CHAPTER 1

Overview of Cisco Unified Communications Manager Business Edition 3000

This chapter contains information on the following topics:

Benefits of Deploying Cisco Unified Communications Manager Business Edition 3000

Cisco Unified Communications Manager Business Edition 3000, a system under the Cisco Unified Communications family of products, provides an IP telephony solution that enables:

- Easy setup of deployments
- Easy provisioning of users, phones, lines, and phone features
- Easy monitoring and troubleshooting
- Easy maintenance of your system (simplified backups, simplified restores, and so on)

The Cisco Unified Communications Manager Business Edition 3000 software is preinstalled on the server that is supported with your system so that you do not have to perform a software installation to get your server up and running. Deployment of the Cisco Unified Communications Manager Business Edition 3000 server, phones, and the gateway across an IP network provides a distributed, virtual telephony network. Quality of service is maintained across constricted WAN links, Internet, or VPN connections.

Your Cisco Unified Communications Manager Business Edition 3000 system is designed to support up to 300 users and 400 phones. Supplementary and enhanced services such as hold, transfer, forward, conference, multiple-line appearances, speed dials, last-number redial, and other features extend to the phones.

Web-browser interfaces allow configuration of the system. These interfaces also provide access to online help.

Components of the Cisco Unified Communications Manager Business Edition 3000 System

Your Cisco Unified Communications Manager Business Edition 3000 system consists of the following components:

- The Cisco Unified Communications Manager Business Edition 3000 Server, page 1-3
- USB Support, page 1-4
- Cisco-Provided .xls Data Configuration File, page 1-5
- Phones, page 1-6
- Attendant Console, page 1-7
- Video Support, page 1-8
- Voicemail, page 1-8
- Auto Attendant, page 1-9
- Gateway, page 1-10
- DHCP Usage for Acquiring IP Addresses, page 1-31
- DNS and Hostname Resolution, page 1-32
- SFTP Server, page 1-33

The Cisco Unified Communications Manager Business Edition 3000 Server

Cisco Unified Communications Manager Business Edition 3000 is installed for you on a standalone MCS7890-C1 with 4GB of RAM. When you plug in the server, the Cisco Unified Communications Manager Business Edition 3000 software is installed and ready for use. Cisco Unified Communications Manager, an internal component of the Cisco Unified Communications Manager Business Edition 3000 software that provides call processing for your system, resides on the Cisco Unified Communications Manager Business Edition 3000 server. Cisco Unity Connection, an internal component of the Cisco Unified Communications Manager Business Edition 3000 software that provides voicemail support for your system, also resides on the Cisco Unified Communications Manager Business Edition 3000 server. The Cisco Unified Communications Manager Business Edition 3000 server also contains the database where your configuration records are stored. Internal services that are part of the Cisco Unified Communications Manager Business Edition 3000 software allow you to troubleshoot, monitor, and perform maintenance tasks, such as backups, upgrades, and so on.

Tip The Cisco Unified Communications Manager Business Edition 3000 server must use a static IP address.

Because you use web-browsable graphical user interfaces (GUIs) for configuration, monitoring, and troubleshooting, you need not connect a keyboard and mouse to the Cisco Unified Communications Manager Business Edition 3000 server. The following graphical user interfaces (GUIs) exist on the server so that you can perform tasks to support your system:

Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard

The Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard guides you through the deployment steps that are necessary to complete an initial configuration. From this wizard, you can upload a Cisco-provided .xls data configuration file that contains data that you can use to configure your system, or you can manually configure settings by moving throughout the wizard. After you log in to the Cisco Unified Communications Manager Business Edition 3000 server for the first time, the First Time Setup guides you through the set up of the application through prompts that display in the main content area. You complete the set up by clicking the appropriate responses. The Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard supports forward and back capability through Back and Next buttons that display on every page of the wizard.

Tip If you click Next throughout the wizard without updating any of the settings, your system uses the default settings.

Cisco Unified Communications Manager Business Edition 3000 Administrative Interface

After you complete the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard, the next time that you log in to the server, you can access the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface. The Cisco Unified Communications Manager Business Edition 3000 Administrative Interface allows you to perform the tasks that are described in this chapter. For example, in this GUI, you can monitor and troubleshoot the system, add, edit, delete configuration data, such as phones, users, sites, and so on, and perform maintenance tasks, such as backups, restorations, upgrades, add and view licenses, and so on.

The Cisco Unified Communications Manager Business Edition 3000 Administrative Interface uses a three-section layout, which consists of a top-level header, navigation menus that display on the left of the page that expand and collapse to display individual menu options, and a content section that displays on the right of the page where you can view, add, update, and delete data.

When you click an arrow next to a navigation menu, the navigation section displays the items that the belong to the navigation menu. To display the contents of an item in the navigation menu, click the item. The contents of that item display on the right side of the GUI.

Cisco Unified Communications Manager Business Edition 3000 User Preferences Interface

When users that exist in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface log in to the Cisco Unified Communications Manager Business Edition 3000 User Preferences Interface, a web page displays where the user can manage user preferences for phone features; for example, the user can update Reach Me Anywhere, call forwarding, speed dials, the phone PIN for Cisco Extension Mobility, and the password for the Cisco Unified Communications Manager Business Edition 3000 User Preferences Interface. In addition, the user can use Cisco Web Dialer to place a call to an extension in the corporate directory.

Users can manage their user preferences settings for phone features by selecting check boxes and entering the appropriate information in the provided fields. Each user accesses his own Cisco Unified Communications Manager Business Edition 3000 User Preferences Interface page, and this page is not shared by users.

Most settings that display in the Cisco Unified Communications Manager Business Edition 3000 User Preferences Interface are dynamic; the settings display only if the user is allowed to use the feature (as configured by you, the system administrator). For example, if you do not enable Reach Me Anywhere in the usage profile that is assigned to the user, the user cannot see the Reach Me Anywhere setting in the Cisco Unified Communications Manager Business Edition 3000 User Preferences Interface.

USB Support

Cisco Unified Communications Manager Business Edition 3000 gives you the option of using USB keys or a USB hard disk for the following functionality:

- Performing a reimage of the server through a USB DVD drive—Perform a reimage of the server only when your technical support team advises that you do so. You can copy the answer file, platformConfig.xml (www.cisco.com), to the USB DVD drive key and perform a reimage.

Tip For MCS7890-C1, the default Answer File (platformConfig.xml) required for install configuration is packaged within the DVD.

- Updating the network parameters—You can copy the configure.xml file to a USB key to update the network parameters. The temporary network address allows you to log in to the First Time Setup Wizard through a browser. This is also an alternative method to connect a laptop to the server using a cable. You must update the network parameters before you can access the GUIs.

Note After you run the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard, designate a single USB DVD drive key for this function.

- Uploading a Cisco-provided country pack—You copy the Cisco-provided country pack to the USB key and then install the country pack through the Country/Locale page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard.
- Uploading the Cisco-provided.xls data configuration file—You can copy the Cisco-provided .xls data configuration file to the USB key and then upload the spreadsheet to the system through the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard.

- Backing Up and Restoring Your Data—You may store your backup tar file to a USB hard disk, and if you must restore you data for any reason, you can access the backup tar file on the USB hard disk to restore the data through the Restore page in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface.
- Uploading an audio source file for Music On Hold—You can copy the .wav file that you want to use for music on hold to the USB key; after you insert the USB key in the Cisco Unified Communications Manager Business Edition 3000 server, you can upload the file through the Music On Hold page in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface.
- Uploading Cisco User Connect licenses—Cisco User Connect license allow you to track the users and phones that are in your system. You may use a USB key to upload licenses.

Note Some operating systems do not allow you to copy an entire file that is larger than 4 GB to the USB key. The system silently copies only 4 GB of the file to the USB key. Hence, Cisco recommends that you use USB keys that are formatted as FAT32 in the Cisco Unified Communications Manager Business Edition 3000.

Linux platform supports USB keys formatted with FAT32.

- Exporting your configured data—By using the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface, you can export all of your configured data to a storage device that is connected to a USB port or to a SFTP server. You may store the exported configuration to a USB key or USB hard disk.
- Using the Cisco Diagnostic Tool—The Cisco Diagnostic Tool allows you to diagnose your system if you cannot access the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface. You copy the diagnose.xml file that is used with the Cisco Diagnostic Tool to a USB key.

Note Make sure that you designate a USB key just for this purpose. Do not use the USB key for other functions.

For More Information

- What is a country pack, and where do I install it?, page 2-7
- Working with the Cisco-Provided .xls Data Configuration File, page 3-1
- Using the Cisco Network Configuration USB Key, page 6-4
- Troubleshooting When You Cannot Access the Graphical User Interfaces, page 46-44

Cisco-Provided .xls Data Configuration File

The data configuration file, which is a Cisco-provided .xls spreadsheet template where you can enter the majority of your configuration data, provides the following support:

- Allows you to plan your configuration before you begin your first day of deployment.
- Allows you to insert users and phones in bulk through the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface after your initial deployment.

To quickly import (add) your configuration data to Cisco Unified Communications Manager Business Edition 3000 after you plug in your Cisco Unified Communications Manager Business Edition 3000 server, you can enter your data and then upload the Cisco-provided .xls data configuration file to the server from a USB key or your desktop when you run the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard. If you upload the file, you bypass the configuration pages in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard, and the wizard immediately takes you to the Summary page where you can confirm your data.

After the server restarts at the end of the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard, you can log into the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface and verify that you data got added to Cisco Unified Communications Manager Business Edition 3000. If you include user and phone data in the Cisco-provided .xls data configuration file, the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface allows you to import the users and phones and then informs you of import errors for users and phones.

Tip If you do not want to upload the Cisco-provided .xls data configuration file when you run the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard, consider entering your data in the file and using it as a guide when you manually enter the information in the GUIs.

For example, during your initial deployment, you inserted 25 users and phones; now, you must insert 25 more users and phones. To accomplish this task, you can modify the Cisco-provided .xls data configuration file that you used for automatic set up during the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard or you can obtain a new Cisco-provided .xls data configuration file and add your new users and phones to that new spreadsheet.

Caution Do not use the Cisco-provided .xls data configuration file to modify your configuration data. Cisco Unified Communications Manager Business Edition 3000 only supports the Cisco-provided .xls data configuration file for the initial deployment and for bulk insertion (adding) of users and phones after the initial deployment. For example, if you attempt to update existing user and phone information through the Cisco-provided .xls data configuration file, the updates fail.

For More Information

- Working with the Cisco-Provided .xls Data Configuration File, page 3-1

Phones

Cisco Unified Communications Manager Business Edition 3000 supports a variety of phones that are available through Cisco. If the phone model can support either SIP or SCCP, Cisco Unified Communications Manager Business Edition 3000 uses SIP with the phone.

The Cisco Unified Communications Manager Business Edition 3000 server sends a phone-specific configuration file to each phone in your system. (This file is not the same as the Cisco-provided .xls data configuration file that is described in the "Cisco-Provided .xls Data Configuration File" section on page 1-5.) This configuration file contains data that your phone requires to work; for example, the configuration file specifies whether the phone can use barge, whether phones can use phone applications, what the locale is for the system, and so on.

You can configure the phone for Cisco Unified Communications Manager Business Edition 3000 by using the following methods:

- Through the Cisco-provided .xls data configuration file in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (for initial deployment)
- Through the Cisco-provided .xls data configuration file in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface (after initial deployment)
- Under **Users/Phones > Phones** in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface (after initial deployment)

Your phone requires an IP address and other network settings to work. For information on how your phone obtains its IP address and other network settings, refer to your phone administration documentation.

For your phone to work, you must install licenses. You cannot add a phone to the system if the appropriate license is not installed and available for use.

All features that are available with Cisco Unified Communications Manager Business Edition 3000 are not supported on all phone models. Before you configure your Cisco Unified Communications Manager Business Edition 3000, determine which features are supported on your phone by obtaining the phone administration documentation that is available with your phone and this version of Cisco Unified Communications Manager Business Edition 3000.

For More Information

- Sites, page 1-36 (for information on how phones get associated with a site)
- DHCP Usage for Acquiring IP Addresses, page 1-31
- Users, Departments, Phones, and Lines, page 1-42
- Cisco User Connect Licensing, page 4-1
- Checklists for Users, Departments, Lines, and Phones, page 8-1

Attendant Console

Cisco Unified Communications Manager Business Edition 3000 supports the Cisco Unified IP Phone 8961 which, can be used as an attendant console when a Cisco Unified IP Color Key Expansion Module (KEM) is attached to the phone. For information on connecting a KEM, see *Cisco Unified IP Phone 8961, 9951, and 9971 Administration Guide for Cisco Unified Communications Manager 8.5 (SIP)*.

The addition of the KEM expands the number of buttons that are available to the Cisco Unified IP Phone 8961 to 41 buttons for use as an attendant console. This provides the user with up to 40 buttons that can be used as speed dials, line buttons, or other features as required.

Note Button number 1 is automatically designated as a line by the system because button number 1 is used to correlate the phone and user when the user extension is assigned to line 1 on the phone. You cannot update Line Button 1.

The system administrator uses the Usage Profile of the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface to set up a usage profile for an attendant console. Using the Phone Button Template, the administrator can configure the 40 buttons that are available when a KEM is attached to the Cisco Unified IP Phone 8961.

Note The Phone Button Template is automatically provisioned with speed dials for the 40 buttons that are available. The system administrator can use the Phone Button Template to change the function of the 40 buttons that are available.

Note During migration all speed-dial details are saved and migrated to the new system.

For more information about configuring the Usage Profile, see Chapter 40, "Usage Profiles Settings."

Video Support

Cisco Unified Communications Manager Business Edition 3000 supports point-to-point video calls between two video-capable, nonteleworker phones (for example, Cisco Unified IP Phones 8941 and 8945) within the same site or when calling between sites that are configured with network interfaces of at least T1 capacity or larger and with video services between sites enabled.

Note Point-to-point video is not supported within the teleworker site or between the teleworker site and any other site that is connected to the teleworker site.

Note Cisco Unified Communications Manager Business Edition 3000 does not support video conferencing.

Caution The number of video calls is expected to be small. Because, bandwidth is usually limited between sites, the system does not reserve video bandwidth for infrequent video calls so that this bandwidth can be used for the audio-only calls. Thus, if a large number of video calls are made (relative to the number of video calls between sites as shown on the sites page), audio and video quality can suffer between the sites. If you encounter poor quality due to a large number of video calls, you may find it necessary to disable video to and from that particular site.

The system administrator accesses **System Settings > Sites** on the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface to configure the system for point-to-point video.

For More Information

Sites, page 1-36

Voicemail

Cisco Unity Connection, an internal component of the Cisco Unified Communications Manager Business Edition 3000 software that provides voicemail support for your system, resides on the Cisco Unified Communications Manager Business Edition 3000 server. With Cisco Unified Communications Manager Business Edition 3000, users can perform the following tasks:

- Call into the voice messaging system
- Send voice messages by using the phone keypad

- Check voice messages by using the phone keypad
- Reply to voice messages by using the phone keypad
- Forward voice messages by using the phone keypad
- Manage receipts by using the phone keypad—Receipts indicate when a voice message was played by an intended recipient, when it was received by the intended recipient, and if it was received by the intended recipient.

Tip Voicemail support requires the use of voicemail licenses. You must install one voicemail license for each user that requires voicemail.

For More Information

- Setting Up Voicemail, page 8-12
- Cisco User Connect Licensing, page 4-1

Auto Attendant

In Cisco Unified Communications Manager Business Edition 3000, the auto attendant serves as the "virtual receptionist;" that is, the caller receives an automated greeting and series of prompts in order to successfully transfer the call to a user without the assistance of an operator. The following options describe the auto attendant support.

Note Auto attendant uses the same internal components as voicemail. Auto attendant is turned on by default, and you cannot turn it off. The system can handle up to 12 simultaneous calls to voicemail and auto attendant.

- The auto attendant uses a single menu for both business and closed hours (default); the auto attendant plays the same greeting and set of prompts during both business and nonbusiness hours. Cisco Unified Communications Manager Business Edition 3000 automatically comes with a sample menu that provides the following functionality. If you do not want to use the sample menu, you can upload another menu that can be used by the system.
  - The auto attendant plays a greeting announcing that the corporate directory has been reached.
  - The auto attendant requests that the caller enter the extension on the phone to transfer the call.
  - If the caller does not enter the extension quickly, the auto attendant requests that the caller enter the extension again.
  - The auto attendant transfers the call to the user of the extension.
  - The auto attendant requests that the caller reenter the extension of the user when the system cannot find the extension.
  - The auto attendant plays a farewell prompt.
- The auto attendant uses a different menu for business hours and for closed hours; for example, the auto attendant plays a greeting and set of prompts during regular business hours, but, when the company is closed, the auto attendant plays an announcement that the business is closed and then automatically ends the call. (On the Auto Attendant page, you specify when the company is closed and when it is open.) For this option, you must upload a file that contains your greeting, set of prompts, and your message for nonbusiness hours.

Tip The auto attendant does not support a different menu for holidays.

To use the auto attendant, you must first configure the Voicemail and Auto Attendant Extension setting in the dial plan. You can configure this setting

- Through the Cisco-provided .xls data configuration file in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (for initial deployment)
- On the Dial Plan page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (during initial deployment if you do not use the Cisco-provided .xls data configuration file)
- Under **System Settings > Dial Plan** in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface (after initial deployment)

After you configure the Voicemail and Auto Attendant Extension setting in the dial plan, configure the Auto Attendant page in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface. (Select **System Settings > Auto Attendant**.) After you set it up, remember to test your auto attendant functionality.

Note Auto attendant uses an internal user called operator. You cannot edit or delete this user, and it does not display in the Search User page. In addition, you cannot add a user with the user ID of operator. (User IDs should indicate who the user is, not the functions or tasks that the user perform.)

Do not assign the Voicemail and Auto Attendant Extension that you configure in the dial plan to the user that is your operator.

For More Information

- Auto Attendant Settings, page 12-1
- Setting Up Auto Attendant, page 8-13
- Setting Up the System So that Incoming Calls Reach the Operator, page 8-14
- Setting Up the System So that Incoming Calls Reach the Auto Attendant, page 8-14
- Setting Up the System So that Incoming Calls Reach the Auto Attendant if the Operator is Not Available, page 8-15

Gateway

For all calls that go through the PSTN, the Cisco Unified Communications Manager Business Edition 3000 uses the following gateways:

- Gateway built in to Cisco Media Convergence Server 7890C1 (MCS7890-C1)
- Cisco 2901 Integrated Services Router (ISR2901)
- SPA8800
- SIP Trunk

Table 1-1 shows the supported PSTN connections for Cisco Unified Communications Manager Business Edition 3000.

Table 1-1 Supported PSTN Connections

Sl No.	Gateway Type	Connection Type	Max Number of ports	Usage
1	MCS7890-C1	• MGCP T1 PRI • MGCP T1 CAS	2	Central Site Only
2	Cisco ISR2901	• MGCP T1 PRI • MGCP T1 CAS	Unlimited for Provisioning	Central Site and/or Remote Site
3	SPA8800	FXO	Unlimited for Provisioning	Central Site and/or Remote Site
4	SIP Trunk	SIP trunk	Unlimited for Provisioning	Central Site and/or Remote Site

The gateways serve as your connection to the PSTN; that is, the gateway allows all of your users to place and receive calls that go through the PSTN.

Note For Cisco ISR2901, ensure that you connect the T1/E1 PSTN connections to slot 0 only.

The Cisco ISR2901 that you use with Cisco Unified Communications Manager Business Edition 3000 cannot be used for any IP routing functions other than those that are supported with Cisco Unified Communications Manager Business Edition 3000.

The Cisco Unified Communications (UC) Technology Package License must be purchased with the order of Cisco ISR2901.

Note Install the Cisco Unified Communications Technology Package License before you configure any Voice features on the Cisco Unified Communications Manager Business Edition 3000.

When you order a new router, it is shipped preinstalled with the software image and the corresponding permanent licenses for the packages and features that you specified. You do not need to activate or register the software before use. For more informations, see http://www.cisco.com/en/US/docs/routers/access/sw_activation/SA_on_ISR.html#wp1057952.

To verify if the Cisco Unified Communications Technology Package License is installed and activated, see Chapter 25, “License Settings”.

The Cisco MCS7890-C1 supports approximately 300 users and 400 devices.

For MCS78901-C1, you can create an internal gateway during the First Time Setup using the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard.

For the Cisco MCS7890-C1 gateway, you can configure the general settings, such as the Media Resource IP address and the hostname. The settings that you configure for the gateway allows the gateway, the Cisco Unified Communications Manager Business Edition 3000 server, and the phones to interact with each other for calls that go through the PSTN connection.

Ensure that you assign a static IP address for the Cisco MCS7890-C1 internal gateway. However, there is no such restriction of a static IP address for ISR2901 gateways. If you plan to use DHCP, see the “DHCP Usage for Acquiring IP Addresses” section on page 1-31.

You can configure the gateway for Cisco Unified Communications Manager Business Edition 3000 by using one of the following methods:

- Through the Cisco-provided .xls data configuration file in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (for initial deployment).
- On the Gateway page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (during initial deployment if you do not use the Cisco-provided .xls data configuration file).

Under **Connections > PSTN Connections > Add PSTN Connection > Connection Type > Device > Device > Add Device** in Cisco Unified Communications Manager Business Edition 3000 Administrative Interface (after initial deployment).

Tip After you add the Cisco ISR2901 gateway configuration to Cisco Unified Communications Manager Business Edition 3000, you must update the gateway with the appropriate CLI commands. See **Generate CLI Commands, page 20-1**.

For More Information

- SPA8800 Gateway, page 1-12
- SIP Trunking, page 1-24
- DHCP Usage for Acquiring IP Addresses, page 1-31
- IP Addressing, page 1-32
- DNS and Hostname Resolution, page 1-32
- Working with the Cisco-Provided .xls Data Configuration File, page 3-1
- Checklists for Configuring the Gateway, page 7-1

SPA8800 Gateway

SPA8800 is a small business analog gateway that supports the following:

- Analog trunking (FXO) to the PSTN
- Devices such analog phones and fax machines

Cisco Unified Communications Manager Business Edition 3000 is not responsible for upgrading SPA8800 firmware. Therefore, users must upgrade SPA8800 to the latest firmware (version 6.1.7 or later) prior to setting up analog trunks and lines on the Cisco Unified Communications Manager Business Edition 3000. Firmware can be downloaded from http://wwwin.cisco.com/voice/products/callcontrol/cmbe/3000/index.shtml.

Note The user interface for SPA8800 gateway interface is supported in English only. When the user interface on the Cisco Unified Communications Manager Business Edition 3000 is changed to another language, the options in the Advanced Options drop-down menu for the device or gateway remain in English.

Note Cisco Unified Communications Manager Business Edition 3000 supports connection to SPA8800 using static IP addresses only.

Overview

Perform the following actions, in sequence, to correctly configure the SPA8800:

1. Cisco Unified Communications Manager Business Edition 3000 Configuration
   a. Configure the Related Connections on the Cisco Unified Communications Manager Business Edition 3000 GUI, page 1-13
   b. Configure the SPA8800 Analog Phones on the Cisco Unified Communications Manager Business Edition 3000 GUI, page 1-17
2. SPA8800 Configuration
   a. Perform the Initial Setup on the SPA8800 for IP Addresses using SPA Interactive Voice Response, page 1-21
   b. Configure settings for TFTP on the SPA8800 GUI, page 1-22

Configure the Related Connections on the Cisco Unified Communications Manager Business Edition 3000 GUI

You must configure the SPA8800 device in the Cisco Unified Communications Manager Business Edition 3000 first, and then add Phone 1 or Line 1 for the SPA8800 device.

Use the following procedures to add, edit, or delete SPA8800 connections on the Cisco Unified Communications Manager Business Edition 3000 GUI.

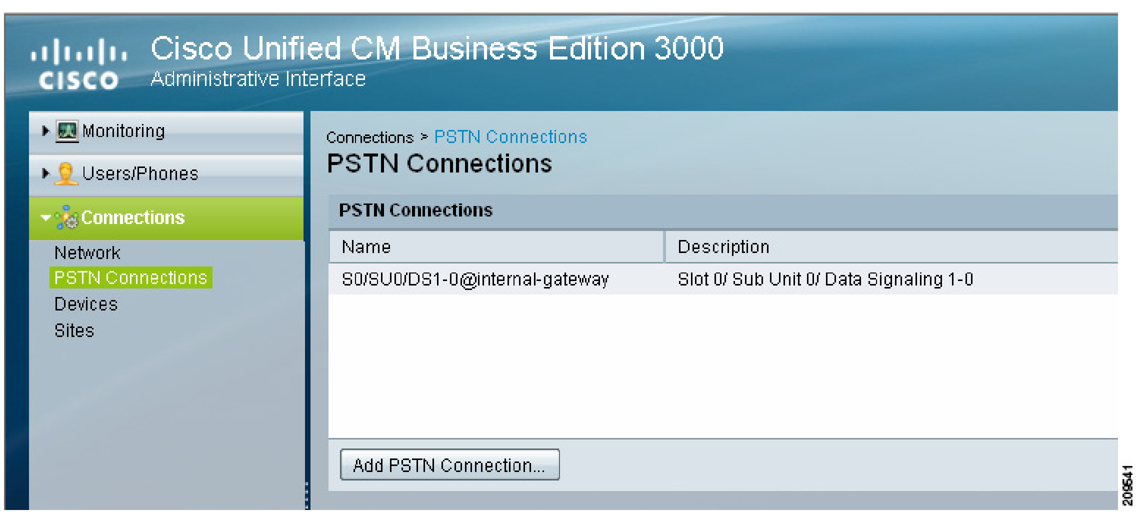
Add SPA8800 connections from the PSTN Connections page

Perform the following procedure to add SPA8800 connections from the PSTN Connections page:

Procedure

Step 1 From the PSTN Connections page, click the **Add PSTN Connection...** button below the PSTN table as shown in Figure 1 on page 1-13.

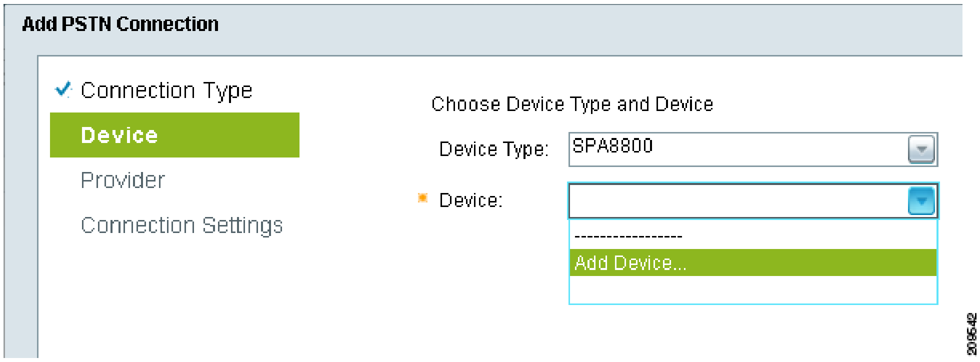
Figure 1 PSTN Connections table



Step 2 The Add PSTN Connection window appears. Choose the connection type **FXO**, and then click **Next**.

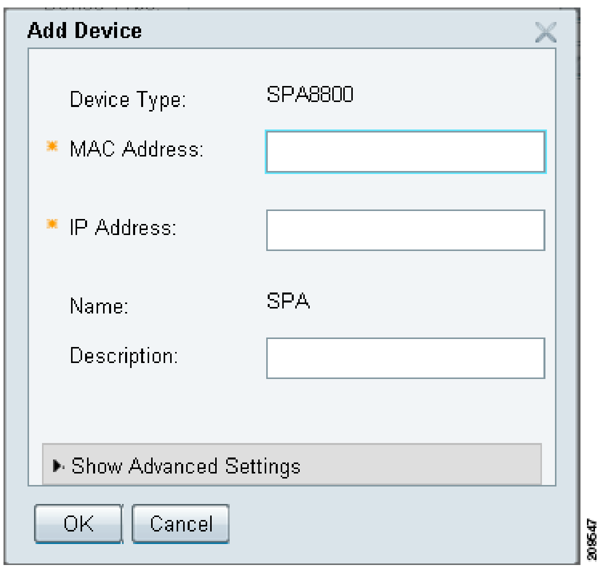
Step 3 The Device options appear. Select **SPA8800** from the Device Type drop-down menu, and choose **Add Device** from the Device drop-down menu.

Figure 1-2 Device options



Step 4 The Add SPA8800 Device window opens, as shown in Figure 3 on page 1-14. Enter the MAC address, IP address, and description. The name is derived from the MAC address. Click **OK**.

Figure 3 Add Device window



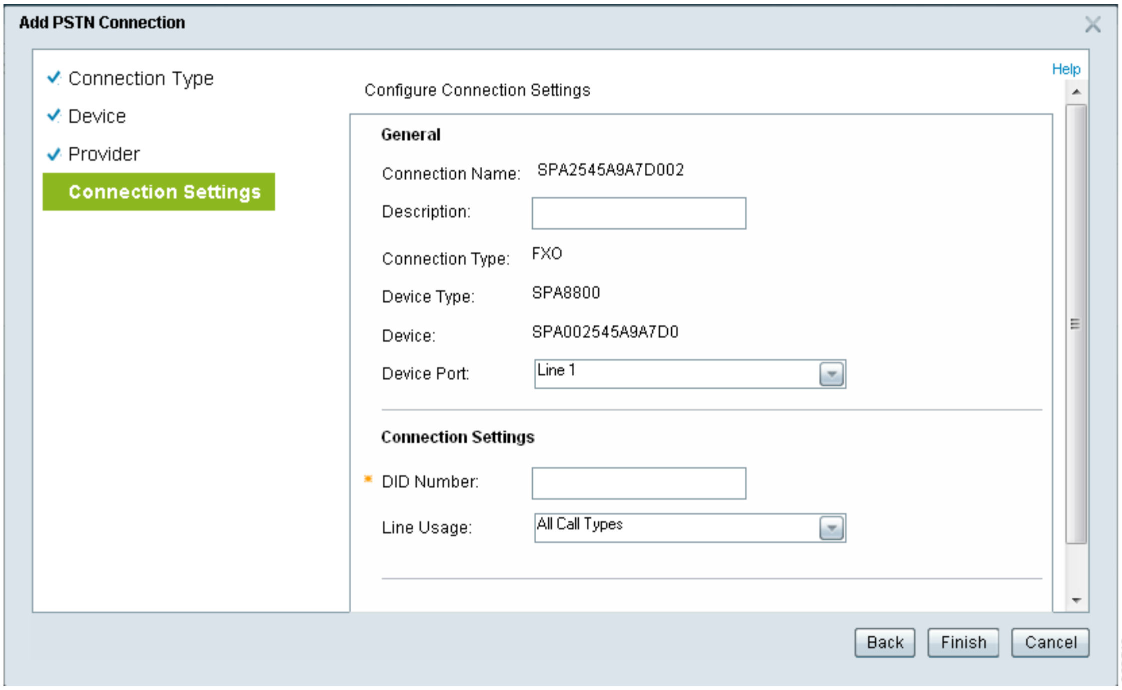
Note If at this point you click Cancel, the connection is cancelled but the device you added in the previous step remains in the system.

Step 5 The new device is now listed as an option under Device in the Add PSTN Connection window. Select the new device from the drop-down list and click **Next**.

Step 6 From the drop-down menu, select a service provider. Click **Next**.

Step 7 The Connection Settings appear, as shown in Figure 4 on page 1-15. Enter the appropriate connection settings and advanced settings. Refer to Table 31-11 on page 31-32 for information on each of these settings.

Figure 4 *Connection Settings*



Step 8 Click **Finish** to complete the addition of the device.

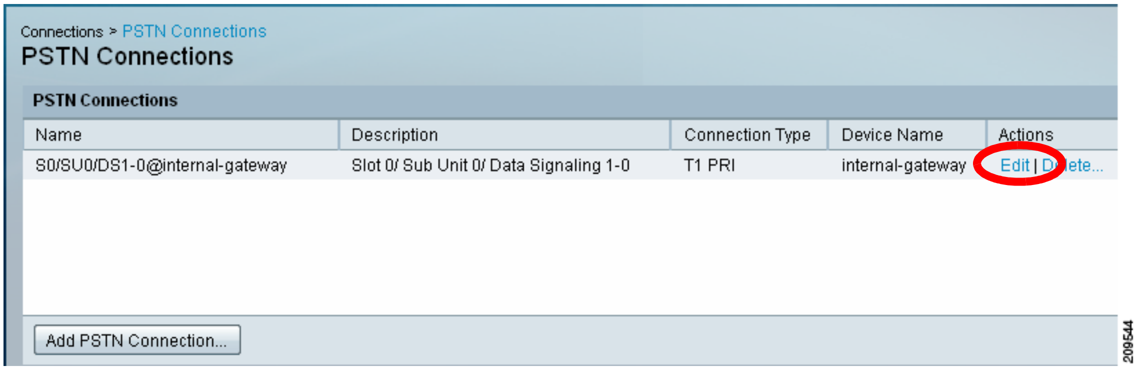
Note Any changes that you make on the SPA8800 connection and phone causes the SPA8800 to reboot. Reboots for configuration changes can take several minutes to take effect.

Edit SPA8800 connections from the PSTN Connections page

Procedure

Step 1 To edit a connection, choose **Edit** for that connection, as shown in Figure 5 on page 1-16.

Figure 5 *Edit PSTN Connections*



Step 2 The Edit window appears. From this window, you can edit the Description, Direct Inward Dial (DID) Number, Line Usage fields, and Advanced Settings. Refer to Table 31-11 on page 31-32 for information on each of these settings.

Step 3 Choose **Save** in the Edit window to save your edits. The device reset dialog appears to notify the user that the SPA8800 device is being reset and that all calls for the associated phones and PSTN connections will be disconnected.

Note Any changes that you make on the SPA8800 connection and phone causes the SPA8800 to reboot. Reboots for configuration changes can take several minutes to take effect.

Delete SPA8800 connections from the PSTN Connections page

Procedure

Step 1 To delete a connection, choose **Delete** for that connection from the PSTN Connection table.

Note A warning appears for connections that are configured for Emergency Calls Only, indicating that the DID used for the connection can no longer be used as an ELIN.

Step 2 The deletion will not occur for Line 1 if Phone 1 is not configured, and the user will get a message saying that the port is a master port and it can be deleted only as part of the SPA8800 device deletion. In all other cases the device reset dialog will appear notifying the user that the SPA8800 device will be reset and all calls of the associated phones and PSTN connections will be disconnected.

The connection is removed from the PSTN Connections list.

Note An associated device is not deleted as a result of removing the connection. A device can be deleted only from the Devices page.

Note Any change made on the SPA8800 connection and phone results in a reboot of the SPA8800. Reboots for configuration changes can take several minutes to take effect.

Configure the SPA8800 Analog Phones on the Cisco Unified Communications Manager Business Edition 3000 GUI

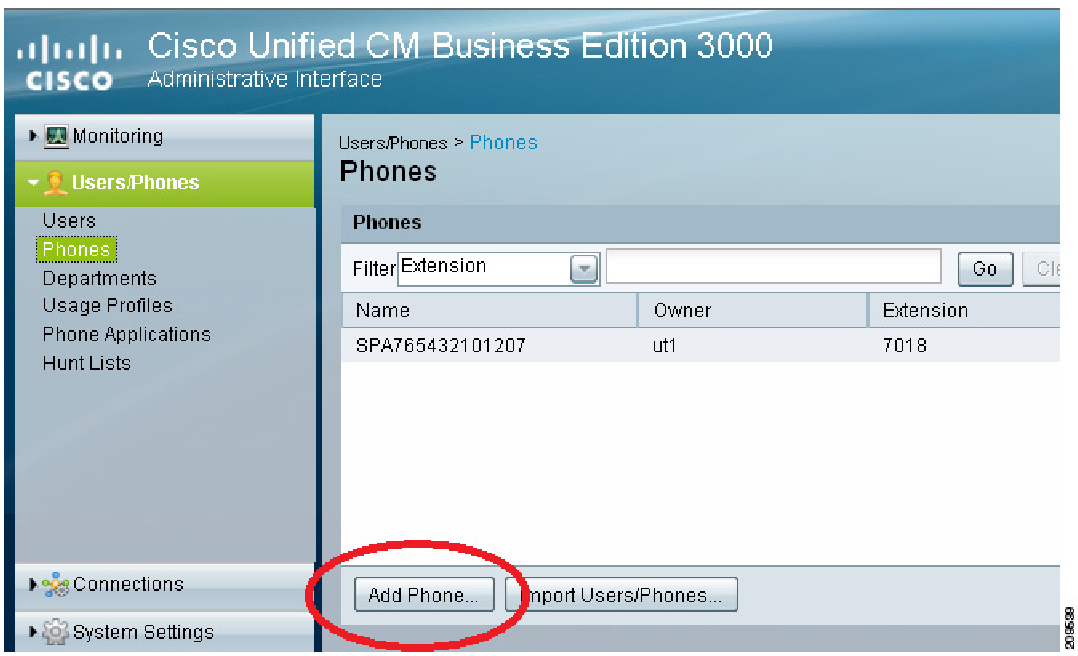
Use the following procedures to add, edit, or delete SPA8800 analog phones and other devices from the Cisco Unified Communications Manager Business Edition 3000 GUI.

Add a SPA8800 analog phone from the Phones page

Procedure

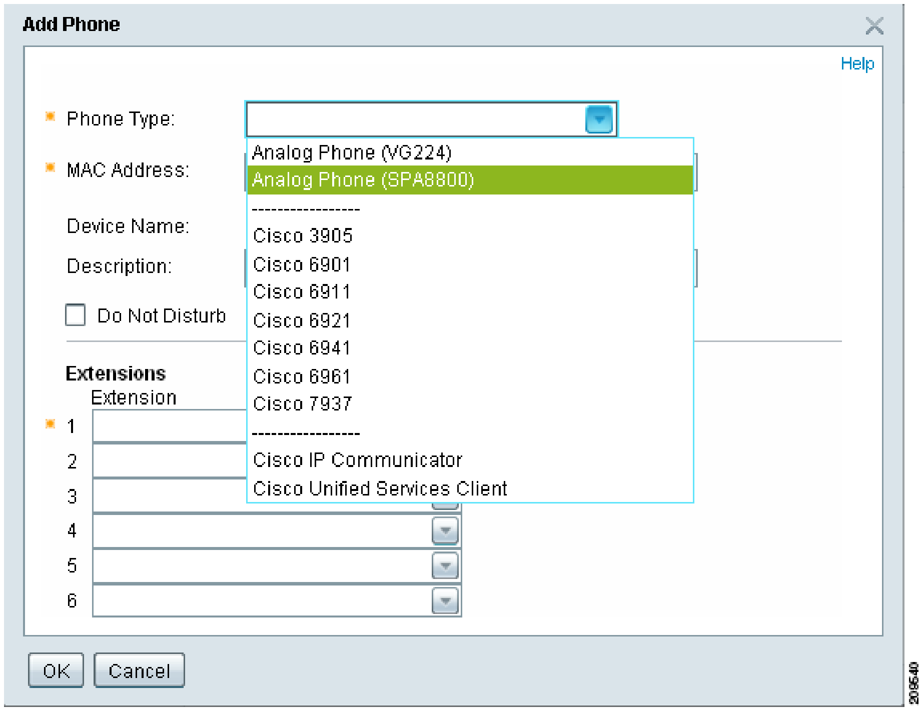
Step 1 From the Phones page, click the **Add Phone** button, as shown in Figure 6 on page 1-17.

Figure 6 Add Phone



Step 2 The Add Phone window appears. Choose **Analog Phone (SPA8800)** from the drop-down menu, as shown in Figure 7 on page 1-18.

Figure 7 Phone Type

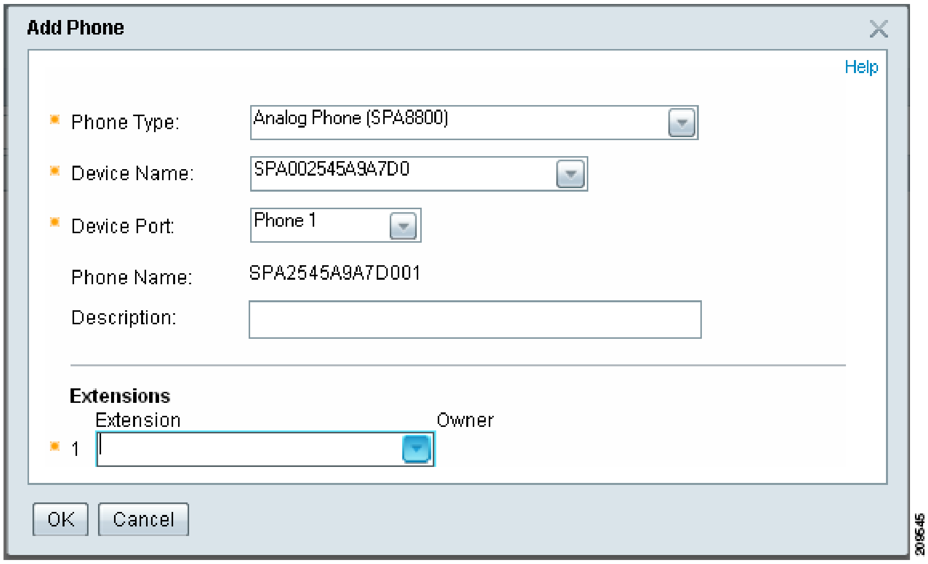


Step 3 Choose **Add Device** from the Device Name drop-down menu. This is necessary only if the gateway does not currently exist. You can choose the **SPA8800 Gateway** if it is already listed in the drop-down menu.

Step 4 The Add Device window appears. Enter the MAC address and the IP address (name is derived from the MAC address). Click **OK**. Devices are added through modal dialog box, with a value returned to Add Phone dialog box. The gateway-specific properties (for example: available/used ports) are then loaded into the Add Phone dialog box.

Step 5 If you added a new device, the Device Name now appears in the Add Phone window, along with the Device Port, as shown in Figure 8 on page 1-19. Only available ports are listed in the Device Port field. If certain ports are unavailable, the Device Port field defaults to the next available port.

Figure 8



Step 6 Choose an extension from the drop-down menu. All configured extensions are listed in the drop-down menu.

Step 7 Click **OK** to add the phone and return to Phones table.

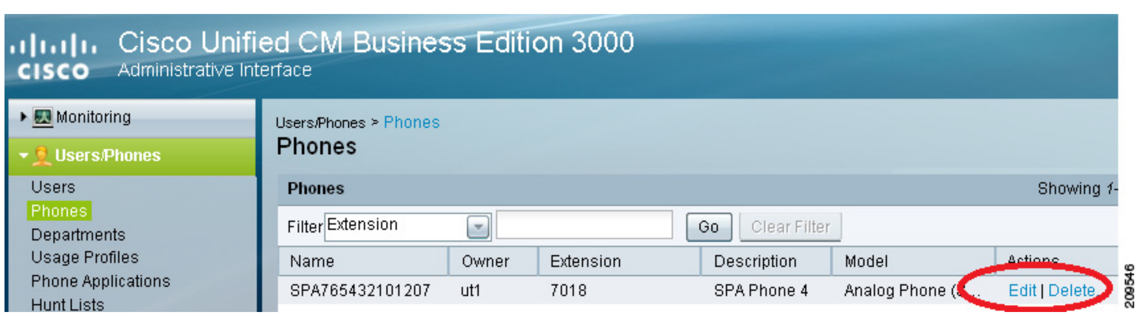
Note Any change made on the SPA8800 connection and phone results in a reboot of the SPA8800. Reboots for configuration changes can take several minutes to take effect.

Edit a SPA8800 analog phone from the Phones page

Procedure

Step 1 To edit an analog phone, click the **Edit** button for that phone in the Phones table, as shown in Figure 9 on page 1-19.

Figure 9



Step 2 The Edit window appears. From this window, you can edit the description and extensions.

Step 3 Click **Save** in the Edit window to return to the Phones table. The device reset dialog box will appear notifying the user that the SPA8800 device will be reset and all calls of the associated phones and PSTN connections will be disconnected.

Note Any change made on the SPA8800 connection and phone will cause the SPA8800 to reboot. Reboots for configuration changes can take several minutes to take effect.

Delete a SPA8800 analog phone from the Phones page

Procedure

Step 1 To delete a phone, click the **Delete** button for that phone in the Phones table.

Step 2 The delete will not occur for Phone 1 if Line 1 is not configured and the user the user will get a message telling them that the port is a master port and it can only be deleted as part of the SPA8800 device deletion. In all other cases the device reset dialog box will appear notifying the user that the SPA8800 device will be reset and all calls of the associated phones and PSTN connections will be disconnected.

The phone is removed from the Phones page.

Note Any change that you make on the SPA8800 connection and phone causes the SPA8800 to reboot. Reboots for configuration changes can take several minutes to take effect.

Edit SPA8800 devices from the Devices page

Procedure

Step 1 In the Devices page, locate the SPA8800 you wish to edit and click the **Edit** button under the Actions column.

Step 2 The Edit window appears. From this window, you can edit the MAC address, IP address, description, and advanced settings.

Step 3 Click **Save** in the Edit window to return to the Devices page. Any change made on the SPA8800 connection and phone will cause the SPA8800 to reboot. Reboots for configuration changes can take several minutes to take effect.

Delete SPA8800 devices from the Devices page

Procedure

Step 1 To delete an SPA8800, locate the SPA8800 you wish to delete and click the **Delete** button in the Actions column.

Step 2 The Confirm Delete window appears. Listed in this window are all phones and PSTN connections associated with this SPA8800.

Step 3 Click **Delete** to confirm deletion of the SPA8800.

Note If the SPA8800 you wish to delete has one or more FXO connections that are configured for emergency calling, any associated ELINs will be removed.

Note Any change made on the SPA8800 connection and phone will cause the SPA8800 to reboot. Reboots for configuration changes can take several minutes to take effect.

DID and ELIN configuration

DIDs for the trunks must be configured elsewhere as translation patterns, attendant numbers, directory numbers, or hunt lists in order to route incoming calls. Simply placing a DID on a trunk does not allow it to associate with a particular station within the system. In the case of the All Call Types option, customers are responsible for determining how calls on a trunk are routed (for example: attendant number or directory number).

In the case of Emergency Calls Only, a call-back pattern is set up that ensures that incoming calls are routed to the last number that called out on that particular gateway.

If an ELIN is associated with an analog trunk, Cisco recommends that it be configured on that analog trunk. If the ELIN is using a digital gateway (example: T1 or PRI connection), Cisco recommends that it be configured on the site.

To confirm that the configuration for emergency calling is successful, after the SPA8800 is configured, go to the site page and confirm that the DID for that trunk is listed as an ELIN. If it is not listed as an ELIN, check the other sites to see if it appears on one of them—it may have been set up on the wrong site. If this happens, check the IP address and subnet settings in Cisco Unified Communications Manager Business Edition 3000 to ensure that the gateway is registering at the correct site.

Note The manner in which a particular SPA8800 is associated with a site is based on the IP address and the subnets that have been set up for that site.

PSTN Connection Settings

Table 31-11 on page 31-32 defines the connection settings and advanced setting that are required to edit SPA8800 connections.

Perform the Initial Setup on the SPA8800 for IP Addresses using SPA Interactive Voice Response

The SPA8800 requires setup through the SPA Interactive Voice Response (IVR) menu to locate the Cisco Unified Communications Manager Business Edition 3000.

Use the following procedure to perform this setup.

Procedure

Step 1 Connect an analog phone to port 1.

Step 2 Go off hook and enter the configuration menu by pressing the * (star) key four times.

Step 3 Enter **101** followed by the # (pound) key to set the Internet connection type, followed by **1#** to set it to static IP addressing.

Step 4 Enter **111#** to set the static IP address.

Note DHCP must be set to Disabled; otherwise you hear "Invalid option" if you try to set this value. A password is required.

Step 5 Enter IP address using numbers on the telephone key pad. Use the * (star) key to enter a decimal point, followed by the # (pound) key. Press 1 to save the configuration change.

Note If, while entering a value (for example, an IP address), you decide to exit without saving any changes, you must press the * (star) key twice within a half-second window of time. Otherwise, the entry of the * (star) key will be treated as a dot (decimal point).

Tip To enter IP address, use numbers 0 – 9 on the telephone key pad and use the * (star) key to enter a decimal point.

Step 6 Check the subnet mask by entering **120#**.

Step 7 If necessary, change the subnet mask by entering **121#**.

Step 8 Set the default gateway IP address by entering **131#**, and then check the gateway IP address by entering **130#**. Hang up the phone for the values to take effect.

Configure settings for TFTP on the SPA8800 GUI

Procedure

Step 1 Enter the web interface using the following URL: **http://IP_Address_Of_SPA/admin/advanced**.

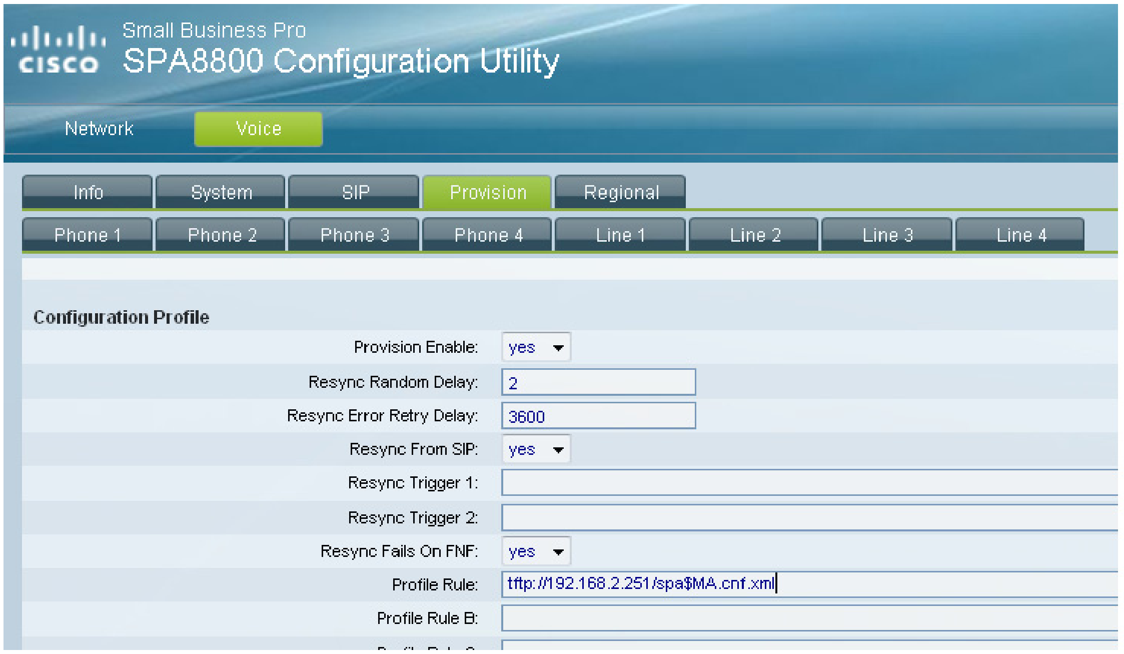
Note If you have problems accessing the SPA8800 device using the web interface, the issue may be a problem with Cisco Discovery Protocol (CDP).

By connecting your laptop directly to the SPA8800 using an Ethernet cable, you can access the SPA8800 administrative interface to enable/disable CDP by choosing **Network > Wan Status > VLAN Settings > Enable CDP**.

Step 2 Set the default password.

Step 3 Set the TFTP address for syncing the SPA8800 configuration from the Cisco Unified Communications Manager Business Edition 3000. Click the **Voice** button on the upper left corner and then click the **Provision** tab, as shown in Figure 10 on page 1-23.

Figure 10



Step 4 In the Profile Rule field, specify the TFTP protocol and the IP address of the Cisco Unified Communications Manager Business Edition 3000. In the example in Figure 10 on page 1-23, **tftp://192.168.2.251/spa$MA.cnf.xml** is the TFTP protocol where 192.168.2.251 is the IP address of the Cisco Unified Communications Manager Business Edition 3000.

Note Only TFTP is supported in the current release.

Step 5 Click the **Submit All Changes** button.

Step 6 Reboot the SPA8800 now. This can be done by unplugging the power cord for the SPA8800 and plugging it back in.

SPA8800 Feature Codes

Table 2 lists the feature codes that are supported for SPA8800.

Table 2 Supported Feature Codes

Feature	Code Number
Call Redial	*07
Call Back	*66

Feature	Code Number
Call Back Deactivate Code	*86
Call Waiting	*56
Call Waiting Deactivate Code	*57
Call Waiting Per Call Act Code	*71
Call Waiting Per Call Deact Code	*70
Call Return	*69
Attn-Transfer	*84
Conference	*85

SPA8800 Limitations

1. The Do Not Disturb (DND) setting cannot be configured for SPA8800 analog phones.
2. Music On Hold (MOH) is not supported on the SPA8800 analog phones.
3. Blind Transfer is not supported on the SPA8800 analog phones.
4. Call Forwarding must be set from the Cisco Unified Communications Manager 3000 and not from the SPA8800.
5. In a clear call scenario, such as the updating of phones or PSTN connections on the SPA8800 gateway, Cisco IP Phones with active PSTN calls receive a fast busy tone when a corresponding active FXO line is cleared.

SIP Trunking

In the Cisco Unified Communications Manager Business Edition 3000, the SIP trunk serves as a connection to the SIP service provider network for PSTN connectivity. To interact with the SIP trunk connection, the Cisco Unified Communications Manager Business Edition 3000 requires SBE (Session Border Elements) to provide security, topology hiding, and ALG (Application Level Gateway) functionalities.

To achieve SIP trunking, use the following operations:

- Provision Cisco Unified Communications Manager Business Edition 3000 to interact with the SBE. You can achieve this by the installation of Connection Packs, and by adding the SIP trunk connection through Cisco Unified Communications Manager Business Edition 3000 Administrative Interface.

  For information on connection packs, see Connection Pack, page 1-27.
- Provision SBE to interact with Cisco Unified Communications Manager Business Edition 3000. You can achieve this using the configurations provided by the service provider.
- Provision SBE to interact with the service provider. You can manage this using the service provider.

The following are the different types of SIP trunk connections for Cisco Unified Communications Manager Business Edition 3000:

- Standard Cisco SIP trunk connection—SIP trunk is connected to the service provider through the Cisco Unified Border Element (CUBE) on Cisco ISR8xx Series (Integrated Services Router). The parameters on the Cisco Unified Communications Manager Business Edition 3000 for the SIP trunk connection are preconfigured to support general SIP trunk functionalities.
- Service Provider SIP trunk connection—SIP trunk is connected to the service provider through other SBEs. The parameters of the Cisco Unified Communications Manager Business Edition 3000 SIP trunk connection is preconfigured based on the inter-operational tests with specific service provider. The service provider for provisioning is available after the installation of the SIP trunk connection pack that is customized for the service provider.

If the Cisco Unified Communications Manager Business Edition 3000 deployment requires SIP trunk connection in your site, deploy an SBE in the site to route PSTN calls. This SIP trunk connection can also be used to route the PSTN calls from other sites.

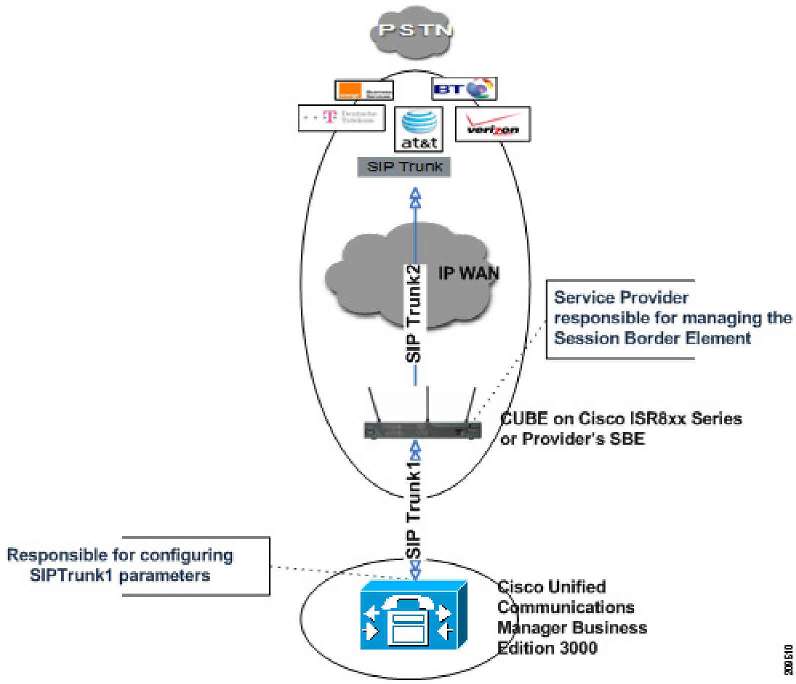
Cisco Unified Communications Manager Business Edition 3000 also supports multiple SIP trunks to the service provider through a single or multiple session border elements.

Note If a remote site needs to deploy a SIP trunk connection to its local SIP service provider, deploy a session border element on the remote site, rather than using the SIP trunk on the central site. The SIP trunk configured on the central site can also be used.

To add, edit, or delete SIP trunk connections, sign in to the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface (**Connections > PSTN Connections**).

Additionally, you must bind a SIP trunk/SBE with a site using the service provider IP address specified on the SIP trunk connection and provision the subnet on the site (**Connections > Sites > General** tab).

Figure 1-11 Provisioning Scope for SIP Trunk with Cisco Unified Communications Manager Business Edition 3000



Note SIP Trunk is another PSTN connection type along with T1/E1 PRI, T1 CAS, and FXO connections.

Based on the requirement of PSTN connections for Cisco Unified Communications Manager Business Edition 3000, provisioning is performed for routing of PSTN calls using the gateway on **Connections > Sites > Add Site > PSTN Access**.

Limitations of SIP Trunking

Cisco Unified Communications Manager Business Edition 3000 does not support the following features on SIP trunk to service providers:

- Digest authentication
- QSIG
- TLS
- IME
- MLPP
- IPV6
- MTP
- Raw DTMF
- RSVP

Cisco Unified Communications Manager Business Edition 3000 does not support the following functionalities on SIP trunk:

- pTime—This media-based parameter cannot be configured on Cisco Unified Communications Manager Business Edition 3000. However, a fixed value will be used.
- DTMF—The PSTN calls with SIP trunk connection encountering DTMF incompatibility requires an MTP to normalize the DTMF. As Cisco Unified Communications Manager Business Edition 3000 does not support MTP, the call fails.
- Early media on 180—For SIP trunk connections, Cisco Unified Communications Manager Business Edition 3000 signals calling phone to play local ringback as it does not receive SDP in the 180 response. However, the system receives as SDP in the 183 response.

Connection Pack

The provisioning of SIP trunks to SBE from Cisco Unified Communications Manager Business Edition 3000 is supported through the installation of connection packs. The connection pack is a Cisco Options Package (COP) file signed by Cisco.The main advantage of using the connection pack is that it allows quick and easy configuration of the SIP Trunk with minimal or no errors.

The connection pack is bundled with the Provider XML file, and Logical Unit Application (LUA) scripts necessary for the operation of SIP Trunks.

- Provider XML— A configuration file detailing the connection definition for a particular SIP trunk on the Cisco Unified Communications Manager Business Edition 3000.

  The provider XML file controls the default values and the display of Administrative Interface elements for PSTN Connection configurations for provisioning the SIP trunk.
- LUA script—(Optional) A mechanism supports transparency and normalization of SIP messages for interacting with the service provider.

For the service provider SIP trunk connection, the provider XML is defined and bundled with the connection pack. The connection pack must be installed before provisioning the service provider SIP trunk.

The mechanism to upgrade the connection pack is through the **Maintenance > Upgrade** page on the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface. For more information, see Installing the Connection Pack File, page 1-28.

The connection pack is used to upgrade the following SIP trunk connections:

- Cisco Standard SIP Connection—The connection pack bundles the provider XML for configuring the SIP trunks using CUBE on Cisco ISR8xx as the service provider.

  You can edit the parameters through the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface.
- Service Provider SIP Connection—The connection pack bundles the provider XML for configuring the SIP trunks using other service providers. This ensures that the Cisco Unified Communications Manager Business Edition 3000 SIP trunk interworks with the service provider through SBEs (including third-party session border elements). The new parameter definitions are effective for the SIP trunks using service provider SIP Connection.

After the installation of a connection pack for a service provider, create SIP trunk using the service provider. The SIP parameters of the SIP trunk connection are preconfigured with the values in the connection pack automatically. When a SIP trunk connection is created, a minimal set of parameters, as designed by the service provider, is editable through the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface (**PSTN Connections > Add PSTN Connection > Connection Settings > Configure Connection Settings**).

Note The parameters in the SIP trunk connection pack are preconfigured based on the inter-operability tests between the Cisco Unified Communications Manager Business Edition 3000 SIP trunk and the service provider. Therefore, the SIP trunk that is created with the service provider connection type will be configured correctly to inter-operate with the service provider.

When you edit a SIP trunk connection, a minimal set of parameters, as designed by the service provider, is editable through the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface (**PSTN Connections > Add PSTN Connection > Connection Settings > Configure Connection Settings**). For more information, see Connection Type: SIP Trunk, page 31-24.

Versioning of the Connection Pack file

The naming convention of the connection pack is based on the Cisco Unified Communications Manager Business Edition 3000 version and the service provider version.

cm-conp-<Cisco Unified Communications Manager Business Edition 3000-version>-<serviceprovidername>-<version>.cop.sgn

```
For example:
    cm-conp-CP-8.6.2-foo-1.cop.sgn
    cm-conp-CP-8.6.2-foo-2.cop.sgn
```

Installing the Connection Pack File

You can upgrade the connection pack through the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface. For instructions on how to import the connection pack file, see Chapter 38, "Upgrade Settings."

Upgrading the Cisco Unified Communications Manager Business Edition 3000 with the SIP trunk connection pack includes the following steps:

1. Download—Download the SIP trunk connection pack (.cop.sgn) file. You can burn the file to a DVD or save it on SFTP location for uploading. You can click Cancel to terminate the upgrade process on the progress bar.

   Note You can get the connection pack (.cop.sgn) file from your Service Provider or from www.cisco.com.

2. Validation—The Cisco Unified Communications Manager Business Edition 3000 validates the connection pack and the version based on the checksum before upgrade.
3. Installation—The connection pack is installed on the Cisco Unified Communications Manager Business Edition 3000. A progress bar depicts the status of the upgrade. You cannot cancel the upgrade process after the installation process is initiated.

The Cisco Unified Communications Manager Business Edition 3000 Administrative Interface notifies the status of installation as a progress bar.

If there are PSTN connections using the connection pack that is upgraded, the Administrative Interface notifies that the connection will be reset and all the calls associated with SIP trunk using that connection pack are dropped. The Cisco Unified Communications Manager Business Edition 3000 Administrative

Interface will request confirmation. If you choose to cancel, the installation process will be terminated and there will be no impact on the current connection and service provider. If you confirm to upgrade, the upgrade process begins.

After you upgrade the connection pack successfully, a confirmation message appears indicating that the connection pack was successfully installed. The connection pack upgrade process takes 2 to 7 minutes if you confirm to upgrade, and 2 to 4 minutes if you choose to cancel the upgrade.

The advance settings of the SIP trunk connections will be updated with the new set of parameters through the connection pack file. After the connection pack upgrade is complete, the SIP trunk connection will be reset and active calls will be dropped.

Upgrade failure handling

If the upgrade fails during validation (for example, if you have a corrupt .cop.sgn file) or installation (for example, if you have an invalid provider XML file), the upgrading process stops and the Administrative Interface displays a message to reboot the system. It is mandatory to reboot the system if this error is encountered. For more information, see Chapter Troubleshooting Issues, page 47-51.

The Cisco Unified Communications Manager Business Edition 3000 allows you to revert to the previous version of the connection pack. You can achieve this by upgrading the system using the connection pack of the desired version.

Impact during Cisco Unified Communications Manager Business Edition 3000 Upgrade

The SIP trunk connection will be reset while the Cisco Unified Communications Manager Business Edition 3000 is upgrading. All the active calls will be dropped.

The upgrading of the new Cisco Unified Communications Manager Business Edition 3000 may result in change of various parameters on the Administrative Interface. Some parameters can be removed and new parameters may be introduced. The user-configured SIP trunk parameter settings will be saved after upgrade also.

Significant Behavior of SIP Trunk

SIP trunk exhibits significant behavior while processing PSTN calls in Cisco Unified Communications Manager Business Edition 3000. The following sections detail the behavior of SIP trunk.

- Incoming 302—Moved Temporarily, page 1-29
- Incoming OOD REFER Message Handling, page 1-30
- Calling Party Transformation, page 1-30
- Connected Party Transformation, page 1-31

Incoming 302—Moved Temporarily

In “Redirect by Application” configuration, the SIP trunk passes the control to the Redirecting Application layer for handling the rerouting. The “Rerouting Calling Search Space” configured on SIP trunk is passed to allow further check on class of service and privilege of the calling user for redirection to the new contact. To set these parameters, refer Connection Type: SIP Trunk, page 31-24.

The Redirect by Application feature of the SIP trunk allows the Cisco Unified Communications Manager Business Edition 3000 to do the following:

- Apply digit analysis to the redirected contacts to ensure that the calls are routed correctly

- Prevent DOS attack by limiting the number of redirection (recursive redirection) that a service parameter can set
- Allow other features to be invoked while the redirection is taking place

Calls get redirected to a restricted phone number (such as an international number) due to handling redirection at the stack level to route the calls without blocking. This behavior occurs when the Redirect by Application check box is not checked in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface.

In case of multiple redirections before final redirection over SIP trunk, the maximum of two Redirection headers will be sent over SIP trunk: the original called party and last called party information.

Incoming OOD REFER Message Handling

Out-of-dialog REFER (OOD-R) enables remote applications to establish calls by sending a REFER message to Cisco Unified Communications Manager Business Edition 3000 without an initial INVITE. After the REFER is sent, the remainder of the call setup is independent of the application and the media stream does not flow through the application. The application using OOD-R triggers a call setup request that specifies the Referee address in the Request-URI and the Refer-Target in the Refer-To header.

Cisco Unified Communications Manager Business Edition 3000 handles the incoming REFER from a SIP trunk service provider.

Calling search spaces determine the partitions that calling devices can search when they attempt to complete a call. The out-of-dialog calling search space is used when a Cisco Unified Communications Manager refers a call (B) that is coming into SIP user (A) to a third party (C) when no involvement of SIP user (A) exists. In this case, the system uses the out-of-dialog calling search space of SIP user (A). The third party (C) is either an internal extension or an auto attendant client.

CUBE on Cisco ISR8xx, the session border element in the Cisco Standard Connection Pack file, does not support the OOD REFER inter-operability. The service provider cannot send OOD REFER through CUBE on Cisco ISR8xx through Cisco Unified Communications Manager Business Edition 3000.

Calling Party Transformation

The mid-call SIP messages, namely reINVITEs, UPDATE, or 200 OK sent from the calling party direction on SIP Trunk from Cisco Unified Communications Manager Business Edition 3000, carry the URI identity containing a number in user portion. This occurs during Hold/Resume, Transfer and so on, which result in transactions inside a SIP dialog.

By default, the Cisco Unified Communications Manager Business Edition 3000 sends configured extension number only, whereas the expected number for inter-operability with the SIP service provider is the DID or full number (e.g: Office code + Subscriber code).

The Cisco Unified Communications Manager Business Edition 3000 preconfigures the “Calling party Transformation” on SIP Trunk used to connect to the session border elements.

By provisioning the “Calling Party Transformations” feature for a SIP Trunk, the SIP dialogs established as part of outbound SIP Trunk calls always use the transformation for upsizing the number sent in P-Asserted ID or Remote Party ID headers of an outbound SIP message.

Note For the calling party transformation to function correctly, ensure that the External Caller ID is defined for the user. You can edit the **External Caller ID** on **Users/Phones > Users > Edit User > General** page.

Connected Party Transformation

The backward direction SIP messages, namely 183, 200 OK or mid-call UPDATE/INVITE messages from the connected party SIP trunk on Cisco Unified Communications Manager Business Edition 3000, carry the URI identity containing a number in user portion.

By default, the Cisco Unified Communications Manager Business Edition 3000 sends configured extension number only, while the expected number for inter-operability with the SIP service provider is the DID or full number (for example: Office code + Subscriber code).

To send the full number (DID and so on) on SIP trunk, the Cisco Unified Communications Manager Business Edition 3000 preconfigures the "Connected Party Transformation" on SIP trunk used to connect to the session border element. By provisioning the "Connected Party Transformation" feature for a SIP trunk, the SIP dialogs that are established as part of inbound SIP trunk calls always use the transformation for upsizing the number sent in PAI/RPID headers of an outbound SIP message.

Note For the Connected Party Transformation to function correctly, ensure that the External Caller ID is defined for the user. You can edit the **External Caller ID** on **Users/Phones > Users > Edit User > General** page.

DHCP Usage for Acquiring IP Addresses

This document does not provide detailed information on DHCP; you should have a thorough understanding of DHCP before you use it with your Cisco Unified Communications Manager Business Edition 3000 system. Typically, the IT support staff for the company or the Internet service provider handles your DHCP setup. DHCP may be run on a computer or on a router. Before you implement DHCP with Cisco Unified Communications Manager Business Edition 3000, consider the following information:

- Use custom option 150 or option 66.
- You can use a DHCP server to issue IP addresses to the phones.

  If DHCP is enabled on a phone, which is the default for the phone, DHCP automatically assigns an IP address to the phone after you connect it to the network. The DHCP server directs the phone to the Cisco Unified Communications Manager Business Edition 3000 server, which serves a phone-specific configuration file to the phone.

  If DHCP is not enabled on a phone, you must manually assign an IP address to the phone and configure the IP address or the hostname of the Cisco Unified Communications Manager Business Edition 3000 server locally on the phone (configure it under the TFTP server option on the phone).
- The Cisco Unified Communications Manager Business Edition 3000 server must use a static IP address that you assign to it. If you use a DHCP server to issue IP addresses to computers and other network devices, make sure that the IP address for the Cisco Unified Communications Manager Business Edition 3000 server is not in the active range of IP addresses on the DHCP server. Make sure that you configure your DHCP server so that it does not hand out the IP address for the Cisco Unified Communications Manager Business Edition 3000 server to a different network device.
- Cisco strongly recommends that you assign a static IP address to the gateway. If you use a DHCP server to issue IP addresses to computers and other network devices, make sure that the IP address for the gateway is not in the active range of IP addresses on the DHCP server. Make sure that you configure your DHCP server so that it does not hand out the IP address for the gateway to a different network device.

If DHCP is enabled on a gateway, DHCP automatically assigns an IP address to the gateway after you connect it to the network.

- Before you configure your sites and DHCP, Cisco strongly recommends that you determine the number of sites that you need and determine how many phones will be located at each site. Configure your DHCP server so that it correctly distributes the IP addresses to the phones at the various sites.

For More Information

- Phone administration documentation that supports your phone model
- Sites, page 1-36 (for subnet and subnet mask information for sites)

IP Addressing

The Cisco MCS7890-C1 uses two external IP addresses. The main IP address is the published IP address on the MCS7890-C1 device. This is the system IP address of the Cisco Unified Communications Manager Business Edition 3000. This IP address is set by your administrator during the First Time Setup of Cisco Unified Communications Manager Business Edition 3000. You can configure this IP address using DHCP or static. The second IP address is the IP address of the media resource used for transcoding and conferencing. Cisco recommends that the media resource IP address be static and unique for MCS7890-C1.

Note For Cisco ISR2901 gateway, the media resource IP address is the same as the ISR2901 gateway IP address.

DNS and Hostname Resolution

This document does not provide detailed information on DNS. You should have a thorough understanding of DNS before you use it with your Cisco Unified Communications Manager Business Edition 3000 system. DNS is optional; you do not have to use a DNS server unless you plan to resolve hostnames for your Cisco Unified Communications Manager Business Edition 3000 server or gateway. If you include a hostname for the server or gateway on the Network or Gateway page and you must use DNS, make sure that you map the hostname(s) to the IP address(es) on the DNS server for both forward and reverse DNS resolution. Cisco recommends that you perform this task before you add or edit the hostname in the Cisco Unified Communications Manager Business Edition 3000 GUIs.

The current version of the Cisco Unified Communications Manager Business Edition 3000 does not require a DNS server; however, it is configured for future requirements such as SIP trunks.

Note Cisco recommends that you do not configure Cisco Unified Communications Manager Business Edition 3000 to use DNS.

SFTP Server

Cisco allows you to use any SFTP server product but recommends SFTP products that are certified with Cisco through the Cisco Technology Developer Partner program (CTDP). CTDP partners, such as GlobalSCAPE, certify their products with a specified release of your software. For information on which vendors have certified their products with your version of software, refer to the following URL:

http://www.cisco.com/pcgi-bin/ctdp/Search.pl

For information on using GlobalSCAPE with supported Cisco Unified Communications versions, refer to the following URL:

http://www.globalscape.com/gsftps/cisco.aspx

Cisco uses the following servers for internal testing. You may use one of the servers, but you must contact the vendor for support:

- Open SSH (refer to http://sshwindows.sourceforge.net/)
- Cygwin (refer to http://www.cygwin.com/)
- Titan (refer to http://www.titanftp.com/)

Caution Cisco does not support using the SFTP product, freeFTPd, because of the file size limit on this SFTP product. For issues with third-party products that have not been certified through the CTDP process, contact the third-party vendor for support.

You can use a SFTP server to complete the following tasks:

- Upload the upgrade file from the SFTP server to the Cisco Unified Communications Manager Business Edition 3000 server before you perform an upgrade
- Store your backup file to a SFTP server, and if you must restore your data, restore the data from the SFTP server
- Export your configuration data to a SFTP server

Support for Computer Telephony Integration

Computer Telephony Integration (CTI) allows you to use computer-processing functions while making, receiving, and managing telephone calls. CTI can allow you to perform such tasks as retrieving customer information from a database on the basis of information that caller ID provides.

Cisco Unified Communications Manager Business Edition 3000 provides user support for CTI applications. Cisco Unified Communications Manager Business Edition 3000 automatically gives all users the ability to run CTI applications, including Cisco Jabber clients. Cisco Jabber clients must be configured as phones in the Phone Configuration window. During the configuration process, the Cisco Jabber client must be given a unique name or identifier and the client must then be associated with a user.

During the Cisco Jabber client registration process, the Cisco Unified Communications Manager Business Edition 3000 TFTP service sends the following three XML files from Cisco Unified Communications Manager Business Edition 3000 to the Cisco Jabber client. These files contain the registration details, application dial rules, and directory lookup dial rules:

- Identifier.cnf.xml (file is named according to naming of Identifier field during configuration)
- AppDialRules.xml
- DirLookupDialRules.xml

Cisco Unified Communications Manager Business Edition 3000 listens on port 2748 for requests from CTI applications. All telephone calls that are placed through the CTI application must use the E.164 number format. The dial rules that are sent during registration convert the following phone number formats to an E.164 format of +{country code}{area code}{local number}, thereby allowing the CTI application to support the Click to Call phone feature.

- {area code}{local number} for example, 972 813 0000
- {country code}{area code}{local number} for example, 1 972 813 0000
- {national access code}{area code}{local number} for example, 1 972 813 0000
- {out of country code}{country code}{area code}{local number} for example, 011 8621 972 813 0000

Common Configuration Concepts in Cisco Unified Communications Manager Business Edition 3000

This section describes common configuration concepts in Cisco Unified Communications Manager Business Edition 3000. In addition, this section describes considerations that you should review before you configure the items:

- Network Settings, page 1-34
- Dial Plans, page 1-35
- Sites, page 1-36
- Usage Profiles, page 1-39
- Users, Departments, Phones, and Lines, page 1-42
- Example of Typical Deployment Model, page 1-47

Network Settings

Your network settings include the IPv4 address or hostname of the server, the subnet mask and default gateway for the server, the primary and secondary DNS server (if you use DNS), the link speed for the Network Interface Controller (NIC) on the server, and the Message Transmission Units (Maximum Transmission Units, MTU) for the network.

- Through the Cisco-provided .xls data configuration file in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (for initial deployment)
- On the Network page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (if you are not using the Cisco-provided .xls data configuration file during initial deployment)
- Under **System Settings > Network** in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface (after initial deployment)
- Through the configure.xml file (used primarily for troubleshooting when you cannot access the GUIs)

For More Information

- Working with the Cisco-Provided .xls Data Configuration File, page 3-1

- Network Settings, page 27-1 (for the Network page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard and Cisco Unified Communications Manager Business Edition 3000 Administrative Interface)
- Using the Cisco Network Configuration USB Key, page 6-4

Dial Plans

Your Cisco Unified Communications Manager Business Edition 3000 Dial Plan allows you to allocate phone numbers and translation rules for your system. You can choose the country where you are installing the Cisco Unified Communications Manager Business Edition 3000. The Cisco Unified Communications Manager Business Edition 3000 installs a Dial Plan based on the regulations in your country.

Cisco Unified Communications Manager Business Edition 3000 supports a local gateway in every site and multiple gateways in any site; therefore the installation of Routing Dial Plan is required.

For load balancing and backup purposes, the routing is set up such that if a local gateway is not available, the calls can be routed to the gateway on the other site.

You can install or update the Dial Plan during the following instances:

- First Time Setup
- Upgrade
- Country Pack installation
- Site creation, update and deletion

Information about Dial Plans is available in the following files:

- Numbering Plan file—Specifies information about Dial Plan tags.
- Cisco-provided .xml file—Specifies the metadata information to set up a Dial Plan for your country. It includes XML elements for Route Filters, Route Filter Members, Translation Patterns, Route Patterns, and Called and Calling Party Transformation Patterns.
- Route Filters .xsd—Specifies XML definition.

For the default countries, the United States, India, and Canada, the dial plans are available with the installation file. For all other countries, you can install Dial Plans using the Country Pack.

For more information, see Country/Locale Settings, page 15-1.

Phone Numbers

For your dial plan, you specify the main business number, the area codes, the length of the extension, the extension ranges, and dialing prefixes for the outside dialing code (access code), operator dialing code, and the feature codes, which a user presses on the phone for certain features, including Meet-Me Conferences, call pickup, and so on.

After you set up your default extension range, you cannot change it. You can change other settings for your dial plan in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface.

The extension length and range that you set in the dial plan impact the extensions that you can assign to your users, departments, and pilot extensions for your hunt lists and voicemail and auto attendant. You cannot add an extension to a user or department that does not belong in the extension range. (Pilot extensions for hunt lists must also be in the extension range, but pilot extensions do not get assigned to users or departments.)

Your system can support internal, local, long distance, and international calls. Toll-free calls are also supported. You can set the level of access for a site on the Sites page; you can set the level of access authorized for the user in the Usage Profile. When a user places a call from a phone in a particular site, the call gets connected if the phone at that site is allowed to make that level of call and if the user that owns the line is also authorized to make that level of call.

Translation Rules

Translation rules allow Cisco Unified Communications Manager Business Edition 3000 to manipulate an incoming phone number that is part of your system and transform it to an extension before routing the call. The following list provides examples of when you would configure translation rules:

- To translate the Meet-Me conference number to an extension
- To translate a toll-free number, such as an 800 number, to an extension
- To translate an extension to a pilot extension in a hunt list

For More Information

- Working with the Cisco-Provided .xls Data Configuration File, page 3-1
- Dial Plan Settings, page 22-1 (for the Dial Plan page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard and Cisco Unified Communications Manager Business Edition 3000 Administrative Interface)
- Checklists for Users, Departments, Lines, and Phones, page 8-1

Sites

Cisco Unified Communications Manager Business Edition 3000 supports PSTN calls from every site through a local Gateway or a gateway located in other sites.

Your Cisco Unified Communications Manager Business Edition 3000 system may contain multiple sites, which are geographical locations that define where the users are working.

- Central Site—The central site contains the Cisco Unified Communications Manager Business Edition 3000 server and the gateway that allows access to the PSTN. In most cases, the central site is the location where the majority of users work; in most cases, the company headquarters is the central site. You can have only one central site, and you cannot delete the central site.
- Remote Sites—Remote sites are branch offices that work with the central site; a dedicated WAN link or Internet connection must exist between the remote and central sites. You can have up to nine remote sites.
- Teleworker Site—The teleworker site is for workers who do not work only at the central site or branch offices; these employees (users) use VPN connections to connect to the central site. A router is not required to contact the central site because their Internet connection provides access to the central site. You can have only one teleworker site.

When you configure a site, you must specify the maximum bandwidth that is required between sites, the maximum bandwidth that is required for internal calls that take place within the site, calling privileges for the site as a whole, and so on. You can configure the bandwidth between the remote site and the central site.

You can configure a site through the following methods:

- Through the Cisco-provided .xls data configuration file in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (for initial deployment)

- On the Sites page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (if you are not using the Cisco-provided .xls data configuration file during initial deployment)
- Under **Connections > Sites** in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface (after initial deployment)

A phone gets associated with a site through the subnet and subnet masks that are configured on the Sites page. If you are using DHCP to assign IP addresses to the phones, the values that you enter for the subnet and subnet mask depend on your DHCP configuration. In this case, the subnet and subnet mask translate to a range of IP addresses that get distributed to the phones that are associated with the site.

Considerations for Configuring Sites

Before you configure your sites and DHCP, Cisco strongly recommends that you determine the number of sites that you need and how many phones will be located at each site. Configure your DHCP server so that it correctly distributes the IP addresses to the phones at the various sites.

If you do not configure the subnet and subnet masks for any site, the phones automatically get assigned to the central site.

If no teleworker site exists and Cisco Unified Communications Manager Business Edition 3000 cannot determine where the phone is located, Cisco Unified Communications Manager Business Edition 3000 automatically places the phone in the central site.

By default, subnet address 192.168.1.0 with a subnet mask of 24 are displayed on the central site page. You may update this information if it does not apply to your setup.

If you enable Reach Me Anywhere in the usage profile, the call privileges for the Reach Me Anywhere call are always based on the highest calling privileges that are selected for the central site.

To associate phones with the remote sites, you must configure the subnet and subnet masks on the Sites pages for the remote sites.

If you have a teleworker site, you must configure the subnet and subnet masks for the central site and all remote sites. For the teleworker site, you cannot specify the subnet and subnet masks.

You cannot enter the same subnet and subnet mask for multiple sites. In addition, Cisco recommends that your subnet and subnet masks do not overlap. If you configure multiple subnet and subnet masks on the Sites page, Cisco Unified Communications Manager Business Edition 3000 selects the subnet and subnet mask that is most clearly defined and associates the phone with that site.

It is not uncommon to disallow emergency calls for the teleworker site. If you disallow emergency calls for the teleworker site, make sure that the users understand that they cannot place emergency calls from their work phone that is outside of the office.

Device Mobility

When you do not configure any remote site in the Cisco Unified Communications Manager Business Edition 3000, the endpoints will register with the central site even when the endpoint is situated in a different subnet. The gateway configured in a different subnet will not register with the central site automatically. You can configure the gateway manually in the central site.

Routing Calls Through Gateways

The calls are routed through the Cisco Unified Communications Manager Business Edition 3000 sites based on the PSTN access settings at **Connections > Sites > Add Site > PSTN Access**. Setting the gateway usage allows you to control the routing of PSTN calls through various gateways.

You can select certain local gateways or all the gateways to route the PSTN calls as required. You can also custom select the gateways. The PSTN calls are routed in a top-down order among the gateway groups.

"All Gateways" feature is available in countries that allow toll bypass. When All Gateways is selected, the order of the gateways for routing the PSTN calls is as follows:

1. Local Site gateways
2. All nonlocal gateways of the same type ordered based on the Site name
3. Gateway within a site
   a. SIP trunk
   b. E1/T1 PRI
   c. T1 CAS
   d. FXO
4. Multiple gateways of the same type sorted based on Connection Description

When Local Gateways is selected, the order of the gateways for routing the PSTN calls is as follows:

1. SIP trunk
2. E1/T1 PRI
3. T1 CAS
4. FXO

When Custom selection is selected, the options allows you to select and order the gateway connection from the complete gateway list in Cisco Unified Communications Manager Business Edition 3000.

Note Ensure that you understand the regulations of your country before you configure call routing. Configuring call routing incorrectly may violate the toll bypass rules in your country.

The list of the gateways selected is sorted in the following order of precedence:

1. Site
2. Connection
3. Description

Note The connection and the description that correspond to a gateway are input in the PSTN Connection flow while provisioning PSTN Connections.

When a site does not have access to any PSTN gateway, the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface displays an error.

Error Message `This site does not have access to any PSTN gateway. Phones on this site will be unable to make/receive PSTN calls.`

Recommended Action Ensure that you select the gateway correctly.

Logical Partitioning

The Logical Partitioning feature is required for countries with telecom restrictions, such as India.

Logical Partitioning prevents toll bypass of the PSTN calls through the Cisco Unified Communications Manager Business Edition 3000 for countries that adhere to telecom regulations. Logical Partitioning is enabled in the Country Pack for countries that have regulations.

The Logical Partitioning feature is not available for countries that do not have regulations. Currently, the Logical Partitioning feature is enabled for India.

You can select one of the following features for Logical Partitioning:

- Default—Allows you to enable default policies such that each site can route calls using the local gateway. This is a basic requirement for deployments and is provided by default.
- Custom—Allows you to choose the required gateways such that multiple sites located in the same trunk area can share a common gateway. This allows provisioning of a Policy matrix between any sites.

For More Information

- DHCP Usage for Acquiring IP Addresses, page 1-31
- Example of Typical Deployment Model, page 1-47
- Working with the Cisco-Provided .xls Data Configuration File, page 3-1
- Sites Settings, page 36-1 (for the Sites page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard and Cisco Unified Communications Manager Business Edition 3000 Administrative Interface)

Usage Profiles

A usage profile allows you to configure most of the user settings for a phone in one place. You can edit an existing usage profile, duplicate an existing usage profile to create a new profile, or add an entirely new usage profile. Each usage profile has a unique name. After you configure your usage profiles, you can assign them to users or to departments, so that the settings in the usage profile apply to the phones that belong to an individual user or to a department.

In the usage profile, you can configure calling privileges for users, phone features, such as barge, Cisco Extension Mobility, and so on, phone hardware functionality, phone applications that may display on the phone, and the phone button template, which controls the order of the buttons and the feature buttons that display on the phone.

You can configure a usage profile through the following methods:

- Through the Cisco-provided .xls data configuration file in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (for initial deployment)
- On the Usage Profile page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (if you are not using the Cisco-provided .xls data configuration file during initial deployment)
- Under **Users/Phones > Usage Profiles** in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface (after initial deployment)

The usage profiles are not phone model specific; all settings that are available in the usage profile do not support all phone models. Before you add or edit an existing usage profile and assign it to a user, determine whether the phone model that is assigned to the user supports the features and functionality that is available in the usage profile. If you configure a setting that is not supported by the phone, the phone ignores the value in the configuration file, and the user cannot use the feature or functionality on the phone.

Note Cisco Unified Communications Manager Business Edition 3000 supports a maximum of 30 usage profiles.

The usage profiles in Table 1-3 come with your system by default; that is, they display on the Usage Profile page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard (and in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface, unless you delete them from the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard). You can delete these usage profiles, you can edit these profiles, or you assign them to your users without modification.

Table 1-3 Default Usage Profiles

Type	Considerations
Standard	Consider applying this usage profile to most of your users. If you use the default values in the usage profile: • The user can place and receive local and internal calls; if the site where the phone resides allows these types of calls. • The user can make emergency calls to the local center that handles emergencies for the municipality; if the site where the phone resides allows emergency calls. • The user can use barge if the phone supports barge; barge allows a user to interrupt a call without permission from other participants on the call. • The user can use call pickup if the phone supports call pickup; call pickup allows the user to pick up calls for another user. • The user can create speed dials if the phone has 3 or more buttons on it. • The user can use the speakerphone and a headset if the phone supports that functionality.
Manager	Consider applying this usage profile to the managers in the company. If you use the default values in the usage profile: • The user can place and receive all types of calls, including long distance and international calls; if the site where the phone resides allows these types of calls. • The user can make emergency calls to the local center that handles emergencies for the municipality; if the site where the phone resides allows emergency calls. • The user can use barge, call pickup, and call park if the phone supports these features. Call park allows a user to park a call on one phone and then pick up the call on another phone. • The user can create speed dials if the phone has 3 or more buttons on it. • The user can use the speakerphone and a headset if the phone supports that functionality.

Table 1-3 Default Usage Profiles (continued)

Type	Considerations
Assistants	Consider applying this usage profile to the assistants that support the managers in the company. If you use the default values in the usage profile: • The user can place and receive internal, local, toll free, and long distance calls; if the site where the phone resides allows these types of calls. • The user can make emergency calls to the local center that handles emergencies for the municipality; if the site where the phone resides allows emergency calls. • The user can use barge and call pickup if the phone supports these features. • The user can create speed dials if the phone has 3 or more buttons on it. • The user can use the speakerphone and a headset if the phone supports that functionality.
Power	Consider applying this usage profile to the users that assist with administering the system; for example, to the IT support staff. If you use the default values in the usage profile: • The user can place and receive internal, local, toll free, and long distance calls; if the site where the phone resides allows these types of calls. • The user can make emergency calls to the local center that handles emergencies for the municipality; if the site where the phone resides allows emergency calls. • The user can use barge and call pickup if the phone supports these features. • The user can create speed dials if the phone has 3 or more buttons on it. • The user can use the speakerphone and a headset if the phone supports that functionality.
Common Area	Consider applying this usage profile to departments, which are for phones that are used in public spaces, such as break rooms, and so on. If you use the default values in the usage profile: • On the public space phone, the user that can place and receive internal calls. • The user can make emergency calls to the local center that handles emergencies for the municipality; if the site where the phone resides allows emergency calling. • On the public space phone, the user can use barge if the phone supports this feature. • As the administrator, you can add speed dials if the public space phone has 3 or more buttons on it. • On the public space phone, the user can use the speakerphone and a headset if the phone supports that functionality.

For More Information

- Working with the Cisco-Provided .xls Data Configuration File, page 3-1
- Usage Profiles Settings, page 40-7 (for the Usage Profile page in the Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard and Cisco Unified Communications Manager Business Edition 3000 Administrative Interface)
- Checklists for Users, Departments, Lines, and Phones, page 8-1

Users, Departments, Phones, and Lines

Phones, users, and lines are closely related in Cisco Unified Communications Manager Business Edition 3000. A user, which is an employee from the company, uses the usage profile on a phone that is supported in Cisco Unified Communications Manager Business Edition 3000. Because phones and users are closely related, you cannot configure a phone without first configuring a user or department that has an extension (line) from the dial plan assigned to it.

In the Cisco Unified Communications Manager Business Edition 3000 system, a user becomes an owner of a phone when you assign the user extension to line 1 on the phone. If the user is an owner of the phone, the phone uses the usage profile that is assigned to the user.

Tip A department is a special kind of user that is reserved for public-space phones; this user is reserved for phones in cafeterias, lobbies, break rooms, and so on. A public-space phone cannot support Reach Me Anywhere. You do not configure passwords or phone PINs for departments, unlike users (**Users/Phones > Users**).

Example of How User or Department Ownership Works for a Phone

Step 1 If you have not already done so, add the user or department configuration; for example, add the user by selecting **Users/Phones > Users (or Department)** in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface.

- When you add the user or department, you must assign a usage profile that you want to be available to the phone.
- When you add the user or department, assign an extension to the user or department, based on the dial plan that is set up for your system.

Step 2 Add the phone and assign the extension to line 1 on the phone. The user or department becomes the owner of the phone, and a user can use the features and functionality from the usage profile on the phone if the phone supports the functionality.

Tip You can create a shared line between an IP phone and an analog phone using the same user extension.

Note Refer to http://www.cisco.com/en/US/docs/routers/access/vg224/software/configuration/guide/scgvoip.html for information on VG224 configuration.

On the User or Department page, you create lines for the user or department based on the dial plan that you set up; for example, if your dial plan is set up for 4-digit dialing with an extension range of 2000-2999, you can assign an extension such as 2555 to the user or department. You can set up to 6 extensions in a prioritized list for each user or department. The usage profile that you assign to the user or department applies to all extensions that are in the prioritized list.

You assign user extensions (lines) to the phone on the Phone page. You can set up to 6 lines as a prioritized list on the Phone page, even if the phone does not support 6 lines. For each line, you can define Call Forward All and External Caller ID.

The phone button template that is configured in the usage profile determines the order of line buttons and the types of functionality that displays next to the line buttons on the phone; for example, for all lines except line 1, which must be a line because of user-phone ownership, you can designate a line as a speed dial, line, or feature button (Mobility, Meet-Me Conference, and so on). When the phone button template that is configured in the usage profile designates the buttons as lines, the system orders the lines that are assigned on the Phone page based on the prioritized list, with the first line being designated for line 1, the second line in the list being used for the next button on the phone that is designated as line 2, and so on. In the Usage Profile, you must establish the purpose for 12 line buttons on the phone, even when the phone does not support 12 buttons.

The users can create up to 12 speed dials in a prioritized list in the Cisco Unified Communications Manager Business Edition 3000 User Preferences Interface, even when the phone does not support 12 speed dials.

Significant Behavior of SIP Trunk

SIP trunk exhibits significant behavior while processing PSTN calls in Cisco Unified Communications Manager Business Edition 3000.

The SIP trunk supports the following features:

- Basic Outgoing/Incoming Call
- Early and Delayed Offer
- PRACK
- Session Timers
- PAI and RPID for Identity
- DTMF using RFC 2833, KPML, and Unsolicited Notify
- Hold/Resume, Transfer, Conference, Forwarding
- Diversion Header
- MWI
- Options Ping

The following sections details the behavior of SIP trunk specific to Cisco Unified Communications Manager Business Edition 3000.

- Incoming 302—Moved Temporarily, page 1-43
- Incoming OOD REFER Message Handling, page 1-44
- Calling Party Transformation, page 1-45
- Connected Party Transformation, page 1-45

Incoming 302—Moved Temporarily

The SIP service provider sends 302 message to the Cisco Unified Communications Manager Business Edition 3000 to reach to SIP Uniform Resource Identifier (URI) as specified in the Contact header field. By default, the SIP trunk will use the “Redirect by Application” feature to reroute the PSTN calls. The SIP trunk checks the class of service and the privilege of the calling user before redirecting the calls.

In a SIP environment, the two possible ways of forwarding or redirecting a call are sending SIP “302 Moved Temporarily” with one or more “Contact:” headers as response to an INVITE or are sending a new INVITE to the new destination.

Cisco Unified Communications Manager Business Edition 3000 sends an INVITE to the new destination upon reception of “302 Moved temporarily” but it cannot generate such a response. The “302 Moved temporarily” response is received when “call forward” is enabled on a SIP endpoint.

The service provider can be configured to react in different ways when a SIP call is forwarded or redirected by one of the call legs. Most often, a SIP peer sends a SIP response “302 Moved temporarily” with the new destination URI appearing in the “Contact:” header of the message.

Similar to Call Transfer Supplementary service, the service provider can be configured to pass along the “302 Moved Temporarily” to the originating call leg or to react to it and send a new INVITE on behalf of the forwarded (FWED) party.

In “Redirect by Application” configuration, the SIP trunk passes the control to Redirecting Application layer for handling the rerouting. The “Rerouting Calling Search Space” configured on SIP trunk is passed to allow further check on class of service and privilege of the calling user for redirection to the new Contact. To set these parameters, refer to Connection Type: SIP Trunk, page 31-24.

The Redirect by Application feature of the SIP trunk allows the Cisco Unified Communications Manager Business Edition 3000 to do the following:

- Apply digit analysis to the redirected contacts to ensure that the calls are routed correctly
- Prevent DOS attack by limiting the number of redirection (recursive redirection) that a service parameter can set
- Allow other features to be invoked while the redirection is taking place

Calls get redirected to a restricted phone number (such as an international number) due to handling redirection at the stack level to route the calls without blocking. This behavior occurs when the Redirect by Application check box is not checked in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface.

Note In case of multiple redirections before final redirection over SIP trunk, the maximum of two Redirection headers will be sent over SIP trunk, that is the original called party and last called party information.

Incoming OOD REFER Message Handling

Out-of-dialog REFER (OOD-R) enables remote applications to establish calls by sending a REFER message to Cisco Unified Communications Manager Business Edition 3000 without an initial INVITE. After the REFER is sent, the remainder of the call setup is independent of the application and the media stream does not flow through the application. The application using OOD-R triggers a call setup request that specifies the Referee address in the Request-URI and the Refer-Target in the Refer-To header.

Cisco Unified Communications Manager Business Edition 3000 handles the incoming REFER from a SIP trunk service provider.

Calling search spaces determine the partitions that calling devices can search when they attempt to complete a call. The out-of-dialog calling search space is used when a Cisco Unified Communications Manager refers a call (B) that is coming into SIP user (A) to a third party (C) when no involvement of SIP user (A) exists. In this case, the system uses the out-of-dialog calling search space of SIP user (A). The third party (C) is either an internal extension or an auto attendant client.

CUBE on Cisco ISR8xx, the session border element in the Cisco Unified Communications Manager Business Edition 3000 SIP trunking solution, does not support the OOD REFER inter-operability. The service provider cannot send OOD REFER through CUBE on Cisco ISR8xx through Cisco Unified Communications Manager Business Edition 3000.

Calling Party Transformation

The mid-call SIP messages, namely reINVITEs, UPDATE or 200 OK sent from the calling party direction on SIP Trunk from Cisco Unified Communications Manager Business Edition 3000, carry the URI identity containing a number in user portion. This occurs during Hold/Resume, Transfer and so on, which result in transactions inside a SIP dialog.

By default, the Cisco Unified Communications Manager Business Edition 3000 sends configured extension number only, while expected number for inter-operability with the SIP service provider is the DID or full number (for example: Office code + Subscriber code).

The Cisco Unified Communications Manager Business Edition 3000 preconfigures the “Calling party Transformation” on SIP Trunk used to connect to the session border elements.

By provisioning the “Calling Party Transformations” feature for a SIP Trunk, the SIP dialogs established as part of outbound SIP Trunk calls always use the transformation for upsizing the number sent in P-Asserted ID or Remote Party ID headers of an outbound SIP message.

Note For the calling party transformation to function correctly, ensure that the External Caller ID is defined for the user. You can edit the **External Caller ID** on **Users/Phones > Users > Edit User > General** page.

Connected Party Transformation

The backward direction SIP messages namely 183, 200 Ok or mid-call UPDATE/INVITE messages from the connected party SIP trunk on Cisco Unified Communications Manager Business Edition 3000 carry the URI identity containing a number in user portion.

By default, the Cisco Unified Communications Manager Business Edition 3000 sends configured extension number only, while the expected number for inter-operability with the SIP service provider is the DID or full number (for example: Office code + Subscriber code).

To send the full number (DID and so on) on SIP trunk, the Cisco Unified Communications Manager Business Edition 3000 preconfigures the “Connected Party Transformation” on SIP trunk used to connect to the session border element. By provisioning the “Connected Party Transformation” feature for a SIP trunk, the SIP dialogs established as part of inbound SIP trunk calls always use the transformation for upsizing the number sent in PAI/RPID headers of an outbound SIP message.

Note For the connected party transformation to function correctly, ensure that the External Caller ID is defined for the user. You can edit the **External Caller ID** on **Users/Phones > Users > Edit User > General** page.

Attendant Group

The Attendant Group page allows you to add or remove users who will be associated with all the phones in the phone list of Cisco Unified Communications Manager Business Edition 3000 (choose **Users/Phones > Attendant Group**). You can add until ten users to the Attendant Group. The system displays an error message when you click to add more than ten users.

Note You must have administrator account, browser access, and Internet access to add or remove users.

Note Attendant Group requires an additional enhanced user license for each group member. If the number of licenses is insufficient, a new user will not get associated to the Attendant Group.

Attendant Group has an Available list of users, which displays all the users in the Cisco Unified Communications Manager Business Edition 3000 who are not associated with Attendant Group, and a Selected list, which displays all users who are associated to Attendant Group. You can move users from Available to Selected to associate a user to Attendant Group.

After adding a user to the Selected list you must save the user in the Selected list. You can also remove users from the Selected list. The users removed from the Selected list are moved to the Available list. For more information, see (Setting Up Attendant Group, page 8-13 and Attendant Group Settings, page 11-1).

Hunt Lists

A hunt list consists of a group of extensions that can answer calls. You set up hunt lists for the purpose of distributing calls amongst the users that belong to the group. For example, if the company does not have a receptionist and several users must answer calls, consider setting up a hunt list to ensure that calls are evenly distributed amongst the users that belong to the hunt list. For example, if several administrative assistants must share the call load for several managers, consider setting up a hunt list to ensure that all calls are answered quickly. You configure hunt lists on the Hunt Lists page in the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface. (Select **Users/Phones > Hunt Lists**.) You can create as many hunt lists as you want.

Extensions in a hunt list can belong to users or departments. You can assign any user or department extension to a hunt list, but only those extensions that are assigned to phones can actually answer the calls. An extension may belong to more than one hunt list.

Pilot extensions for hunt lists must be in the extension range(s), but pilot extensions do not get assigned to users or departments.

For the members (extensions) that belong to the group, you can select one of the following distribution methods:

Tip An idle member is not servicing any calls. An available member is on an active call but is available to accept a new call. A busy member cannot accept calls.

- Top Down—Cisco Unified Communications Manager Business Edition 3000 distributes a call to idle or available members (extensions) starting from the first idle or available member of a hunt list to the last idle or available member.
- Circular—Cisco Unified Communications Manager Business Edition 3000 distributes a call to idle or available members starting from the (n+1)th member of a hunt list, where the nth member is the member to which Cisco Unified Communications Manager Business Edition 3000 most recently extended a call. If the nth member is the last member of a hunt list, Cisco Unified Communications Manager Business Edition 3000 distributes a call starting from the top of the hunt list.
- Longest Idle Time—Cisco Unified Communications Manager Business Edition 3000 only distributes a call to idle members, starting from the longest idle member to the least idle member of a hunt list.
- Broadcast—Cisco Unified Communications Manager Business Edition 3000 distributes a call to all idle or available members of a hunt list simultaneously.

Caution Do not put extensions that are shared lines in a hunt list that uses the Broadcast distribution algorithm. Cisco Unified Communications Manager Business Edition 3000 cannot display shared lines correctly on the phone if the extensions are members of a hunt list that uses the Broadcast distribution algorithm.

For More Information

Setting Up the System So that Incoming Calls Reach the Auto Attendant if the Operator is Not Available, page 8-15

Example of Typical Deployment Model

Figure 1-12 shows an example of a typical deployment model that includes the central site, two remote sites, and the teleworker site. The central site includes the gateway, server, several routers, a Cisco VG224 Analog Phone Gateway, and IP and analog phones. The remote sites include phones and routers. For more information on the example, see the following:

- Figure 1-12
- Table 1-4
- Central Site (Example)
- Teleworker Site (Example)
- Remote Site 1 (Example)
- Remote Site 2 (Example)

Figure 1-12 Example of a Typical Deployment Model

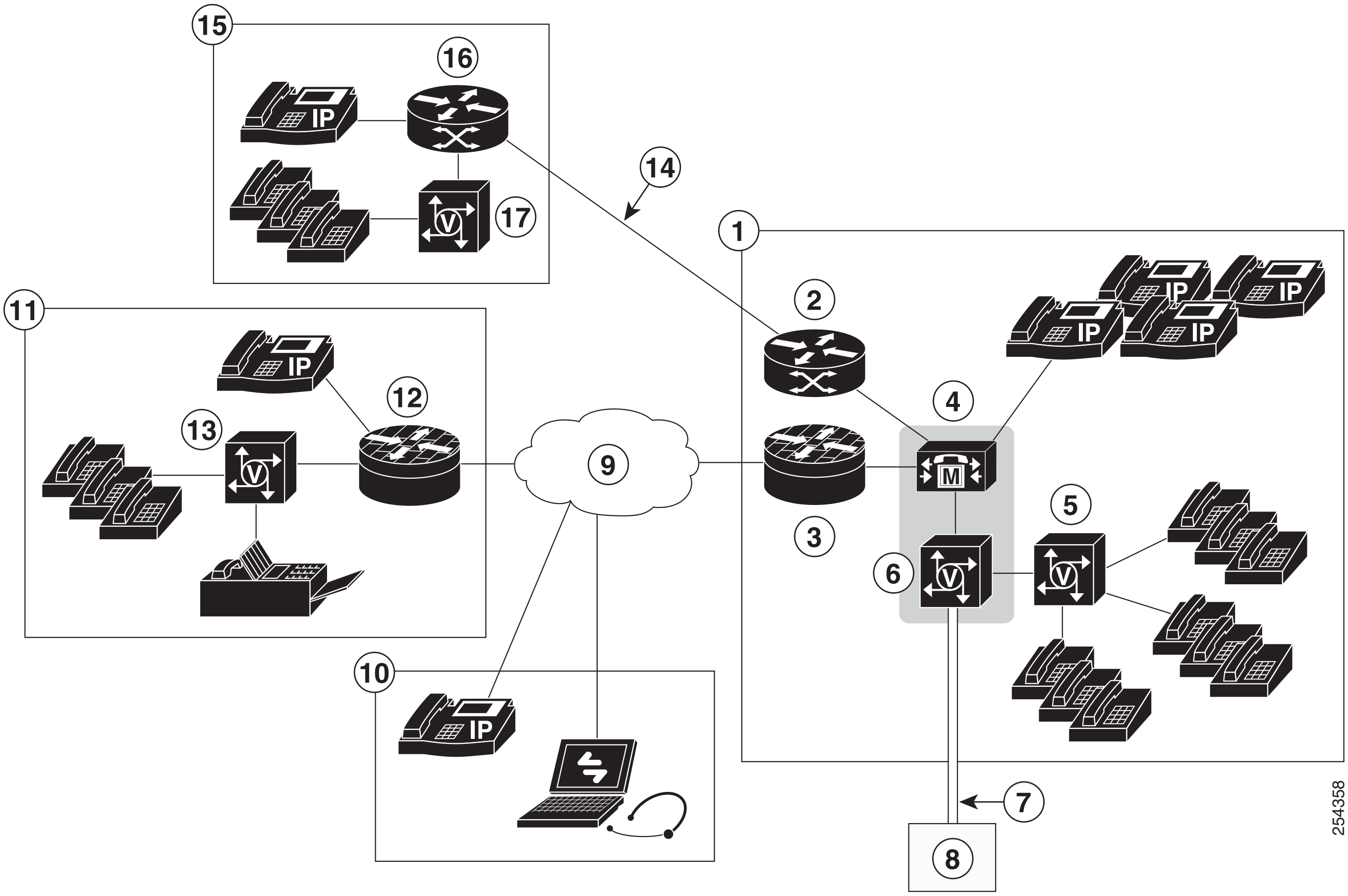


Table 1-4 Components of a Typical Deployment Model

Number	Description
1	Central Site
2	Router
3	Router with VPN
4	Cisco Unified Communications Manager Business Edition 3000 Server
5	Cisco VG224 Analog Phone Gateway
6	Cisco 2901 Integrated Services Router (ISR) (Gateway)
7	Dual E1/TI connection
8	Service Provider (Telecommunications Company)
9	Internet
10	Teleworkers Site
11	Remote Site 1
12	Router with VPN
13	Cisco VG224 Analog Phone Gateway
14	Dedicated WAN link

Table 1-4 Components of a Typical Deployment Model (continued)

Number	Description
15	Remote Site 2
16	Router
17	Cisco VG224 Analog Phone Gateway

Central Site (Example)

Figure 1-12 illustrates that the central site is where your server, gateway, and the majority of your phones/users are located. To use fax or analog phones at the central site, the Cisco VG224 Analog Phone Gateway is set up specifically for the central site. The gateway allows access to the PSTN through a dual T1/E1 connection that is provided by the service provider (telecommunications company).

Although not included in Figure 1-12, a switch exists between the server and the Cisco Unified IP Phones.

Teleworker Site (Example)

Figure 1-12 illustrates that the teleworker site has phones and personal computers that connect to the central site through the Internet. A router that supports VPN connects the teleworker site to the central site.

Tip Quality of service (QoS) may not be available for the teleworker site.

Remote Site 1 (Example)

Figure 1-12 illustrates that remote site 1 connects to the central site through a router that supports VPN (and connects to the Internet). Analog phones and fax are used at remote site 1, so the Cisco VG224 Analog Phone Gateway is set up for these phones and functionality specifically for this site. Fewer phones are included in remote site 1 than at the central site.

Remote Site 2 (Example)

Figure 1-12 illustrates that remote site 2 connects to the central site through a dedicated WAN link. Analog phones are used at remote site 2, so the Cisco VG224 Analog Phone Gateway is set up to support the analog phones at this site.