip:914083831059@192.168.26.180:6060;maddr=192.168.26.180>  
Header: Contact: <sip:6715@192.168.26.12:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 220

-----  
SDP Headers  
-----

Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 14800 9904 IN IP4 192.168.26.12  
Header: s=SIP Call  
Header: c=IN IP4 192.168.26.12  
Header: t=0 0  
Header: m=audio 24246 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers  
-----

sip-res: SIP/2.0 100 Trying [192.168.26.220:6072->192.168.26.180:6060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:6060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.26.12:5060  
Header: From: <sip:6715@192.168.26.12:5060>  
Header: To: <sip:914083831059@192.168.26.180:5060>  
Header: Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12  
Header: CSeq: 100 INVITE  
Header: Content-Length: 0

-----  
SIP Headers  
-----

```
sip-res: SIP/2.0 403 Forbidden [192.168.26.220:6072->192.168.26.180:6060]
Header: Via: SIP/2.0/UDP 192.168.26.180:6060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.12:5060
Header: From: <sip:6715@192.168.26.12:5060>
Header: To: <sip:914083831059@192.168.26.180:5060>
Header: Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12
Header: CSeq: 100 INVITE
Header: Content-Length: 0
```

-----  
SIP Headers

```
sip-req: ACK sip:914083831059@192.168.26.220:6072;user=phone SIP/2.0
[192.168.26.180:6060->192.168.26.220:6072]
Header: Via: SIP/2.0/UDP 192.168.26.180:6060;branch=2
Header: From: <sip:6715@192.168.26.12:5060>
Header: To: <sip:914083831059@192.168.26.180:5060>
Header: Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12
Header: CSeq: 100 ACK
Header: Content-Length: 0
```

-----  
SIP Headers

```
sip-res: SIP/2.0 403 Forbidden [192.168.26.180:6060->192.168.26.12:5060]
Header: Via: SIP/2.0/UDP 192.168.26.12:5060
Header: From: <sip:6715@192.168.26.12:5060>
Header: To: <sip:914083831059@192.168.26.180:5060>
Header: Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12
Header: CSeq: 100 INVITE
Header: Content-Length: 0
```

-----  
SIP Headers

```
sip-req: ACK sip:914083831059@192.168.26.180 SIP/2.0 [192.168.26.12:50420-
>192.168.26.180:6060]
Header: Via: SIP/2.0/UDP 192.168.26.12:5060
Header: From: sip:6715@192.168.26.12
Header: To: <sip:914083831059@192.168.26.180:5060>
Header: Call-ID: c2943000-d061261-221d950-2e323931@192.168.26.12
Header: CSeq: 100 ACK
Header: Content-Length: 0
-----
```

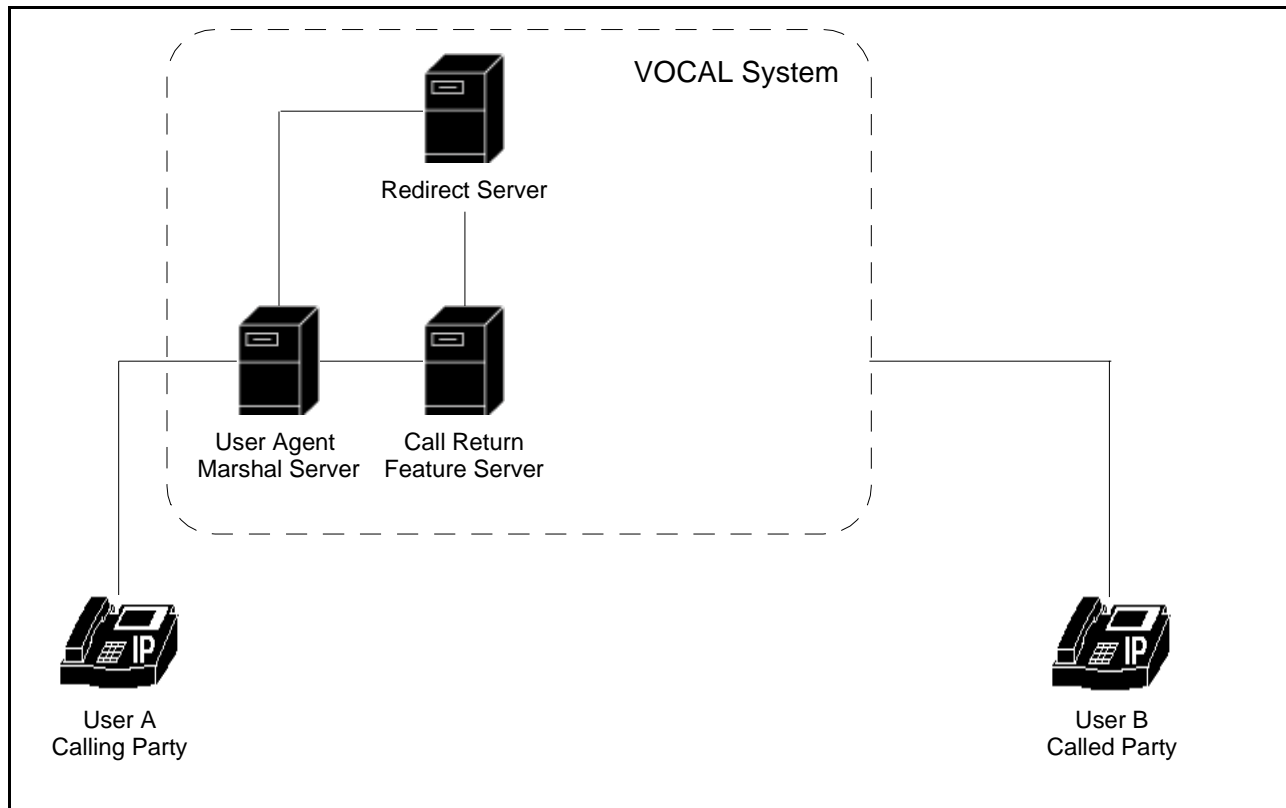
---

## SIP IP Phone to SIP IP Phone: Call Return

**Call Scenario**

Figure C-16 illustrates the following call scenario:

- User A dials \*69 to determine the last number that was called, User B
- User A calls User B



**Figure C-16. SIP Phone to SIP Phone: Call Return**

**Call Flow Diagram**

Figures C-17, C-18 and C-19 show a call return request leading to an established call between two on-network SIP IP phones.

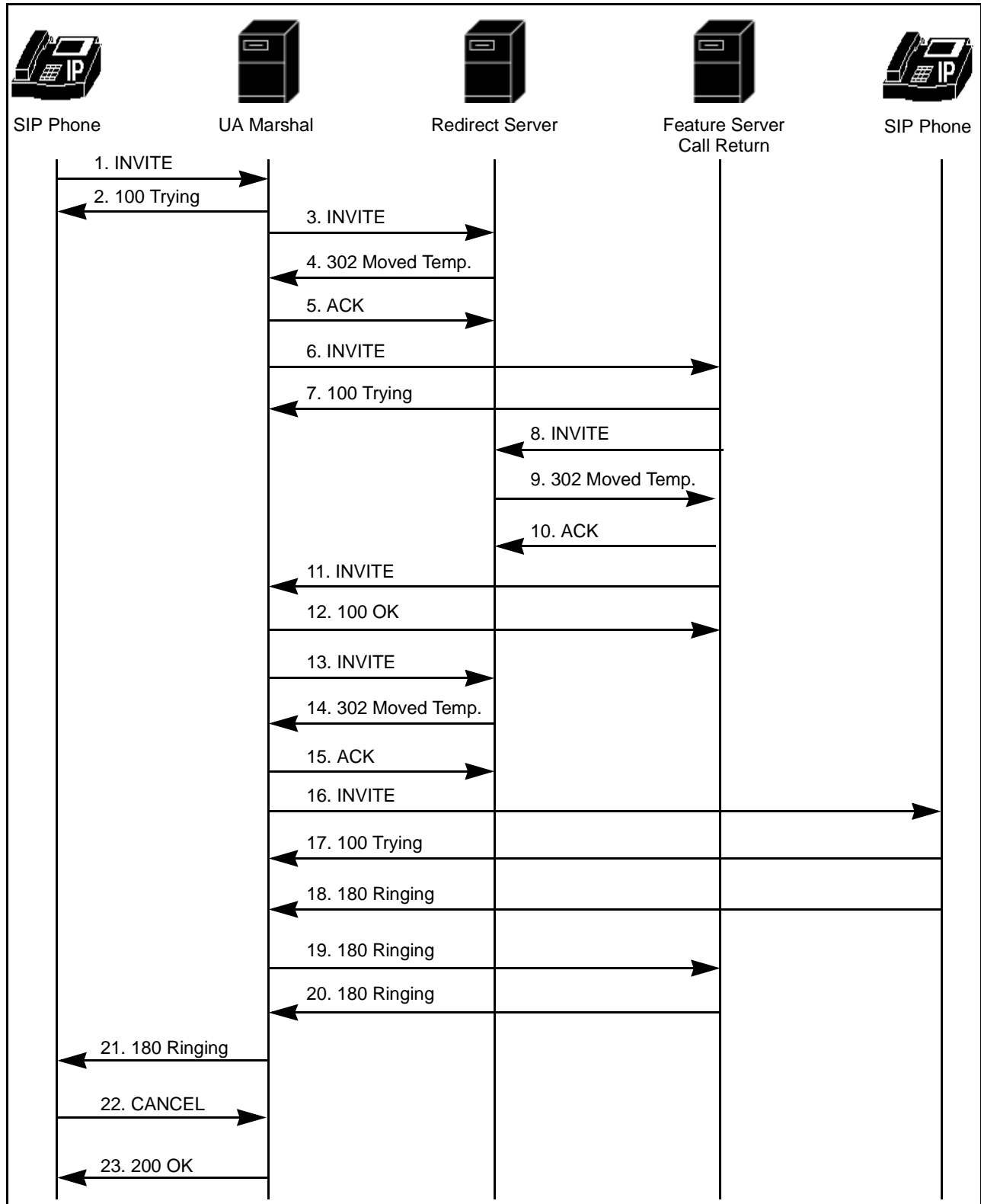
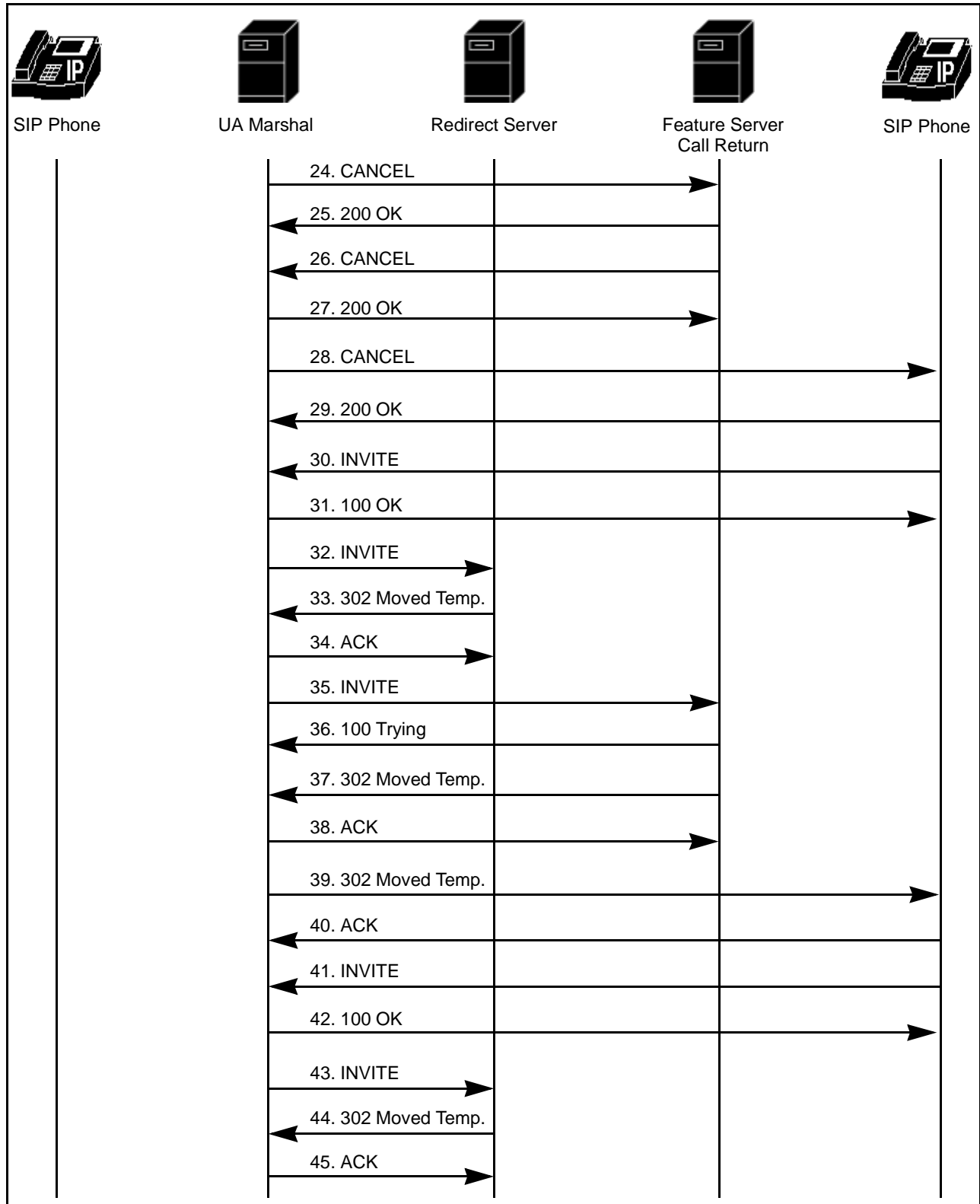


Figure C-17. SIP IP Phone to SIP IP Phone: Call Return — Diagram 1

*SIP IP Phone to SIP IP Phone: Call Return*



**Figure C-18. SIP IP Phone to SIP IP Phone: Call Return — Diagram 2**

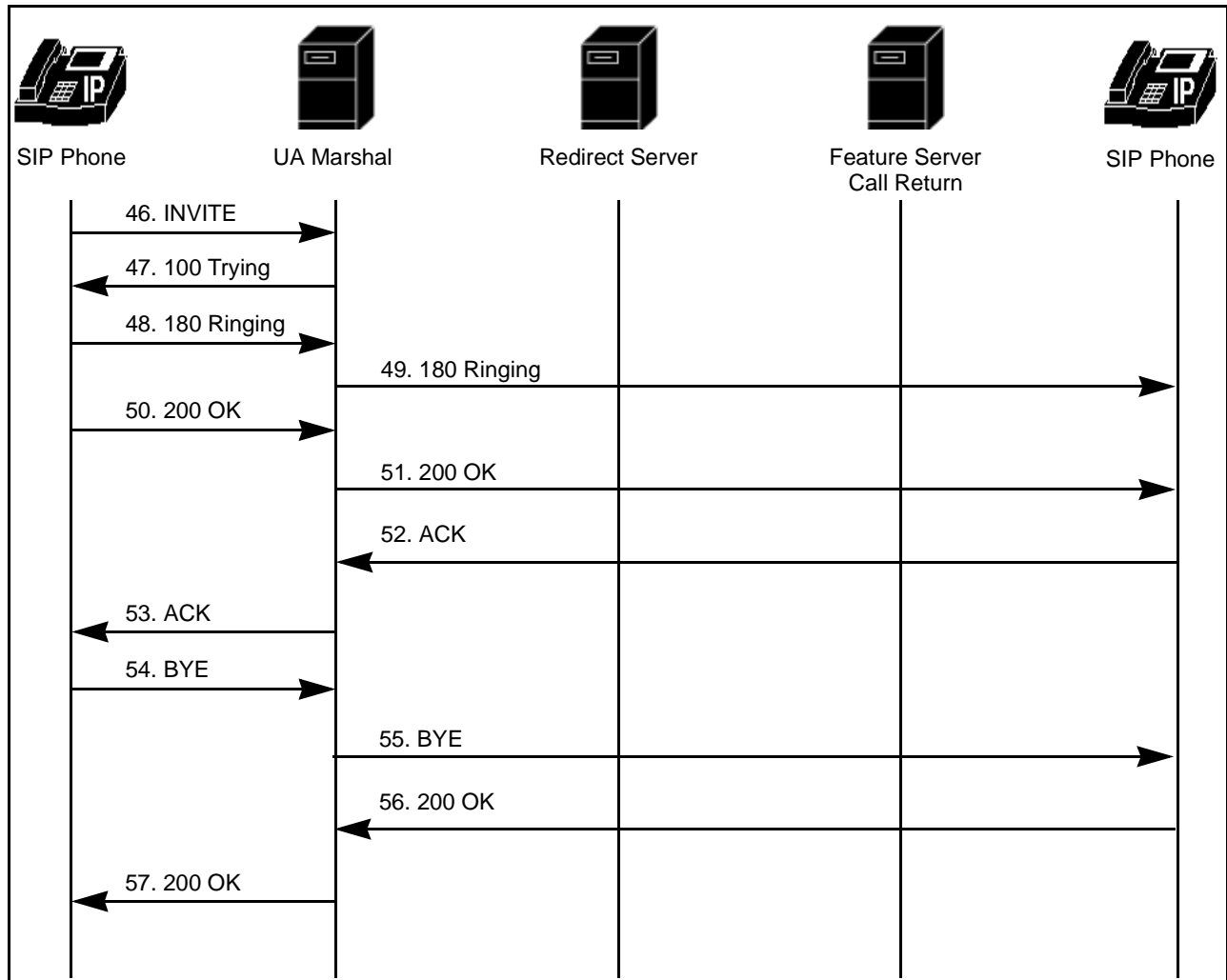


Figure C-19. SIP IP Phone to SIP IP Phone: Call Return — Diagram 3

### Call Trace

The following call trace shows a call return request leading to an established call between two on-network SIP IP phones.

```

-----
                        SIP Headers
-----
sip-req:  INVITE sip:6711@192.168.26.180 SIP/2.0      [192.168.26.11:50783-
>192.168.26.180:5060]
Header:   Via: SIP/2.0/UDP 192.168.26.11:5060
Header:   From: sip:6715@192.168.26.11
Header:   To: <sip:6711@192.168.26.180>
Header:   Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:   CSeq: 100 INVITE
Header:   Expires: 180
Header:   User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:   Accept: application/sdp
Header:   Contact: sip:6715@192.168.26.11:5060
Header:   Content-Type: application/sdp
Header:   Content-Length: 220
-----
                        SDP Headers
-----

```

## SIP IP Phone to SIP IP Phone: Call Return

```
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 25077 6500 IN IP4 192.168.26.11
Header: s=SIP Call
Header: c=IN IP4 192.168.26.11
Header: t=0 0
Header: m=audio 27724 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

---

### SIP Headers

```
-----
sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.26.11:5060]
Header: Via: SIP/2.0/UDP 192.168.26.11:5060
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header: CSeq: 100 INVITE
Header: Content-Length: 0
```

---

### SIP Headers

```
-----
sip-req: INVITE sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1
Header: Via: SIP/2.0/UDP 192.168.26.11:5060
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header: CSeq: 100 INVITE
Header: Proxy-Authorization: Basic 123
Header: Expires: 180
Header: Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header: Contact: <sip:6715@192.168.26.11:5060>
Header: Content-Type: application/sdp
Header: Content-Length: 220
```

---

### SDP Headers

```
-----
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 25077 6500 IN IP4 192.168.26.11
Header: s=SIP Call
Header: c=IN IP4 192.168.26.11
Header: t=0 0
Header: m=audio 27724 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

---

### SIP Headers

```
-----
sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1
Header: Via: SIP/2.0/UDP 192.168.26.11:5060
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header: CSeq: 100 INVITE
Header: Contact: <sip:6711@192.168.26.220:6074;user=phone>
Header: Content-Length: 0
```

---

### SIP Headers

```
-----
sip-req: ACK sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header: CSeq: 100 ACK
```



```

Header:                               Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    INVITE sip:6711@192.168.26.220:6074;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.220:6074]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.26.11:5060
Header:    From: <sip:6715@192.168.26.11:5060>
Header:    To: <sip:6711@192.168.26.180:5060>
Header:    Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:    CSeq: 100 INVITE
Header:    Proxy-Authorization: Basic 123
Header:    Expires: 180
Header:    Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header:    Contact: <sip:6715@192.168.26.11:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 220
-----
                                SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 25077 6500 IN IP4 192.168.26.11
Header:    s=SIP Call
Header:    c=IN IP4 192.168.26.11
Header:    t=0 0
Header:    m=audio 27724 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000
Header:    a=fmtp:101 0-11
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.26.220:6074->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.26.11:5060
Header:    From: <sip:6715@192.168.26.11:5060>
Header:    To: <sip:6711@192.168.26.180:5060>
Header:    Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:    CSeq: 100 INVITE
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    INVITE sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.220:6074-
>192.168.26.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.220:6074;branch=101
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.26.11:5060
Header:    From: <sip:6715@192.168.26.11:5060>
Header:    To: <sip:6711@192.168.26.180:5060>
Header:    Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:    CSeq: 100 INVITE
Header:    Proxy-Authorization: Basic 123
Header:    Expires: 180
Header:    Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header:    Contact: <sip:6715@192.168.26.11:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 220
-----
                                SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 25077 6500 IN IP4 192.168.26.11
Header:    s=SIP Call
Header:    c=IN IP4 192.168.26.11
Header:    t=0 0
Header:    m=audio 27724 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000

```

## SIP IP Phone to SIP IP Phone: Call Return

```
Header:          a=fmtp:101 0-11
-----
SIP Headers
-----
sip-res:        SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.220:6074]
Header:         Via: SIP/2.0/UDP 192.168.26.220:6074;branch=101
Header:         Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:         Via: SIP/2.0/UDP 192.168.26.11:5060
Header:         From: <sip:6715@192.168.26.11:5060>
Header:         To: <sip:6711@192.168.26.180:5060>
Header:         Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:         CSeq: 100 INVITE
Header:         Contact: <sip:6711@192.168.26.180:5060>
Header:         Content-Length: 0
-----
SIP Headers
-----
sip-req:        ACK sip:6711@192.168.26.200:5060 SIP/2.0 [192.168.26.220:6074-
>192.168.26.200:5060]
Header:         Via: SIP/2.0/UDP 192.168.26.220:6074;branch=101
Header:         From: <sip:6715@192.168.26.11:5060>
Header:         To: <sip:6711@192.168.26.180:5060>
Header:         Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:         CSeq: 100 ACK
Header:         Content-Length: 0
-----
SIP Headers
-----
sip-req:        INVITE sip:6711@192.168.26.180:5060 SIP/2.0 [192.168.26.220:6074-
>192.168.26.180:5060]
Header:         Via: SIP/2.0/UDP 192.168.26.220:6074;branch=102
Header:         Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:         Via: SIP/2.0/UDP 192.168.26.11:5060
Header:         From: <sip:6715@192.168.26.11:5060>
Header:         To: <sip:6711@192.168.26.180:5060>
Header:         Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:         CSeq: 100 INVITE
Header:         Proxy-Authorization: Basic 123
Header:         Expires: 180
Header:         Record-Route:
<sip:6711@192.168.26.220:6074;maddr=192.168.26.220>,<sip:6711@192.168.26.180:5060;maddr=1
92.168.26.180>
Header:         Contact: <sip:6715@192.168.26.11:5060>
Header:         Content-Type: application/sdp
Header:         Content-Length: 220
Header:         CC-Redirect: <sip:6711@192.168.26.180:5060>;redir-
reason=unconditional;redir-counter=0;redir-limit=99
-----
SDP Headers
-----
Header:         v=0
Header:         o=CiscoSystemsSIP-IPPhone-UserAgent 25077 6500 IN IP4 192.168.26.11
Header:         s=SIP Call
Header:         c=IN IP4 192.168.26.11
Header:         t=0 0
Header:         m=audio 27724 RTP/AVP 0 101
Header:         a=rtpmap:0 pcmu/8000
Header:         a=rtpmap:101 telephone-event/8000
Header:         a=fmtp:101 0-11
-----
SIP Headers
-----
sip-res:        SIP/2.0 100 Trying [192.168.26.180:5060->192.168.26.220:6074]
Header:         Via: SIP/2.0/UDP 192.168.26.220:6074;branch=102
Header:         Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:         Via: SIP/2.0/UDP 192.168.26.11:5060
Header:         From: <sip:6715@192.168.26.11:5060>
Header:         To: <sip:6711@192.168.26.180:5060>
Header:         Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:         CSeq: 100 INVITE
```

```

Header:                               Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    INVITE sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=3
Header:    Via: SIP/2.0/UDP 192.168.26.220:6074;branch=102
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.26.11:5060
Header:    From: <sip:6715@192.168.26.11:5060>
Header:    To: <sip:6711@192.168.26.180:5060>
Header:    Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:    CSeq: 100 INVITE
Header:    Expires: 180
Header:    Record-Route:
<sip:6711@192.168.26.180:5060;maddr=192.168.26.180>, <sip:6711@192.168.26.220:6074;maddr=1
92.168.26.220>, <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header:    Contact: <sip:6715@192.168.26.11:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 220
Header:    CC-Redirect: <sip:6711@192.168.26.180:5060>;redir-
reason=unconditional;redir-counter=0;redir-limit=99
-----
                                SDP Headers
-----
Header:    v=0
Header:    o=CiscoSystemsSIP-IPPhone-UserAgent 25077 6500 IN IP4 192.168.26.11
Header:    s=SIP Call
Header:    c=IN IP4 192.168.26.11
Header:    t=0 0
Header:    m=audio 27724 RTP/AVP 0 101
Header:    a=rtpmap:0 pcmu/8000
Header:    a=rtpmap:101 telephone-event/8000
Header:    a=fmtp:101 0-11
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=3
Header:    Via: SIP/2.0/UDP 192.168.26.220:6074;branch=102
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.26.11:5060
Header:    From: <sip:6715@192.168.26.11:5060>
Header:    To: <sip:6711@192.168.26.180:5060>
Header:    Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:    CSeq: 100 INVITE
Header:    Contact: <sip:6711@192.168.26.10:5060>
Header:    Content-Length: 0
Header:    CC-Redirect: <sip:6711@192.168.26.180:5060>;redir-
reason=unconditional;redir-counter=0;redir-limit=99
-----
                                SIP Headers
-----
sip-req:    ACK sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=3
Header:    From: <sip:6715@192.168.26.11:5060>
Header:    To: <sip:6711@192.168.26.180:5060>
Header:    Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header:    CSeq: 100 ACK
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    INVITE sip:6711@192.168.26.10:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.26.10:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.26.220:6074;branch=102
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2

```

## SIP IP Phone to SIP IP Phone: Call Return

```
Header: Via: SIP/2.0/UDP 192.168.26.11:5060
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header: CSeq: 100 INVITE
Header: Expires: 180
Header: Record-Route:
<sip:6711@192.168.26.180:5060;maddr=192.168.26.180>,<sip:6711@192.168.26.220:6074;maddr=1
92.168.26.220>,<sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header: Contact: <sip:6715@192.168.26.11:5060>
Header: Content-Type: application/sdp
Header: Content-Length: 220
Header: CC-Redirect: <sip:6711@192.168.26.180:5060>;redir-
reason=unconditional;redir-counter=0;redir-limit=99
```

### SDP Headers

```
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 25077 6500 IN IP4 192.168.26.11
Header: s=SIP Call
Header: c=IN IP4 192.168.26.11
Header: t=0 0
Header: m=audio 27724 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

### SIP Headers

```
sip-res: SIP/2.0 100 Trying [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.26.220:6074;branch=102,SIP/2.0/UDP 192.168.26.180:5060;branch=2,SIP/2.0/UDP
192.168.26.11:5060
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>;tag=c29430001e2620-0
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: CSeq: 100 INVITE
Header: Content-Length: 0
```

### SIP Headers

```
sip-res: SIP/2.0 180 Ringing [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.26.220:6074;branch=102,SIP/2.0/UDP 192.168.26.180:5060;branch=2,SIP/2.0/UDP
192.168.26.11:5060
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>;tag=c29430001e2620-0
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: CSeq: 100 INVITE
Header: Content-Length: 0
```

### SIP Headers

```
sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.26.220:6074]
Header: Via: SIP/2.0/UDP 192.168.26.220:6074;branch=102
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.11:5060
Header: From: <sip:6715@192.168.26.11:5060>
Header: To: <sip:6711@192.168.26.180:5060>;tag=c29430001e2620-0
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11
Header: CSeq: 100 INVITE
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Content-Length: 0
```

### SIP Headers

```
sip-res: SIP/2.0 180 Ringing [192.168.26.220:6074->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
```

Header: Via: SIP/2.0/UDP 192.168.26.11:5060  
Header: From: <sip:6715@192.168.26.11:5060>  
Header: To: <sip:6711@192.168.26.180:5060>;tag=c29430001e2620-0  
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11  
Header: CSeq: 100 INVITE  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.26.11:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.11:5060  
Header: From: <sip:6715@192.168.26.11:5060>  
Header: To: <sip:6711@192.168.26.180:5060>;tag=c29430001e2620-0  
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11  
Header: CSeq: 100 INVITE  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: CANCEL sip:6711@192.168.26.180 SIP/2.0 [192.168.26.11:50783->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.11:5060  
Header: From: sip:6715@192.168.26.11  
Header: To: <sip:6711@192.168.26.180>  
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11  
Header: User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: CSeq: 100 CANCEL  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.26.11:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.11:5060  
Header: From: <sip:6715@192.168.26.11:5060>  
Header: To: <sip:6711@192.168.26.180:5060>  
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11  
Header: CSeq: 100 CANCEL  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: CANCEL sip:6711@192.168.26.220:6074;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.220:6074]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: From: <sip:6715@192.168.26.11:5060>  
Header: To: <sip:6711@192.168.26.180:5060>  
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11  
Header: CSeq: 100 CANCEL  
Header: Proxy-Authorization: Basic 123  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-res: SIP/2.0 200 OK [192.168.26.220:6074->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: From: <sip:6715@192.168.26.11:5060>  
Header: To: <sip:6711@192.168.26.180:5060>  
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11  
Header: CSeq: 100 CANCEL  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: CANCEL sip:6711@192.168.26.180:5060 SIP/2.0 [192.168.26.220:6074->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.220:6074;branch=102  
Header: From: <sip:6715@192.168.26.11:5060>  
Header: To: <sip:6711@192.168.26.180:5060>

## SIP IP Phone to SIP IP Phone: Call Return

Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11  
Header: CSeq: 100 CANCEL  
Header: Proxy-Authorization: Basic 123  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.26.220:6074]  
Header: Via: SIP/2.0/UDP 192.168.26.220:6074;branch=102  
Header: From: <sip:6715@192.168.26.11:5060>  
Header: To: <sip:6711@192.168.26.180:5060>  
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11  
Header: CSeq: 100 CANCEL  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: CANCEL sip:6711@192.168.26.10:5060 SIP/2.0 [192.168.26.180:5060->192.168.26.10:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: From: <sip:6715@192.168.26.11:5060>  
Header: To: <sip:6711@192.168.26.180:5060>  
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11  
Header: CSeq: 100 CANCEL  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.26.10:50373->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: From: <sip:6715@192.168.26.11:5060>  
Header: To: <sip:6711@192.168.26.180:5060>;tag=c29430001e2620-0  
Header: Call-ID: c3943000-2978b-aa0e-2e323931@192.168.26.11  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: CSeq: 100 CANCEL  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:\*69@192.168.26.180 SIP/2.0 [192.168.26.10:50373->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: sip:6711@192.168.26.10  
Header: To: <sip:\*69@192.168.26.180>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Expires: 180  
Header: User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: Accept: application/sdp  
Header: Contact: sip:6711@192.168.26.10:5060  
Header: Content-Type: application/sdp  
Header: Content-Length: 219

-----  
SDP Headers

-----  
Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 6339 9080 IN IP4 192.168.26.10  
Header: s=SIP Call  
Header: c=IN IP4 192.168.26.10  
Header: t=0 0  
Header: m=audio 29850 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.26.10:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>

Header: To: <sip:\*69@192.168.26.180:5060>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:\*69@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:\*69@192.168.26.180:5060>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Proxy-Authorization: Basic 123  
Header: Expires: 180  
Header: Record-Route: <sip:\*69@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:6711@192.168.26.10:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 219

-----  
SDP Headers

-----  
Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 6339 9080 IN IP4 192.168.26.10  
Header: s=SIP Call  
Header: c=IN IP4 192.168.26.10  
Header: t=0 0  
Header: m=audio 29850 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:\*69@192.168.26.180:5060>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:\*69@192.168.26.220:6074;user=phone>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: ACK sip:\*69@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:\*69@192.168.26.180:5060>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:\*69@192.168.26.220:6074;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.220:6074]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:\*69@192.168.26.180:5060>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Proxy-Authorization: Basic 123  
Header: Expires: 180  
Header: Record-Route: <sip:\*69@192.168.26.180:5060;maddr=192.168.26.180>



## SIP IP Phone to SIP IP Phone: Call Return

Header: Contact: <sip:6711@192.168.26.10:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 219

-----  
SDP Headers

Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 6339 9080 IN IP4 192.168.26.10  
Header: s=SIP Call  
Header: c=IN IP4 192.168.26.10  
Header: t=0 0  
Header: m=audio 29850 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.26.220:6074->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:\*69@192.168.26.180:5060>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.220:6074->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:\*69@192.168.26.180:5060>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:6715@192.168.26.180:5060;user=phone>  
Header: Content-Length: 0  
Header: CC-Redirect: <sip:6715@192.168.26.180:5060;user=phone>;redir-  
reason=unconditional;redir-counter=0;redir-limit=99

-----  
SIP Headers

-----  
sip-req: ACK sip:\*69@192.168.26.220:6074;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.220:6074]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:\*69@192.168.26.180:5060>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.180:5060->192.168.26.10:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:\*69@192.168.26.180:5060>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:6715@192.168.26.180:5060;user=phone>  
Header: Content-Length: 0  
Header: CC-Redirect: <sip:6715@192.168.26.180:5060;user=phone>;redir-  
reason=unconditional;redir-counter=0;redir-limit=99

-----  
SIP Headers

-----  
sip-req: ACK sip:\*69@192.168.26.180 SIP/2.0 [192.168.26.10:50373->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060



Header: From: sip:6711@192.168.26.10  
Header: To: <sip:\*69@192.168.26.180>  
Header: Call-ID: c2943000-2e262-b27e-2e323931@192.168.26.10  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:6715@192.168.26.180:5060;user=phone SIP/2.0 [192.168.26.10:50373->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: sip:6711@192.168.26.10  
Header: To: sip:6715@192.168.26.180:5060;user=phone  
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10  
Header: CSeq: 101 INVITE  
Header: Expires: 180  
Header: User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: Accept: application/sdp  
Header: Contact: sip:6711@192.168.26.10:5060  
Header: Content-Type: application/sdp  
Header: Content-Length: 219

-----  
SDP Headers

-----  
Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 8962 2811 IN IP4 192.168.26.10  
Header: s=SIP Call  
Header: c=IN IP4 192.168.26.10  
Header: t=0 0  
Header: m=audio 29850 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.26.10:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6715@192.168.26.180:5060>;user=phone  
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10  
Header: CSeq: 101 INVITE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:6715@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6715@192.168.26.180:5060>;user=phone  
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10  
Header: CSeq: 101 INVITE  
Header: Proxy-Authorization: Basic 123  
Header: Expires: 180  
Header: Record-Route: <sip:6715@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:6711@192.168.26.10:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 219

-----  
SDP Headers

-----  
Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 8962 2811 IN IP4 192.168.26.10  
Header: s=SIP Call  
Header: c=IN IP4 192.168.26.10  
Header: t=0 0  
Header: m=audio 29850 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000

## SIP IP Phone to SIP IP Phone: Call Return

Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6715@192.168.26.180:5060>;user=phone  
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10  
Header: CSeq: 101 INVITE  
Header: Contact: <sip:6715@192.168.26.180:5060>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: ACK sip:6715@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6715@192.168.26.180:5060>;user=phone  
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10  
Header: CSeq: 101 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:6715@192.168.26.11:5060 SIP/2.0 [192.168.26.180:5060->192.168.26.11:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6715@192.168.26.180:5060>;user=phone  
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10  
Header: CSeq: 101 INVITE  
Header: Expires: 180  
Header: Record-Route:  
<sip:6715@192.168.26.180:5060;maddr=192.168.26.180>,<sip:6715@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:6711@192.168.26.10:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 219

-----  
SDP Headers

-----  
Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 8962 2811 IN IP4 192.168.26.10  
Header: s=SIP Call  
Header: c=IN IP4 192.168.26.10  
Header: t=0 0  
Header: m=audio 29850 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.26.11:50783->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP 192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.10:5060  
Header: From: <sip:6711@192.168.26.10:5060>  
Header: To: <sip:6715@192.168.26.180:5060>;user=phone;tag=c39430001978b0-0  
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: CSeq: 101 INVITE  
Header: Content-Length: 0

-----  
SIP Headers

```
-----
sip-res: SIP/2.0 180 Ringing [192.168.26.11:50783->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6715@192.168.26.180:5060>;user=phone;tag=c39430001978b0-0
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: CSeq: 101 INVITE
Header: Content-Length: 0
-----
```

SIP Headers

```
-----
sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.26.10:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6715@192.168.26.180:5060>;tag=c39430001978b0-0;user=phone
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10
Header: CSeq: 101 INVITE
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Content-Length: 0
-----
```

SIP Headers

```
-----
sip-res: SIP/2.0 200 OK [192.168.26.11:50783->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6715@192.168.26.180:5060>;user=phone;tag=c39430001978b0-0
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Contact: sip:6715@192.168.26.11:5060
Header: Record-Route:
<sip:6715@192.168.26.180:5060;maddr=192.168.26.180>,<sip:6715@192.168.26.180:5060;maddr=1
92.168.26.180>
Header: CSeq: 101 INVITE
Header: Content-Type: application/sdp
Header: Content-Length: 220
-----
```

SDP Headers

```
-----
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 9104 19747 IN IP4 192.168.26.11
Header: s=SIP Call
Header: c=IN IP4 192.168.26.11
Header: t=0 0
Header: m=audio 29956 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
-----
```

SIP Headers

```
-----
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.26.10:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6715@192.168.26.180:5060>;tag=c39430001978b0-0;user=phone
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10
Header: CSeq: 101 INVITE
Header: Contact: <sip:6715@192.168.26.11:5060>
Header: Record-Route:
<sip:6715@192.168.26.180:5060;maddr=192.168.26.180>,<sip:6715@192.168.26.180:5060;maddr=1
92.168.26.180>
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Content-Type: application/sdp
Header: Content-Length: 220
-----
```

SDP Headers

```
-----
Header: v=0
-----
```

## SIP IP Phone to SIP IP Phone: Call Return

```
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 9104 19747 IN IP4 192.168.26.11
Header: s=SIP Call
Header: c=IN IP4 192.168.26.11
Header: t=0 0
Header: m=audio 29956 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

---

### SIP Headers

```
-----
sip-req: ACK sip:6715@192.168.26.180:5060 SIP/2.0 [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6715@192.168.26.180:5060>;tag=c39430001978b0-0;user=phone
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10
Header: Route: <sip:6715@192.168.26.180:5060;maddr=192.168.26.180>,
<sip:6715@192.168.26.11:5060>
Header: CSeq: 101 ACK
Header: Content-Length: 0
```

---

### SIP Headers

```
-----
sip-req: ACK sip:6715@192.168.26.11:5060 SIP/2.0 [192.168.26.180:5060->192.168.26.11:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From: <sip:6711@192.168.26.10:5060>
Header: To: <sip:6715@192.168.26.180:5060>;tag=c39430001978b0-0;user=phone
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10
Header: CSeq: 101 ACK
Header: Content-Length: 0
```

---

### SIP Headers

```
-----
sip-req: BYE sip:6715@192.168.26.180:5060 SIP/2.0 [192.168.26.11:50783->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.11:5060
Header: From:
Header: <sip:6715@192.168.26.180:5060>;user=phone;tag=c39430001978b0-0
Header: To: <sip:6711@192.168.26.10:5060>
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10
Header: User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: CSeq: 102 BYE
Header: Route: <sip:6715@192.168.26.180:5060;maddr=192.168.26.180>,
<sip:6711@192.168.26.10:5060>
Header: Content-Length: 0
```

---

### SIP Headers

```
-----
sip-req: BYE sip:6711@192.168.26.10:5060 SIP/2.0 [192.168.26.180:5060->192.168.26.10:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.11:5060
Header: From: <sip:6715@192.168.26.180:5060>;tag=c39430001978b0-0;user=phone
Header: To: <sip:6711@192.168.26.10:5060>
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10
Header: CSeq: 102 BYE
Header: Content-Length: 0
```

---

### SIP Headers

```
-----
sip-res: SIP/2.0 200 OK [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP 192.168.26.180:5060;branch=2,SIP/2.0/UDP 192.168.26.11:5060
```

## SIP IP Phone to SIP IP Phone: Call Return

Header: From: <sip:6715@192.168.26.180:5060>;tag=c39430001978b0-  
0;user=phone  
Header: To: <sip:6711@192.168.26.10:5060>  
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: CSeq: 102 BYE  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.26.11:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.11:5060  
Header: From: <sip:6715@192.168.26.180:5060>;tag=c39430001978b0-  
0;user=phone  
Header: To: <sip:6711@192.168.26.10:5060>  
Header: Call-ID: c2943000-3e262-b4b1-2e323931@192.168.26.10  
Header: CSeq: 102 BYE  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: Content-Length: 0  
-----

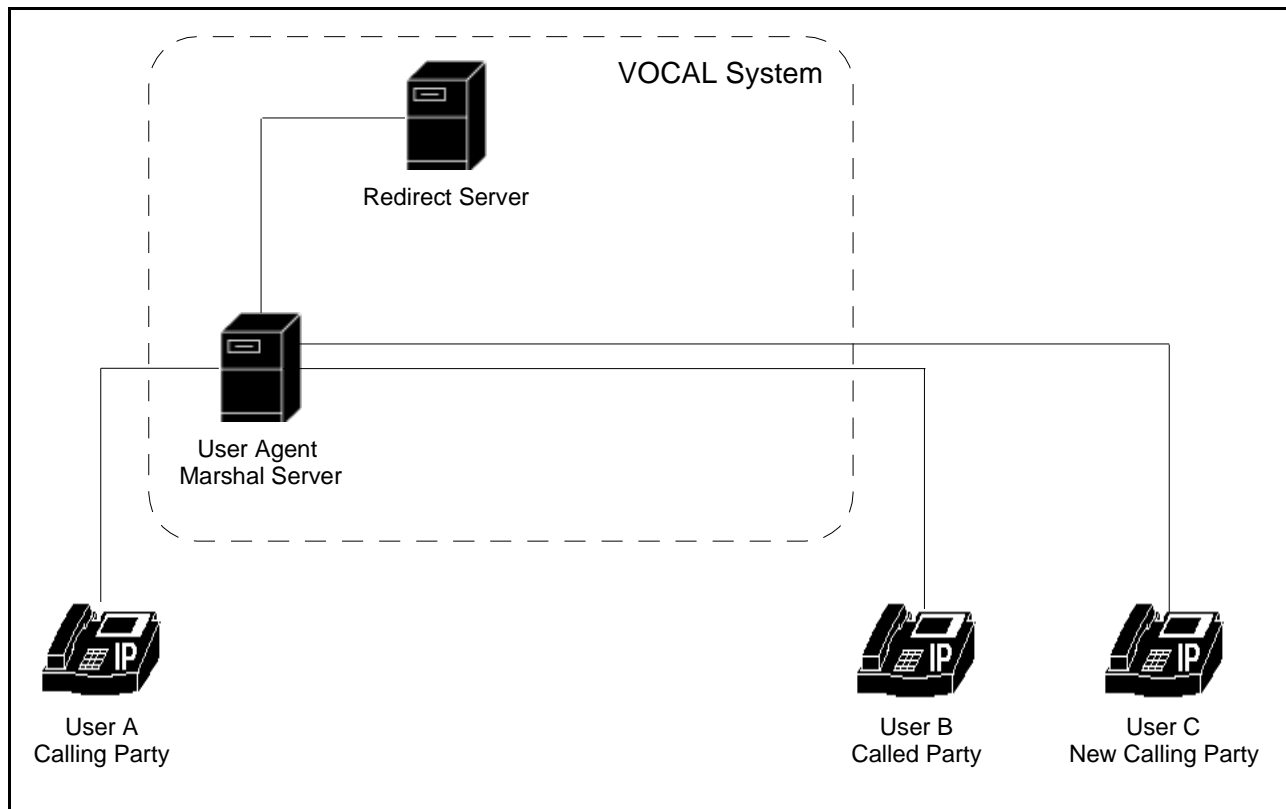
---

## User Agent to User Agent: Call Waiting

### Call Scenario

Figure C-20 illustrates the following call scenario:

- User A calls User B
- While Users A and B are in conversation, User C calls User A
- User A is notified that another caller attempting to connect



**Figure C-20. User Agent to User Agent: Call Waiting**

### Call Flow Diagram

Figures C-21 and C-22 show a third party attempting to connect to a phone that is engaged in conversation with another phone.

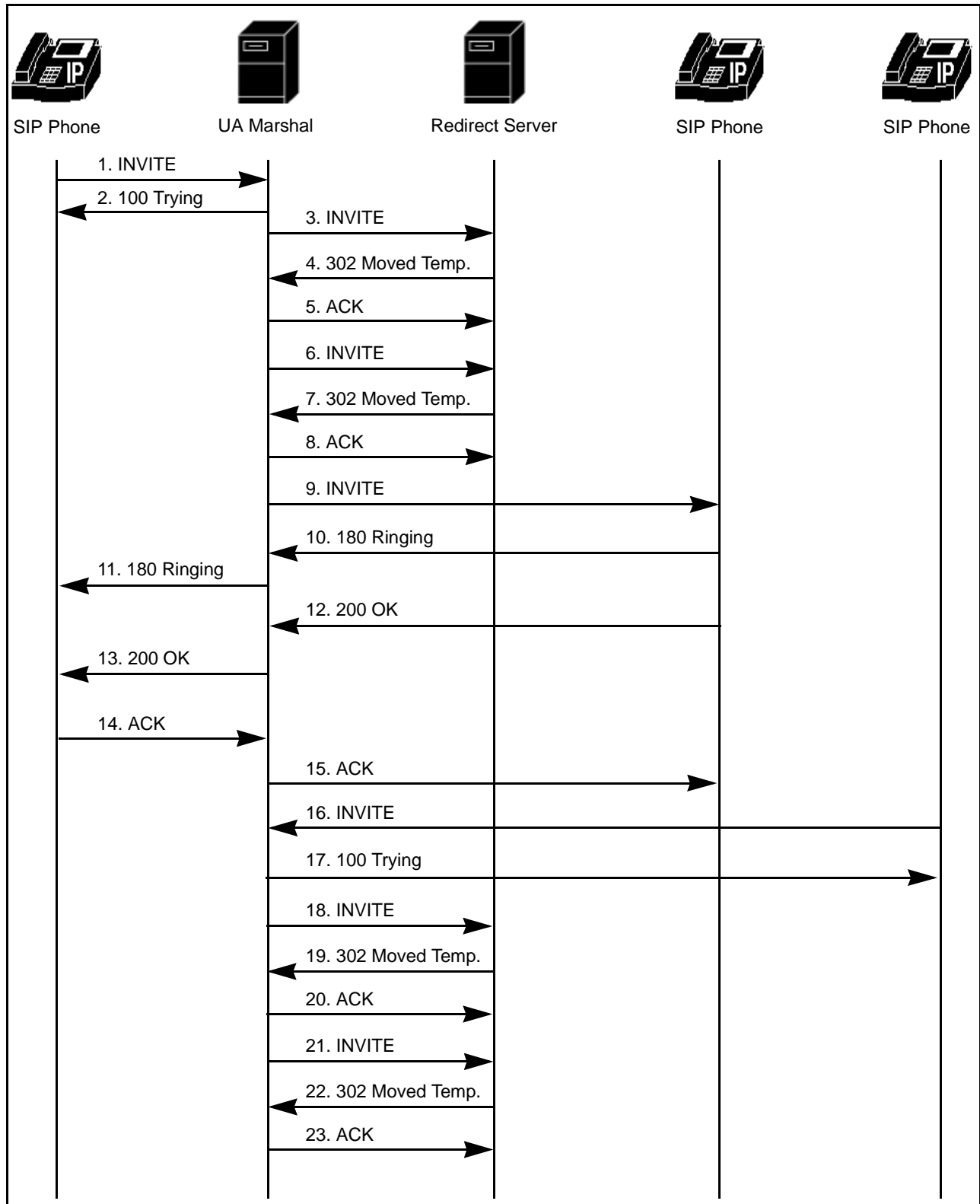
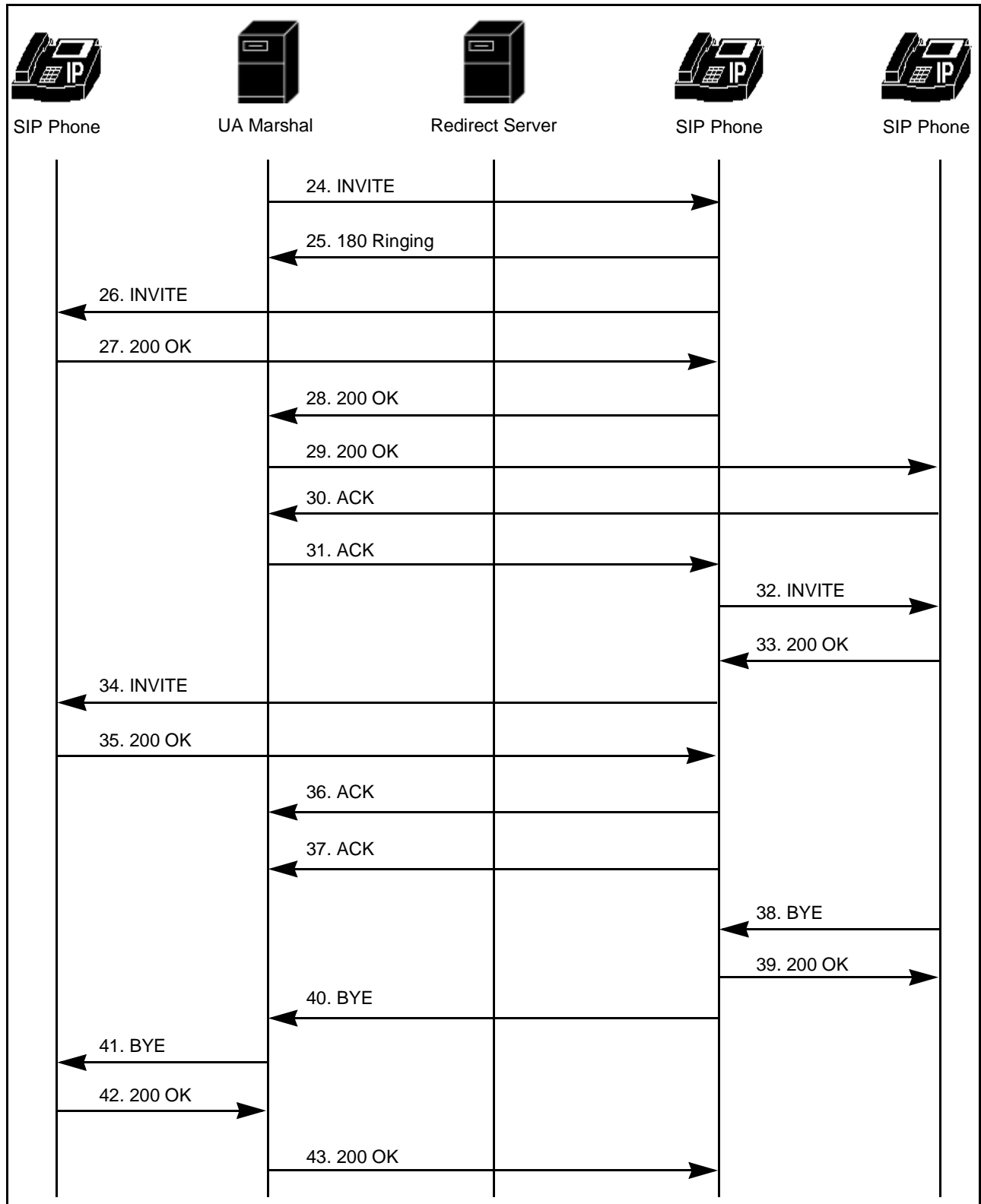


Figure C-21. User Agent to User Agent: Call Waiting — Diagram 1

*User Agent to User Agent: Call Waiting*



**Figure C-22. User Agent to User Agent: Call Waiting — Diagram 2**



Call Trace

The following call trace shows a third party attempting to connect to a phone that is engaged in conversation with another phone.

```

=====
                        A calls B
-----
                        SIP Headers
-----
sip-req:    INVITE sip:5220@192.168.66.180:5060;user=phone SIP/2.0 [192.168.66.1:5060->192.168.66.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header:    To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:    Call-ID: 4732a6465cfdfbdc0d38708c0728708@192.168.66.1
Header:    CSeq: 1 INVITE
Header:    Subject: VovidaINVITE
Header:    Contact: <sip:5221@192.168.66.1:5060;user=phone>
Header:    Content-Type: application/sdp
Header:    Content-Length: 168
-----
                        SDP Headers
-----
Header:    v=0
Header:    o=- 1528076688 1528076688 IN IP4 192.168.66.1
Header:    s=VOVIDA Session
Header:    c=IN IP4 192.168.66.1
Header:    t=3177769010 0
Header:    m=audio 56104 RTP/AVP 0
Header:    a=rtpmap:0 PCMU/8000
Header:    a=ptime:20
Header:    7
-----
                        SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.66.180:5060->192.168.66.1:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header:    To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:    Call-ID: 4732a6465cfdfbdc0d38708c0728708@192.168.66.1
Header:    CSeq: 1 INVITE
Header:    Content-Length: 0
-----
                        SIP Headers
-----
sip-req:    INVITE sip:5220@192.168.66.200:5060;user=phone SIP/2.0 [192.168.66.180:5060->192.168.66.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.180:5060;branch=1
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header:    To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:    Call-ID: 4732a6465cfdfbdc0d38708c0728708@192.168.66.1
Header:    CSeq: 1 INVITE
Header:    Proxy-Authorization: Basic 123
Header:    Subject: VovidaINVITE
Header:    Record-Route: <sip:5220@192.168.66.180:5060;maddr=192.168.66.180>
Header:    Contact: <sip:5221@192.168.66.1:5060;user=phone>
Header:    Content-Type: application/sdp
Header:    Content-Length: 168
-----
                        SDP Headers
-----
Header:    v=0
Header:    o=- 1528076688 1528076688 IN IP4 192.168.66.1
Header:    s=VOVIDA Session
Header:    c=IN IP4 192.168.66.1
Header:    t=3177769010 0
Header:    m=audio 56104 RTP/AVP 0
Header:    a=rtpmap:0 PCMU/8000
Header:    a=ptime:20

```

## User Agent to User Agent: Call Waiting

```
Header: F
-----
SIP Headers
-----
sip-res: SIP/2.0 302 Moved Temporarily [192.168.66.200:5060->192.168.66.180:5060]
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=1
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1
Header: CSeq: 1 INVITE
Header: Contact: <sip:5220@192.168.66.180:5060>
Header: Content-Length: 0
-----
SIP Headers
-----
sip-req: ACK sip:5220@192.168.66.200:5060;user=phone SIP/2.0 [192.168.66.180:5060->192.168.66.200:5060]
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=1
Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1
Header: CSeq: 1 ACK
Header: Content-Length: 0
-----
SIP Headers
-----
sip-req: INVITE sip:5220@192.168.66.200:5060;user=phone SIP/2.0 [192.168.66.180:5060->192.168.66.200:5060]
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=3
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1
Header: CSeq: 1 INVITE
Header: Subject: VovidaINVITE
Header: Record-Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>, <sip:5220@192.168.66.180:5060;maddr=192.168.66.180>
Header: Contact: <sip:5221@192.168.66.1:5060;user=phone>
Header: Content-Type: application/sdp
Header: Content-Length: 168
-----
SDP Headers
-----
Header: v=0
Header: o=- 1528076688 1528076688 IN IP4 192.168.66.1
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.66.1
Header: t=3177769010 0
Header: m=audio 56104 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
Header: u
-----
SIP Headers
-----
sip-res: SIP/2.0 302 Moved Temporarily [192.168.66.200:5060->192.168.66.180:5060]
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=3
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1
Header: CSeq: 1 INVITE
Header: Contact: <sip:5220@192.168.66.2:5060>
Header: Content-Length: 0
-----
SIP Headers
-----
```

-----  
sip-req: ACK sip:5220@192.168.66.200:5060;user=phone SIP/2.0 [192.168.66.180:5060->192.168.66.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=3  
Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1  
Header: CSeq: 1 ACK  
Header: Content-Length: 0  
-----

SIP Headers

-----  
sip-req: INVITE sip:5220@192.168.66.2:5060 SIP/2.0 [192.168.66.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Subject: VovidaINVITE  
Header: Record-Route:  
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>,<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>  
Header: Contact: <sip:5221@192.168.66.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168  
-----

SDP Headers

-----  
Header: v=0  
Header: o=- 1528076688 1528076688 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.1  
Header: t=3177769010 0  
Header: m=audio 56104 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header:  
a  
-----

SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.66.2:5060->192.168.66.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Content-Length: 0  
-----

SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.66.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Content-Length: 0  
-----

SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.66.2:5060->192.168.66.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
-----

## User Agent to User Agent: Call Waiting

Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Contact: <sip:5220@192.168.66.2:5060>  
Header: Record-Route:  
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>, <sip:5220@192.168.66.180:5060;maddr=192.168.66.180>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

-----  
SDP Headers

Header: v=0  
Header: o=- 1528076688 1528076688 IN IP4 192.168.66.2  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.2  
Header: t=3177769010 0  
Header: m=audio 56114 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: aD

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.66.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Contact: <sip:5220@192.168.66.2:5060>  
Header: Record-Route:  
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>, <sip:5220@192.168.66.180:5060;maddr=192.168.66.180>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

-----  
SDP Headers

Header: v=0  
Header: o=- 1528076688 1528076688 IN IP4 192.168.66.2  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.2  
Header: t=3177769010 0  
Header: m=audio 56114 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: ;

-----  
SIP Headers

-----  
sip-req: ACK sip:5220@192.168.66.180:5060;maddr=192.168.66.180 SIP/2.0  
[192.168.66.1:5060->192.168.66.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1  
Header: CSeq: 1 ACK  
Header: Route:  
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>, <sip:5220@192.168.66.2:5060>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: ACK sip:5220@192.168.66.2:5060 SIP/2.0 [192.168.66.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060

## User Agent to User Agent: Call Waiting

```
Header:      From: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header:      To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:      Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1
Header:      CSeq: 1 ACK
Header:      Content-Length: 0
-----
A and B are talking
=====
C now calls B
-----
SIP Headers
-----
sip-req:     INVITE sip:5220@192.168.66.180:5060;user=phone SIP/2.0 [192.168.66.3:5060->192.168.66.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.66.3:5060
Header:      From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header:      To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:      Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header:      CSeq: 1 INVITE
Header:      Subject: VovidaINVITE
Header:      Contact: <sip:5222@192.168.66.3:5060;user=phone>
Header:      Content-Type: application/sdp
Header:      Content-Length: 168
-----
SDP Headers
-----
Header:      v=0
Header:      o=- 1089208644 1089208644 IN IP4 192.168.66.3
Header:      s=VOVIDA Session
Header:      c=IN IP4 192.168.66.3
Header:      t=3177769022 0
Header:      m=audio 60335 RTP/AVP 0
Header:      a=rtpmap:0 PCMU/8000
Header:      a=ptime:20
-----
SIP Headers
-----
sip-res:     SIP/2.0 100 Trying [192.168.66.180:5060->192.168.66.3:5060]
Header:      Via: SIP/2.0/UDP 192.168.66.3:5060
Header:      From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header:      To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:      Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header:      CSeq: 1 INVITE
Header:      Content-Length: 0
-----
SIP Headers
-----
sip-req:     INVITE sip:5220@192.168.66.200:5060;user=phone SIP/2.0 [192.168.66.180:5060->192.168.66.200:5060]
Header:      Via: SIP/2.0/UDP 192.168.66.180:5060;branch=1
Header:      Via: SIP/2.0/UDP 192.168.66.3:5060
Header:      From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header:      To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:      Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header:      CSeq: 1 INVITE
Header:      Proxy-Authorization: Basic 123
Header:      Subject: VovidaINVITE
Header:      Record-Route: <sip:5220@192.168.66.180:5060;maddr=192.168.66.180>
Header:      Contact: <sip:5222@192.168.66.3:5060;user=phone>
Header:      Content-Type: application/sdp
Header:      Content-Length: 168
-----
SDP Headers
-----
Header:      v=0
Header:      o=- 1089208644 1089208644 IN IP4 192.168.66.3
Header:      s=VOVIDA Session
Header:      c=IN IP4 192.168.66.3
Header:      t=3177769022 0
```

## User Agent to User Agent: Call Waiting

Header: m=audio 60335 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: \*

---

### SIP Headers

---

sip-res: SIP/2.0 302 Moved Temporarily [192.168.66.200:5060->192.168.66.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.66.3:5060  
Header: From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3  
Header: CSeq: 1 INVITE  
Header: Contact: <sip:5220@192.168.66.180:5060>  
Header: Content-Length: 0

---

### SIP Headers

---

sip-req: ACK sip:5220@192.168.66.200:5060;user=phone SIP/2.0 [192.168.66.180:5060->192.168.66.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=1  
Header: From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3  
Header: CSeq: 1 ACK  
Header: Content-Length: 0

---

### SIP Headers

---

sip-req: INVITE sip:5220@192.168.66.200:5060;user=phone SIP/2.0 [192.168.66.180:5060->192.168.66.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=3  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.3:5060  
Header: From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3  
Header: CSeq: 1 INVITE  
Header: Subject: VovidaINVITE  
Header: Record-Route:  
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>, <sip:5220@192.168.66.180:5060;maddr=192.168.66.180>  
Header: Contact: <sip:5222@192.168.66.3:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

---

### SDP Headers

---

Header: v=0  
Header: o=- 1089208644 1089208644 IN IP4 192.168.66.3  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.3  
Header: t=3177769022 0  
Header: m=audio 60335 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: u

---

### SIP Headers

---

sip-res: SIP/2.0 302 Moved Temporarily [192.168.66.200:5060->192.168.66.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=3  
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.3:5060  
Header: From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>  
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>  
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3  
Header: CSeq: 1 INVITE  
Header: Contact: <sip:5220@192.168.66.2:5060>

```
Header:          Content-Length: 0
-----
SIP Headers
-----
sip-req:  ACK sip:5220@192.168.66.200:5060;user=phone SIP/2.0 [192.168.66.180:5060-
>192.168.66.200:5060]
Header:          Via: SIP/2.0/UDP 192.168.66.180:5060;branch=3
Header:          From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header:          To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:          Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header:          CSeq: 1 ACK
Header:          Content-Length: 0
-----
SIP Headers
-----
sip-req:  INVITE sip:5220@192.168.66.2:5060 SIP/2.0 [192.168.66.180:5060-
>192.168.66.2:5060]
Header:          Via: SIP/2.0/UDP 192.168.66.180:5060;branch=4
Header:          Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2
Header:          Via: SIP/2.0/UDP 192.168.66.3:5060
Header:          From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header:          To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:          Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header:          CSeq: 1 INVITE
Header:          Subject: VovidaINVITE
Header:          Record-Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>, <sip:5220@192.168.66.180:5060;maddr=1
92.168.66.180>
Header:          Contact: <sip:5222@192.168.66.3:5060;user=phone>
Header:          Content-Type: application/sdp
Header:          Content-Length: 168
-----
SDP Headers
-----
Header:          v=0
Header:          o=- 1089208644 1089208644 IN IP4 192.168.66.3
Header:          s=VOVIDA Session
Header:          c=IN IP4 192.168.66.3
Header:          t=3177769022 0
Header:          m=audio 60335 RTP/AVP 0
Header:          a=rtpmap:0 PCMU/8000
Header:          a=ptime:20
-----
SIP Headers
-----
sip-res:  SIP/2.0 180 Ringing [192.168.66.2:5060->192.168.66.180:5060]
Header:          Via: SIP/2.0/UDP 192.168.66.180:5060;branch=4
Header:          Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2
Header:          Via: SIP/2.0/UDP 192.168.66.3:5060
Header:          From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header:          To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:          Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header:          CSeq: 1 INVITE
Header:          Content-Length: 0
=====
B places A on hold
-----
SIP Headers
-----
sip-req:  INVITE sip:5221@192.168.66.1:5060;user=phone SIP/2.0 [192.168.66.2:5060-
>192.168.66.1:5060]
Header:          Via: SIP/2.0/UDP 192.168.66.2:5060
Header:          From: <sip:5220@192.168.66.180:5060;user=phone>
Header:          To: 5221<sip:5221@192.168.66.1:5060;user=phone>
Header:          Call-ID: 4732a6465cfdffbd60c98708c0728708@192.168.66.1
Header:          CSeq: 2 INVITE
Header:          Subject: VovidaINVITE
Header:          Contact: <sip:5220@192.168.66.2:5060;user=phone>
```



## User Agent to User Agent: Call Waiting

```
Header:          Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>, <sip:5220@192.168.66.180:5060;maddr=1
92.168.66.180>, <sip:5221@192.168.66.1:5060>
Header:          Content-Type: application/sdp
Header:          Content-Length: 126
-----
                        SDP Headers
-----
Header:          v=0
Header:          o=- 944563072 944563072 IN IP4 192.168.66.2
Header:          s=VOVIDA Session
Header:          c=IN IP4 0.0.0.0
Header:          t=3177769026 0
Header:          m=audio 3456 RTP/AVP 0
Header:          VI
-----
                        SIP Headers
-----
sip-res:        SIP/2.0 200 OK [192.168.66.1:5060->192.168.66.2:5060]
Header:          Via: SIP/2.0/UDP 192.168.66.1:5060
Header:          From: <sip:5220@192.168.66.180:5060;user=phone>
Header:          To: 5221<sip:5221@192.168.66.2:5060;user=phone>
Header:          Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1
Header:          CSeq: 2 INVITE
Header:          Contact: <sip:5220@192.168.66.2:5060;user=phone>
Header:          Content-Type: application/sdp
Header:          Content-Length: 131
-----
                        SDP Headers
-----
Header:          v=0
Header:          o=- 944563072 944563072 IN IP4 192.168.66.1
Header:          s=VOVIDA Session
Header:          c=IN IP4 192.168.66.1
Header:          t=3177769026 0
Header:          m=audio 3456 RTP/AVP 0
Header:          60
=====
                        B acknowledges C
-----
                        SIP Headers
-----
sip-res:        SIP/2.0 200 OK [192.168.66.2:5060->192.168.66.180:5060]
Header:          Via: SIP/2.0/UDP 192.168.66.180:5060;branch=4
Header:          Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2
Header:          Via: SIP/2.0/UDP 192.168.66.3:5060
Header:          From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header:          To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:          Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header:          CSeq: 1 INVITE
Header:          Contact: <sip:5220@192.168.66.2:5060>
Header:          Record-Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>, <sip:5220@192.168.66.180:5060;maddr=1
92.168.66.180>
Header:          Content-Type: application/sdp
Header:          Content-Length: 168
-----
                        SDP Headers
-----
Header:          v=0
Header:          o=- 1089208644 1089208644 IN IP4 192.168.66.2
Header:          s=VOVIDA Session
Header:          c=IN IP4 192.168.66.2
Header:          t=3177769022 0
Header:          m=audio 60345 RTP/AVP 0
Header:          a=rtpmap:0 PCMU/8000
Header:          a=ptime:20
-----
                        SIP Headers
-----
```



```
sip-res: SIP/2.0 200 OK [192.168.66.180:5060-
>192.168.66.3:5060]
Header: Via: SIP/2.0/UDP 192.168.66.3:5060
Header: From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header: CSeq: 1 INVITE
Header: Contact: <sip:5220@192.168.66.2:5060>
Header: Record-Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>,<sip:5220@192.168.66.180:5060;maddr=1
92.168.66.180>
Header: Content-Type: application/sdp
Header: Content-Length: 168
```

-----
SDP Headers

```
Header: v=0
Header: o=- 1089208644 1089208644 IN IP4 192.168.66.2
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.66.2
Header: t=3177769022 0
Header: m=audio 60345 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
```

-----
SIP Headers

```
-----
sip-req: ACK sip:5220@192.168.66.180:5060;maddr=192.168.66.180 SIP/2.0
[192.168.66.3:5060->192.168.66.180:5060]
Header: Via: SIP/2.0/UDP 192.168.66.3:5060
Header: From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header: CSeq: 1 ACK
Header: Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>,<sip:5220@192.168.66.2:5060>
Header: Content-Length: 0
```

-----
SIP Headers

```
-----
sip-req: ACK sip:5220@192.168.66.2:5060 SIP/2.0 [192.168.66.180:5060-
>192.168.66.2:5060]
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.66.3:5060
Header: From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header: CSeq: 1 ACK
Header: Content-Length: 0
```

=====
B leaves C and resume talking to A

-----
SIP Headers

```
-----
sip-req: INVITE sip:5222@192.168.66.3:5060;user=phone SIP/2.0 [192.168.66.2:5060-
>192.168.66.3:5060]
Header: Via: SIP/2.0/UDP 192.168.66.2:5060
Header: From: <sip:5220@192.168.66.180:5060;user=phone>
Header: To: 5222<sip:5222@192.168.66.3:5060;user=phone>
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header: CSeq: 2 INVITE
Header: Subject: VovidaINVITE
Header: Contact: <sip:5220@192.168.66.2:5060;user=phone>
Header: Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>,<sip:5220@192.168.66.180:5060;maddr=1
92.168.66.180>,<sip:5222@192.168.66.3:5060>
Header: Content-Type: application/sdp
Header: Content-Length: 128
```

## User Agent to User Agent: Call Waiting

```
-----
SDP Headers
-----
Header:          v=0
Header:          o=- 1573383876 1573383876 IN IP4 192.168.66.2
Header:          s=VOVIDA Session
Header:          c=IN IP4 0.0.0.0
Header:          t=3177769031 0
Header:          m=audio 3456 RTP/AVP 0
Header:
-----
SIP Headers
-----
sip-res:        SIP/2.0 200 OK [192.168.66.3:5060->192.168.66.2:5060]
Header:        Via: SIP/2.0/UDP 192.168.66.2:5060
Header:        From: <sip:5220@192.168.66.180:5060;user=phone>
Header:        To: 5222<sip:5222@192.168.66.3:5060;user=phone>
Header:        Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header:        CSeq: 2 INVITE
Header:        Contact: <sip:5222@192.168.66.3:5060;user=phone>
Header:        Content-Type: application/sdp
Header:        Content-Length: 133
-----
SDP Headers
-----
Header:          v=0
Header:          o=- 1573383876 1573383876 IN IP4 192.168.66.3
Header:          s=VOVIDA Session
Header:          c=IN IP4 192.168.66.3
Header:          t=3177769031 0
Header:          m=audio 3456 RTP/AVP 0
Header:          68
-----
SIP Headers
-----
sip-req:        INVITE sip:5221@192.168.66.1:5060;user=phone SIP/2.0 [192.168.66.2:5060->192.168.66.1:5060]
Header:        Via: SIP/2.0/UDP 192.168.66.2:5060
Header:        From: <sip:5220@192.168.66.180:5060;user=phone>
Header:        To: 5221<sip:5221@192.168.66.1:5060;user=phone>
Header:        Call-ID: 4732a6465cfdffbd60c0728708@192.168.66.1
Header:        CSeq: 3 INVITE
Header:        Subject: VovidaINVITE
Header:        Contact: <sip:5220@192.168.66.2:5060;user=phone>
Header:        Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>,<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>,<sip:5221@192.168.66.1:5060>
Header:        Content-Type: application/sdp
Header:        Content-Length: 168
-----
SDP Headers
-----
Header:          v=0
Header:          o=- 1528076688 1528076688 IN IP4 192.168.66.2
Header:          s=VOVIDA Session
Header:          c=IN IP4 192.168.66.2
Header:          t=3177769010 0
Header:          m=audio 56114 RTP/AVP 0
Header:          a=rtpmap:0 PCMU/8000
Header:          a=ptime:20
Header:
-----
SIP Headers
-----
sip-res:        SIP/2.0 200 OK [192.168.66.1:5060->192.168.66.2:5060]
Header:        Via: SIP/2.0/UDP 192.168.66.2:5060
Header:        From: <sip:5220@192.168.66.180:5060;user=phone>
Header:        To: 5221<sip:5221@192.168.66.1:5060;user=phone>
Header:        Call-ID: 4732a6465cfdffbd60c0728708@192.168.66.1
Header:        CSeq: 3 INVITE
```

```
Header: Contact: <sip:5221@192.168.66.1:5060;user=phone>
Header: Content-Type: application/sdp
Header: Content-Type: application/sdp
Header: Content-Length: 168
Header: Content-Length: 168
-----
SDP Headers
-----
Header: v=0
Header: o=- 1528076688 1528076688 IN IP4 192.168.66.1
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.66.1
Header: t=3177769010 0
Header: m=audio 56124 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
-----
SIP Headers
-----
sip-req: ACK sip:5220@192.168.66.180:5060;maddr=192.168.66.180 SIP/2.0
[192.168.66.2:5060->192.168.66.180:5060]
Header: Via: SIP/2.0/UDP 192.168.66.2:5060
Header: From: <sip:5220@192.168.66.180:5060;user=phone>
Header: To: 5222<sip:5222@192.168.66.3:5060;user=phone>
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header: CSeq: 2 ACK
Header: Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>,<sip:5222@192.168.66.3:5060>
Header: Content-Length: 0
-----
SIP Headers
-----
sip-req: ACK sip:5220@192.168.66.180:5060;maddr=192.168.66.180 SIP/2.0
[192.168.66.2:5060->192.168.66.180:5060]
Header: Via: SIP/2.0/UDP 192.168.66.2:5060
Header: From: <sip:5220@192.168.66.180:5060;user=phone>
Header: To: 5221<sip:5221@192.168.66.1:5060;user=phone>
Header: Call-ID: 4732a6465cfdffbd60c98708c0728708@192.168.66.1
Header: CSeq: 3 ACK
Header: Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>,<sip:5221@192.168.66.1:5060>
Header: Content-Length: 0
=====
C hangs up on B
-----
SIP Headers
-----
sip-req: BYE sip:5220@192.168.66.2:5060;user=phone SIP/2.0 [192.168.66.3:5060-
>192.168.66.2:5060]
Header: Via: SIP/2.0/UDP 192.168.66.3:5060
Header: From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header: CSeq: 2 BYE
Header: Content-Length: 0
-----
SIP Headers
-----
sip-res: SIP/2.0 200 OK [192.168.66.2:5060->192.168.66.3:5060]
Header: Via: SIP/2.0/UDP 192.168.66.3:5060
Header: From: UserAgent<sip:5222@192.168.66.3:5060;user=phone>
Header: To: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header: Call-ID: 52f78a2b5cfdffbd60c98708a0688708@192.168.66.3
Header: CSeq: 2 BYE
Header: Content-Length: 0
=====
B hangs up on A
-----
SIP Headers
-----
```

## User Agent to User Agent: Call Waiting

```
sip-req:    BYE sip:5220@192.168.66.180:5060;maddr=192.168.66.180 SIP/2.0
[192.168.66.2:5060->192.168.66.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.2:5060
Header:    From: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:    To: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header:    Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1
Header:    CSeq: 4 BYE
Header:    Route:
<sip:5220@192.168.66.180:5060;maddr=192.168.66.180>,<sip:5221@192.168.66.1:5060>
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req:    BYE sip:5221@192.168.66.1:5060 SIP/2.0          [192.168.66.180:5060-
>192.168.66.1:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.66.2:5060
Header:    From: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:    To: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header:    Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1
Header:    CSeq: 4 BYE
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res:    SIP/2.0 200 OK [192.168.66.1:5060->192.168.66.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.66.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.66.2:5060
Header:    From: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:    To: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header:    Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1
Header:    CSeq: 4 BYE
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----

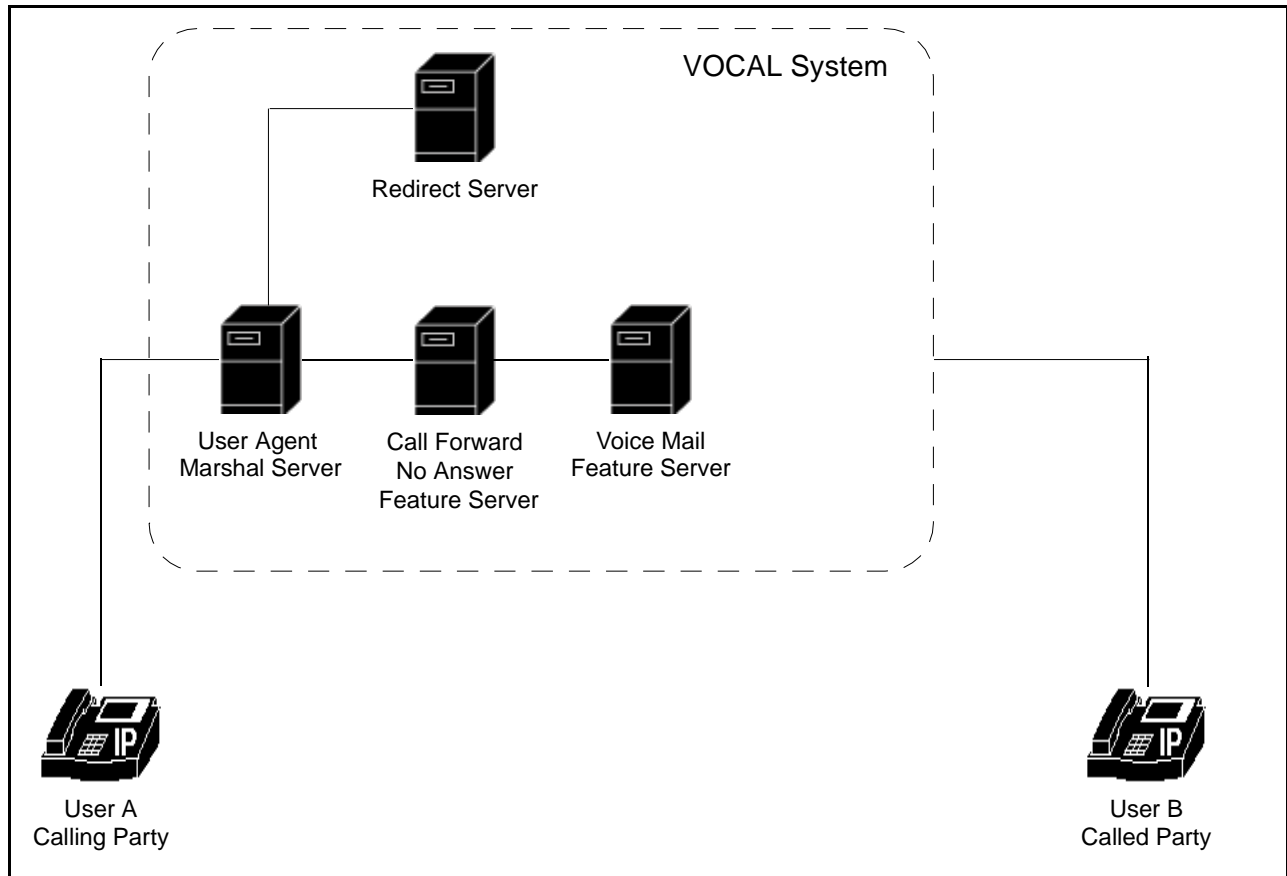
```
sip-res:    SIP/2.0 200 OK [192.168.66.180:5060->192.168.66.2:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.2:5060
Header:    From: 5220<sip:5220@192.168.66.180:5060;user=phone>
Header:    To: UserAgent<sip:5221@192.168.66.1:5060;user=phone>
Header:    Call-ID: 4732a6465cfdffbd0d38708c0728708@192.168.66.1
Header:    CSeq: 4 BYE
Header:    Content-Length: 0
```

## SIP IP Phone to SIP IP Phone: Forward to Voice Mail

### Call Scenario

Figure C-23 illustrates the following call scenario:

- User A calls User B
- User B does not answer the call
- The call is forwarded to the voice mail feature server

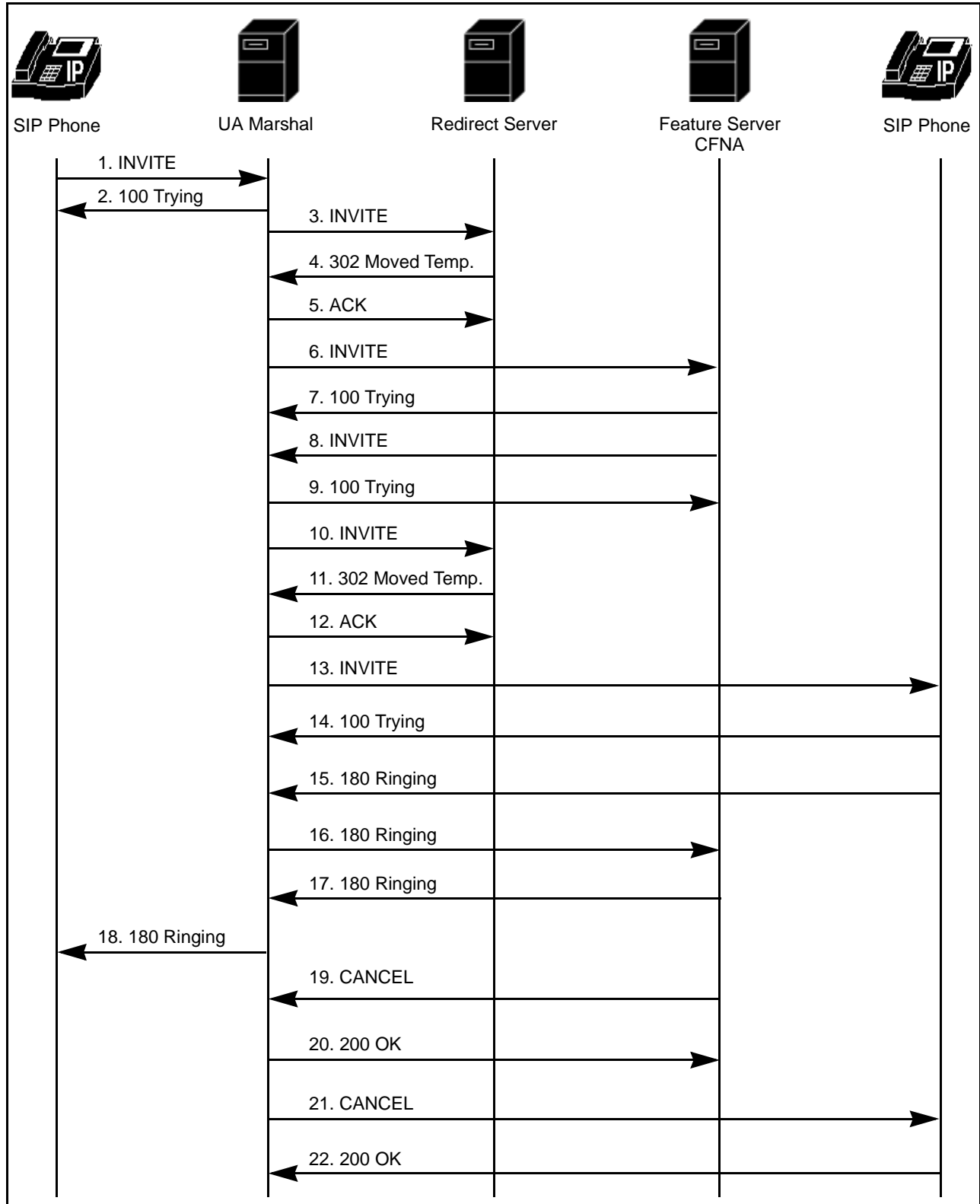


**Figure C-23. SIP IP Phone to SIP IP Phone: Forward to Voice Mail**

### Call Flow Diagram

Figures C-24 and C-25 show a SIP IP phone attempting to call another on-network SIP IP phone. The second phone is unanswred and the call is re-initiated with the Voice Mail server.

*SIP IP Phone to SIP IP Phone: Forward to Voice Mail*



**Figure C-24. User Agent to User Agent: Call Waiting — Diagram 1**

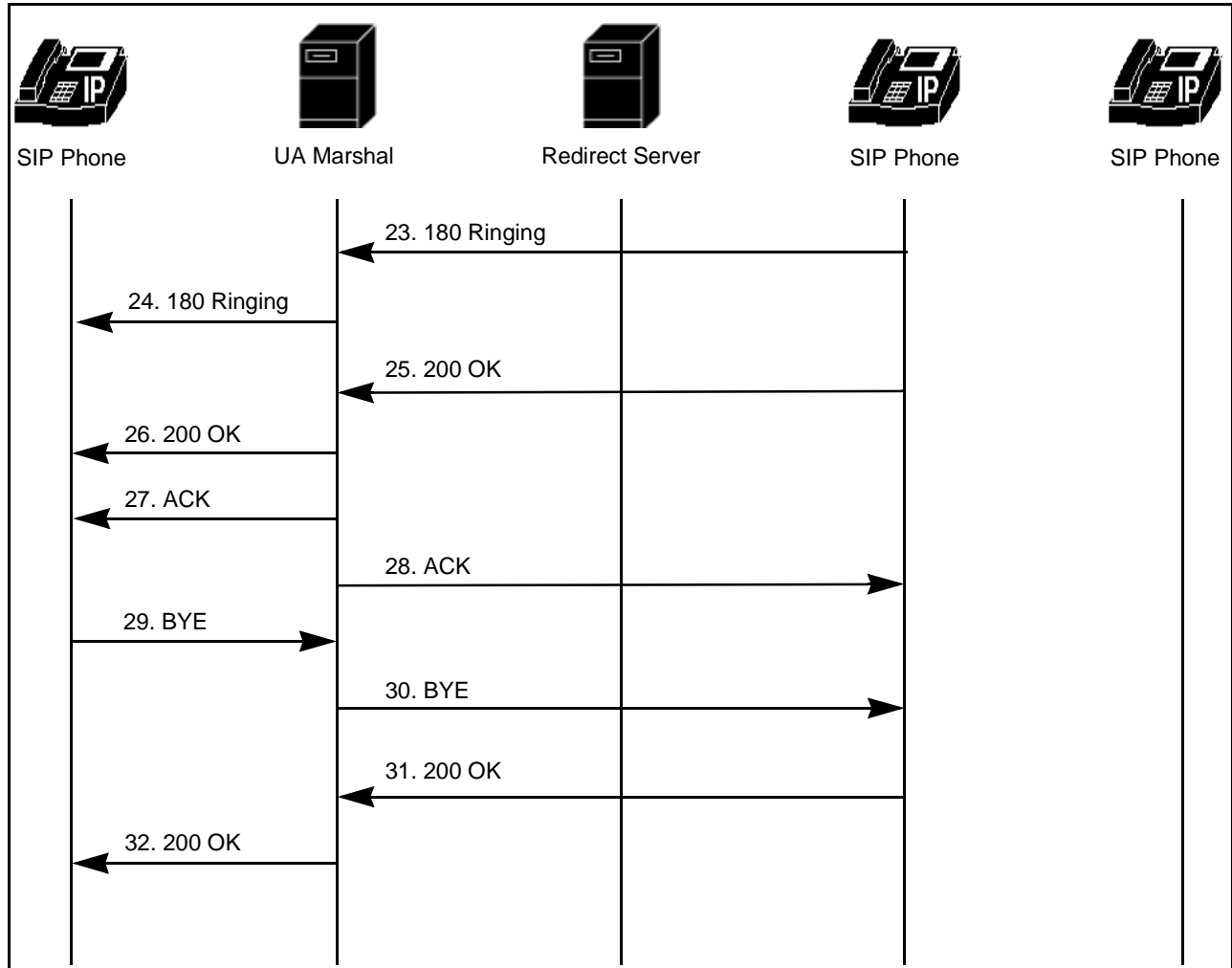


Figure C-25. User Agent to User Agent: Call Waiting — Diagram 2

**Call Trace**

The following call trace shows a SIP IP phone attempting to call another on-network SIP IP phone. The second phone is unanswred and the call is re-initiated with the Voice Mail server.

```

-----
SIP Headers
-----
sip-req:  INVITE sip:5210@192.168.56.180 SIP/2.0      [192.168.10.18:50443-
>192.168.56.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.10.18:5060
Header:      From: sip:5218@192.168.10.18
Header:      To: <sip:5210@192.168.56.180>
Header:      Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18
Header:      CSeq: 100 INVITE
Header:      Expires: 180
Header:      User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:      Accept: application/sdp
Header:      Contact: sip:5218@192.168.10.18:5060
Header:      Content-Type: application/sdp
Header:      Content-Length: 221
-----
SDP Headers
-----

```

## SIP IP Phone to SIP IP Phone: Forward to Voice Mail

```
-----
Header:          v=0
Header:          o=CiscoSystemsSIP-IPPhone-UserAgent 25678 28140 IN IP4
192.168.10.18
Header:          s=SIP Call
Header:          c=IN IP4 192.168.10.18
Header:          t=0 0
Header:          m=audio 23994 RTP/AVP 0 101
Header:          a=rtpmap:0 pcmu/8000
Header:          a=rtpmap:101 telephone-event/8000
Header:          a=fmtp:101 0-11
-----
```

### SIP Headers

```
-----
sip-res:        SIP/2.0 100 Trying [192.168.56.180:5060->192.168.10.18:5060]
Header:         Via: SIP/2.0/UDP 192.168.10.18:5060
Header:         From: <sip:5218@192.168.10.18:5060>
Header:         To: <sip:5210@192.168.56.180:5060>
Header:         Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18
Header:         CSeq: 100 INVITE
Header:         Content-Length: 0
-----
```

### SIP Headers

```
-----
sip-req:        INVITE sip:5210@192.168.56.200:5060;user=phone SIP/2.0 [192.168.56.180:5060-
>192.168.56.200:5060]
Header:         Via: SIP/2.0/UDP 192.168.56.180:5060;branch=1
Header:         Via: SIP/2.0/UDP 192.168.10.18:5060
Header:         From: <sip:5218@192.168.10.18:5060>
Header:         To: <sip:5210@192.168.56.180:5060>
Header:         Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18
Header:         CSeq: 100 INVITE
Header:         Proxy-Authorization: Basic VovidaClassXSwitch
Header:         Expires: 180
Header:         Record-Route: <sip:5210@192.168.56.180:5060;maddr=192.168.56.180>
Header:         Contact: <sip:5218@192.168.10.18:5060>
Header:         Content-Type: application/sdp
Header:         Content-Length: 221
-----
```

### SDP Headers

```
-----
Header:          v=0
Header:          o=CiscoSystemsSIP-IPPhone-UserAgent 25678 28140 IN IP4
192.168.10.18
Header:          s=SIP Call
Header:          c=IN IP4 192.168.10.18
Header:          t=0 0
Header:          m=audio 23994 RTP/AVP 0 101
Header:          a=rtpmap:0 pcmu/8000
Header:          a=rtpmap:101 telephone-event/8000
Header:          a=fmtp:101 0-11
-----
```

### SIP Headers

```
-----
sip-res:        SIP/2.0 302 Moved Temporarily [192.168.56.200:5060->192.168.56.180:5060]
Header:         Via: SIP/2.0/UDP 192.168.56.180:5060;branch=1
Header:         Via: SIP/2.0/UDP 192.168.10.18:5060
Header:         From: <sip:5218@192.168.10.18:5060>
Header:         To: <sip:5210@192.168.56.180:5060>
Header:         Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18
Header:         CSeq: 100 INVITE
Header:         Contact: <sip:5210@192.168.56.220:5074;user=phone>
Header:         Content-Length: 0
-----
```

### SIP Headers

```
-----
sip-req:        ACK sip:5210@192.168.56.200:5060;user=phone SIP/2.0 [192.168.56.180:5060-
>192.168.56.200:5060]
Header:         Via: SIP/2.0/UDP 192.168.56.180:5060;branch=1
Header:         From: <sip:5218@192.168.10.18:5060>
-----
```



## SIP IP Phone to SIP IP Phone: Forward to Voice Mail

Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

---

### SIP Headers

---

sip-req: INVITE sip:5210@192.168.56.220:5074;user=phone SIP/2.0 [192.168.56.180:5060->192.168.56.220:5074]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Proxy-Authorization: Basic VovidaClassXSwitch  
Header: Expires: 180  
Header: Record-Route: <sip:5210@192.168.56.180:5060;maddr=192.168.56.180>  
Header: Contact: <sip:5218@192.168.10.18:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 221

---

### SDP Headers

---

Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 25678 28140 IN IP4  
192.168.10.18  
Header: s=SIP Call  
Header: c=IN IP4 192.168.10.18  
Header: t=0 0  
Header: m=audio 23994 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

---

### SIP Headers

---

sip-res: SIP/2.0 100 Trying [192.168.56.220:5074->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Content-Length: 0

---

### SIP Headers

---

sip-req: INVITE sip:5210@192.168.56.180:5060 SIP/2.0 [192.168.56.220:5074->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.220:5074;branch=102  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Proxy-Authorization: Basic VovidaClassXSwitch  
Header: Expires: 180  
Header: Record-Route: <sip:5210@192.168.56.220:5074;maddr=192.168.56.220>,<sip:5210@192.168.56.180:5060;maddr=192.168.56.180>  
Header: Contact: <sip:5218@192.168.10.18:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 221  
Header: CC-Redirect: <sip:5210@192.168.56.180:5060>;redir-reason=unconditional;redir-counter=0;redir-limit=99

---

### SDP Headers

---

## SIP IP Phone to SIP IP Phone: Forward to Voice Mail

```
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 25678 28140 IN IP4
192.168.10.18
Header: s=SIP Call
Header: c=IN IP4 192.168.10.18
Header: t=0 0
Header: m=audio 23994 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

---

### SIP Headers

```
-----
sip-res: SIP/2.0 100 Trying [192.168.56.180:5060->192.168.56.220:5074]
Header: Via: SIP/2.0/UDP 192.168.56.220:5074;branch=102
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.10.18:5060
Header: From: <sip:5218@192.168.10.18:5060>
Header: To: <sip:5210@192.168.56.180:5060>
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18
Header: CSeq: 100 INVITE
Header: Content-Length: 0
```

---

### SIP Headers

```
-----
sip-req: INVITE sip:5210@192.168.56.200:5060;user=phone SIP/2.0 [192.168.56.180:5060-
>192.168.56.200:5060]
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=3
Header: Via: SIP/2.0/UDP 192.168.56.220:5074;branch=102
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.10.18:5060
Header: From: <sip:5218@192.168.10.18:5060>
Header: To: <sip:5210@192.168.56.180:5060>
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18
Header: CSeq: 100 INVITE
Header: Expires: 180
Header: Record-Route:
<sip:5210@192.168.56.180:5060;maddr=192.168.56.180>, <sip:5210@192.168.56.220:5074;maddr=1
92.168.56.220>, <sip:5210@192.168.56.180:5060;maddr=192.168.56.180>
Header: Contact: <sip:5218@192.168.10.18:5060>
Header: Content-Type: application/sdp
Header: Content-Length: 221
Header: CC-Redirect: <sip:5210@192.168.56.180:5060>;redir-
reason=unconditional;redir-counter=0;redir-limit=99
```

---

### SDP Headers

```
-----
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 25678 28140 IN IP4
192.168.10.18
Header: s=SIP Call
Header: c=IN IP4 192.168.10.18
Header: t=0 0
Header: m=audio 23994 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
```

---

### SIP Headers

```
-----
sip-res: SIP/2.0 302 Moved Temporarily [192.168.56.200:5060->192.168.56.180:5060]
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=3
Header: Via: SIP/2.0/UDP 192.168.56.220:5074;branch=102
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.10.18:5060
Header: From: <sip:5218@192.168.10.18:5060>
Header: To: <sip:5210@192.168.56.180:5060>
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18
Header: CSeq: 100 INVITE
Header: Contact: <sip:5210@192.168.10.10:5060>
```

## SIP IP Phone to SIP IP Phone: Forward to Voice Mail

Header: Content-Length: 0  
Header: CC-Redirect: <sip:5210@192.168.56.180:5060>;redir-  
reason=unconditional;redir-counter=0;redir-limit=99

-----  
SIP Headers

-----  
sip-req: ACK sip:5210@192.168.56.200:5060;user=phone SIP/2.0 [192.168.56.180:5060->192.168.56.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=3  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:5210@192.168.10.10:5060 SIP/2.0 [192.168.56.180:5060->192.168.10.10:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.56.220:5074;branch=102  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Expires: 180  
Header: Record-Route:  
<sip:5210@192.168.56.180:5060;maddr=192.168.56.180>, <sip:5210@192.168.56.220:5074;maddr=192.168.56.220>, <sip:5210@192.168.56.180:5060;maddr=192.168.56.180>  
Header: Contact: <sip:5218@192.168.10.18:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 221  
Header: CC-Redirect: <sip:5210@192.168.56.180:5060>;redir-  
reason=unconditional;redir-counter=0;redir-limit=99

-----  
SDP Headers

-----  
Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 25678 28140 IN IP4  
192.168.10.18  
Header: s=SIP Call  
Header: c=IN IP4 192.168.10.18  
Header: t=0 0  
Header: m=audio 23994 RTP/AVP 0 101  
Header: a=rtpmap:0 pcmu/8000  
Header: a=rtpmap:101 telephone-event/8000  
Header: a=fmtp:101 0-11

-----  
SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.10.10:50783->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=4,SIP/2.0/UDP  
192.168.56.220:5074;branch=102,SIP/2.0/UDP 192.168.56.180:5060;branch=2,SIP/2.0/UDP  
192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>;tag=c2943000e6fe50-0  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: CSeq: 100 INVITE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.10.10:50783->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=4,SIP/2.0/UDP  
192.168.56.220:5074;branch=102,SIP/2.0/UDP 192.168.56.180:5060;branch=2,SIP/2.0/UDP  
192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>

## SIP IP Phone to SIP IP Phone: Forward to Voice Mail

Header: To: <sip:5210@192.168.56.180:5060>;tag=c2943000e6fe50-0  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: CSeq: 100 INVITE  
Header: Content-Length: 0

### SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.56.180:5060->192.168.56.220:5074]  
Header: Via: SIP/2.0/UDP 192.168.56.220:5074;branch=102  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>;tag=c2943000e6fe50-0  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: Content-Length: 0

### SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.56.220:5074->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>;tag=c2943000e6fe50-0  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: Content-Length: 0

### SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.56.180:5060->192.168.10.18:5060]  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>;tag=c2943000e6fe50-0  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: Content-Length: 0

### SIP Headers

-----  
sip-req: CANCEL sip:5210@192.168.56.180:5060 SIP/2.0 [192.168.56.220:5074->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.220:5074;branch=102  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 CANCEL  
Header: Proxy-Authorization: Basic VovidaClassXSwitch  
Header: Content-Length: 0

### SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.56.180:5060->192.168.56.220:5074]  
Header: Via: SIP/2.0/UDP 192.168.56.220:5074;branch=102  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 CANCEL  
Header: Content-Length: 0

### SIP Headers

-----  
sip-req: CANCEL sip:5210@192.168.10.10:5060 SIP/2.0 [192.168.56.180:5060->192.168.10.10:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=4  
Header: From: <sip:5218@192.168.10.18:5060>

## SIP IP Phone to SIP IP Phone: Forward to Voice Mail

Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 CANCEL  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.10.10:50783->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=4  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>;tag=c2943000e6fe50-0  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: CSeq: 100 CANCEL  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.56.220:5074->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.56.180:5060->192.168.10.18:5060]  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.56.220:5074->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:6500@192.168.56.220:5082>  
Header: Record-Route:  
<sip:192.168.56.220:5060;maddr=192.168.56.220>,<sip:5210@192.168.56.220:5074;maddr=192.168.56.220>,<sip:5210@192.168.56.180:5060;maddr=192.168.56.180>  
Header: Content-Type: application/sdp  
Header: Content-Length: 179

---

### SDP Headers

-----  
Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 25678 28140 IN IP4  
192.168.56.220  
Header: s=SIP Call  
Header: c=IN IP4 192.168.56.220  
Header: t=0 0  
Header: m=audio 24004 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20

---

### SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.56.180:5060->192.168.10.18:5060]  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>

## SIP IP Phone to SIP IP Phone: Forward to Voice Mail

Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 INVITE  
Header: Contact: <sip:6500@192.168.56.220:5082>  
Header: Record-Route:  
<sip:192.168.56.220:5060;maddr=192.168.56.220>,<sip:5210@192.168.56.220:5074;maddr=192.168.56.220>,<sip:5210@192.168.56.180:5060;maddr=192.168.56.180>  
Header: Content-Type: application/sdp  
Header: Content-Length: 179

---

### SDP Headers

Header: v=0  
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 25678 28140 IN IP4 192.168.56.220  
Header: s=SIP Call  
Header: c=IN IP4 192.168.56.220  
Header: t=0 0  
Header: m=audio 24004 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20

---

### SIP Headers

-----  
sip-req: ACK sip:5210@192.168.56.180:5060 SIP/2.0 [192.168.10.18:50443->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: Route:  
<sip:5210@192.168.56.220:5074;maddr=192.168.56.220>,<sip:192.168.56.220:5060;maddr=192.168.56.220>,<sip:6500@192.168.56.220:5082>  
Header: CSeq: 100 ACK  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-req: ACK sip:5210@192.168.56.220:5074;maddr=192.168.56.220 SIP/2.0 [192.168.56.180:5060->192.168.56.220:5074]  
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: CSeq: 100 ACK  
Header: Route:  
<sip:192.168.56.220:5060;maddr=192.168.56.220>,<sip:6500@192.168.56.220:5082>  
Header: Proxy-Authorization: Basic VovidaClassXSwitch  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-req: BYE sip:5210@192.168.56.180:5060 SIP/2.0 [192.168.10.18:50443->192.168.56.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.10.18:5060  
Header: From: <sip:5218@192.168.10.18:5060>  
Header: To: <sip:5210@192.168.56.180:5060>  
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18  
Header: User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled  
Header: CSeq: 101 BYE  
Header: Route:  
<sip:5210@192.168.56.220:5074;maddr=192.168.56.220>,<sip:192.168.56.220:5060;maddr=192.168.56.220>,<sip:6500@192.168.56.220:5082>  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-req: BYE sip:5210@192.168.56.220:5074;maddr=192.168.56.220 SIP/2.0 [192.168.56.180:5060->192.168.56.220:5074]

## SIP IP Phone to SIP IP Phone: Forward to Voice Mail

```
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.10.18:5060
Header: From: <sip:5218@192.168.10.18:5060>
Header: To: <sip:5210@192.168.56.180:5060>
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18
Header: CSeq: 101 BYE
Header: Proxy-Authorization: Basic VovidaClassXSwitch
Header: Route:
<sip:192.168.56.220:5060;maddr=192.168.56.220>,<sip:6500@192.168.56.220:5082>
Header: Content-Length: 0
```

---

### SIP Headers

```
-----
sip-res: SIP/2.0 200 OK [192.168.56.220:5074->192.168.56.180:5060]
Header: Via: SIP/2.0/UDP 192.168.56.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.10.18:5060
Header: From: <sip:5218@192.168.10.18:5060>
Header: To: <sip:5210@192.168.56.180:5060>
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18
Header: CSeq: 101 BYE
Header: Content-Length: 0
```

---

### SIP Headers

```
-----
sip-res: SIP/2.0 200 OK [192.168.56.180:5060->192.168.10.18:5060]
Header: Via: SIP/2.0/UDP 192.168.10.18:5060
Header: From: <sip:5218@192.168.10.18:5060>
Header: To: <sip:5210@192.168.56.180:5060>
Header: Call-ID: c2943000-482e7-61caa-2e323931@192.168.10.18
Header: CSeq: 101 BYE
Header: Content-Length: 0
-----
```

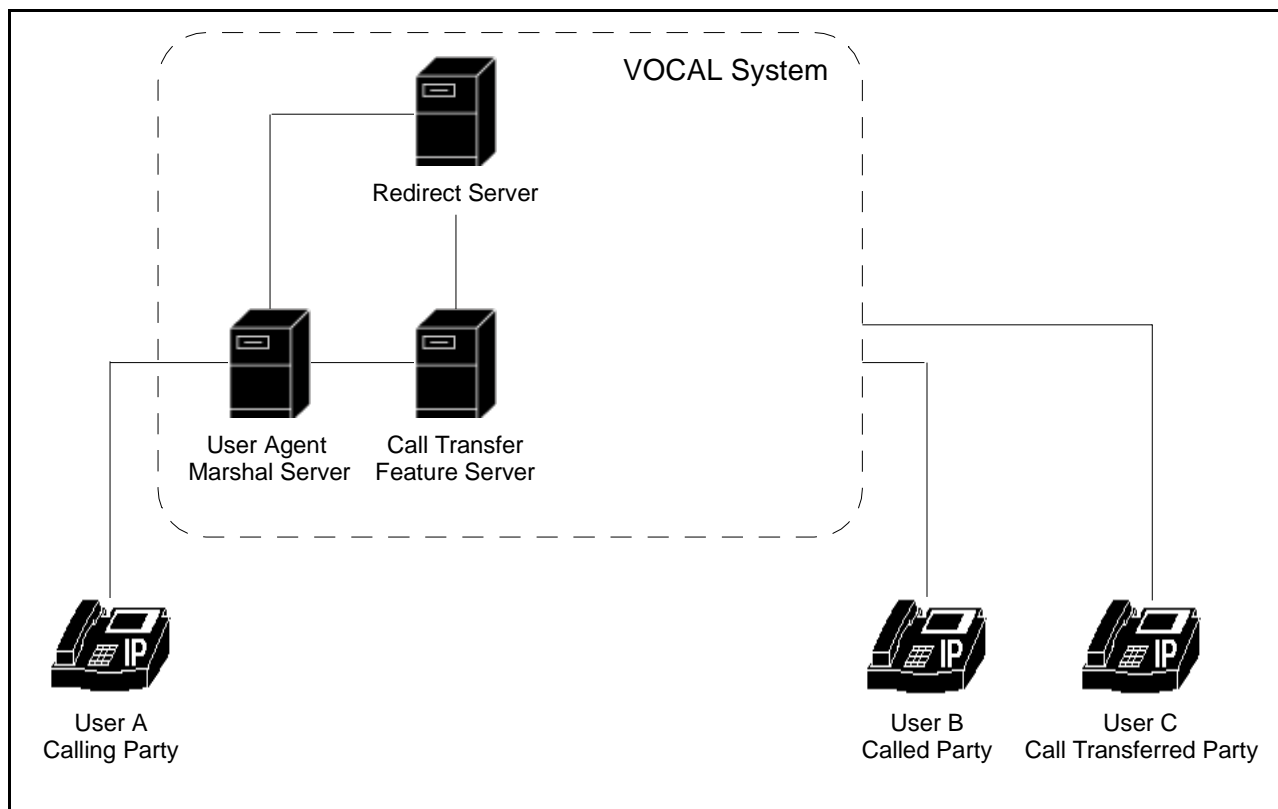
---

## User Agent to User Agent: Consulted Transfer

### Call Scenario

Figure C-26 illustrates the following call scenario:

- User A calls User B.
- User B puts User B on hold and notifies User C about User A's call.
- User B transfers the call to User C.



**Figure C-26. User Agent to User Agent: Consulted Transfer**

#### ■Note

To simplify the call flow diagrams, the call flows to the redirect server are not shown.

### Call Flow Diagram

Figures C-27, C-28 and C-29 show a consulted call transfer between two SIP IP phones.

#### ■Note

This example is taken from VOCAL 1.2.0 and uses the now, deprecated TRANSFER method. VOCAL 1.4.0 supports the REFER method. This call flow will be updated soon.



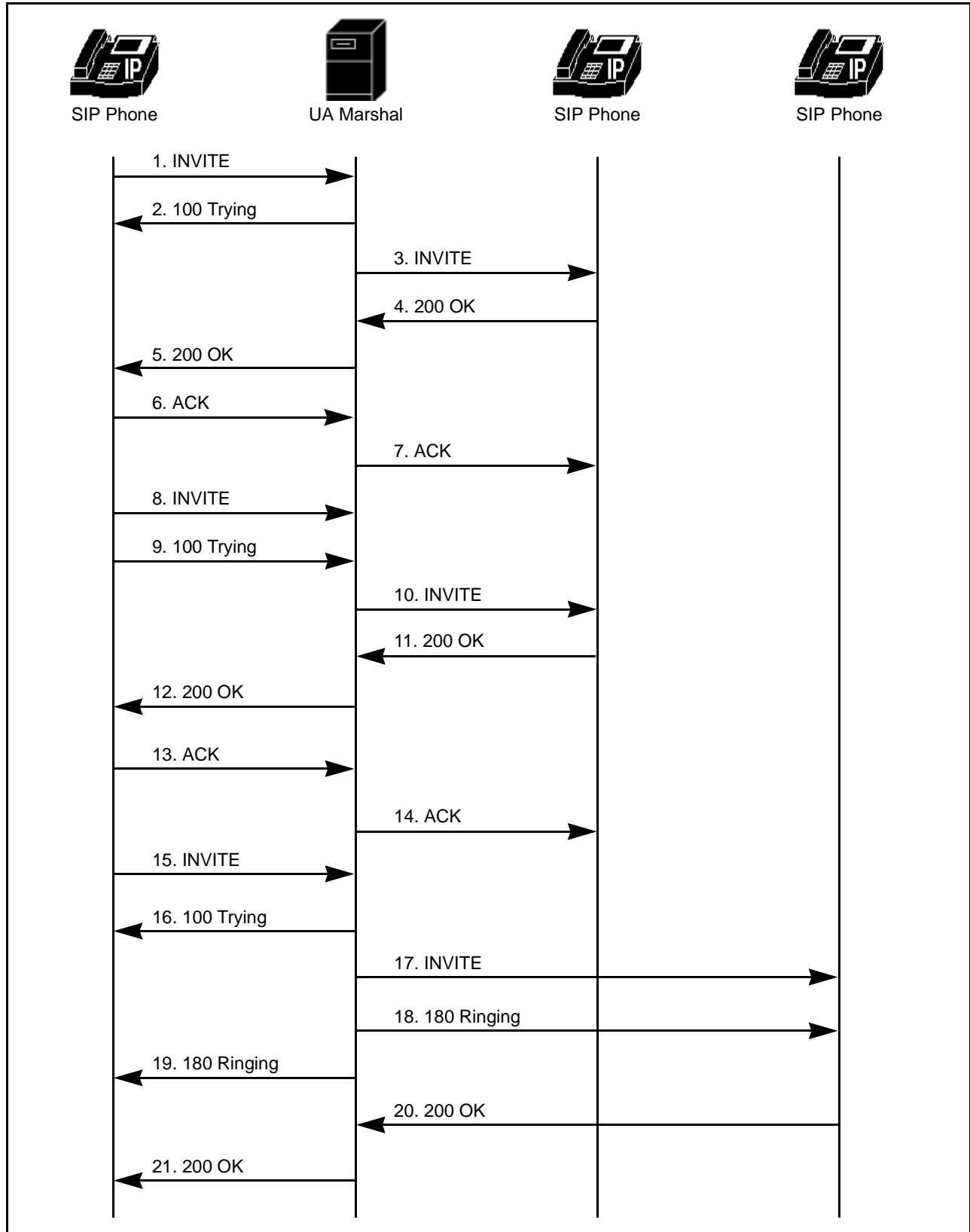


Figure C-27. UA to UA: Consulted Transfer — Diagram 1

User Agent to User Agent: Consulted Transfer

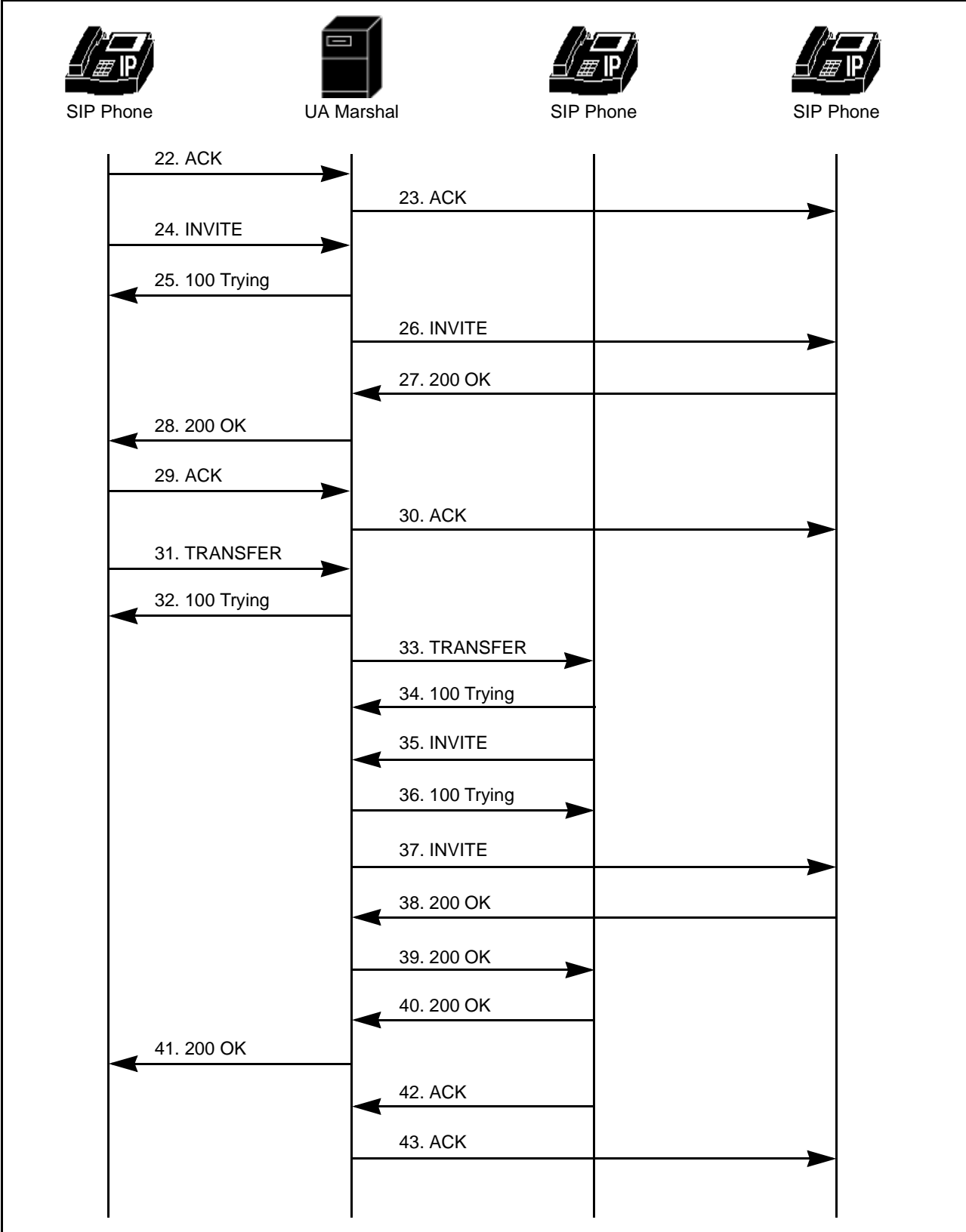


Figure C-28. UA to UA: Consulted Transfer — Diagram 2

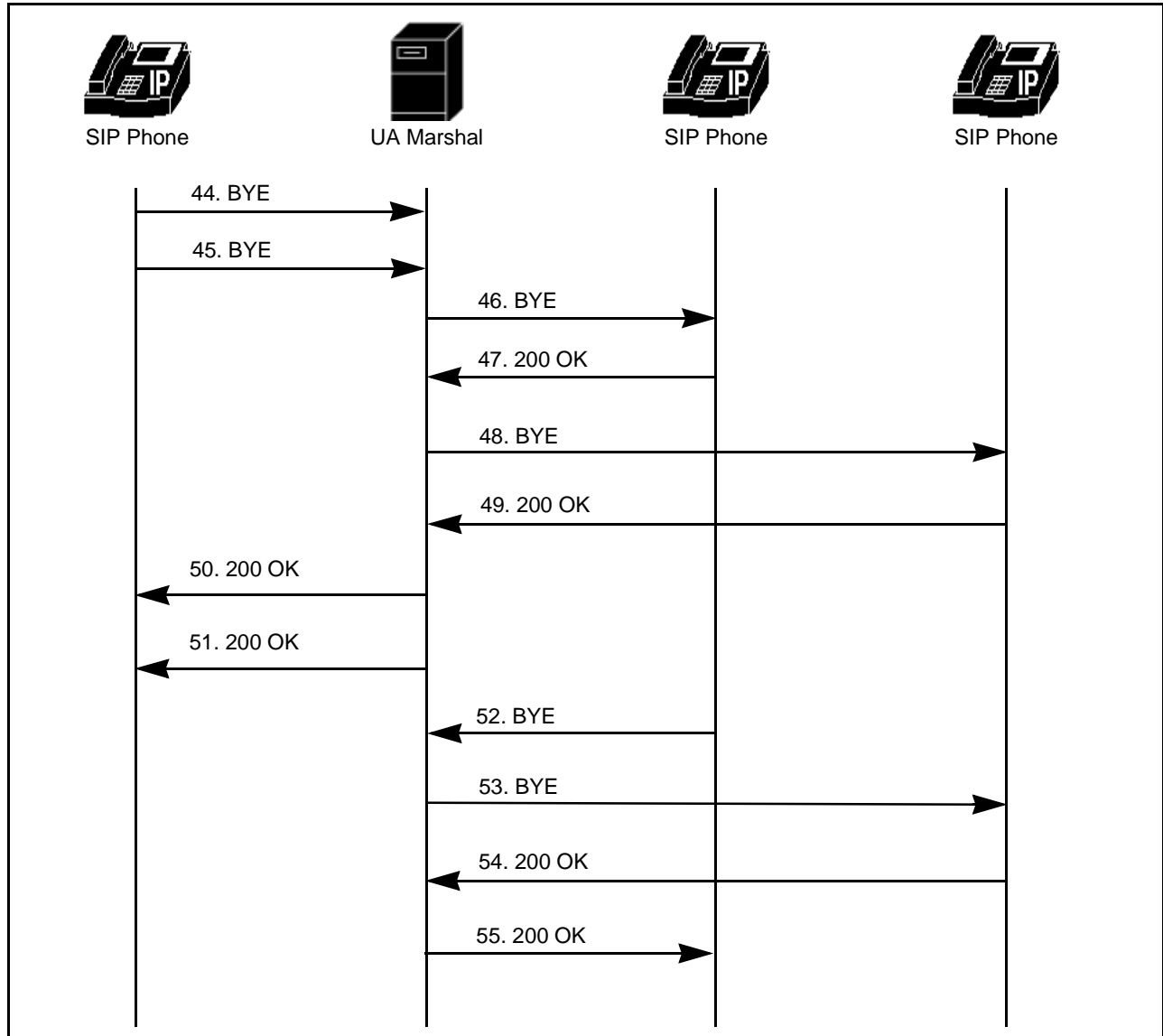


Figure C-29. UA to UA: Consulted Transfer — Diagram 3

**Call Trace**

The following call trace shows a consulted call transfer between two SIP IP phones.

```

-----
SIP Headers
-----
sip-req: INVITE sip:5221@192.168.26.180:5060;user=phone SIP/2.0 [192.168.66.1:5060-
>192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header: Call-ID: 1614879410580032@192.168.66.1
Header: CSeq: 1 INVITE
Header: Subject: VovidaINVITE
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>
Header: Content-Type: application/sdp
Header: Content-Length: 166

```

## User Agent to User Agent: Consulted Transfer

```
-----
                                SDP Headers
-----
Header:          v=0
Header:          o=- 698447251 698447251 IN IP4 192.168.66.1
Header:          s=VOVIDA Session
Header:          c=IN IP4 192.168.66.1
Header:          t=3174939344 0
Header:          m=audio 23456 RTP/AVP 0
Header:          a=rtpmap:0 PCMU/8000
Header:          a=ptime:20
Header:          ;
-----
                                SIP Headers
-----
sip-res:   SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.1:5060]
Header:   Via: SIP/2.0/UDP 192.168.66.1:5060
Header:   From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:   To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:   Call-ID: 1614879410580032@192.168.66.1
Header:   CSeq: 1 INVITE
Header:   Content-Length: 0
-----
                                SIP Headers
-----
sip-req:   INVITE sip:5221@192.168.66.2:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.2:5060]
Header:   Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:   Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:   Via: SIP/2.0/UDP 192.168.66.1:5060
Header:   From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:   To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:   Call-ID: 1614879410580032@192.168.66.1
Header:   CSeq: 1 INVITE
Header:   Subject: VovidaINVITE
Header:   Record-Route:
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>
Header:   Contact: <sip:5220@192.168.66.1:5060;user=phone>
Header:   Content-Type: application/sdp
Header:   Content-Length: 166
-----
                                SDP Headers
-----
Header:          v=0
Header:          o=- 698447251 698447251 IN IP4 192.168.66.1
Header:          s=VOVIDA Session
Header:          c=IN IP4 192.168.66.1
Header:          t=3174939344 0
Header:          m=audio 23456 RTP/AVP 0
Header:          a=rtpmap:0 PCMU/8000
Header:          a=ptime:20
Header:          ;
-----
                                SIP Headers
-----
sip-res:   SIP/2.0 200 OK [192.168.66.2:5060->192.168.26.180:5060]
Header:   Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:   Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:   Via: SIP/2.0/UDP 192.168.66.1:5060
Header:   From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:   To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:   Call-ID: 1614879410580032@192.168.66.1
Header:   CSeq: 1 INVITE
Header:   Contact: <sip:5221@192.168.66.2:5060>
Header:   Record-Route:
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>
Header:   Content-Type: application/sdp
Header:   Content-Length: 166
```

```
-----
                                SDP Headers
-----
Header:      v=0
Header:      o=- 698447251 698447251 IN IP4 192.168.66.2
Header:      s=VOVIDA Session
Header:      c=IN IP4 192.168.66.2
Header:      t=3174939344 0
Header:      m=audio 23466 RTP/AVP 0
Header:      a=rtpmap:0 PCMU/8000
Header:      a=ptime:20
-----
                                SIP Headers
-----
sip-res:     SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]
Header:     Via: SIP/2.0/UDP 192.168.66.1:5060
Header:     From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:     To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:     Call-ID: 1614879410580032@192.168.66.1
Header:     CSeq: 1 INVITE
Header:     Contact: <sip:5221@192.168.66.2:5060>
Header:     Record-Route:
Header:     <sip:5221@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5221@192.168.26.180:5060;maddr=1
92.168.26.180>
Header:     Content-Type: application/sdp
Header:     Content-Length: 166
-----
                                SDP Headers
-----
Header:      v=0
Header:      o=- 698447251 698447251 IN IP4 192.168.66.2
Header:      s=VOVIDA Session
Header:      c=IN IP4 192.168.66.2
Header:      t=3174939344 0
Header:      m=audio 23466 RTP/AVP 0
Header:      a=rtpmap:0 PCMU/8000
Header:      a=ptime:20
Header:      T-
-----
                                SIP Headers
-----
sip-req:     ACK sip:5221@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.1:5060->192.168.26.180:5060]
Header:     Via: SIP/2.0/UDP 192.168.66.1:5060
Header:     From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:     To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:     Call-ID: 1614879410580032@192.168.66.1
Header:     CSeq: 1 ACK
Header:     Route:
Header:     <sip:5221@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5221@192.168.66.2:5060>
Header:     Content-Length: 0
-----
                                SIP Headers
-----
sip-req:     ACK sip:5221@192.168.66.2:5060 SIP/2.0          [192.168.26.180:5060-
>192.168.66.2:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:     Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:     Via: SIP/2.0/UDP 192.168.66.1:5060
Header:     From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:     To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:     Call-ID: 1614879410580032@192.168.66.1
Header:     CSeq: 1 ACK
Header:     Content-Length: 0
-----
                                SIP Headers
-----
sip-req:     INVITE sip:5221@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.1:5060->192.168.26.180:5060]
Header:     Via: SIP/2.0/UDP 192.168.66.1:5060
```

## User Agent to User Agent: Consulted Transfer

Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 INVITE  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Route:  
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.66.2:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 161

---

### SDP Headers

Header: v=0  
Header: o=- 698447251 698447251 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 0.0.0.0  
Header: t=3174939344 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: iK

---

### SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 INVITE  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-req: INVITE sip:5221@192.168.66.2:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 INVITE  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 161

---

### SDP Headers

Header: v=0  
Header: o=- 698447251 698447251 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 0.0.0.0  
Header: t=3174939344 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: jE

---

### SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 INVITE  
Header: Contact: <sip:5221@192.168.66.2:5060;user=phone>

## User Agent to User Agent: Consulted Transfer

Header: Content-Type: application/sdp  
Header: Content-Length: 166

-----  
SDP Headers

Header: v=0  
Header: o=- 698447251 698447251 IN IP4 192.168.66.2  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.2  
Header: t=3174939344 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 INVITE  
Header: Contact: <sip:5221@192.168.66.2:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 166

-----  
SDP Headers

Header: v=0  
Header: o=- 698447251 698447251 IN IP4 192.168.66.2  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.2  
Header: t=3174939344 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: ,

-----  
SIP Headers

-----  
sip-req: ACK sip:5221@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0  
[192.168.66.1:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 ACK  
Header: Route:  
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.66.2:5060>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: ACK sip:5221@192.168.66.2:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:5222@192.168.26.180:5060;user=phone SIP/2.0 [192.168.66.1:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>

## User Agent to User Agent: Consulted Transfer

Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Subject: VovidaINVITE  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

---

### SDP Headers

Header: v=0  
Header: o=- 1113249245 1113249245 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.1  
Header: t=3174939385 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: ^ã

---

### SIP Headers

sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Content-Length: 0

---

### SIP Headers

sip-req: INVITE sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.3:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Subject: VovidaINVITE  
Header: Record-Route: <sip:5222@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

---

### SDP Headers

Header: v=0  
Header: o=- 1113249245 1113249245 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.1  
Header: t=3174939385 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: om

---

### SIP Headers

sip-res: SIP/2.0 180 Ringing [192.168.66.3:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1



## User Agent to User Agent: Consulted Transfer

Header: CSeq: 3 INVITE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.66.3:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Contact: <sip:5222@192.168.66.3:5060>  
Header: Record-Route:  
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

-----  
SDP Headers

-----  
Header: v=0  
Header: o=- 1113249245 1113249245 IN IP4 192.168.66.3  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.3  
Header: t=3174939385 0  
Header: m=audio 23466 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header:

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Contact: <sip:5222@192.168.66.3:5060>  
Header: Record-Route:  
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

-----  
SDP Headers

-----  
Header: v=0  
Header: o=- 1113249245 1113249245 IN IP4 192.168.66.3  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.3  
Header: t=3174939385 0  
Header: m=audio 23466 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: 8U

-----  
SIP Headers

## User Agent to User Agent: Consulted Transfer

```
-----
sip-req:    ACK sip:5222@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.1:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 1614879410580032@192.168.66.1
Header:    CSeq: 3 ACK
Header:    Route:
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5222@192.168.66.3:5060>
Header:    Content-Length: 0
-----
```

### SIP Headers

```
-----
sip-req:    ACK sip:5222@192.168.66.3:5060 SIP/2.0          [192.168.26.180:5060-
>192.168.66.3:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 1614879410580032@192.168.66.1
Header:    CSeq: 3 ACK
Header:    Content-Length: 0
-----
```

### SIP Headers

```
-----
sip-req:    INVITE sip:5222@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.1:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 1614879410580032@192.168.66.1
Header:    CSeq: 4 INVITE
Header:    Route:
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5222@192.168.66.3:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 161
-----
```

### SDP Headers

```
-----
Header:    v=0
Header:    o=- 698447251 698447251 IN IP4 192.168.66.1
Header:    s=VOVIDA Session
Header:    c=IN IP4 0.0.0.0
Header:    t=3174939344 0
Header:    m=audio 23456 RTP/AVP 0
Header:    a=rtpmap:0 PCMU/8000
Header:    a=ptime:20
Header:    †
-----
```

### SIP Headers

```
-----
sip-res:    SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.1:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 1614879410580032@192.168.66.1
Header:    CSeq: 4 INVITE
Header:    Content-Length: 0
-----
```

### SIP Headers

```
-----
sip-req:    INVITE sip:5222@192.168.66.3:5060 SIP/2.0          [192.168.26.180:5060-
>192.168.66.3:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5222<sip:5222@192.168.26.180:5060;user=phone>
-----
```

## User Agent to User Agent: Consulted Transfer

Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 4 INVITE  
Header: Content-Type: application/sdp  
Header: Content-Length: 161

-----  
SDP Headers

Header: v=0  
Header: o=- 698447251 698447251 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 0.0.0.0  
Header: t=3174939344 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: n@

-----  
SIP Headers

sip-res: SIP/2.0 200 OK [192.168.66.3:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 4 INVITE  
Header: Contact: <sip:5222@192.168.66.3:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 166

-----  
SDP Headers

Header: v=0  
Header: o=- 698447251 698447251 IN IP4 192.168.66.3  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.3  
Header: t=3174939344 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: F

-----  
SIP Headers

sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 4 INVITE  
Header: Contact: <sip:5222@192.168.66.3:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 166

-----  
SDP Headers

Header: v=0  
Header: o=- 698447251 698447251 IN IP4 192.168.66.3  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.3  
Header: t=3174939344 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: ,

-----  
SIP Headers

## User Agent to User Agent: Consulted Transfer

```
sip-req: ACK sip:5222@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.1:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header: Call-ID: 1614879410580032@192.168.66.1
Header: CSeq: 4 ACK
Header: Route:
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5222@192.168.66.3:5060>
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: ACK sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.66.3:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header: Call-ID: 1614879410580032@192.168.66.1
Header: CSeq: 4 ACK
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: TRANSFER sip:5221@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.1:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header: Transfer-To: <sip:5222@192.168.26.180:5060;user=phone>
Header: Call-ID: 1614879410580032@192.168.66.1
Header: CSeq: 2 TRANSFER
Header: Require: cc Transfer
Header: Route:
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.66.2:5060>
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.1:5060]
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header: Call-ID: 1614879410580032@192.168.66.1
Header: CSeq: 2 TRANSFER
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: TRANSFER sip:5221@192.168.66.2:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.66.2:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header: Transfer-To: <sip:5222@192.168.26.180:5060;user=phone>
Header: Call-ID: 1614879410580032@192.168.66.1
Header: CSeq: 2 TRANSFER
Header: Require: cc Transfer
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res: SIP/2.0 100 Trying [192.168.66.2:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
```

## User Agent to User Agent: Consulted Transfer

Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 TRANSFER  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-req: INVITE sip:5222@192.168.26.180:5060;user=phone SIP/2.0 [192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Subject: VovidaINVITE  
Header: Contact: <sip:5221@192.168.66.2:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

---

### SDP Headers

-----  
Header: v=0  
Header: o=- 1277546069 1277546069 IN IP4 192.168.66.2  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.2  
Header: t=3174939395 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20

---

### SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-req: INVITE sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.3:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Subject: VovidaINVITE  
Header: Record-Route: <sip:5222@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:5221@192.168.66.2:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

---

### SDP Headers

-----  
Header: v=0  
Header: o=- 1277546069 1277546069 IN IP4 192.168.66.2  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.2  
Header: t=3174939395 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20

## User Agent to User Agent: Consulted Transfer

```
Header:                )u
-----
                        SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.66.3:5060->192.168.26.180:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:     Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:     Via: SIP/2.0/UDP 192.168.66.2:5060
Header:     From: UserAgent<sip:5221@192.168.66.2:5060>
Header:     To: <sip:5222@192.168.26.180:5060;user=phone>
Header:     Call-ID: 1614879410580032@192.168.66.1
Header:     CSeq: 1 INVITE
Header:     Contact: <sip:5222@192.168.66.3:5060;user=phone>
Header:     Record-Route:
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5222@192.168.26.180:5060;maddr=1
92.168.26.180>
Header:     Content-Type: application/sdp
Header:     Content-Type: application/sdp
Header:     Content-Length: 168
Header:     Content-Length: 168
-----
                        SDP Headers
-----
Header:     v=0
Header:     o=- 1277546069 1277546069 IN IP4 192.168.66.3
Header:     s=VOVIDA Session
Header:     c=IN IP4 192.168.66.3
Header:     t=3174939395 0
Header:     m=audio 23466 RTP/AVP 0
Header:     a=rtpmap:0 PCMU/8000
Header:     a=ptime:20
-----
                        SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.2:5060]
Header:     Via: SIP/2.0/UDP 192.168.66.2:5060
Header:     From: UserAgent<sip:5221@192.168.66.2:5060>
Header:     To: <sip:5222@192.168.26.180:5060;user=phone>
Header:     Call-ID: 1614879410580032@192.168.66.1
Header:     CSeq: 1 INVITE
Header:     Contact: <sip:5222@192.168.66.3:5060;user=phone>
Header:     Record-Route:
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5222@192.168.26.180:5060;maddr=1
92.168.26.180>
Header:     Content-Type: application/sdp
Header:     Content-Length: 168
-----
                        SDP Headers
-----
Header:     v=0
Header:     o=- 1277546069 1277546069 IN IP4 192.168.66.3
Header:     s=VOVIDA Session
Header:     c=IN IP4 192.168.66.3
Header:     t=3174939395 0
Header:     m=audio 23466 RTP/AVP 0
Header:     a=rtpmap:0 PCMU/8000
Header:     a=ptime:20
Header:     bc
-----
                        SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.66.2:5060->192.168.26.180:5060]
Header:     Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:     Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:     Via: SIP/2.0/UDP 192.168.66.1:5060
Header:     From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:     To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:     Call-ID: 1614879410580032@192.168.66.1
Header:     CSeq: 2 TRANSFER
Header:     Content-Length: 0
```

```
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:    Call-ID: 1614879410580032@192.168.66.1
Header:    CSeq: 2 TRANSFER
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    ACK sip:5222@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.2:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.2:5060
Header:    From: UserAgent<sip:5221@192.168.66.2:5060>
Header:    To: <sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 1614879410580032@192.168.66.1
Header:    CSeq: 1 ACK
Header:    Route:
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5222@192.168.66.3:5060>
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    ACK sip:5222@192.168.66.3:5060 SIP/2.0          [192.168.26.180:5060-
>192.168.66.3:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.66.2:5060
Header:    From: UserAgent<sip:5221@192.168.66.2:5060>
Header:    To: <sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 1614879410580032@192.168.66.1
Header:    CSeq: 1 ACK
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    BYE sip:5221@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.1:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:    Call-ID: 1614879410580032@192.168.66.1
Header:    CSeq: 4 BYE
Header:    Route:
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.66.2:5060>
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    BYE sip:5222@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.1:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 1614879410580032@192.168.66.1
Header:    CSeq: 5 BYE
Header:    Route:
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5222@192.168.66.3:5060>
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    BYE sip:5221@192.168.66.2:5060 SIP/2.0          [192.168.26.180:5060-
>192.168.66.2:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
```



## User Agent to User Agent: Consulted Transfer

Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 4 BYE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 4 BYE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: BYE sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.3:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 5 BYE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.66.3:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 5 BYE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 4 BYE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 5 BYE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: BYE sip:5222@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0  
[192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>



## User Agent to User Agent: Consulted Transfer

Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 BYE  
Header: Route:  
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5222@192.168.66.3:5060>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: BYE sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.3:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 BYE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.66.3:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 BYE  
Header: Content-Length: 0  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 1614879410580032@192.168.66.1  
Header: CSeq: 2 BYE  
Header: Content-Length: 0

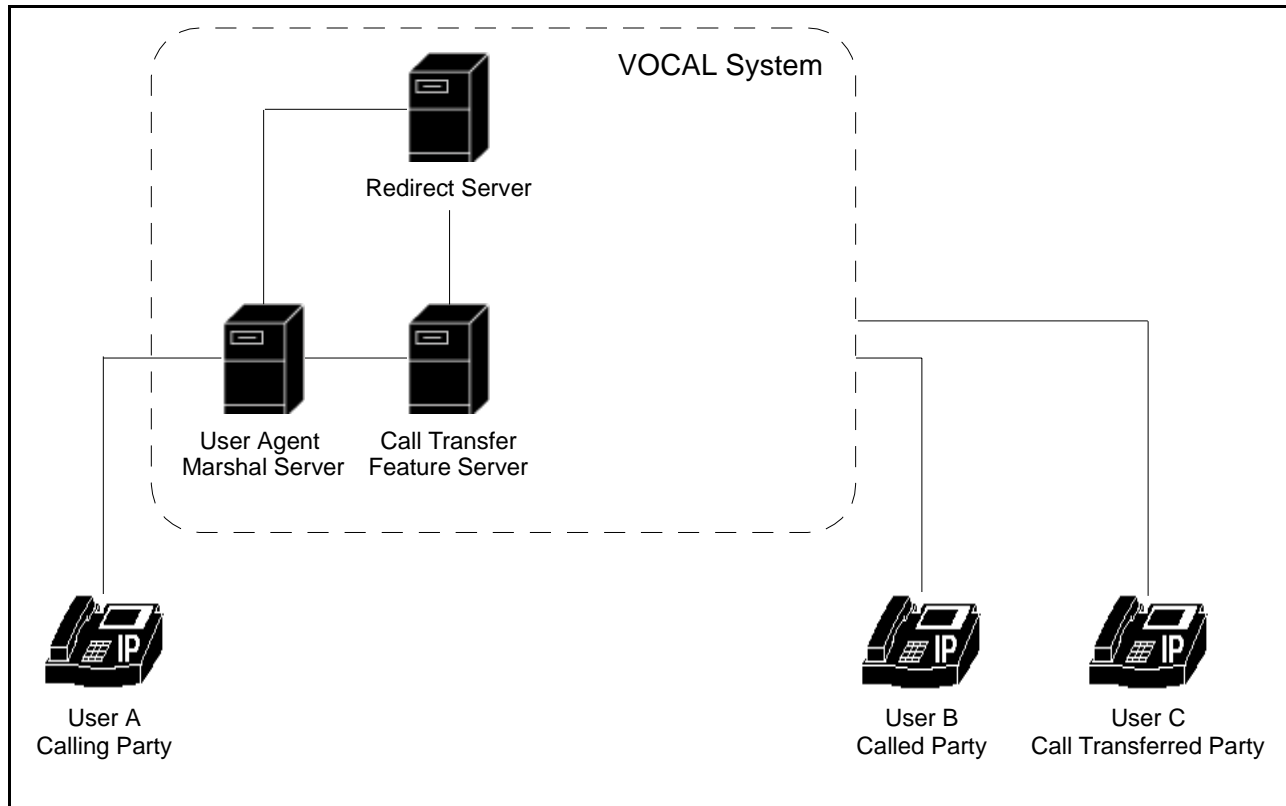
---

## User Agent to User Agent: Blind Transfer

### Call Scenario

Figure C-30 illustrates a call scenario in which:

- User A calls User B.
- User B transfers the call to User C without consulting User C.



**Figure C-30. User Agent to User Agent: Blind Transfer**

### Call Flow Diagram

Figures C-31, C-32 and C-33 show an unconsulted call transfer, also known as a "blind" transfer.

#### ■Note

This example is taken from VOCAL 1.2.0 and uses the now, deprecated TRANSFER method. VOCAL 1.4.0 supports the REFER method. This call flow will be updated soon.

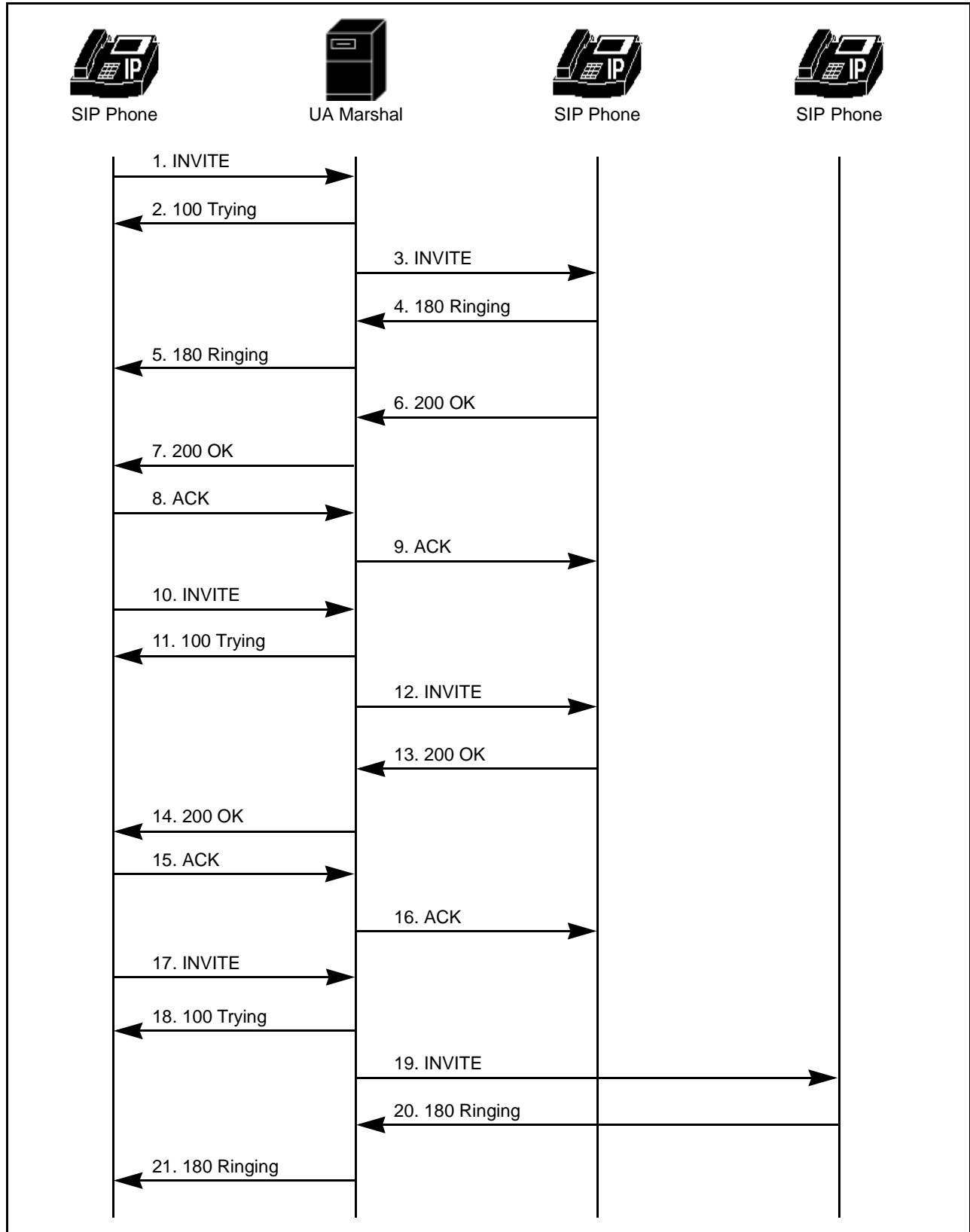


Figure C-31. UA to UA: Blind Transfer — Diagram 1

User Agent to User Agent: Blind Transfer

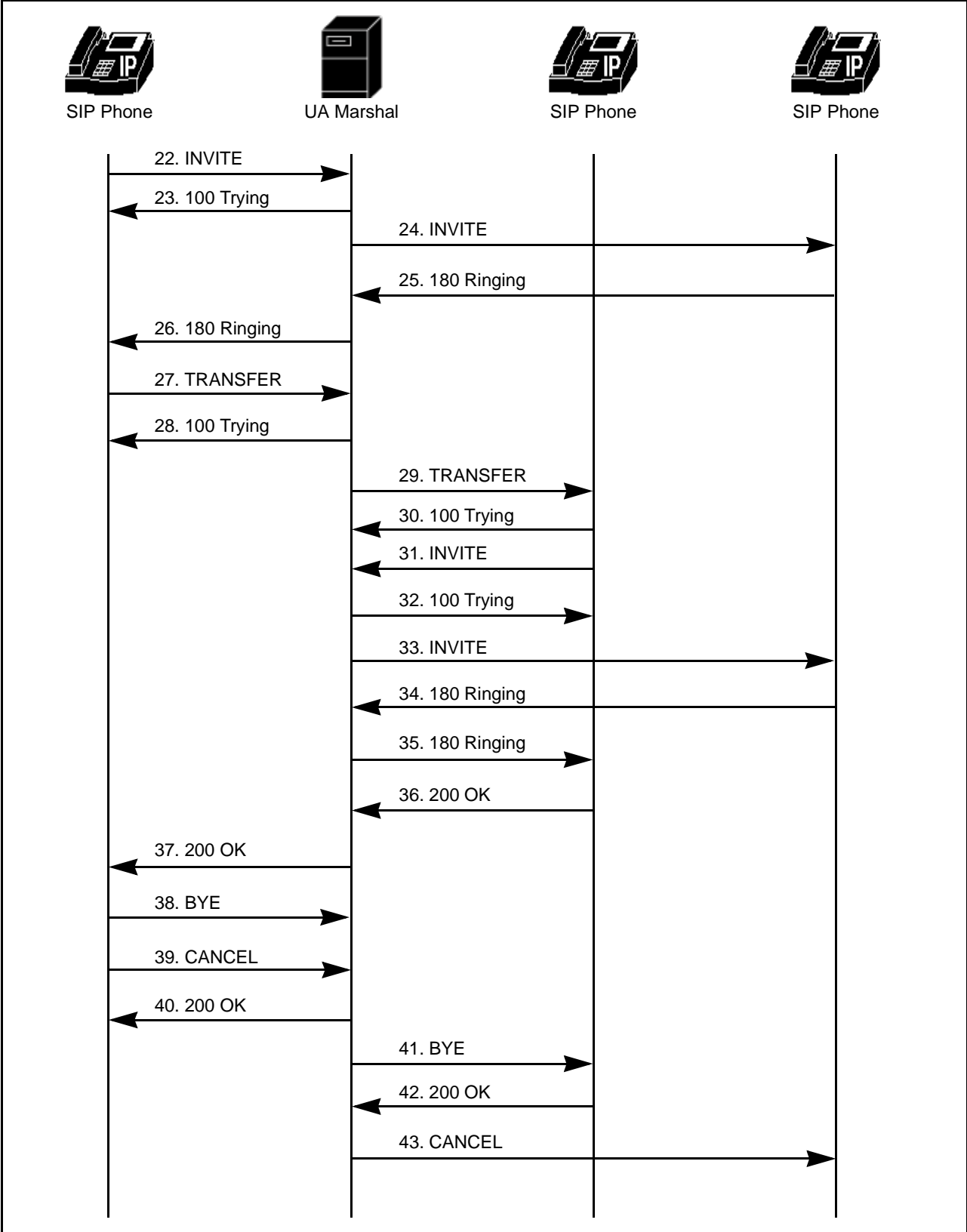


Figure C-32. UA to UA: Blind Transfer — Diagram 2

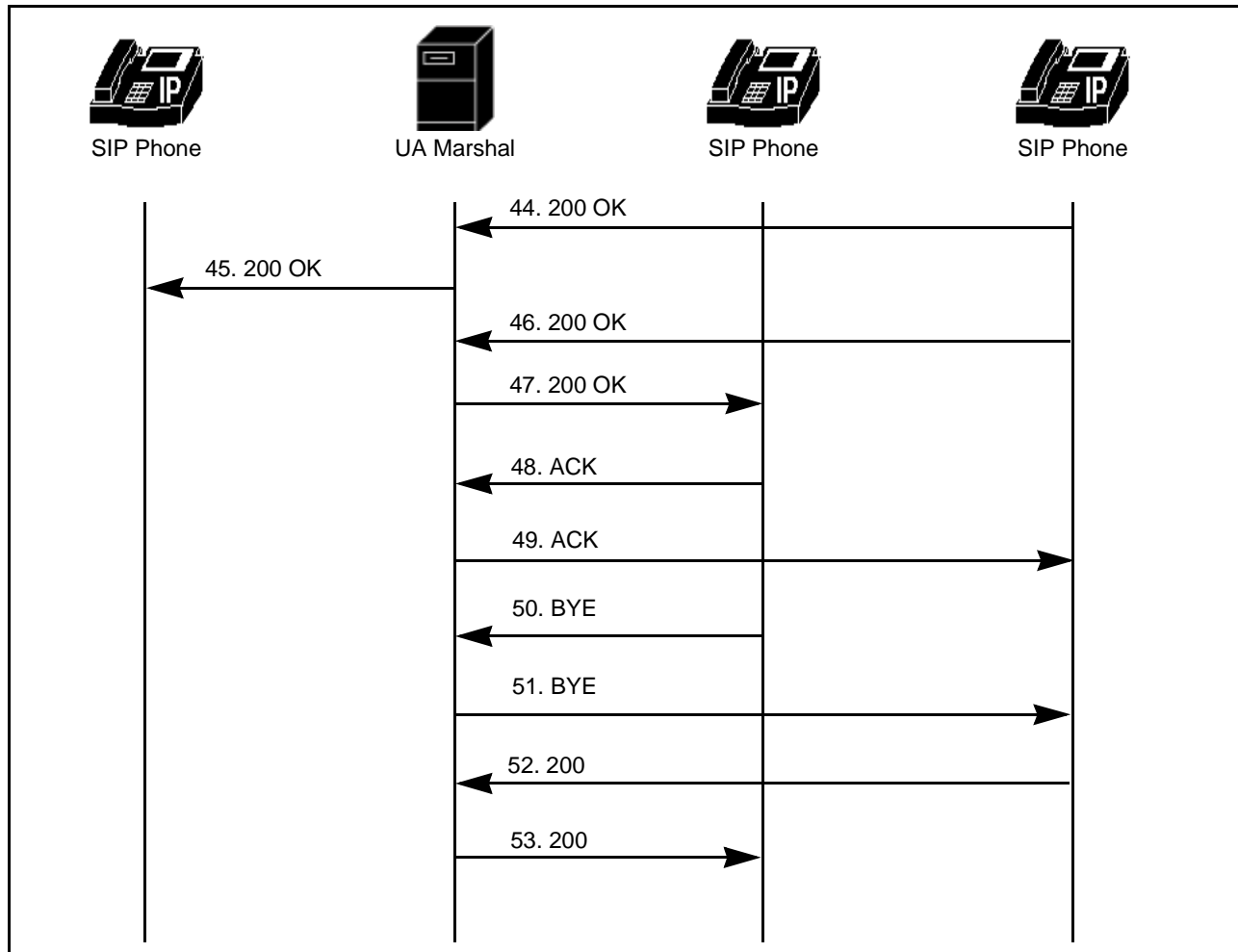


Figure C-33. UA to UA: Blind Transfer — Diagram 3

### Call Trace

The following call trace shows an unconsulted call transfer.

```

-----
SIP Headers
-----
sip-req:    INVITE sip:5221@192.168.26.180:5060;user=phone SIP/2.0 [192.168.66.1:5060-
>192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:    Call-ID: 507936623238514@192.168.66.1
Header:    CSeq: 1 INVITE
Header:    Subject: VovidaINVITE
Header:    Contact: <sip:5220@192.168.66.1:5060;user=phone>
Header:    Content-Type: application/sdp
Header:    Content-Length: 168
-----
SDP Headers
-----
Header:    v=0
Header:    o=- 1914312474 1914312474 IN IP4 192.168.66.1
Header:    s=VOVIDA Session
Header:    c=IN IP4 192.168.66.1
Header:    t=3174939460 0

```

## User Agent to User Agent: Blind Transfer

Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: :

---

### SIP Headers

---

sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Content-Length: 0

---

### SIP Headers

---

sip-req: INVITE sip:5221@192.168.66.2:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Subject: VovidaINVITE  
Header: Record-Route:  
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

---

### SDP Headers

---

Header: v=0  
Header: o=- 1914312474 1914312474 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.1  
Header: t=3174939460 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: 0

---

### SIP Headers

---

sip-res: SIP/2.0 180 Ringing [192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Content-Length: 0

---

### SIP Headers

---

sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Content-Length: 0

---

### SIP Headers

---

## User Agent to User Agent: Blind Transfer

```
sip-res: SIP/2.0 200 OK [192.168.66.2:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header: Call-ID: 507936623238514@192.168.66.1
Header: CSeq: 1 INVITE
Header: Contact: <sip:5221@192.168.66.2:5060>
Header: Record-Route:
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.26.180:5060;maddr=1
92.168.26.180>
Header: Content-Type: application/sdp
Header: Content-Length: 168
```

### SDP Headers

```
Header: v=0
Header: o=- 1914312474 1914312474 IN IP4 192.168.66.2
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.66.2
Header: t=3174939460 0
Header: m=audio 23466 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
```

### SIP Headers

```
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header: Call-ID: 507936623238514@192.168.66.1
Header: CSeq: 1 INVITE
Header: Contact: <sip:5221@192.168.66.2:5060>
Header: Record-Route:
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.26.180:5060;maddr=1
92.168.26.180>
Header: Content-Type: application/sdp
Header: Content-Length: 168
```

### SDP Headers

```
Header: v=0
Header: o=- 1914312474 1914312474 IN IP4 192.168.66.2
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.66.2
Header: t=3174939460 0
Header: m=audio 23466 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
Header: }!
```

### SIP Headers

```
sip-req: ACK sip:5221@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.1:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header: Call-ID: 507936623238514@192.168.66.1
Header: CSeq: 1 ACK
Header: Route:
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.66.2:5060>
Header: Content-Length: 0
```

### SIP Headers

```
sip-req: ACK sip:5221@192.168.66.2:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.66.2:5060]
```

## User Agent to User Agent: Blind Transfer

Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:5221@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0  
[192.168.66.1:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 INVITE  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Route:  
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5221@192.168.66.2:5060>  
Header: Content-Type: application/sdp  
Header: Content-Length: 163

-----  
SDP Headers

-----  
Header: v=0  
Header: o=- 1914312474 1914312474 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 0.0.0.0  
Header: t=3174939460 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: AA

-----  
SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 INVITE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:5221@192.168.66.2:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 INVITE  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 163

-----  
SDP Headers

-----  
Header: v=0  
Header: o=- 1914312474 1914312474 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 0.0.0.0  
Header: t=3174939460 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000



Header: aptime:20  
Header: jô

-----  
SIP Headers  
-----

sip-res: SIP/2.0 200 OK [192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 INVITE  
Header: Contact: <sip:5221@192.168.66.2:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

-----  
SDP Headers  
-----

Header: v=0  
Header: o=- 1914312474 1914312474 IN IP4 192.168.66.2  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.2  
Header: t=3174939460 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: aptime:20

-----  
SIP Headers  
-----

sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 INVITE  
Header: Contact: <sip:5221@192.168.66.2:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

-----  
SDP Headers  
-----

Header: v=0  
Header: o=- 1914312474 1914312474 IN IP4 192.168.66.2  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.2  
Header: t=3174939460 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: aptime:20  
Header: ,

-----  
SIP Headers  
-----

sip-req: ACK sip:5221@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0  
[192.168.66.1:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 ACK  
Header: Route:  
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.66.2:5060>  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: ACK sip:5221@192.168.66.2:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4

## User Agent to User Agent: Blind Transfer

Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:5222@192.168.26.180:5060;user=phone SIP/2.0 [192.168.66.1:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Subject: VovidaINVITE  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

-----  
SDP Headers

-----  
Header: v=0  
Header: o=- 1669800512 1669800512 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.1  
Header: t=3174939482 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: G

-----  
SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.3:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Subject: VovidaINVITE  
Header: Record-Route: <sip:5222@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5222@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

-----  
SDP Headers

-----  
Header: v=0  
Header: o=- 1669800512 1669800512 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.1  
Header: t=3174939482 0

Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header:

-----  
SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.66.3:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Content-Length: 0  
-----

-----  
SIP Headers

-----  
sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 3 INVITE  
Header: Content-Length: 0  
-----

-----  
SIP Headers

-----  
sip-req: INVITE sip:5222@192.168.26.180:5060;user=phone SIP/2.0 [192.168.66.1:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 4 INVITE  
Header: Subject: VovidaINVITE  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 163  
-----

-----  
SDP Headers

-----  
Header: v=0  
Header: o=- 1669800512 1669800512 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 0.0.0.0  
Header: t=3174939482 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: AÌ  
-----

-----  
SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 4 INVITE  
Header: Content-Length: 0  
-----

-----  
SIP Headers

-----  
sip-req: INVITE sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.3:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
-----

## User Agent to User Agent: Blind Transfer

Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 4 INVITE  
Header: Subject: VovidaINVITE  
Header: Record-Route:  
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5222@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:5220@192.168.66.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 163

---

### SDP Headers

Header: v=0  
Header: o=- 1669800512 1669800512 IN IP4 192.168.66.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 0.0.0.0  
Header: t=3174939482 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: 52

---

### SIP Headers

sip-res: SIP/2.0 180 Ringing [192.168.66.3:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 4 INVITE  
Header: Content-Length: 0

---

### SIP Headers

sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 4 INVITE  
Header: Content-Length: 0

---

### SIP Headers

sip-req: TRANSFER sip:5221@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0  
[192.168.66.1:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Transfer-To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 TRANSFER  
Header: Require: cc Transfer  
Header: Route:  
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5221@192.168.66.2:5060>  
Header: Content-Length: 0

---

### SIP Headers

sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 TRANSFER  
Header: Content-Length: 0

-----  
SIP Headers  
-----  
sip-req: TRANSFER sip:5221@192.168.66.2:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Transfer-To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 TRANSFER  
Header: Require: cc Transfer  
Header: Content-Length: 0  
-----

-----  
SIP Headers  
-----  
sip-res: SIP/2.0 100 Trying [192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 TRANSFER  
Header: Content-Length: 0  
-----

-----  
SIP Headers  
-----  
sip-req: INVITE sip:5222@192.168.26.180:5060;user=phone SIP/2.0 [192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Subject: VovidaINVITE  
Header: Contact: <sip:5221@192.168.66.2:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168  
-----

-----  
SDP Headers  
-----  
Header: v=0  
Header: o=- 1986829226 1986829226 IN IP4 192.168.66.2  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.2  
Header: t=3174939488 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
-----

-----  
SIP Headers  
-----  
sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Content-Length: 0  
-----

-----  
SIP Headers  
-----  
sip-req: INVITE sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.3:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
-----

## User Agent to User Agent: Blind Transfer

Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Subject: VovidaINVITE  
Header: Record-Route:  
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5222@192.168.26.180:5060;maddr=192.168.26.180>  
Header: Contact: <sip:5221@192.168.66.2:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 168

---

### SDP Headers

Header: v=0  
Header: o=- 1986829226 1986829226 IN IP4 192.168.66.2  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.66.2  
Header: t=3174939488 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: gu

---

### SIP Headers

sip-res: SIP/2.0 180 Ringing [192.168.66.3:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Content-Length: 0

---

### SIP Headers

sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.66.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 INVITE  
Header: Content-Length: 0

---

### SIP Headers

sip-res: SIP/2.0 200 OK [192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 TRANSFER  
Header: Content-Length: 0

---

### SIP Headers

sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.1:5060  
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>  
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 TRANSFER  
Header: Content-Length: 0

---

### SIP Headers

---

## User Agent to User Agent: Blind Transfer

```
sip-req:    BYE sip:5221@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.66.1:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:    Call-ID: 507936623238514@192.168.66.1
Header:    CSeq: 4 BYE
Header:    Route:
<sip:5221@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5221@192.168.66.2:5060>
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req:    CANCEL sip:5222@192.168.26.180:5060;user=phone SIP/2.0 [192.168.66.1:5060-
>192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 507936623238514@192.168.66.1
Header:    CSeq: 5 CANCEL
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res:    SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 507936623238514@192.168.66.1
Header:    CSeq: 5 CANCEL
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req:    BYE sip:5221@192.168.66.2:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.66.2:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:    Call-ID: 507936623238514@192.168.66.1
Header:    CSeq: 4 BYE
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res:    SIP/2.0 200 OK [192.168.66.2:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.66.1:5060
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header:    Call-ID: 507936623238514@192.168.66.1
Header:    CSeq: 4 BYE
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req:    CANCEL sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.66.3:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:    From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header:    To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 507936623238514@192.168.66.1
Header:    CSeq: 5 CANCEL
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----



## User Agent to User Agent: Blind Transfer

```
sip-res: SIP/2.0 200 OK [192.168.66.3:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5222<sip:5222@192.168.26.180:5060;user=phone>
Header: Call-ID: 507936623238514@192.168.66.1
Header: CSeq: 5 CANCEL
Header: Content-Length: 0
-----
SIP Headers
-----
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.1:5060]
Header: Via: SIP/2.0/UDP 192.168.66.1:5060
Header: From: UserAgent<sip:5220@192.168.66.1:5060;user=phone>
Header: To: 5221<sip:5221@192.168.26.180:5060;user=phone>
Header: Call-ID: 507936623238514@192.168.66.1
Header: CSeq: 4 BYE
Header: Content-Length: 0
-----
SIP Headers
-----
sip-res: SIP/2.0 200 OK [192.168.66.3:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.66.2:5060
Header: From: UserAgent<sip:5221@192.168.66.2:5060>
Header: To: <sip:5222@192.168.26.180:5060;user=phone>
Header: Call-ID: 507936623238514@192.168.66.1
Header: CSeq: 1 INVITE
Header: Contact: <sip:5222@192.168.66.3:5060>
Header: Record-Route:
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5222@192.168.26.180:5060;maddr=1
92.168.26.180>
Header: Content-Type: application/sdp
Header: Content-Length: 168
-----
SDP Headers
-----
Header: v=0
Header: o=- 1986829226 1986829226 IN IP4 192.168.66.3
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.66.3
Header: t=3174939488 0
Header: m=audio 23466 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
Header: ÷I
-----
SIP Headers
-----
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.2:5060]
Header: Via: SIP/2.0/UDP 192.168.66.2:5060
Header: From: UserAgent<sip:5221@192.168.66.2:5060>
Header: To: <sip:5222@192.168.26.180:5060;user=phone>
Header: Call-ID: 507936623238514@192.168.66.1
Header: CSeq: 1 INVITE
Header: Contact: <sip:5222@192.168.66.3:5060>
Header: Record-Route:
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>, <sip:5222@192.168.26.180:5060;maddr=1
92.168.26.180>
Header: Content-Type: application/sdp
Header: Content-Length: 168
-----
SDP Headers
-----
Header: v=0
Header: o=- 1986829226 1986829226 IN IP4 192.168.66.3
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.66.3
Header: t=3174939488 0
Header: m=audio 23466 RTP/AVP 0
```



Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
Header: xb

-----  
SIP Headers  
-----

sip-req: ACK sip:5222@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0  
[192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 ACK  
Header: Route:  
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5222@192.168.66.3:5060>  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: ACK sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.3:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 1 ACK  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: BYE sip:5222@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0  
[192.168.66.2:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 BYE  
Header: Route:  
<sip:5222@192.168.26.180:5060;maddr=192.168.26.180>,<sip:5222@192.168.66.3:5060>  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-req: BYE sip:5222@192.168.66.3:5060 SIP/2.0 [192.168.26.180:5060->192.168.66.3:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 BYE  
Header: Content-Length: 0

-----  
SIP Headers  
-----

sip-res: SIP/2.0 200 OK [192.168.66.3:5060->192.168.26.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.66.2:5060  
Header: From: UserAgent<sip:5221@192.168.66.2:5060>  
Header: To: <sip:5222@192.168.26.180:5060;user=phone>  
Header: Call-ID: 507936623238514@192.168.66.1  
Header: CSeq: 2 BYE  
Header: Content-Length: 0

-----  
SIP Headers  
-----

## *User Agent to User Agent: Blind Transfer*

```
sip-res:    SIP/2.0 200 OK [192.168.26.180:5060->192.168.66.2:5060]
Header:    Via: SIP/2.0/UDP 192.168.66.2:5060
Header:    From: UserAgent<sip:5221@192.168.66.2:5060>
Header:    To: <sip:5222@192.168.26.180:5060;user=phone>
Header:    Call-ID: 507936623238514@192.168.66.1
Header:    CSeq: 2 BYE
Header:    Content-Length: 0
-----
```

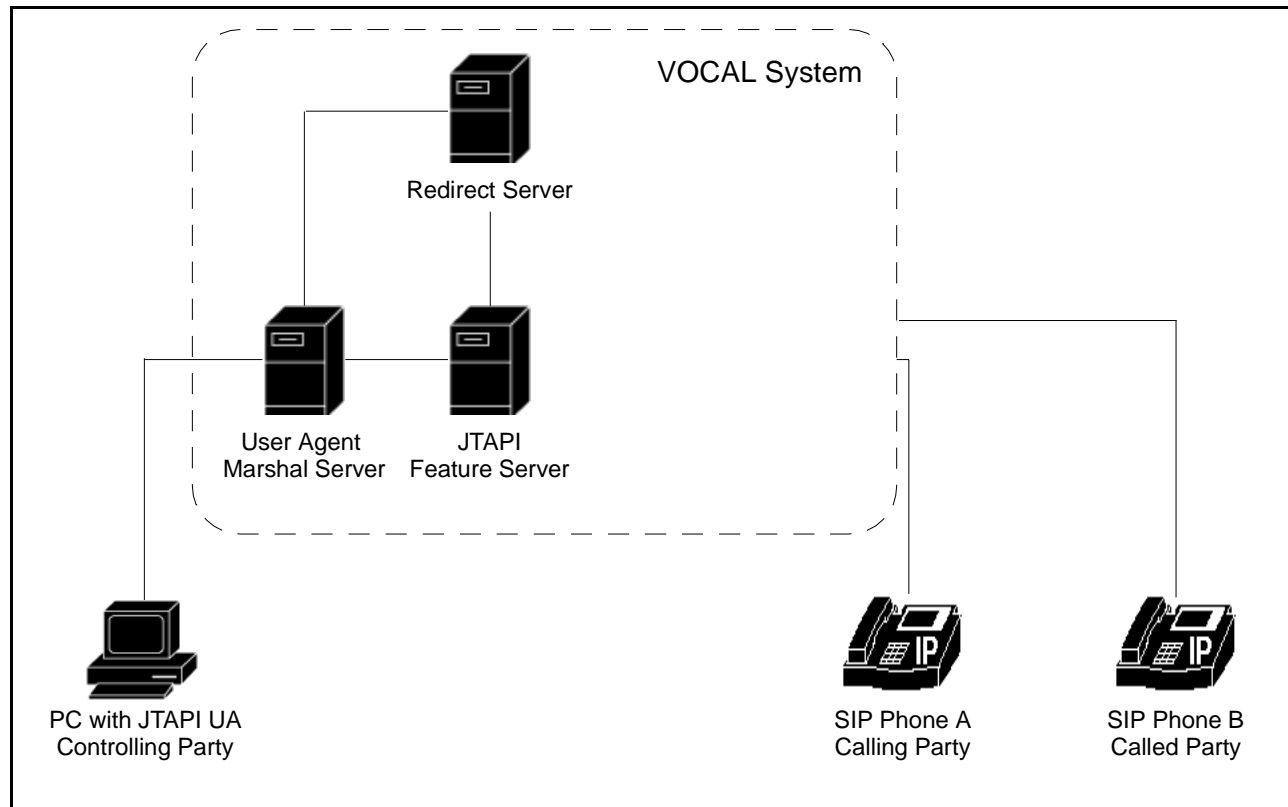
---

# JTAPI

## Call Scenario

Figure C-34 illustrates the following call scenario:

- A user uses a JTAPI User Agent on a PC to remotely instruct SIP Phone A to call SIP Phone B.



**Figure C-34. JTAPI**

## Call Flow Diagram

Figures C-35, C-36 and C-37 show third party call control through a JTAPI server.

### ■ Note

The JTAPI server has been deprecated in VOCAL version 1.4.0 in favor of the Back-to-Back User Agent, which is available separately on [Vovida.org](http://Vovida.org). This call flow shows how the JTAPI server worked with older versions of VOCAL.

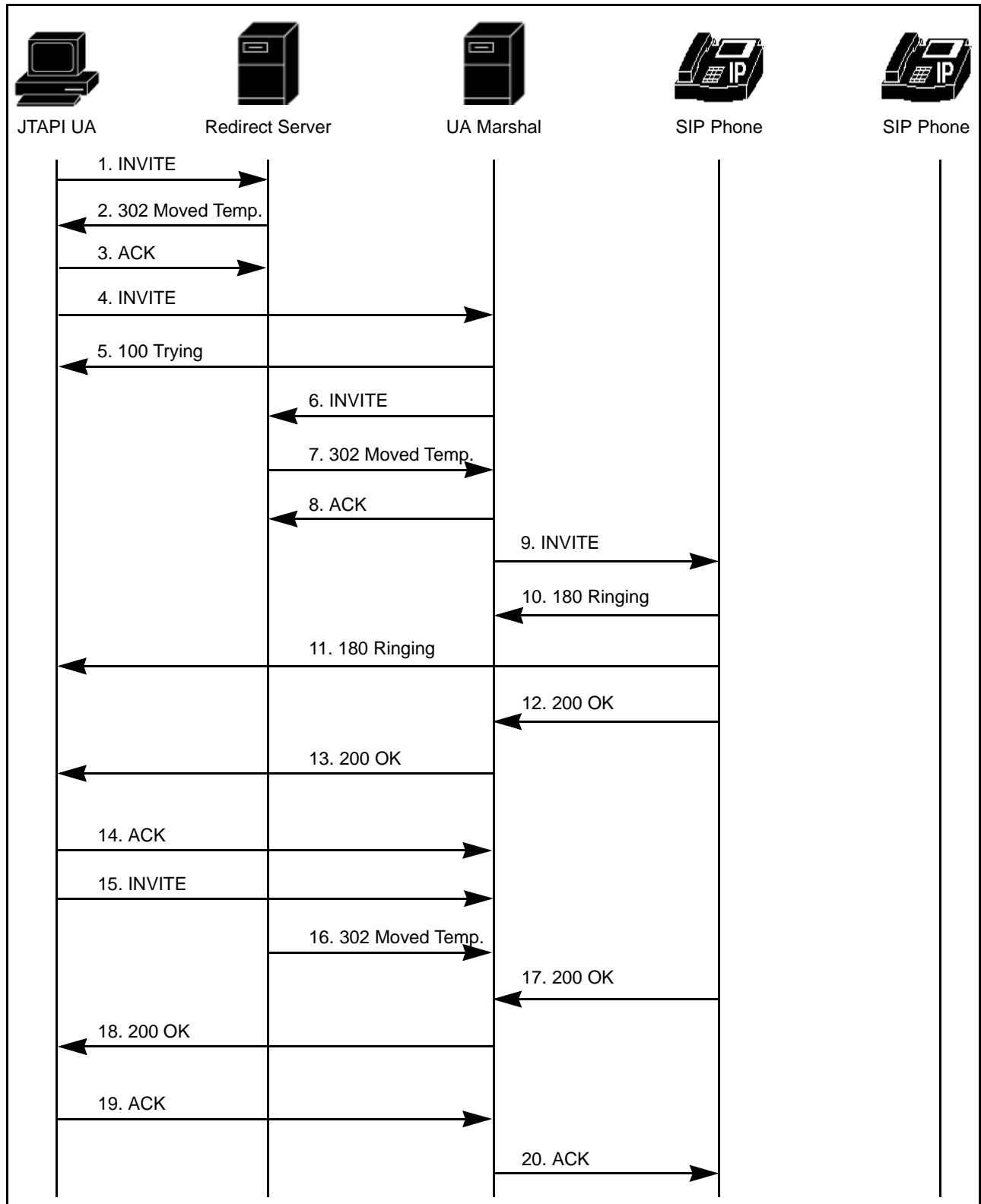


Figure C-35. User Agent to User Agent: Call Waiting — Diagram 1

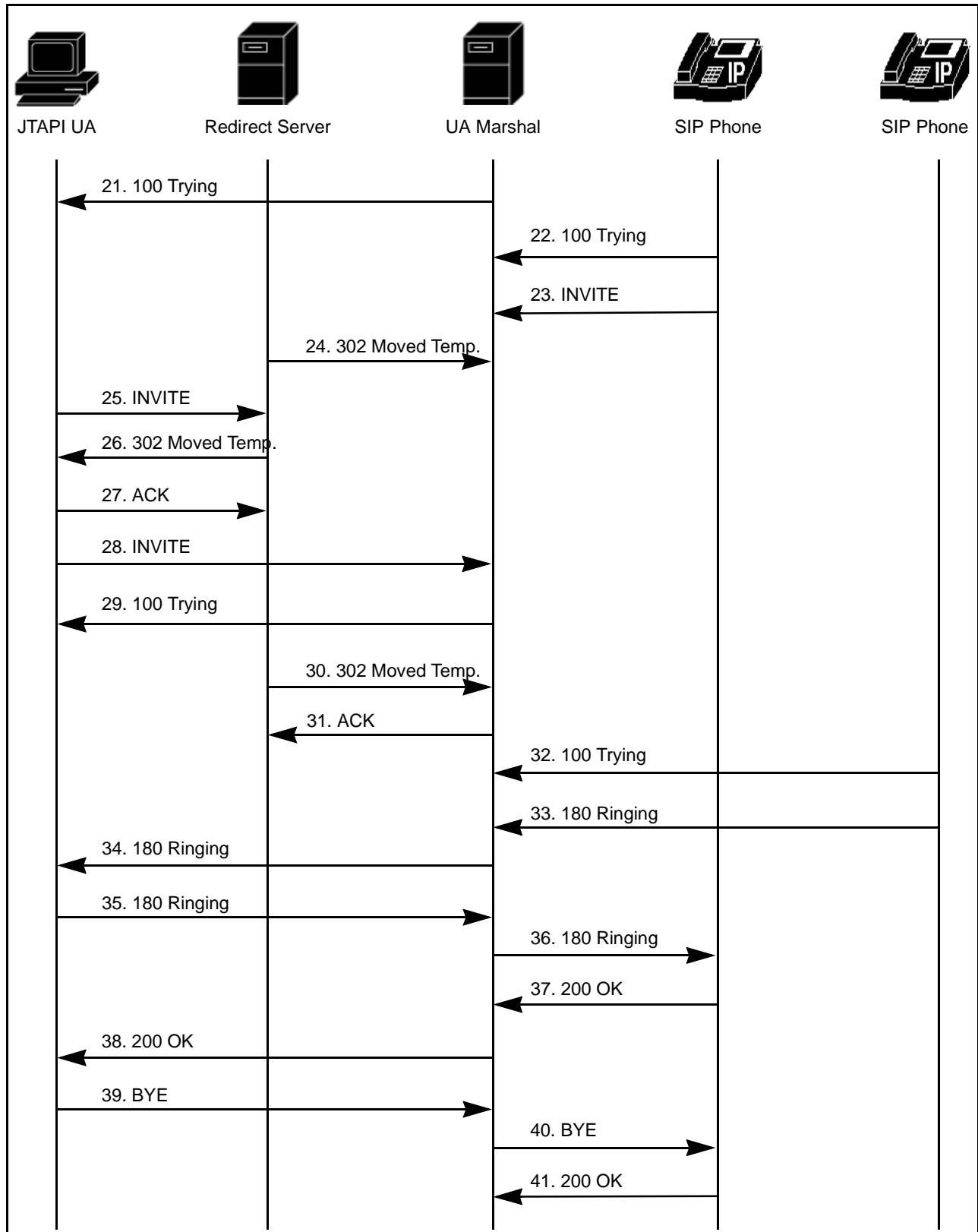


Figure C-36. User Agent to User Agent: Call Waiting — Diagram 2

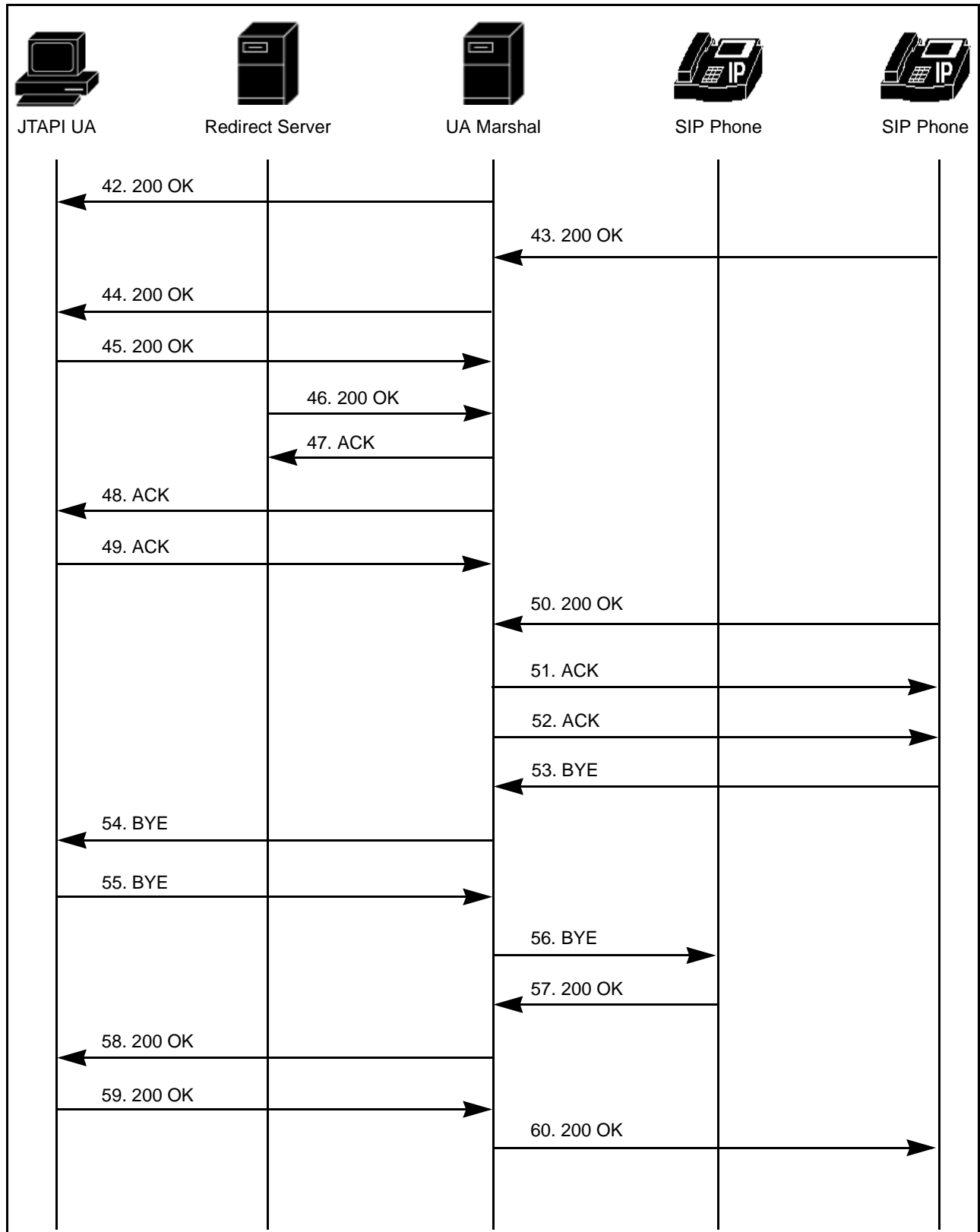


Figure C-37. User Agent to User Agent: Call Waiting — Diagram 3

## Call Trace

The following call trace shows third party call control through a JTAPI server.

```

-----
                        SIP Headers
-----
sip-reg:    INVITE sip:6710@192.168.26.200:5060;user=phone SIP/2.0 [192.168.5.11:25060-
>192.168.26.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.5.11:25060;branch=201
Header:    From: <sip:-@192.168.5.11:25060>
Header:    To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:    Call-ID: 1087893762978930@192.168.5.11
Header:    CSeq: 1 INVITE
Header:    Proxy-Authorization: Basic vovidaClassXswitch
Header:    Subject: VovidaINVITE
Header:    Contact: <sip:js@192.168.5.11:25060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 0
-----
                        SDP Headers
-----
-----
                        SIP Headers
-----
sip-res:    SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.5.11:25060]
Header:    Via: SIP/2.0/UDP 192.168.5.11:25060;branch=201
Header:    From: <sip:-@192.168.5.11:25060>
Header:    To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:    Call-ID: 1087893762978930@192.168.5.11
Header:    CSeq: 1 INVITE
Header:    Contact: <sip:6710@192.168.26.180:5060>
Header:    Content-Length: 0
-----
                        SIP Headers
-----
sip-reg:    ACK sip:6710@192.168.26.200:5060;user=phone SIP/2.0 [192.168.5.11:25060-
>192.168.26.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.5.11:25060;branch=201
Header:    From: <sip:-@192.168.5.11:25060>
Header:    To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:    Call-ID: 1087893762978930@192.168.5.11
Header:    CSeq: 1 ACK
Header:    Content-Length: 0
-----
                        SIP Headers
-----
sip-reg:    INVITE sip:6710@192.168.26.180:5060 SIP/2.0 [192.168.5.11:25060-
>192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header:    From: <sip:-@192.168.5.11:25060>
Header:    To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:    Call-ID: 1087893762978930@192.168.5.11
Header:    CSeq: 1 INVITE
Header:    Proxy-Authorization: Basic vovidaClassXswitch
Header:    Subject: VovidaINVITE
Header:    Contact: <sip:js@192.168.5.11:25060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 0
-----
                        SDP Headers
-----
-----
                        SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.26.180:5060->192.168.5.11:25060]
Header:    Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header:    From: <sip:-@192.168.5.11:25060>
Header:    To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:    Call-ID: 1087893762978930@192.168.5.11
Header:    CSeq: 1 INVITE

```

```

Header:                               Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    INVITE sip:6710@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=3
Header:    Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header:    From: <sip:-@192.168.5.11:25060>
Header:    To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:    Call-ID: 1087893762978930@192.168.5.11
Header:    CSeq: 1 INVITE
Header:    Subject: VovidaINVITE
Header:    Record-Route: <sip:6710@192.168.26.180:5060;maddr=192.168.26.180>
Header:    Contact: <sip:js@192.168.5.11:25060>
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=3
Header:    Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header:    From: <sip:-@192.168.5.11:25060>
Header:    To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:    Call-ID: 1087893762978930@192.168.5.11
Header:    CSeq: 1 INVITE
Header:    Contact: <sip:6710@192.168.22.36:5060>
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    ACK sip:6710@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060-
>192.168.26.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=3
Header:    From: <sip:-@192.168.5.11:25060>
Header:    To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:    Call-ID: 1087893762978930@192.168.5.11
Header:    CSeq: 1 ACK
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    INVITE sip:6710@192.168.22.36:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.22.36:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header:    From: <sip:-@192.168.5.11:25060>
Header:    To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:    Call-ID: 1087893762978930@192.168.5.11
Header:    CSeq: 1 INVITE
Header:    Subject: VovidaINVITE
Header:    Record-Route: <sip:6710@192.168.26.180:5060;maddr=192.168.26.180>
Header:    Contact: <sip:js@192.168.5.11:25060>
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 180 Ringing [192.168.22.36:5060->192.168.26.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header:    From: <sip:-@192.168.5.11:25060>
Header:    To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:    Call-ID: 1087893762978930@192.168.5.11
Header:    CSeq: 1 INVITE
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 180 Ringing [192.168.26.180:5060-
>192.168.5.11:25060]

```



```

Header:      Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header:      From: <sip:-@192.168.5.11:25060>
Header:      To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:      Call-ID: 1087893762978930@192.168.5.11
Header:      CSeq: 1 INVITE
Header:      Content-Length: 0

```

-----  
SIP Headers

```

sip-res:     SIP/2.0 200 OK [192.168.22.36:5060->192.168.26.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header:      Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header:      From: <sip:-@192.168.5.11:25060>
Header:      To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:      Call-ID: 1087893762978930@192.168.5.11
Header:      CSeq: 1 INVITE
Header:      Contact: <sip:6710@192.168.22.36:5060>
Header:      Record-Route: <sip:6710@192.168.26.180:5060;maddr=192.168.26.180>
Header:      Content-Type: application/sdp
Header:      Content-Length: 170

```

-----  
SDP Headers

```

Header:      v=0
Header:      o=- 1735072859 1735072859 IN IP4 192.168.22.36
Header:      s=VOVIDA Session
Header:      c=IN IP4 192.168.22.36
Header:      t=3174942917 0
Header:      m=audio 23456 RTP/AVP 0
Header:      a=rtpmap:0 PCMU/8000
Header:      a=ptime:20

```

-----  
SIP Headers

```

sip-res:     SIP/2.0 200 OK [192.168.26.180:5060->192.168.5.11:25060]
Header:      Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header:      From: <sip:-@192.168.5.11:25060>
Header:      To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:      Call-ID: 1087893762978930@192.168.5.11
Header:      CSeq: 1 INVITE
Header:      Contact: <sip:6710@192.168.22.36:5060>
Header:      Record-Route: <sip:6710@192.168.26.180:5060;maddr=192.168.26.180>
Header:      Content-Type: application/sdp
Header:      Content-Length: 170

```

-----  
SDP Headers

```

Header:      v=0
Header:      o=- 1735072859 1735072859 IN IP4 192.168.22.36
Header:      s=VOVIDA Session
Header:      c=IN IP4 192.168.22.36
Header:      t=3174942917 0
Header:      m=audio 23456 RTP/AVP 0
Header:      a=rtpmap:0 PCMU/8000
Header:      a=ptime:20

```

-----  
SIP Headers

```

sip-req:     ACK sip:6710@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.5.11:25060->192.168.26.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header:      From: <sip:-@192.168.5.11:25060>
Header:      To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header:      Call-ID: 1087893762978930@192.168.5.11
Header:      CSeq: 1 ACK
Header:      Route: <sip:6710@192.168.22.36:5060>
Header:      Proxy-Authorization: Basic vovidaClassXswitch
Header:      Content-Type: application/sdp
Header:      Content-Length: 170

```

## SDP Headers

```

-----
Header: v=0
Header: o=- 1735072859 1735072859 IN IP4 192.168.22.36
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.22.36
Header: t=3174942917 0
Header: m=audio 23456 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
-----

```

## SIP Headers

```

-----
sip-req: INVITE sip:6710@192.168.26.180:5060 SIP/2.0 [192.168.5.11:25060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header: From: <sip:-@192.168.5.11:25060>
Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 INVITE
Header: Proxy-Authorization: Basic vovidaClassXswitch
Header: Subject: VovidaINVITE
Header: Contact: <sip:js@192.168.5.11:25060>
Header: Content-Type: application/sdp
Header: Content-Length: 164
-----

```

## SDP Headers

```

-----
Header: v=0
Header: o=- 1735072859 1735072859 IN IP4 192.168.22.36
Header: s=VOVIDA Session
Header: c=IN IP4 0.0.0.0
Header: t=3174942917 0
Header: m=audio 23456 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
-----

```

## SIP Headers

```

-----
sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=3
Header: Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header: From: <sip:-@192.168.5.11:25060>
Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 INVITE
Header: Contact: <sip:6710@192.168.22.36:5060>
Header: Content-Length: 0
-----

```

## SIP Headers

```

-----
sip-res: SIP/2.0 200 OK [192.168.22.36:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301
Header: From: <sip:-@192.168.5.11:25060>
Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 INVITE
Header: Contact: <sip:6710@192.168.22.36:5060;user=phone>
Header: Record-Route: <sip:6710@192.168.26.180:5060;maddr=192.168.26.180>
Header: Content-Type: application/sdp
Header: Content-Length: 170
-----

```

## SDP Headers

```

-----
Header: v=0
Header: o=- 1735072859 1735072859 IN IP4 192.168.22.36
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.22.36
Header: t=3174942917 0
-----

```

Header: m=audio 23456 RTP/AVP 0  
 Header: a=rtpmap:0 PCMU/8000  
 Header: a=ptime:20

-----  
 SIP Headers

-----  
 sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.5.11:25060]  
 Header: Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301  
 Header: From: <sip:-@192.168.5.11:25060>  
 Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>  
 Header: Call-ID: 1087893762978930@192.168.5.11  
 Header: CSeq: 2 INVITE  
 Header: Contact: <sip:6710@192.168.22.36:5060;user=phone>  
 Header: Record-Route: <sip:6710@192.168.26.180:5060;maddr=192.168.26.180>  
 Header: Content-Type: application/sdp  
 Header: Content-Length: 170  
 -----

SDP Headers

-----  
 Header: v=0  
 Header: o=- 1735072859 1735072859 IN IP4 192.168.22.36  
 Header: s=VOVIDA Session  
 Header: c=IN IP4 192.168.22.36  
 Header: t=3174942917 0  
 Header: m=audio 23456 RTP/AVP 0  
 Header: a=rtpmap:0 PCMU/8000  
 Header: a=ptime:20  
 -----

SIP Headers

-----  
 sip-req: ACK sip:6710@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0  
 [192.168.5.11:25060->192.168.26.180:5060]  
 Header: Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301  
 Header: From: <sip:-@192.168.5.11:25060>  
 Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>  
 Header: Call-ID: 1087893762978930@192.168.5.11  
 Header: CSeq: 2 ACK  
 Header: Route: <sip:6710@192.168.22.36:5060>  
 Header: Proxy-Authorization: Basic vovidaClassXswitch  
 Header: Content-Length: 0  
 -----

SIP Headers

-----  
 sip-req: ACK sip:6710@192.168.22.36:5060 SIP/2.0 [192.168.26.180:5060->192.168.22.36:5060]  
 Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
 Header: Via: SIP/2.0/UDP 192.168.5.11:25060;branch=301  
 Header: From: <sip:-@192.168.5.11:25060>  
 Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>  
 Header: Call-ID: 1087893762978930@192.168.5.11  
 Header: CSeq: 2 ACK  
 Header: Content-Length: 0  
 -----

SIP Headers

-----  
 sip-res: SIP/2.0 100 Trying [192.168.26.180:5060->192.168.5.11:25060]  
 Header: Via: SIP/2.0/UDP 192.168.5.11:25060  
 Header: From: <sip:-@192.168.5.11:25060>  
 Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>  
 Header: Call-ID: 1087893762978930@192.168.5.11  
 Header: CSeq: 2 TRANSFER  
 Header: Content-Length: 0  
 -----

SIP Headers

-----  
 sip-res: SIP/2.0 100 Trying [192.168.22.36:5060->192.168.26.180:5060]  
 Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4  
 Header: Via: SIP/2.0/UDP 192.168.5.11:25060  
 Header: From: <sip:-@192.168.5.11:25060>  
 Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>  
 -----

Header: Call-ID: 1087893762978930@192.168.5.11  
 Header: CSeq: 2 TRANSFER  
 Header: Content-Length: 0

-----  
 SIP Headers  
 -----

sip-req: INVITE sip:6711@192.168.26.180:5060;user=phone SIP/2.0 [192.168.22.36:5060->192.168.26.180:5060]  
 Header: Via: SIP/2.0/UDP 192.168.22.36:5060  
 Header: From: UserAgent<sip:6710@192.168.22.36:5060>  
 Header: To: <sip:6711@192.168.26.200:5060;user=phone>  
 Header: Call-ID: 1087893762978930@192.168.5.11  
 Header: CSeq: 1 INVITE  
 Header: Subject: VovidaINVITE  
 Header: Contact: <sip:6710@192.168.22.36:5060;user=phone>  
 Header: Content-Type: application/sdp  
 Header: Content-Length: 168

-----  
 SDP Headers  
 -----

Header: v=0  
 Header: o=- 356310517 356310517 IN IP4 192.168.22.36  
 Header: s=VOVIDA Session  
 Header: c=IN IP4 192.168.22.36  
 Header: t=3174942918 0  
 Header: m=audio 23456 RTP/AVP 0  
 Header: a=rtpmap:0 PCMU/8000  
 Header: a=ptime:20

-----  
 SIP Headers  
 -----

sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]  
 Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=1  
 Header: Via: SIP/2.0/UDP 192.168.22.36:5060  
 Header: From: UserAgent<sip:6710@192.168.22.36:5060>  
 Header: To: <sip:6711@192.168.26.200:5060;user=phone>  
 Header: Call-ID: 1087893762978930@192.168.5.11  
 Header: CSeq: 1 INVITE  
 Header: Contact: <sip:6711@192.168.5.11:15060;user=phone>  
 Header: Content-Length: 0

-----  
 SIP Headers  
 -----

sip-req: INVITE sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.5.11:15060->192.168.26.200:5060]  
 Header: Via: SIP/2.0/UDP 192.168.5.11:15060;branch=202  
 Header: From: UserAgent<sip:6710@192.168.22.36:5060>  
 Header: To: 6711<sip:6711@192.168.26.200:5060;user=phone>  
 Header: Call-ID: 458898268105186@192.168.5.11  
 Header: CSeq: 1 INVITE  
 Header: Proxy-Authorization: Basic vovidaClassXswitch  
 Header: Subject: VovidaINVITE  
 Header: Contact: <sip:js@192.168.5.11:15060>  
 Header: Content-Type: application/sdp  
 Header: Content-Length: 168

-----  
 SDP Headers  
 -----

Header: v=0  
 Header: o=- 356310517 356310517 IN IP4 192.168.22.36  
 Header: s=VOVIDA Session  
 Header: c=IN IP4 192.168.22.36  
 Header: t=3174942918 0  
 Header: m=audio 23456 RTP/AVP 0  
 Header: a=rtpmap:0 PCMU/8000  
 Header: a=ptime:20

-----  
 SIP Headers  
 -----

sip-res: SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.5.11:15060]

```

Header:      Via: SIP/2.0/UDP 192.168.5.11:15060;branch=202
Header:      From: UserAgent<sip:6710@192.168.22.36:5060>
Header:      To: 6711<sip:6711@192.168.26.200:5060;user=phone>
Header:      Call-ID: 458898268105186@192.168.5.11
Header:      CSeq: 1 INVITE
Header:      Contact: <sip:6711@192.168.26.180:5060>
Header:      Content-Length: 0

```

-----  
SIP Headers

```

sip-req:     ACK sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.5.11:15060-
>192.168.26.200:5060]
Header:      Via: SIP/2.0/UDP 192.168.5.11:15060;branch=202
Header:      From: UserAgent<sip:6710@192.168.22.36:5060>
Header:      To: 6711<sip:6711@192.168.26.200:5060;user=phone>
Header:      Call-ID: 458898268105186@192.168.5.11
Header:      CSeq: 1 ACK
Header:      Content-Length: 0

```

-----  
SIP Headers

```

sip-req:     INVITE sip:6711@192.168.26.180:5060 SIP/2.0 [192.168.5.11:15060-
>192.168.26.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.5.11:15060;branch=302
Header:      From: UserAgent<sip:6710@192.168.22.36:5060>
Header:      To: 6711<sip:6711@192.168.26.200:5060;user=phone>
Header:      Call-ID: 458898268105186@192.168.5.11
Header:      CSeq: 1 INVITE
Header:      Proxy-Authorization: Basic vovidaClassXswitch
Header:      Subject: VovidaINVITE
Header:      Contact: <sip:js@192.168.5.11:15060>
Header:      Content-Type: application/sdp
Header:      Content-Length: 168

```

-----  
SDP Headers

```

Header:      v=0
Header:      o=- 356310517 356310517 IN IP4 192.168.22.36
Header:      s=VOVIDA Session
Header:      c=IN IP4 192.168.22.36
Header:      t=3174942918 0
Header:      m=audio 23456 RTP/AVP 0
Header:      a=rtpmap:0 PCMU/8000
Header:      a=ptime:20

```

-----  
SIP Headers

```

sip-res:     SIP/2.0 100 Trying [192.168.26.180:5060->192.168.5.11:15060]
Header:      Via: SIP/2.0/UDP 192.168.5.11:15060;branch=302
Header:      From: UserAgent<sip:6710@192.168.22.36:5060>
Header:      To: 6711<sip:6711@192.168.26.200:5060;user=phone>
Header:      Call-ID: 458898268105186@192.168.5.11
Header:      CSeq: 1 INVITE
Header:      Content-Length: 0

```

-----  
SIP Headers

```

sip-res:     SIP/2.0 302 Moved Temporarily [192.168.26.200:5060->192.168.26.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.26.180:5060;branch=3
Header:      Via: SIP/2.0/UDP 192.168.5.11:15060;branch=302
Header:      From: UserAgent<sip:6710@192.168.22.36:5060>
Header:      To: 6711<sip:6711@192.168.26.200:5060;user=phone>
Header:      Call-ID: 458898268105186@192.168.5.11
Header:      CSeq: 1 INVITE
Header:      Contact: <sip:6711@192.168.26.10:5060>
Header:      Content-Length: 0

```

-----  
SIP Headers

```
sip-res: ACK sip:6711@192.168.26.200:5060;user=phone SIP/2.0 [192.168.26.180:5060->192.168.26.200:5060]
```

```
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=3
Header: From: UserAgent<sip:6710@192.168.22.36:5060>
Header: To: 6711<sip:6711@192.168.26.200:5060;user=phone>
Header: Call-ID: 458898268105186@192.168.5.11
Header: CSeq: 1 ACK
Header: Content-Length: 0
```

```
-----
SIP Headers
-----
```

```
sip-res: SIP/2.0 100 Trying [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.5.11:15060;branch=302
Header: From: UserAgent<sip:6710@192.168.22.36:5060>
Header: To:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header: Call-ID: 458898268105186@192.168.5.11
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: CSeq: 1 INVITE
Header: Content-Length: 0
```

```
-----
SIP Headers
-----
```

```
sip-res: SIP/2.0 180 Ringing [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.5.11:15060;branch=302
Header: From: UserAgent<sip:6710@192.168.22.36:5060>
Header: To:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header: Call-ID: 458898268105186@192.168.5.11
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: CSeq: 1 INVITE
Header: Content-Length: 0
```

```
-----
SIP Headers
-----
```

```
sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.5.11:15060]
Header: Via: SIP/2.0/UDP 192.168.5.11:15060;branch=302
Header: From: UserAgent<sip:6710@192.168.22.36:5060>
Header: To:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header: Call-ID: 458898268105186@192.168.5.11
Header: CSeq: 1 INVITE
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Content-Length: 0
```

```
-----
SIP Headers
-----
```

```
sip-res: SIP/2.0 180 Ringing [192.168.5.11:15060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.22.36:5060
Header: From: UserAgent<sip:6710@192.168.22.36:5060>
Header: To: <sip:6711@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 1 INVITE
Header: Content-Length: 0
```

```
-----
SIP Headers
-----
```

```
sip-res: SIP/2.0 180 Ringing [192.168.26.180:5060->192.168.22.36:5060]
Header: Via: SIP/2.0/UDP 192.168.22.36:5060
Header: From: UserAgent<sip:6710@192.168.22.36:5060>
Header: To: <sip:6711@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 1 INVITE
Header: Content-Length: 0
```

```
-----
SIP Headers
-----
```

```

sip-res: SIP/2.0 200 OK [192.168.22.36:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.5.11:25060
Header: From: <sip:-@192.168.5.11:25060>
Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 TRANSFER
Header: Content-Length: 0

```

-----  
SIP Headers

```

sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.5.11:25060]
Header: Via: SIP/2.0/UDP 192.168.5.11:25060
Header: From: <sip:-@192.168.5.11:25060>
Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 TRANSFER
Header: Content-Length: 0

```

-----  
SIP Headers

```

sip-req: BYE sip:6710@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.5.11:25060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.5.11:25060
Header: From: <sip:-@192.168.5.11:25060>
Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 BYE
Header: Proxy-Authorization: Basic vovidaClassXswitch
Header: Route: <sip:6710@192.168.22.36:5060>
Header: Content-Length: 0

```

-----  
SIP Headers

```

sip-req: BYE sip:6710@192.168.22.36:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.22.36:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.5.11:25060
Header: From: <sip:-@192.168.5.11:25060>
Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 BYE
Header: Content-Length: 0

```

-----  
SIP Headers

```

sip-res: SIP/2.0 200 OK [192.168.22.36:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.5.11:25060
Header: From: <sip:-@192.168.5.11:25060>
Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 BYE
Header: Content-Length: 0

```

-----  
SIP Headers

```

sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.5.11:25060]
Header: Via: SIP/2.0/UDP 192.168.5.11:25060
Header: From: <sip:-@192.168.5.11:25060>
Header: To: 6710<sip:6710@192.168.26.200:5060;user=phone>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 BYE
Header: Content-Length: 0

```

-----  
SIP Headers

```

sip-res: SIP/2.0 200 OK [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.5.11:15060;branch=302

```

```

Header:      From: UserAgent<sip:6710@192.168.22.36:5060>
Header:      To:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header:      Call-ID: 458898268105186@192.168.5.11
Header:      Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:      Contact: sip:6711@192.168.26.10:5060
Header:      Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header:      CSeq: 1 INVITE
Header:      Content-Type: application/sdp
Header:      Content-Length: 221

```

-----  
SDP Headers

```

Header:      v=0
Header:      o=CiscoSystemsSIP-IPPhone-UserAgent 26487 28247 IN IP4
192.168.26.10
Header:      s=SIP Call
Header:      c=IN IP4 192.168.26.10
Header:      t=0 0
Header:      m=audio 23206 RTP/AVP 0 101
Header:      a=rtpmap:0 pcmu/8000
Header:      a=rtpmap:101 telephone-event/8000
Header:      a=fmtp:101 0-11

```

-----  
SIP Headers

```

sip-res:     SIP/2.0 200 OK [192.168.26.180:5060->192.168.5.11:15060]
Header:      Via: SIP/2.0/UDP 192.168.5.11:15060;branch=302
Header:      From: UserAgent<sip:6710@192.168.22.36:5060>
Header:      To:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header:      Call-ID: 458898268105186@192.168.5.11
Header:      CSeq: 1 INVITE
Header:      Contact: <sip:6711@192.168.26.10:5060>
Header:      Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header:      Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header:      Content-Type: application/sdp
Header:      Content-Length: 221

```

-----  
SDP Headers

```

Header:      v=0
Header:      o=CiscoSystemsSIP-IPPhone-UserAgent 26487 28247 IN IP4
192.168.26.10
Header:      s=SIP Call
Header:      c=IN IP4 192.168.26.10
Header:      t=0 0
Header:      m=audio 23206 RTP/AVP 0 101
Header:      a=rtpmap:0 pcmu/8000
Header:      a=rtpmap:101 telephone-event/8000
Header:      a=fmtp:101 0-11

```

-----  
SIP Headers

```

sip-res:     SIP/2.0 200 OK [192.168.5.11:15060->192.168.26.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:      Via: SIP/2.0/UDP 192.168.22.36:5060
Header:      From: UserAgent<sip:6710@192.168.22.36:5060>
Header:      To: <sip:6711@192.168.26.200:5060;user=phone>
Header:      Call-ID: 1087893762978930@192.168.5.11
Header:      CSeq: 1 INVITE
Header:      Contact: <sip:js@192.168.5.11:15060>
Header:      Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header:      Content-Type: application/sdp
Header:      Content-Length: 221

```

-----  
SDP Headers

```

Header:      v=0

```



```

Header:          o=CiscoSystemsSIP-IPPhone-UserAgent 26487 28247 IN IP4
192.168.26.10
Header:          s=SIP Call
Header:          c=IN IP4 192.168.26.10
Header:          t=0 0
Header:          m=audio 23206 RTP/AVP 0 101
Header:          a=rtpmap:0 pcmu/8000
Header:          a=rtpmap:101 telephone-event/8000
Header:          a=fmtp:101 0-11

```

-----  
SIP Headers

```

sip-res:        SIP/2.0 200 OK [192.168.26.180:5060->192.168.22.36:5060]
Header:         Via: SIP/2.0/UDP 192.168.22.36:5060
Header:         From: UserAgent<sip:6710@192.168.22.36:5060>
Header:         To: <sip:6711@192.168.26.200:5060;user=phone>
Header:         Call-ID: 1087893762978930@192.168.5.11
Header:         CSeq: 1 INVITE
Header:         Contact: <sip:js@192.168.5.11:15060>
Header:         Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header:         Content-Type: application/sdp
Header:         Content-Length: 221

```

-----  
SDP Headers

```

Header:         v=0
Header:         o=CiscoSystemsSIP-IPPhone-UserAgent 26487 28247 IN IP4
192.168.26.10
Header:         s=SIP Call
Header:         c=IN IP4 192.168.26.10
Header:         t=0 0
Header:         m=audio 23206 RTP/AVP 0 101
Header:         a=rtpmap:0 pcmu/8000
Header:         a=rtpmap:101 telephone-event/8000
Header:         a=fmtp:101 0-11

```

-----  
SIP Headers

```

sip-req:        ACK sip:6711@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.22.36:5060->192.168.26.180:5060]
Header:         Via: SIP/2.0/UDP 192.168.22.36:5060
Header:         From: UserAgent<sip:6710@192.168.22.36:5060>
Header:         To: <sip:6711@192.168.26.200:5060;user=phone>
Header:         Call-ID: 1087893762978930@192.168.5.11
Header:         CSeq: 1 ACK
Header:         Route: <sip:js@192.168.5.11:15060>
Header:         Content-Length: 0

```

-----  
SIP Headers

```

sip-req:        ACK sip:js@192.168.5.11:15060 SIP/2.0          [192.168.26.180:5060-
>192.168.5.11:15060]
Header:         Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header:         Via: SIP/2.0/UDP 192.168.22.36:5060
Header:         From: UserAgent<sip:6710@192.168.22.36:5060>
Header:         To: <sip:6711@192.168.26.200:5060;user=phone>
Header:         Call-ID: 1087893762978930@192.168.5.11
Header:         CSeq: 1 ACK
Header:         Proxy-Authorization: Basic vovidaClassXswitch
Header:         Content-Length: 0

```

-----  
SIP Headers

```

sip-req:        ACK sip:6711@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0
[192.168.5.11:15060->192.168.26.180:5060]
Header:         Via: SIP/2.0/UDP 192.168.5.11:15060;branch=302
Header:         From: UserAgent<sip:6710@192.168.22.36:5060>
Header:         To:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header:         Call-ID: 458898268105186@192.168.5.11

```

```
Header: CSeq: 1 ACK
Header: Route: <sip:6711@192.168.26.10:5060>
Header: Proxy-Authorization: Basic vovidaClassXswitch
Header: Content-Length: 0
```

-----  
SIP Headers

```
-----
sip-res: SIP/2.0 200 OK [192.168.26.10:50373->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4,SIP/2.0/UDP
192.168.5.11:15060;branch=302
Header: From: UserAgent<sip:6710@192.168.22.36:5060>
Header: To:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header: Call-ID: 458898268105186@192.168.5.11
Header: Server: Cisco IP Phone/ Rev. 1/ SIP enabled
Header: Contact: sip:6711@192.168.26.10:5060
Header: Record-Route: <sip:6711@192.168.26.180:5060;maddr=192.168.26.180>
Header: CSeq: 1 INVITE
Header: Content-Type: application/sdp
Header: Content-Length: 221
-----
```

-----  
SDP Headers

```
-----
Header: v=0
Header: o=CiscoSystemsSIP-IPPhone-UserAgent 26487 28247 IN IP4
192.168.26.10
Header: s=SIP Call
Header: c=IN IP4 192.168.26.10
Header: t=0 0
Header: m=audio 23206 RTP/AVP 0 101
Header: a=rtpmap:0 pcmu/8000
Header: a=rtpmap:101 telephone-event/8000
Header: a=fmtp:101 0-11
-----
```

-----  
SIP Headers

```
-----
sip-req: ACK sip:6711@192.168.26.10:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.26.10:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.5.11:15060;branch=302
Header: From: UserAgent<sip:6710@192.168.22.36:5060>
Header: To:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header: Call-ID: 458898268105186@192.168.5.11
Header: CSeq: 1 ACK
Header: Content-Length: 0
-----
```

-----  
SIP Headers

```
-----
sip-req: ACK sip:6711@192.168.26.10:5060 SIP/2.0 [192.168.26.180:5060-
>192.168.26.10:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.5.11:15060;branch=302
Header: From: UserAgent<sip:6710@192.168.22.36:5060>
Header: To:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header: Call-ID: 458898268105186@192.168.5.11
Header: CSeq: 1 ACK
Header: Content-Length: 0
-----
```

-----  
SIP Headers

```
-----
sip-req: BYE sip:6711@192.168.26.180:5060 SIP/2.0 [192.168.26.10:50373-
>192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header: To: UserAgent<sip:6710@192.168.22.36:5060>
Header: Call-ID: 458898268105186@192.168.5.11
Header: User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
-----
```

```
Header: CSeq: 2 BYE
Header: Route: <sip:js@192.168.5.11:15060>
Header: Content-Length: 0
```

-----  
SIP Headers

```
-----
sip-req: BYE sip:js@192.168.5.11:15060 SIP/2.0 [192.168.26.180:5060->192.168.5.11:15060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
Header: From:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header: To: UserAgent<sip:6710@192.168.22.36:5060>
Header: Call-ID: 458898268105186@192.168.5.11
Header: CSeq: 2 BYE
Header: Proxy-Authorization: Basic vovidaClassXswitch
Header: Content-Length: 0
```

-----  
SIP Headers

```
-----
sip-req: BYE sip:6711@192.168.26.180:5060;maddr=192.168.26.180 SIP/2.0 [192.168.5.11:15060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.5.11:15060
Header: From: <sip:6711@192.168.26.200:5060;user=phone>
Header: To: UserAgent<sip:6710@192.168.22.36:5060>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 BYE
Header: Route: <sip:6710@192.168.22.36:5060>
Header: Content-Length: 0
```

-----  
SIP Headers

```
-----
sip-req: BYE sip:6710@192.168.22.36:5060 SIP/2.0 [192.168.26.180:5060->192.168.22.36:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.5.11:15060
Header: From: <sip:6711@192.168.26.200:5060;user=phone>
Header: To: UserAgent<sip:6710@192.168.22.36:5060>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 BYE
Header: Proxy-Authorization: Basic vovidaClassXswitch
Header: Content-Length: 0
```

-----  
SIP Headers

```
-----
sip-res: SIP/2.0 200 OK [192.168.22.36:5060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.5.11:15060
Header: From: <sip:6711@192.168.26.200:5060;user=phone>
Header: To: UserAgent<sip:6710@192.168.22.36:5060>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 BYE
Header: Content-Length: 0
```

-----  
SIP Headers

```
-----
sip-res: SIP/2.0 200 OK [192.168.26.180:5060->192.168.5.11:15060]
Header: Via: SIP/2.0/UDP 192.168.5.11:15060
Header: From: <sip:6711@192.168.26.200:5060;user=phone>
Header: To: UserAgent<sip:6710@192.168.22.36:5060>
Header: Call-ID: 1087893762978930@192.168.5.11
Header: CSeq: 2 BYE
Header: Content-Length: 0
```

-----  
SIP Headers

```
-----
sip-res: SIP/2.0 200 OK [192.168.5.11:15060->192.168.26.180:5060]
Header: Via: SIP/2.0/UDP 192.168.26.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.26.10:5060
```

## JTAPI

```
Header:          From:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header:          To: UserAgent<sip:6710@192.168.22.36:5060>
Header:          Call-ID: 458898268105186@192.168.5.11
Header:          CSeq: 2 BYE
Header:          Contact: <sip:js@192.168.5.11:15060>
Header:          Content-Length: 0
-----
                          SIP Headers
-----
sip-res:      SIP/2.0 200 OK [192.168.26.180:5060->192.168.26.10:5060]
Header:      Via: SIP/2.0/UDP 192.168.26.10:5060
Header:      From:
6711<sip:6711@192.168.26.200:5060;user=phone>;tag=c29430003e2620-0
Header:      To: UserAgent<sip:6710@192.168.22.36:5060>
Header:      Call-ID: 458898268105186@192.168.5.11
Header:      CSeq: 2 BYE
Header:      Contact: <sip:js@192.168.5.11:15060>
Header:      Content-Length: 0
-----
```

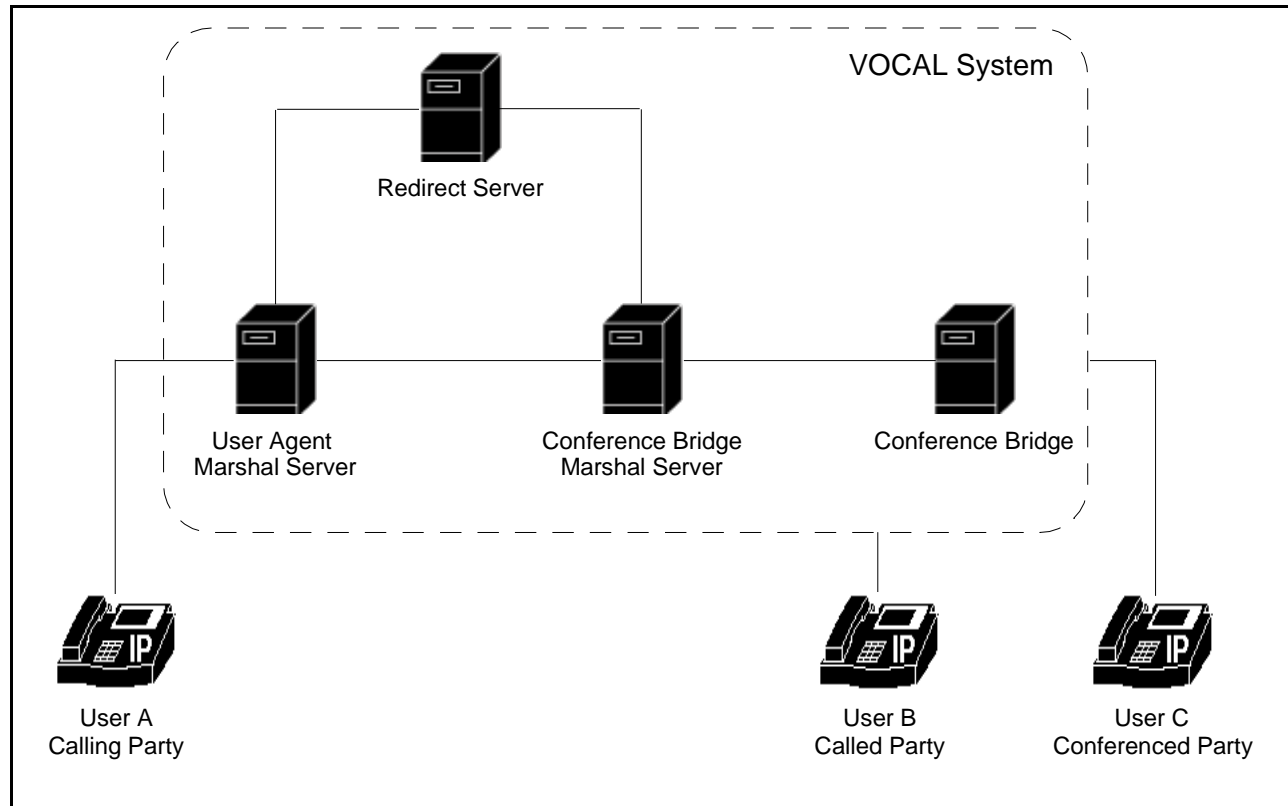
---

## Ad Hoc Conference Call Between User Agents

### Call Scenario

Figure C-38 illustrates the following call scenario:

- User A calls User B
- User A brings User C into the conversation via ad hoc conferencing.



**Figure C-38. User Agent to User Agent to User Agent: Ad Hoc Conference Call: Disconnect Last Party**

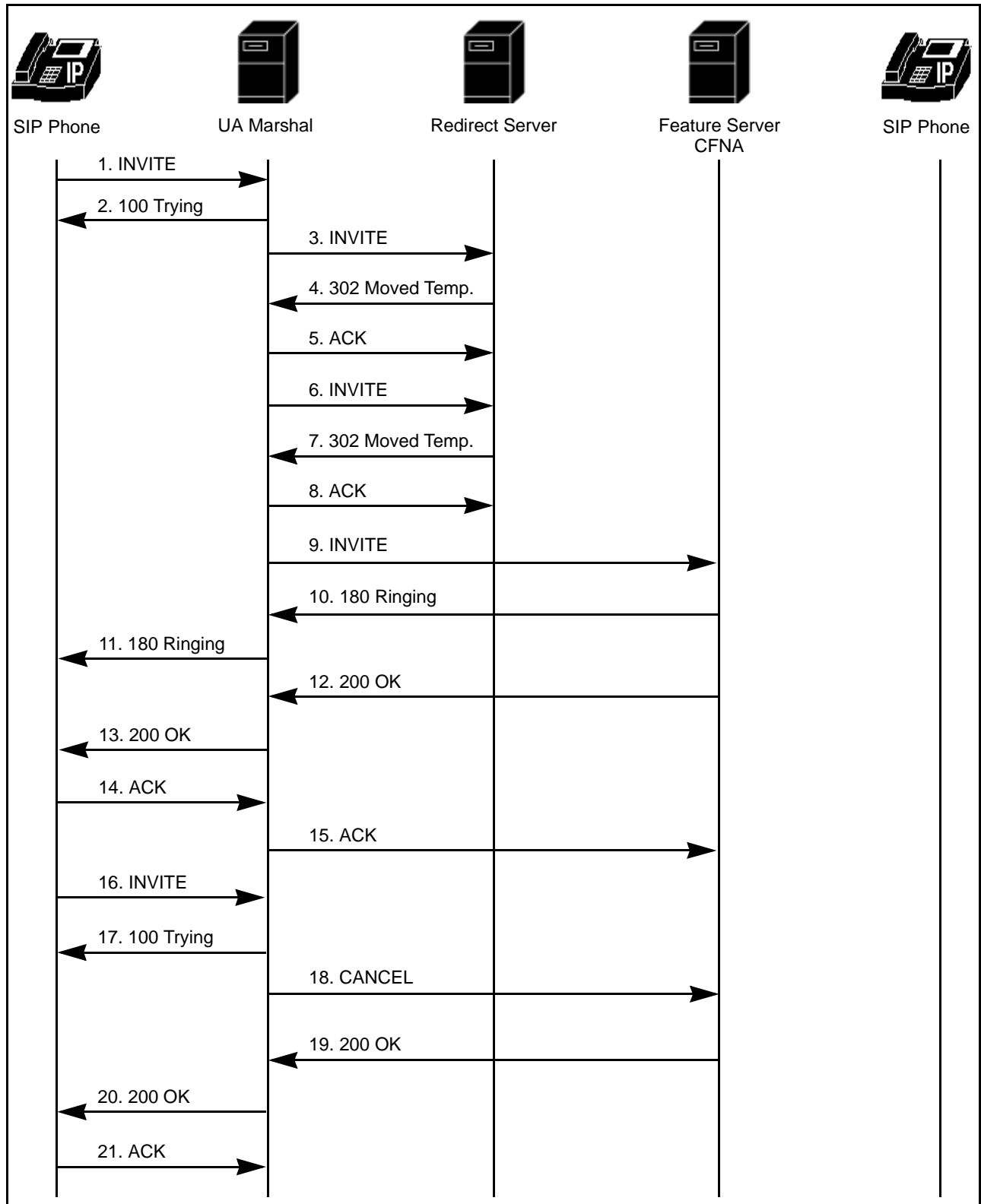
### ■ Note

To simplify the call flow diagrams, the call flows to the redirect server are not shown.

### Call Flow Diagram

Figures C-39, C-40, C-41 and C-42 show an ad hoc conference call between three users.

*Ad Hoc Conference Call Between User Agents*



**Figure C-39. User Agent to User Agent to User Agent: Ad Hoc Conference Call: Disconnect Last Party — Diagram 1**

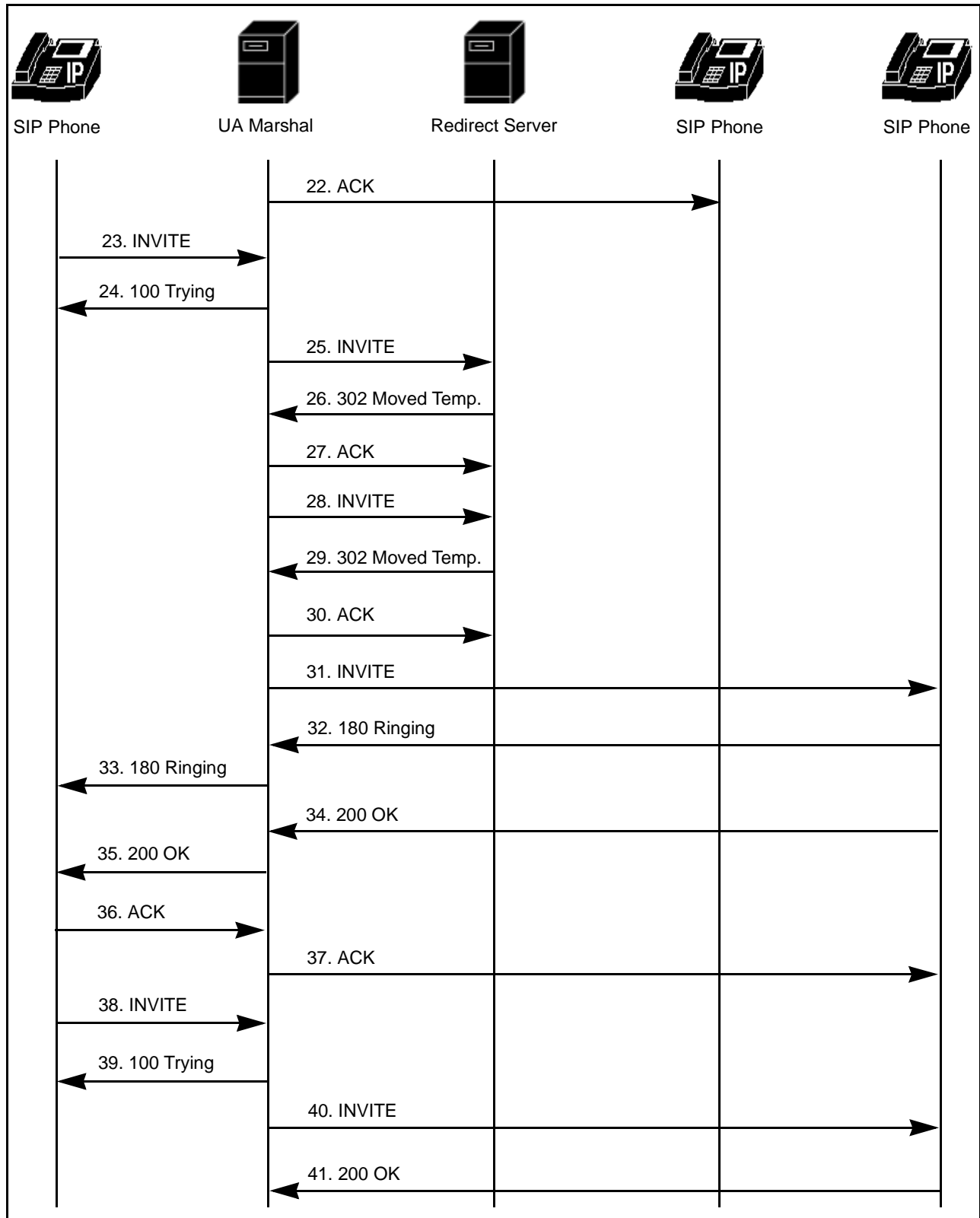
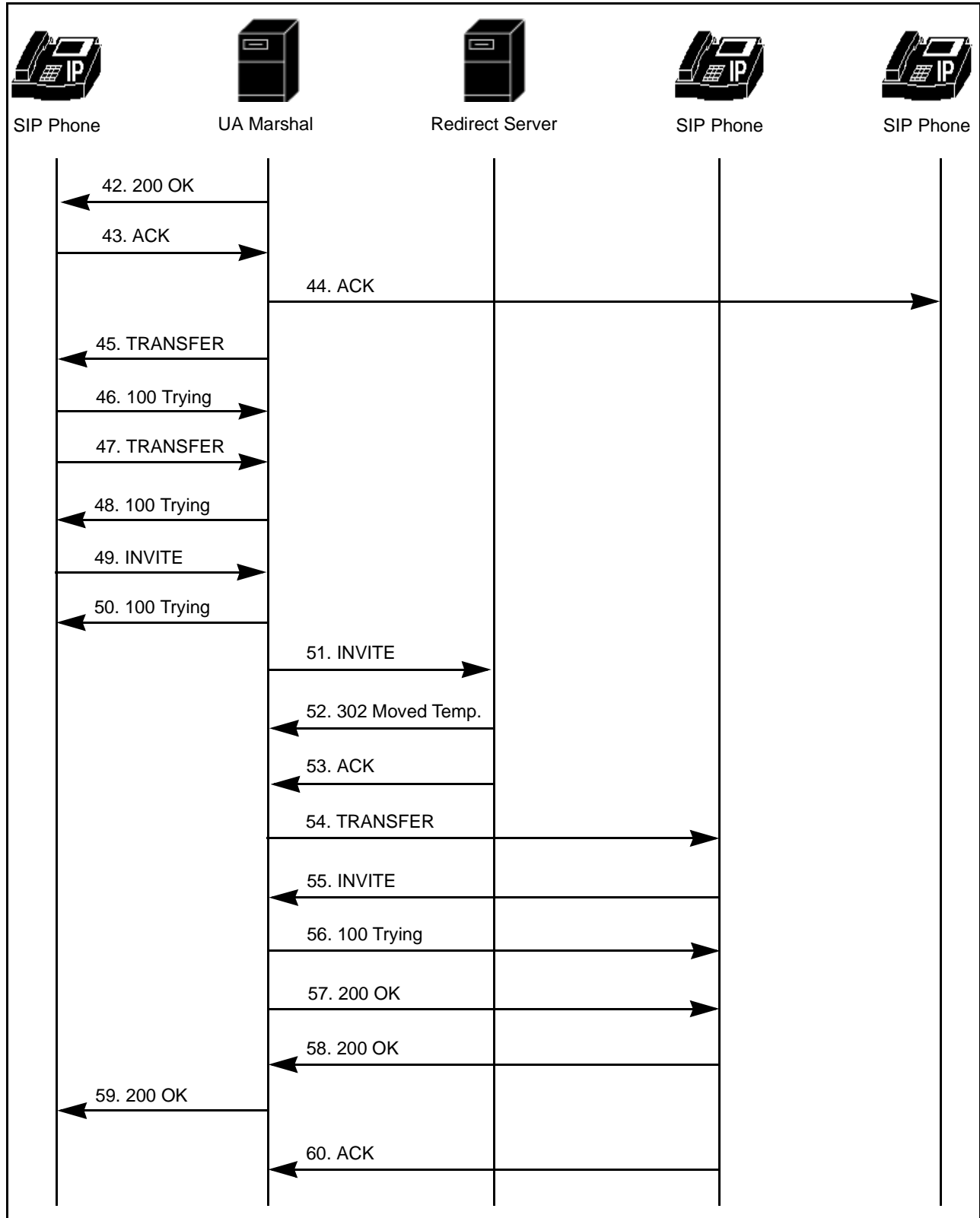


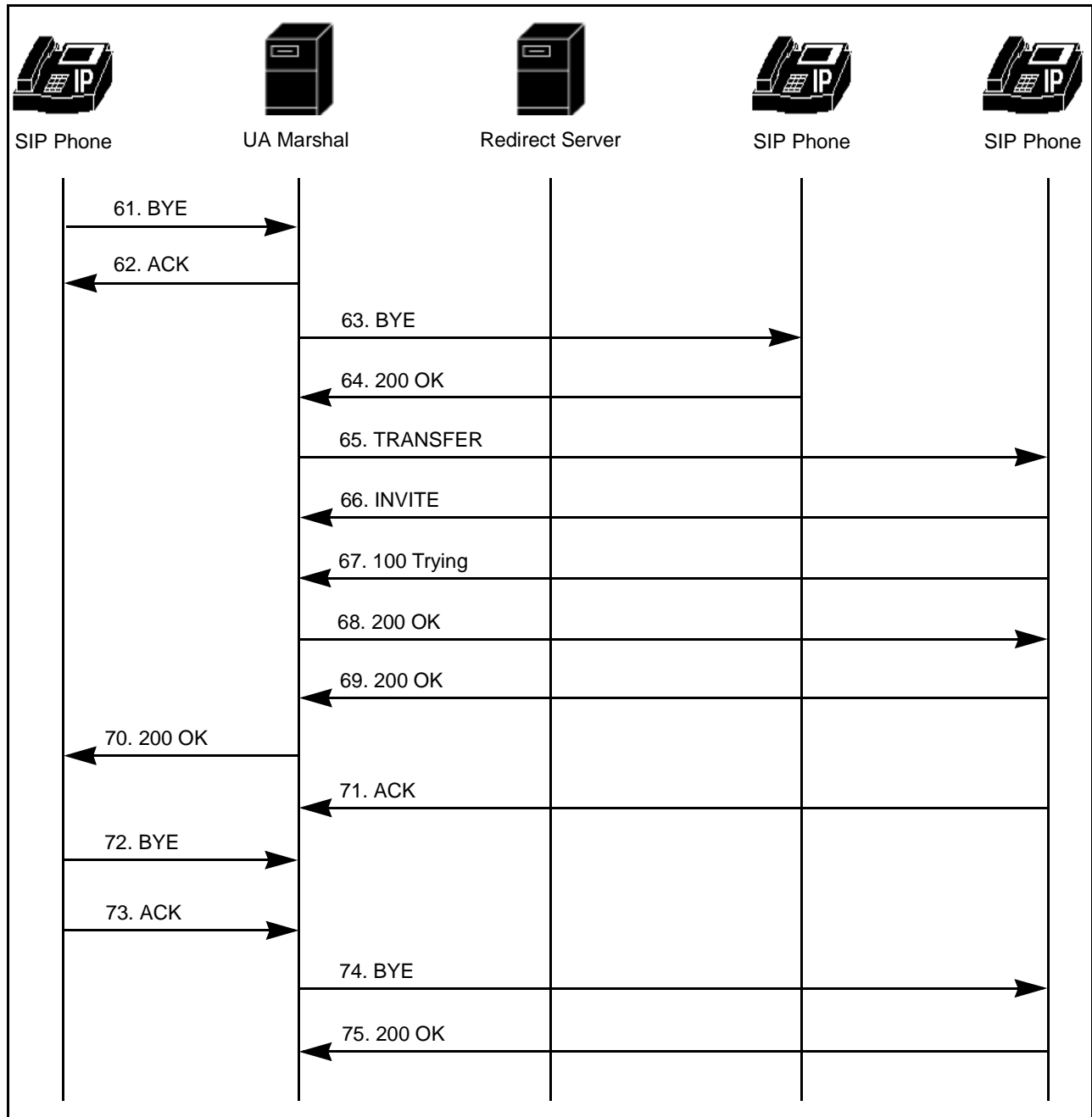
Figure C-40. User Agent to User Agent to User Agent: Ad Hoc Conference Call: Disconnect Last Party — Diagram 2

*Ad Hoc Conference Call Between User Agents*



**Figure C-41. User Agent to User Agent to User Agent: Ad Hoc Conference Call: Disconnect Last Party — Diagram 3**





**Figure C-42. User Agent to User Agent to User Agent: Ad Hoc Conference Call: Disconnect Last Party — Diagram 4**

## Ad Hoc Conference Call Between User Agents

### Call Trace

The following call trace shows an ad hoc conference call between three users.

```
-----
                          SIP Headers
-----
sip-reg:    INVITE sip:5203@192.168.46.180:5060;user=phone SIP/2.0 [192.168.46.1:5060->192.168.46.180:5060]
Header:     Via: SIP/2.0/UDP 192.168.46.1:5060
Header:     From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:     To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:     Call-ID: 64133833318102@192.168.46.1
Header:     CSeq: 1 INVITE
Header:     Subject: VovidaINVITE
Header:     Contact: <sip:5201@192.168.46.1:5060;user=phone>
Header:     Content-Type: application/sdp
Header:     Content-Length: 168
-----
                          SDP Headers
-----
Header:     v=0
Header:     o=- 1554681096 1554681096 IN IP4 192.168.46.1
Header:     s=VOVIDA Session
Header:     c=IN IP4 192.168.46.1
Header:     t=3177798649 0
Header:     m=audio 23456 RTP/AVP 0
Header:     a=rtpmap:0 PCMU/8000
Header:     a=ptime:20
-----
                          SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.46.180:5060->192.168.46.1:5060]
Header:     Via: SIP/2.0/UDP 192.168.46.1:5060
Header:     From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:     To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:     Call-ID: 64133833318102@192.168.46.1
Header:     CSeq: 1 INVITE
Header:     Content-Length: 0
-----
                          SIP Headers
-----
sip-reg:    INVITE sip:5203@192.168.46.200:5060;user=phone SIP/2.0 [192.168.46.180:5060->192.168.46.200:5060]
Header:     Via: SIP/2.0/UDP 192.168.46.180:5060;branch=1
Header:     Via: SIP/2.0/UDP 192.168.46.1:5060
Header:     From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:     To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:     Call-ID: 64133833318102@192.168.46.1
Header:     CSeq: 1 INVITE
Header:     Proxy-Authorization: Basic VovidaClassXSwitch
Header:     Subject: VovidaINVITE
Header:     Record-Route: <sip:5203@192.168.46.180:5060;maddr=192.168.46.180>
Header:     Contact: <sip:5201@192.168.46.1:5060;user=phone>
Header:     Content-Type: application/sdp
Header:     Content-Length: 168
-----
                          SDP Headers
-----
Header:     v=0
Header:     o=- 1554681096 1554681096 IN IP4 192.168.46.1
Header:     s=VOVIDA Session
Header:     c=IN IP4 192.168.46.1
Header:     t=3177798649 0
Header:     m=audio 23456 RTP/AVP 0
Header:     a=rtpmap:0 PCMU/8000
Header:     a=ptime:20
-----
                          SIP Headers
-----
```

## Ad Hoc Conference Call Between User Agents

```
sip-res: SIP/2.0 302 Moved Temporarily [192.168.46.200:5060-
>192.168.46.180:5060]
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=1
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 1 INVITE
Header: Contact: <sip:5203@192.168.46.180:5060>
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: ACK sip:5203@192.168.46.200:5060;user=phone SIP/2.0 [192.168.46.180:5060-
>192.168.46.200:5060]
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=1
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 1 ACK
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: INVITE sip:5203@192.168.46.200:5060;user=phone SIP/2.0 [192.168.46.180:5060-
>192.168.46.200:5060]
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=3
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 1 INVITE
Header: Subject: VovidaINVITE
Header: Record-Route:
<sip:5203@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5203@192.168.46.180:5060;maddr=1
92.168.46.180>
Header: Contact: <sip:5201@192.168.46.1:5060;user=phone>
Header: Content-Type: application/sdp
Header: Content-Length: 168
```

-----  
SDP Headers  
-----

```
Header: v=0
Header: o=- 1554681096 1554681096 IN IP4 192.168.46.1
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.46.1
Header: t=3177798649 0
Header: m=audio 23456 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
```

-----  
SIP Headers  
-----

```
sip-res: SIP/2.0 302 Moved Temporarily [192.168.46.200:5060->192.168.46.180:5060]
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=3
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 1 INVITE
Header: Contact: <sip:5203@192.168.46.3:5060>
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: ACK sip:5203@192.168.46.200:5060;user=phone SIP/2.0 [192.168.46.180:5060-
>192.168.46.200:5060]
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=3
```

## Ad Hoc Conference Call Between User Agents

```
Header:      From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:      To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:      Call-ID: 64133833318102@192.168.46.1
Header:      CSeq: 1 ACK
Header:      Content-Length: 0
-----
                        SIP Headers
-----
sip-req:     INVITE sip:5203@192.168.46.3:5060 SIP/2.0      [192.168.46.180:5060-
>192.168.46.3:5060]
Header:      Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:      Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:      Via: SIP/2.0/UDP 192.168.46.1:5060
Header:      From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:      To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:      Call-ID: 64133833318102@192.168.46.1
Header:      CSeq: 1 INVITE
Header:      Subject: VovidaINVITE
Header:      Record-Route:
<sip:5203@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5203@192.168.46.180:5060;maddr=1
92.168.46.180>
Header:      Contact: <sip:5201@192.168.46.1:5060;user=phone>
Header:      Content-Type: application/sdp
Header:      Content-Length: 168
-----
                        SDP Headers
-----
Header:      v=0
Header:      o=- 1554681096 1554681096 IN IP4 192.168.46.1
Header:      s=VOVIDA Session
Header:      c=IN IP4 192.168.46.1
Header:      t=3177798649 0
Header:      m=audio 23456 RTP/AVP 0
Header:      a=rtpmap:0 PCMU/8000
Header:      a=ptime:20
-----
                        SIP Headers
-----
sip-res:     SIP/2.0 180 Ringing [192.168.46.3:5060->192.168.46.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:      Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:      Via: SIP/2.0/UDP 192.168.46.1:5060
Header:      From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:      To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:      Call-ID: 64133833318102@192.168.46.1
Header:      CSeq: 1 INVITE
Header:      Content-Length: 0
-----
                        SIP Headers
-----
sip-res:     SIP/2.0 180 Ringing [192.168.46.180:5060->192.168.46.1:5060]
Header:      Via: SIP/2.0/UDP 192.168.46.1:5060
Header:      From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:      To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:      Call-ID: 64133833318102@192.168.46.1
Header:      CSeq: 1 INVITE
Header:      Content-Length: 0
-----
                        SIP Headers
-----
sip-res:     SIP/2.0 200 OK [192.168.46.3:5060->192.168.46.180:5060]
Header:      Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:      Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:      Via: SIP/2.0/UDP 192.168.46.1:5060
Header:      From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:      To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:      Call-ID: 64133833318102@192.168.46.1
Header:      CSeq: 1 INVITE
Header:      Contact: <sip:5203@192.168.46.3:5060>
```

## Ad Hoc Conference Call Between User Agents

```
Header: Record-Route:
<sip:5203@192.168.46.180:5060;maddr=192.168.46.180>, <sip:5203@192.168.46.180:5060;maddr=1
92.168.46.180>
Header: Content-Type: application/sdp
Header: Content-Length: 168
-----
SDP Headers
-----
Header: v=0
Header: o=- 1554681096 1554681096 IN IP4 192.168.46.3
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.46.3
Header: t=3177798649 0
Header: m=audio 23466 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
-----
SIP Headers
-----
sip-res: SIP/2.0 200 OK [192.168.46.180:5060->192.168.46.1:5060]
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 1 INVITE
Header: Contact: <sip:5203@192.168.46.3:5060>
Header: Record-Route:
<sip:5203@192.168.46.180:5060;maddr=192.168.46.180>, <sip:5203@192.168.46.180:5060;maddr=1
92.168.46.180>
Header: Content-Type: application/sdp
Header: Content-Length: 168
-----
SDP Headers
-----
Header: v=0
Header: o=- 1554681096 1554681096 IN IP4 192.168.46.3
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.46.3
Header: t=3177798649 0
Header: m=audio 23466 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
-----
SIP Headers
-----
sip-req: ACK sip:5203@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0
[192.168.46.1:5060->192.168.46.180:5060]
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 1 ACK
Header: Route:
<sip:5203@192.168.46.180:5060;maddr=192.168.46.180>, <sip:5203@192.168.46.3:5060>
Header: Content-Length: 0
-----
SIP Headers
-----
sip-req: ACK sip:5203@192.168.46.3:5060 SIP/2.0 [192.168.46.180:5060-
>192.168.46.3:5060]
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 1 ACK
Header: Content-Length: 0
```

## Ad Hoc Conference Call Between User Agents

```
=====
                          First call established
-----
                          SIP Headers
-----
sip-req:    INVITE sip:5203@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0
[192.168.46.1:5060->192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 2 INVITE
Header:    Contact: <sip:5201@192.168.46.1:5060;user=phone>
Header:    Route:
<sip:5203@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5203@192.168.46.3:5060>
Header:    Content-Type: application/sdp
Header:    Content-Length: 163
-----
                          SDP Headers
-----
Header:    v=0
Header:    o=- 1554681096 1554681096 IN IP4 192.168.46.1
Header:    s=VOVIDA Session
Header:    c=IN IP4 0.0.0.0
Header:    t=3177798649 0
Header:    m=audio 23456 RTP/AVP 0
Header:    a=rtpmap:0 PCMU/8000
Header:    a=ptime:20
-----
                          SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.46.180:5060->192.168.46.1:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 2 INVITE
Header:    Content-Length: 0
-----
                          SIP Headers
-----
sip-req:    INVITE sip:5203@192.168.46.3:5060 SIP/2.0    [192.168.46.180:5060-
>192.168.46.3:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 2 INVITE
Header:    Contact: <sip:5201@192.168.46.1:5060;user=phone>
Header:    Content-Type: application/sdp
Header:    Content-Length: 163
-----
                          SDP Headers
-----
Header:    v=0
Header:    o=- 1554681096 1554681096 IN IP4 192.168.46.1
Header:    s=VOVIDA Session
Header:    c=IN IP4 0.0.0.0
Header:    t=3177798649 0
Header:    m=audio 23456 RTP/AVP 0
Header:    a=rtpmap:0 PCMU/8000
Header:    a=ptime:20
-----
                          SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.46.3:5060->192.168.46.180:5060]
```

## Ad Hoc Conference Call Between User Agents

```
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 2 INVITE
Header: Contact: <sip:5203@192.168.46.3:5060;user=phone>
Header: Content-Type: application/sdp
Header: Content-Length: 168
-----
SDP Headers
-----
Header: v=0
Header: o=- 1554681096 1554681096 IN IP4 192.168.46.3
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.46.3
Header: t=3177798649 0
Header: m=audio 23456 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: aptime:20
-----
SIP Headers
-----
sip-res: SIP/2.0 200 OK [192.168.46.180:5060->192.168.46.1:5060]
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 2 INVITE
Header: Contact: <sip:5203@192.168.46.3:5060;user=phone>
Header: Content-Type: application/sdp
Header: Content-Length: 168
-----
SDP Headers
-----
Header: v=0
Header: o=- 1554681096 1554681096 IN IP4 192.168.46.3
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.46.3
Header: t=3177798649 0
Header: m=audio 23456 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: aptime:20
-----
SIP Headers
-----
sip-req: ACK sip:5203@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0
[192.168.46.1:5060->192.168.46.180:5060]
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 2 ACK
Header: Route:
<sip:5203@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5203@192.168.46.3:5060>
Header: Content-Length: 0
-----
SIP Headers
-----
sip-req: ACK sip:5203@192.168.46.3:5060 SIP/2.0 [192.168.46.180:5060-
>192.168.46.3:5060]
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 2 ACK
Header: Content-Length: 0
```

## Ad Hoc Conference Call Between User Agents

```
-----
                          SIP Headers
-----
sip-req:    INVITE sip:5202@192.168.46.180:5060;user=phone SIP/2.0 [192.168.46.1:5060-
>192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 3 INVITE
Header:    Subject: VovidaINVITE
Header:    Contact: <sip:5201@192.168.46.1:5060;user=phone>
Header:    Content-Type: application/sdp
Header:    Content-Length: 166
-----
                          SDP Headers
-----
Header:    v=0
Header:    o=- 979501686 979501686 IN IP4 192.168.46.1
Header:    s=VOVIDA Session
Header:    c=IN IP4 192.168.46.1
Header:    t=3177798665 0
Header:    m=audio 23456 RTP/AVP 0
Header:    a=rtpmap:0 PCMU/8000
Header:    a=ptime:20
-----
                          SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.46.180:5060->192.168.46.1:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 3 INVITE
Header:    Content-Length: 0
-----
                          SIP Headers
-----
sip-req:    INVITE sip:5202@192.168.46.200:5060;user=phone SIP/2.0 [192.168.46.180:5060-
>192.168.46.200:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=1
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 3 INVITE
Header:    Proxy-Authorization: Basic VovidaClassXSwitch
Header:    Subject: VovidaINVITE
Header:    Record-Route: <sip:5202@192.168.46.180:5060;maddr=192.168.46.180>
Header:    Contact: <sip:5201@192.168.46.1:5060;user=phone>
Header:    Content-Type: application/sdp
Header:    Content-Length: 166
-----
                          SDP Headers
-----
Header:    v=0
Header:    o=- 979501686 979501686 IN IP4 192.168.46.1
Header:    s=VOVIDA Session
Header:    c=IN IP4 192.168.46.1
Header:    t=3177798665 0
Header:    m=audio 23456 RTP/AVP 0
Header:    a=rtpmap:0 PCMU/8000
Header:    a=ptime:20
-----
                          SIP Headers
-----
sip-res:    SIP/2.0 302 Moved Temporarily [192.168.46.200:5060->192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=1
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
```



## Ad Hoc Conference Call Between User Agents

Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 3 INVITE  
Header: Contact: <sip:5202@192.168.46.180:5060>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: ACK sip:5202@192.168.46.200:5060;user=phone SIP/2.0 [192.168.46.180:5060->192.168.46.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=1  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 3 ACK  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: INVITE sip:5202@192.168.46.200:5060;user=phone SIP/2.0 [192.168.46.180:5060->192.168.46.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=3  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 3 INVITE  
Header: Subject: VovidaINVITE  
Header: Record-Route:  
<sip:5202@192.168.46.180:5060;maddr=192.168.46.180>, <sip:5202@192.168.46.180:5060;maddr=192.168.46.180>  
Header: Contact: <sip:5201@192.168.46.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 166

-----  
SDP Headers

Header: v=0  
Header: o=- 979501686 979501686 IN IP4 192.168.46.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.46.1  
Header: t=3177798665 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20

-----  
SIP Headers

-----  
sip-res: SIP/2.0 302 Moved Temporarily [192.168.46.200:5060->192.168.46.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=3  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 3 INVITE  
Header: Contact: <sip:5202@192.168.46.2:5060>  
Header: Content-Length: 0

-----  
SIP Headers

-----  
sip-req: ACK sip:5202@192.168.46.200:5060;user=phone SIP/2.0 [192.168.46.180:5060->192.168.46.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=3  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 3 ACK  
Header: Content-Length: 0

## Ad Hoc Conference Call Between User Agents

```
-----
                                SIP Headers
-----
sip-req:    INVITE sip:5202@192.168.46.2:5060 SIP/2.0    [192.168.46.180:5060-
>192.168.46.2:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 3 INVITE
Header:    Subject: VovidaINVITE
Header:    Record-Route:
<sip:5202@192.168.46.180:5060;maddr=192.168.46.180>, <sip:5202@192.168.46.180:5060;maddr=1
92.168.46.180>
Header:    Contact: <sip:5201@192.168.46.1:5060;user=phone>
Header:    Content-Type: application/sdp
Header:    Content-Length: 166
-----
                                SDP Headers
-----
Header:    v=0
Header:    o=- 979501686 979501686 IN IP4 192.168.46.1
Header:    s=VOVIDA Session
Header:    c=IN IP4 192.168.46.1
Header:    t=3177798665 0
Header:    m=audio 23456 RTP/AVP 0
Header:    a=rtpmap:0 PCMU/8000
Header:    a=ptime:20
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 180 Ringing [192.168.46.2:5060->192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 3 INVITE
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 180 Ringing [192.168.46.180:5060->192.168.46.1:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 3 INVITE
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.46.2:5060->192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 3 INVITE
Header:    Contact: <sip:5202@192.168.46.2:5060>
Header:    Record-Route:
<sip:5202@192.168.46.180:5060;maddr=192.168.46.180>, <sip:5202@192.168.46.180:5060;maddr=1
92.168.46.180>
Header:    Content-Type: application/sdp
Header:    Content-Length: 166
-----
```

## Ad Hoc Conference Call Between User Agents

```
-----
SDP Headers
-----
Header:          v=0
Header:          o=- 979501686 979501686 IN IP4 192.168.46.2
Header:          s=VOVIDA Session
Header:          c=IN IP4 192.168.46.2
Header:          t=3177798665 0
Header:          m=audio 23466 RTP/AVP 0
Header:          a=rtpmap:0 PCMU/8000
Header:          a=ptime:20
-----
SIP Headers
-----
sip-res:        SIP/2.0 200 OK [192.168.46.180:5060->192.168.46.1:5060]
Header:        Via: SIP/2.0/UDP 192.168.46.1:5060
Header:        From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:        To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:        Call-ID: 64133833318102@192.168.46.1
Header:        CSeq: 3 INVITE
Header:        Contact: <sip:5202@192.168.46.2:5060>
Header:        Record-Route:
<sip:5202@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5202@192.168.46.180:5060;maddr=1
92.168.46.180>
Header:        Content-Type: application/sdp
Header:        Content-Length: 166
-----
SDP Headers
-----
Header:          v=0
Header:          o=- 979501686 979501686 IN IP4 192.168.46.2
Header:          s=VOVIDA Session
Header:          c=IN IP4 192.168.46.2
Header:          t=3177798665 0
Header:          m=audio 23466 RTP/AVP 0
Header:          a=rtpmap:0 PCMU/8000
Header:          a=ptime:20
-----
SIP Headers
-----
sip-req:        ACK sip:5202@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0
[192.168.46.1:5060->192.168.46.180:5060]
Header:        Via: SIP/2.0/UDP 192.168.46.1:5060
Header:        From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:        To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:        Call-ID: 64133833318102@192.168.46.1
Header:        CSeq: 3 ACK
Header:        Route:
<sip:5202@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5202@192.168.46.2:5060>
Header:        Content-Length: 0
-----
SIP Headers
-----
sip-req:        ACK sip:5202@192.168.46.2:5060 SIP/2.0          [192.168.46.180:5060-
>192.168.46.2:5060]
Header:        Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:        Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:        Via: SIP/2.0/UDP 192.168.46.1:5060
Header:        From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:        To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:        Call-ID: 64133833318102@192.168.46.1
Header:        CSeq: 3 ACK
Header:        Content-Length: 0

=====
sip-req:        Second call completed, now place second call on hold
-----
SIP Headers
-----
```

## Ad Hoc Conference Call Between User Agents

```
sip-req: INVITE sip:5202@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0
[192.168.46.1:5060->192.168.46.180:5060]
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 4 INVITE
Header: Route:
<sip:5202@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5202@192.168.46.2:5060>
Header: Content-Type: application/sdp
Header: Content-Length: 163
```

---

### SDP Headers

```
Header: v=0
Header: o=- 1554681096 1554681096 IN IP4 192.168.46.1
Header: s=VOVIDA Session
Header: c=IN IP4 0.0.0.0
Header: t=3177798649 0
Header: m=audio 23456 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
```

---

### SIP Headers

```
sip-res: SIP/2.0 100 Trying [192.168.46.180:5060->192.168.46.1:5060]
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 4 INVITE
Header: Content-Length: 0
```

---

### SIP Headers

```
sip-req: INVITE sip:5202@192.168.46.2:5060 SIP/2.0 [192.168.46.180:5060-
>192.168.46.2:5060]
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 4 INVITE
Header: Content-Type: application/sdp
Header: Content-Length: 163
```

---

### SDP Headers

```
Header: v=0
Header: o=- 1554681096 1554681096 IN IP4 192.168.46.1
Header: s=VOVIDA Session
Header: c=IN IP4 0.0.0.0
Header: t=3177798649 0
Header: m=audio 23456 RTP/AVP 0
Header: a=rtpmap:0 PCMU/8000
Header: a=ptime:20
```

---

### SIP Headers

```
sip-res: SIP/2.0 200 OK [192.168.46.2:5060->192.168.46.180:5060]
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 4 INVITE
Header: Contact: <sip:5202@192.168.46.2:5060;user=phone>
Header: Content-Type: application/sdp
```

## Ad Hoc Conference Call Between User Agents

```
Header:                               Content-Length: 168
-----
                                SDP Headers
-----
Header:                               v=0
Header:                               o=- 1554681096 1554681096 IN IP4 192.168.46.2
Header:                               s=VOVIDA Session
Header:                               c=IN IP4 192.168.46.2
Header:                               t=3177798649 0
Header:                               m=audio 23456 RTP/AVP 0
Header:                               a=rtpmap:0 PCMU/8000
Header:                               a=ptime:20
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.46.180:5060->192.168.46.1:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 4 INVITE
Header:    Contact: <sip:5202@192.168.46.2:5060;user=phone>
Header:    Content-Type: application/sdp
Header:    Content-Length: 168
-----
                                SDP Headers
-----
Header:                               v=0
Header:                               o=- 1554681096 1554681096 IN IP4 192.168.46.2
Header:                               s=VOVIDA Session
Header:                               c=IN IP4 192.168.46.2
Header:                               t=3177798649 0
Header:                               m=audio 23456 RTP/AVP 0
Header:                               a=rtpmap:0 PCMU/8000
Header:                               a=ptime:20
-----
                                SIP Headers
-----
sip-req:    ACK sip:5202@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0
[192.168.46.1:5060->192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 4 ACK
Header:    Route:
<sip:5202@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5202@192.168.46.2:5060>
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    ACK sip:5202@192.168.46.2:5060 SIP/2.0          [192.168.46.180:5060-
>192.168.46.2:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 4 ACK
Header:    Content-Length: 0

=====
sip-req:    Initiate Conference
-----
                                SIP Headers
-----
sip-req:    TRANSFER sip:5203@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0
[192.168.46.1:5060->192.168.46.180:5060]
```

## Ad Hoc Conference Call Between User Agents

```
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Transfer-To: <sip:818883831000@192.168.46.180:5060>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 2 TRANSFER
Header: Require: cc Transfer
Header: Route:
<sip:5203@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5203@192.168.46.3:5060>
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res: SIP/2.0 100 Trying [192.168.46.180:5060->192.168.46.1:5060]
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 2 TRANSFER
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req: TRANSFER sip:5202@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0
[192.168.46.1:5060->192.168.46.180:5060]
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header: Transfer-To: <sip:818883831000@192.168.46.180:5060>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 4 TRANSFER
Header: Require: cc Transfer
Header: Route:
<sip:5202@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5202@192.168.46.2:5060>
Header: Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res: SIP/2.0 100 Trying [192.168.46.180:5060->192.168.46.1:5060]
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 4 TRANSFER
Header: Content-Length: 0
```

```
=====  
sip-req: Invite self into conference room  
-----
```

-----  
SIP Headers  
-----

```
sip-req: INVITE sip:818883831000@192.168.46.180:5060 SIP/2.0 [192.168.46.1:5060-
>192.168.46.180:5060]
Header: Via: SIP/2.0/UDP 192.168.46.1:5060
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header: To: 818883831000<sip:818883831000@192.168.46.180:5060>
Header: Call-ID: 64133833318102@192.168.46.1
Header: CSeq: 3 INVITE
Header: Subject: VovidaINVITE
Header: Contact: <sip:5201@192.168.46.1:5060;user=phone>
Header: Content-Type: application/sdp
Header: Content-Length: 166
```

-----  
SDP Headers  
-----

```
Header: v=0
Header: o=- 979501686 979501686 IN IP4 192.168.46.1
Header: s=VOVIDA Session
Header: c=IN IP4 192.168.46.1
```

## Ad Hoc Conference Call Between User Agents

Header: t=3177798665 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20

-----  
SIP Headers

-----  
sip-res: SIP/2.0 100 Trying [192.168.46.180:5060->192.168.46.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 818883831000<sip:818883831000@192.168.46.180:5060>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 3 INVITE  
Header: Content-Length: 0  
-----

SIP Headers

-----  
sip-req: INVITE sip:818883831000@192.168.46.200:5060;user=phone SIP/2.0  
[192.168.46.180:5060->192.168.46.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 818883831000<sip:818883831000@192.168.46.180:5060>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 3 INVITE  
Header: Proxy-Authorization: Basic VovidaClassXSwitch  
Header: Subject: VovidaINVITE  
Header: Record-Route:  
<sip:818883831000@192.168.46.180:5060;maddr=192.168.46.180>  
Header: Contact: <sip:5201@192.168.46.1:5060;user=phone>  
Header: Content-Type: application/sdp  
Header: Content-Length: 166  
-----

SDP Headers

-----  
Header: v=0  
Header: o=- 979501686 979501686 IN IP4 192.168.46.1  
Header: s=VOVIDA Session  
Header: c=IN IP4 192.168.46.1  
Header: t=3177798665 0  
Header: m=audio 23456 RTP/AVP 0  
Header: a=rtpmap:0 PCMU/8000  
Header: a=ptime:20  
-----

SIP Headers

-----  
sip-res: SIP/2.0 302 Moved Temporarily [192.168.46.200:5060->192.168.46.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=1  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 818883831000<sip:818883831000@192.168.46.180:5060>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 3 INVITE  
Header: Contact: <sip:818883831000@192.168.46.180:5062;user=phone>  
Header: Content-Length: 0  
-----

SIP Headers

-----  
sip-req: ACK sip:818883831000@192.168.46.200:5060;user=phone SIP/2.0  
[192.168.46.180:5060->192.168.46.200:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=1  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 818883831000<sip:818883831000@192.168.46.180:5060>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 3 ACK  
Header: Content-Length: 0  
-----

=====  
sip-req: Add second party to conference room



## Ad Hoc Conference Call Between User Agents

```
-----
                                SIP Headers
-----
sip-req:    TRANSFER sip:5203@192.168.46.3:5060 SIP/2.0 [192.168.46.180:5060-
>192.168.46.3:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:    Transfer-To: <sip:818883831000@192.168.46.180:5060>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 2 TRANSFER
Header:    Require: cc Transfer
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-req:    INVITE sip:818883831000@192.168.46.180:5060 SIP/2.0 [192.168.46.3:5060-
>192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.3:5060
Header:    From: apex<sip:5203@192.168.46.3:5060>
Header:    To: <sip:818883831000@192.168.46.180:5060>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 1 INVITE
Header:    Subject: VovidaINVITE
Header:    Contact: <sip:5203@192.168.46.3:5060;user=phone>
Header:    Content-Type: application/sdp
Header:    Content-Length: 168
-----
                                SDP Headers
-----
Header:    v=0
Header:    o=- 1304377875 1304377875 IN IP4 192.168.46.3
Header:    s=VOVIDA Session
Header:    c=IN IP4 192.168.46.3
Header:    t=3177798671 0
Header:    m=audio 23456 RTP/AVP 0
Header:    a=rtpmap:0 PCMU/8000
Header:    a=ptime:20
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 100 Trying [192.168.46.180:5060->192.168.46.3:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 2 TRANSFER
Header:    Content-Length: 0
-----
                                SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.46.180:5060->192.168.46.3:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.3:5060
Header:    From: apex<sip:5203@192.168.46.3:5060>
Header:    To: <sip:818883831000@192.168.46.180:5060>;tag=49FD04D0-25BA
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 1 INVITE
Header:    Contact: <sip:818883831000@192.168.5.169:5060;user=phone>
Header:    Date: Thu, 09 Nov 2000 18:40:48 PST
Header:    Record-Route:
<sip:818883831000@192.168.46.180:5060;maddr=192.168.46.180>,<sip:818883831000@192.168.46.
180:5060;maddr=192.168.46.180>
Header:    Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled
Header:    Content-Type: application/sdp
Header:    Content-Length: 136
-----
                                SDP Headers
-----
```



## Ad Hoc Conference Call Between User Agents

```
-----  
Header: v=0  
Header: o=CiscoSystemsSIP-GW-UserAgent 1397 1625 IN IP4 192.168.5.169  
Header: s=SIP Call  
Header: c=IN IP4 192.168.5.169  
Header: t=0 0  
Header: m=audio 20246 RTP/AVP 0  
-----
```

### SIP Headers

```
-----  
sip-res: SIP/2.0 200 OK [192.168.46.3:5060->192.168.46.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 2 TRANSFER  
Header: Content-Length: 0  
-----
```

### SIP Headers

```
-----  
sip-res: SIP/2.0 200 OK [192.168.46.180:5060->192.168.46.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5203<sip:5203@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 2 TRANSFER  
Header: Content-Length: 0  
-----
```

### SIP Headers

```
-----  
sip-req: ACK sip:818883831000@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0  
[192.168.46.3:5060->192.168.46.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.3:5060  
Header: From: apex<sip:5203@192.168.46.3:5060>  
Header: To: <sip:818883831000@192.168.46.180:5060>;tag=49FD04D0-25BA  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 1 ACK  
Header: Route:  
<sip:818883831000@192.168.46.180:5062;maddr=192.168.46.180>,<sip:818883831000@192.168.5.1  
69:5060>  
Header: Content-Length: 0  
-----
```

### SIP Headers

```
-----  
sip-req: BYE sip:5202@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0  
[192.168.46.1:5060->192.168.46.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 5 BYE  
Header: Route:  
<sip:5202@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5202@192.168.46.2:5060>  
Header: Content-Length: 0  
-----
```

### SIP Headers

```
-----  
sip-req: ACK sip:5202@192.168.46.180:5060;user=phone SIP/2.0 [192.168.46.1:5060-  
>192.168.46.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 4 ACK  
Header: Content-Length: 0  
-----
```

### SIP Headers

## Ad Hoc Conference Call Between User Agents

```
sip-req:    BYE sip:5203@192.168.46.3:5060 SIP/2.0      [192.168.46.180:5060-
>192.168.46.3:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 4 BYE
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-res:    SIP/2.0 200 OK [192.168.46.3:5060->192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 4 BYE
Header:    Content-Length: 0
```

```
=====  
sip-req:    Add third party to conference room  
-----
```

SIP Headers  
-----

```
sip-req:    TRANSFER sip:5202@192.168.46.2:5060 SIP/2.0  [192.168.46.180:5060-
>192.168.46.2:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Transfer-To: <sip:818883831000@192.168.46.180:5060>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 4 TRANSFER
Header:    Require: cc Transfer
Header:    Content-Length: 0
```

-----  
SIP Headers  
-----

```
sip-req:    INVITE sip:818883831000@192.168.46.180:5060 SIP/2.0 [192.168.46.2:5060-
>192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.2:5060
Header:    From: grouse<sip:5202@192.168.46.2:5060>
Header:    To: <sip:818883831000@192.168.46.180:5060>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 1 INVITE
Header:    Subject: VovidaINVITE
Header:    Contact: <sip:5202@192.168.46.2:5060;user=phone>
Header:    Content-Type: application/sdp
Header:    Content-Length: 166
```

-----  
SDP Headers  
-----

```
Header:    v=0
Header:    o=- 695905101 695905101 IN IP4 192.168.46.2
Header:    s=VOVIDA Session
Header:    c=IN IP4 192.168.46.2
Header:    t=3177798673 0
Header:    m=audio 23456 RTP/AVP 0
Header:    a=rtpmap:0 PCMU/8000
Header:    a=ptime:20
```

-----  
SIP Headers  
-----

```
sip-res:    SIP/2.0 100 Trying [192.168.46.2:5060->192.168.46.180:5060]
```

## Ad Hoc Conference Call Between User Agents

Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 4 TRANSFER  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.46.180:5060->192.168.46.2:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.2:5060  
Header: From: grouse<sip:5202@192.168.46.2:5060>  
Header: To: <sip:818883831000@192.168.46.180:5060>;tag=49FD06B8-4B7  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 1 INVITE  
Header: Contact: <sip:818883831000@192.168.5.169:5060;user=phone>  
Header: Date: Thu, 09 Nov 2000 18:40:49 PST  
Header: Record-Route:  
<sip:818883831000@192.168.46.180:5062;maddr=192.168.46.180>,<sip:818883831000@192.168.46.180:5060;maddr=192.168.46.180>  
Header: Server: Cisco VoIP Gateway/ IOS 12.x/ SIP enabled  
Header: Content-Type: application/sdp  
Header: Content-Length: 136

---

### SDP Headers

-----  
Header: v=0  
Header: o=CiscoSystemsSIP-GW-UserAgent 4400 7810 IN IP4 192.168.5.169  
Header: s=SIP Call  
Header: c=IN IP4 192.168.5.169  
Header: t=0 0  
Header: m=audio 20390 RTP/AVP 0

---

### SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.46.2:5060->192.168.46.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4  
Header: Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 4 TRANSFER  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-res: SIP/2.0 200 OK [192.168.46.180:5060->192.168.46.1:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.1:5060  
Header: From: seymour<sip:5201@192.168.46.1:5060;user=phone>  
Header: To: 5202<sip:5202@192.168.46.180:5060;user=phone>  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 4 TRANSFER  
Header: Content-Length: 0

---

### SIP Headers

-----  
sip-req: ACK sip:818883831000@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0  
[192.168.46.2:5060->192.168.46.180:5060]  
Header: Via: SIP/2.0/UDP 192.168.46.2:5060  
Header: From: grouse<sip:5202@192.168.46.2:5060>  
Header: To: <sip:818883831000@192.168.46.180:5060>;tag=49FD06B8-4B7  
Header: Call-ID: 64133833318102@192.168.46.1  
Header: CSeq: 1 ACK  
Header: Route:  
<sip:818883831000@192.168.46.180:5062;maddr=192.168.46.180>,<sip:818883831000@192.168.5.169:5060>  
Header: Content-Length: 0

## Ad Hoc Conference Call Between User Agents

```
-----
                        SIP Headers
-----
sip-req:    BYE sip:5203@192.168.46.180:5060;maddr=192.168.46.180 SIP/2.0
[192.168.46.1:5060->192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 4 BYE
Header:    Route:
<sip:5203@192.168.46.180:5060;maddr=192.168.46.180>,<sip:5203@192.168.46.3:5060>
Header:    Content-Length: 0
-----
                        SIP Headers
-----
sip-req:    ACK sip:5203@192.168.46.180:5060;user=phone SIP/2.0 [192.168.46.1:5060-
>192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5203<sip:5203@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 2 ACK
Header:    Content-Length: 0
-----
                        SIP Headers
-----
sip-req:    BYE sip:5202@192.168.46.2:5060 SIP/2.0          [192.168.46.180:5060-
>192.168.46.2:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 5 BYE
Header:    Content-Length: 0
-----
                        SIP Headers
-----
sip-res:    SIP/2.0 200 OK [192.168.46.2:5060->192.168.46.180:5060]
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=4
Header:    Via: SIP/2.0/UDP 192.168.46.180:5060;branch=2
Header:    Via: SIP/2.0/UDP 192.168.46.1:5060
Header:    From: seymour<sip:5201@192.168.46.1:5060;user=phone>
Header:    To: 5202<sip:5202@192.168.46.180:5060;user=phone>
Header:    Call-ID: 64133833318102@192.168.46.1
Header:    CSeq: 5 BYE
Header:    Content-Length: 0
-----
```

---

# Index

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## Symbols

\*69  
    call flow C-52

## Numerics

1xx and 2xx B-3  
3xx Responses B-3  
4xx Responses B-3  
5xx Responses B-4  
6xx Responses B-4  
900 # Admin Block 1-20  
900 # User Block 1-23

## A

Access Level 1-4  
Access list C-4  
ACK B-2  
adding users, see New  
Adhoc conferencing A-6  
Administrator 1-4  
alias 1-29  
Aliases 1-32  
Authentication  
    access list C-4  
    digest C-6  
Authentication Type 1-21

## B

BYE B-2

## C

Call Block Enabled 1-20  
Call Block Group 1-20  
Call blocking 1-32, A-3  
Call flow  
    \*69 C-52  
    ad hoc conference call C-149  
    analog phone C-15  
    blind transfer C-114  
    busy C-24  
    call blocking C-47  
    call return C-52  
    call screening C-41  
    call setup and disconnect C-8  
    call waiting C-70  
    forward all calls C-32  
    JTAPI C-131  
    registration C-3  
    transfer C-96  
    voice mail C-85  
Call return A-4  
Call Return Enabled 1-21

Call Return Group 1-21  
Call Screen Enabled 1-20  
Call Screen Group 1-20  
Call screening 1-33, A-4  
Call waiting A-5  
Caller ID A-5  
Caller ID Block Enabled 1-21  
Caller ID blocking 1-33, A-3  
Caller ID Blocking Set 1-23  
Caller ID Group 1-21  
Calling name delivery A-5  
Calling number delivery A-5  
CANCEL B-2  
Conferencing  
    adhoc A-6  
    meet-me A-4  
Core system features A-3–A-4  
    call blocking A-3  
    call return A-4  
    call screening A-4  
    caller ID blocking A-3  
    forward all calls A-3  
    forward no answer and forward busy A-3  
    meet-me conferencing A-4  
    voice mail A-4

## D

Data fields  
    900 # Admin Block 1-20  
    900 # User Block 1-23  
    Authentication Type 1-21  
    Call Block Enabled 1-20  
    Call Block Group 1-20  
    Call Return Enabled 1-21  
    Call Return Group 1-21  
    Call Screen Enabled 1-20  
    Call Screen Group 1-20  
    Caller ID Block Enabled 1-21  
    Caller ID Blocking Set 1-23  
    Caller ID Group 1-21  
    Failure Case 1-20  
    Forward All Calls Enabled 1-19  
    Forward All Group 1-19  
    Forward All Set 1-22  
    Forward All To 1-22  
    Forward Busy Set 1-22  
    Forward Busy To 1-22  
    Forward Busy/No Ans. Enabled 1-19  
    Forward Busy/No Ans. Group 1-20  
    Forward No Ans To 1-22  
    Forward No Ans. Set 1-22  
    IP 1-19  
    JTAPI Enabled 1-20  
    JTAPI Group 1-20  
    JTAPI Set 1-23

## Index (Continued)

---

- Long Distance Admin Block 1-20
- Long Distance User Block 1-22
- Marshal 1-19
- Name 1-19
- Password 1-21
- Static Reg Enabled 1-21
- Terminating Host 1-21
- Terminating Port 1-21
- User Group 1-19

- Delete 1-8
  - procedure for deleting users 1-27
- Digest C-6

### E

- Edit 1-8
  - multiple users 1-28
  - users 1-28
- Edit user
  - Aliases 1-32
  - call blocking 1-32
  - call screening 1-33
  - caller ID blocking 1-33
  - forward all calls 1-32
  - forward no answer busy 1-33
  - JTAPI 1-32
- Edit User Screen 1-31

### F

- Failure Case 1-20
- Features
  - core system features A-3–A-4
    - call blocking A-3
    - call return A-4
    - call screening A-4
    - caller ID blocking A-3
    - forward all calls A-3
    - forward no answer and forward busy A-3
  - meet-me Conferencing A-4
  - voice mail A-4
  - set-based features A-5–A-6
- Forward all calls 1-32, A-3
- Forward All Calls Enabled 1-19
- Forward All Group 1-19
- Forward All Set 1-22
- Forward All To 1-22
- Forward Busy Set 1-22
- Forward Busy To 1-22
- Forward Busy/No Ans. Enabled 1-19
- Forward Busy/No Ans. Group 1-20
- Forward No Ans To 1-22
- Forward No Ans. Set 1-22
- Forward no answer and forward busy A-3
- Forward no answer busy 1-33

### H

- Hosts & processes 2-6

### I

- INFO B-2
- INVITE B-2
- IP 1-19

### J

- Java plug-in 1-2
- JTAPI 1-32
- JTAPI Enabled 1-20
- JTAPI Group 1-20
- JTAPI Set 1-23

### L

- Load all users 1-18, 1-24–1-27
  - screen capture 1-25
- Logging In 1-3–1-4
  - Access Level 1-4
  - Administrator 1-4
  - Login ID 1-4
  - Password 1-4
  - screen 1-3
  - Technician 1-4
- Login ID 1-4
- Long Distance Admin Block 1-20
- Long Distance User Block 1-22

### M

- Marshal 1-19
- Meet-me conferencing A-4
- Most recent SNMP trap 2-6

### N

- Name 1-19
- Network management
  - statistics 2-6
- New 1-8
  - adding new users 1-10

### O

- option boxes 1-7, 1-8
- OPTIONS B-2

### P

- Password 1-4, 1-21
- Process controller 2-6
- Provisioning
  - accessing 1-2

## Index (Continued)

---

### R

REGISTER B-2  
Registration C-3  
Right mouse click menu 1-16

### S

Screen  
  Edit User  
    Administrator view 1-11  
    User data view 1-31  
  Login 1-3  
  SNMP messages 2-5  
  User Configuration 1-5  
Set-based features A-5–A-6  
  adhoc conferencing A-6  
  call waiting A-5  
  caller ID A-5  
  calling name delivery A-5  
  calling number delivery A-5  
  transfer A-5  
Show admin data 1-7  
Show aliases 1-7  
Show user data 1-7  
SIP  
  Request Messages B-2  
    ACK B-2  
    BYE B-2  
    CANCEL B-2  
    INFO B-2  
    INVITE B-2  
    OPTIONS B-2  
    REGISTER B-2  
  Response Messages B-3–B-4  
    1xx and 2xx B-3  
    3xx Responses B-3  
    4xx Responses B-3  
    5xx Responses B-4  
    6xx Responses B-4  
SNMP  
  GUI 2-5  
    hosts & processes 2-6  
    most recent SNMP trap 2-6  
    process controller 2-6  
    SNMP traps 2-6  
  version 2-2  
SNMP traps 2-6  
Static Reg Enabled 1-21  
Statistics 2-6

### T

Technician 1-4  
Terminating Host 1-21  
Terminating Port 1-21  
Transfer A-5

### U

User  
  Configuration screen  
    buttons 1-6  
    data field descriptions 1-19  
    editing: show alias 1-29  
    features: user controlled 1-30  
  User Configuration  
    Admin data fields  
      900 # Admin Block 1-20  
      Authentication Type 1-21  
      Call Block Enabled 1-20  
      Call Block Group 1-20  
      Call Return Enabled 1-21  
      Call Return Group 1-21  
      Call Screen Enabled 1-20  
      Call Screen Group 1-20  
      Caller ID Block Enabled 1-21  
      Caller ID Group 1-21  
      Failure Case 1-20  
      Forward All Calls Enabled 1-19  
      Forward All Group 1-19  
      Forward Busy/No Ans. Enabled 1-19  
      Forward Busy/No Ans. Group 1-20  
      JTAPI Enabled 1-20  
      JTAPI Group 1-20  
      Long Distance Admin Block 1-20  
      Password 1-21  
      Static Reg Enabled 1-21  
      Terminating Host 1-21  
      Terminating Port 1-21  
    Default data fields  
      IP 1-19  
      Marshal 1-19  
      Name 1-19  
      User Group 1-19  
    Load all users 1-24–1-27  
    User data fields  
      900 # User Block 1-23  
      Caller ID Blocking Set 1-23  
      Forward All Set 1-22  
      Forward All To 1-22  
      Forward Busy Set 1-22  
      Forward Busy To 1-22  
      Forward No Ans To 1-22  
      Forward No Ans. Set 1-22  
      JTAPI Set 1-23  
      Long Distance User Block 1-22  
      viewing users individually 1-16–1-18  
    User Configuration screen 1-5–1-7  
    Right mouse click menu 1-16  
    Show admin data 1-7  
    Show aliases 1-7  
    Show user data 1-7  
  User Group 1-19

## Index (Continued)

---

### Users

- adding new users 1-10
- deleting 1-27
- edit multiple users 1-28
- finding users 1-26
- Load all users 1-24–1-27
- procedure for editing 1-28
- viewing individually 1-16–1-18

### V

- View 1-8
- viewing users 1-16
- Voice mail A-4