

CHAPTER 3

Deploying Devices

After all predeployment tasks are completed, devices can be added to the system. This chapter discusses the tasks required to add certain devices to the CallManager cluster.

A number of types of devices can be added to the CallManager. The devices discussed in this chapter fall into one of two categories, clients or gateways. Other devices, such as gatekeepers, are discussed in future chapters.

We start by looking at clients, more specifically phones. There are a number of different models of Cisco IP phones, but the task of adding each is very similar. Next, gateways are covered. Gateways allow connectivity to another system such as the Public Switch Telephone Network (PSTN) or another private branch exchange (PBX). In CallManager 4.0 there are over 30 different types of gateways, and although the function of each is similar, the configuration of each varies. It is not possible to provide a step-by-step guide for the configuration of each, but this chapter does include detailed steps on how to configure the most popular types.

NOTE In all the steps in this book, I have tried to cover all the parameters that appear on the screen. Because each version of CallManager adds fields, all the fields you see in these steps may not appear on your screen, or additional fields may be present, due to the version of CallManager you are running. Regardless of the version you are running, the steps in this chapter will help you walk through the process. Not all the parameters must be configured. Some can remain at default, whereas others are not required at all. The end of the names of required parameters are marked with an asterisk (*). In some cases you may need to configure only the required parameters for a device to function. It is a good idea, however, to review each parameter so that you can be certain the device is configured exactly the way you want it.

Adding Clients

Depending on the environment, adding a phone can be as simple as plugging it into a port that has connectivity to the CallManager. Although it is possible to configure a CallManager to allow phones to be automatically added simply by connecting them to the network, it is

not always wise or desired. This section explores four ways phones can be added to the system, but before adding phones to a system, there are a few more components to configure.

You are probably thinking, “Wait a minute, the last chapter discussed predeployment tasks.” That’s right, it did, but those were the general or global settings that should be configured. This section looks at settings that are specific to phones.

NOTE Which came first the chicken or the egg? This is a question that has plagued the minds of great thinkers from the beginning of time. Okay, maybe it hasn’t, but a similar issue sometimes occurs when teaching new technologies. When two separate concepts are interdependent, it may be difficult to grasp either concept until you understand both of them. This being said, you will notice throughout this chapter that new concepts and components are mentioned that are not discussed in detail until later in this book. In these cases, a brief description is offered for each new concept, as is a reference to the chapter that lends greater detail. You may choose to jump ahead to gain a better understanding or just accept the brief description, knowing more detail is offered later.

Defining Device Settings

Before adding phones, it is recommended, but not required, that some device settings be configured. Configuring the device setting first will most likely save time. These settings include Phone Button Templates, Softkey Templates, and Device Defaults. The following section explains the function of each of these and how to configure them.

Phone Button Templates

Some Cisco IP phones have buttons that can be configured for various functions. The most popular functions these buttons support are lines and speed dials. When configured as lines, extension numbers can be assigned to the buttons. Each phone must have at least one button configured as a line. When a button is configured as speed dial, administrators or users can assign a speed dial number to it, and because most Cisco phones have only a few of these buttons (one to eight), the number of speed dials that can be defined in this way is limited. The user can access additional speed dials by using a feature known as abbreviated dial, which is discussed later in this chapter.

In addition to lines and speed dial functions, these buttons may be able to be configured for other functions, depending on the phone model. There are also phones on which the buttons are not configurable. Table 3-1 shows which phones have configurable buttons and which do not.

Table 3-1 Configurable/Non-Configurable Phone Buttons

Configurable Button Phones	Fixed Button Phones
30 VIP	7902
12 SP(+)	7905
7910	7912
7920	7935
7940	7936
7960	ATA 186–188
7970	VCG

The exact function that can be assigned to buttons varies depending on the model of the phone. The 7940, 7960, and 7970 buttons can be configured as Service URL, privacy, speed dial, or line buttons. By configuring a button as a Service URL, a user may access a phone service by simply pressing the button. A button that is configured as a privacy button allows a user to make a call private. This feature is used on shared lines and prevents the other shared-line phone from entering the call. It is also possible that a button has no function assigned to it. When a button serves no function, it is labeled as “none.”

NOTE

The 7920 buttons can be configured only as lines or speed dials.

When a phone is added to the system, a phone button template is associated to the phone. The template is used to determine the function each button will serve. If the phone is added via auto-registration, it uses the template defined under device defaults. Device defaults settings are discussed later in this chapter. The creation of these templates is simple because, on most phones, there are only two functions for these buttons.

To create phone button templates follow these steps:

Step 1 From within CCMAadmin, select **Device>Device Settings>Phone Button Template**.

Step 2 Select the **Add a New Phone Button Template** link.

Step 3 On the next page you must select an existing phone button template to copy. This is because a new phone button template must be based on an existing template. From the drop-down list in the field labeled Phone Button Template, select the standard template for the correct model of phone. For example, if you are creating a phone button template for a 7960, select the Standard 7960 Template.

Step 4 Then click the **Copy** button.

Step 5 A screen like that shown in Figure 3-1 displays. At first glance you may think that you have selected the wrong phone type because it shows 34 buttons instead of the six you were expecting. Because the 7960 can support two 7914 expansion modules, which support 14 buttons apiece, 34 total buttons are possible. In the Button Template Name field enter a descriptive name.

Figure 3-1 Phone Button Template Configuration



Note It is recommended that the Button Template Name identify the configuration of this template. For example, if the template is configured with two lines, three speed dials, and a privacy button, a good name would be “7960 2-3 w/Privacy.”

Step 6 Assign the proper function to each line by selecting Service URL, Privacy, Line or Speed Dial, or None from the drop-down list next to the button number.

Step 7 Next to the feature field is a label field. Enter a descriptive label in this field such as Speed Dial 2 or Line 3.

Step 8 Click the **Insert** button to save the new template.

Softkey Template

In addition to the buttons that can be used for lines and speed dials, Cisco IP phones have buttons that are referred to as *softkeys*. These buttons (keys) allow the user to access features of the phone such as hold, transfer, conference as well as many others. The function of each key changes depending on the state of the call. Because there are often more available functions configured than physical softkeys, the last softkey functions as a toggle key allowing the user to scroll through all the available options. The softkey template allows you to determine what functions are available on the phone and in what order they display. Much like phone button templates, the softkey templates can be associated directly with a phone. However, as you saw in the last chapter, the softkeys templates are also associated to device pools, which are in turn associated with phones. By associating the softkey template to a device pool, you can easily and quickly assign the softkey template to a large number of phones. If different softkey templates are associated with a phone at both the phone level and the device pool, the one at the phone level takes precedence.

The following steps take you through the process of creating a softkey template.

Step 1 From within CCMAdmin, select **Device>Device Settings>Softkey Template**.

Step 2 Click the **Add a New Softkey Template** link.

Step 3 On the next page you must select an existing softkey template on which to base the new template. Select a template from the drop-down list in the **Create a softkey template based on** field.

Step 4 Then click the **Copy** button.

Step 5 On the next screen, enter name and description for the new template. The description should help identify the features associated with the template.

The Application field lists the applications that are available in this template and cannot be changed in this screen.

Step 6 Click the **Insert** button to add the new template.

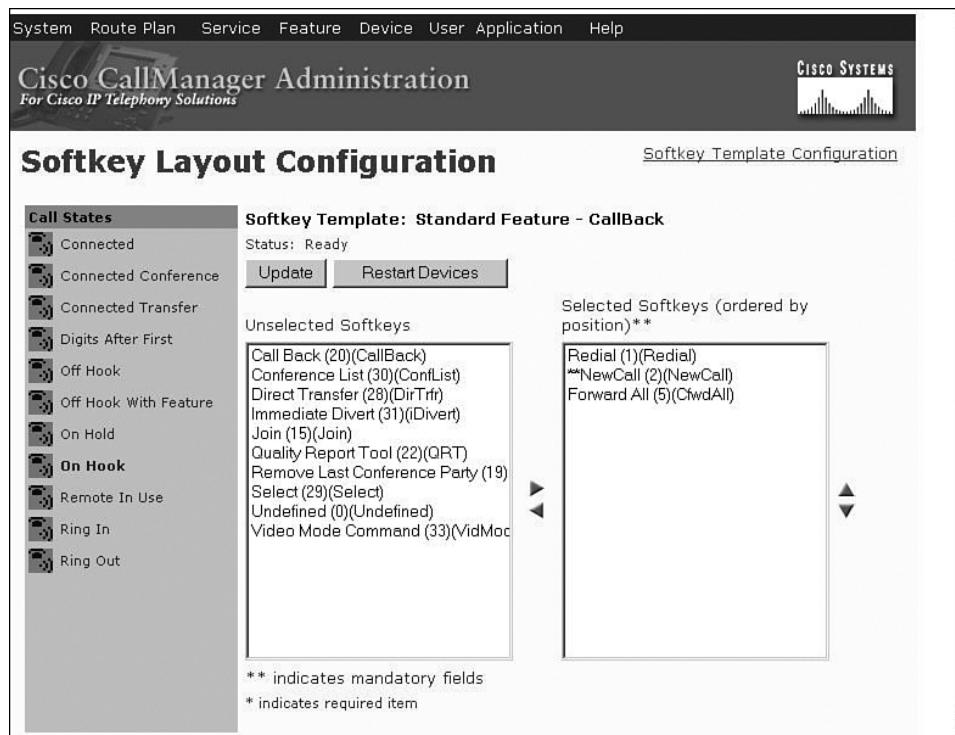
Step 7 After the template is added, you must configure it. If you want to add applications that are found on other softkey templates, click the **Add Application** button. If you do not wish to add applications from an existing template, skip this and the next step. For example, you would not add an application if you are simply moving or adding standard softkeys to the template.

Step 8 If you selected Add Application in the new window that displays, select the standard softkey template that contains the application you want to add to the new template and click **Insert and Close**.

Step 9 Click the **Configure Softkey Layout** link to modify the current layout.

A screen similar to that found in Figure 3-2 displays. On the left side of the screen, all of the states of the phone are listed. Because the softkeys change depending on the state of the phone, each state must be configured separately. In Figure 3-2 the new template allows access to the call-back feature. In this example the call-back softkey must be configured for all states from which it might be accessed which are On Hook and Ring Out.

Figure 3-2 Softkey Layout Configuration



Step 10 Determine which call state you want to modify and select it from the list. For instance, if you wanted to add a softkey that would appear on the phone while you were on a call, you would select **Connected**.

Step 11 The softkey currently assigned displays in the box on the right side of the screen labeled Selected Softkey. The softkeys that may be added display

in the box on the left side labeled Unselected Softkeys. To add unselected softkeys, click once on the **softkey** and then click on the **top arrow** of the two arrows that display between the two boxes.

Note Because there are a limited number of softkeys on the phone, it is common to have more softkeys assigned than buttons on the phone. In this case the last button becomes a “more” button that acts as a toggling mechanism to allow users to access the other features. It is important to understand that the more button will automatically appear on the phone and is not something you configure on the template.

Step 12 To remove a current softkey, click on the softkey you wish to remove and click the **bottom arrow** of the two arrows that display between the boxes.

Step 13 The softkeys display on the phone in the same order that they display in the Selected Softkeys box. To change the order in which they display, highlight the desired softkey and click the **up or down arrow** that displays on the left side of the Selected Softkey Box.

Note If softkeys are used in more than one call state, for instance the callback softkey is available in the On Hook and Ring Out call states, pay attention to the position of the softkey in each of the call states. It's a good idea to keep the softkey in the same position for each call state if possible. If it is desirable to leave a softkey position unused, the Undefined softkey can be used as a placeholder.

Step 14 After you have modified all the desired call states, click the **Update** button.

NOTE When modifying an existing softkey template, you need to reset the devices that are associated with the templates. To do this, click the Restart Devices button on this page. Take care when resetting devices, because resetting causes the phones to be unusable while they reset. This will not affect phones that are currently on a call. The reset will occur once the phone is idle. It is always recommended that whenever a reset must be performed on a large number of phones, it be done off hours to minimize the impact on the end user.

Device Defaults

When a phone boots, it requests a configuration file from the CallManager Trivial File Transfer Protocol (TFTP) server. The file that it requests is associated with the phone's Media Access Control (MAC) address. If the phone has not previously been registered with a CallManager on the system and has not been manually added, no configuration file exists. When auto-registration is being used, a phone uses a default configuration file that defines how it is to attach to the system and register. The Device Defaults settings define the values used.

Three settings are defined under the Device Defaults settings as follows:

Load Information

This is the ID of the firmware load that the device should be running. When CallManager is shipped, it includes a current version of the firmware loaded for each device. From time-to-time the firmware is upgraded to offer additional features or patches. By editing this field you can specify the ID of the new load. The next time the device boots, it downloads the new firmware load.

Device Pool

This field allows you to define which device pool is used when auto-registration takes place.

Phone Template

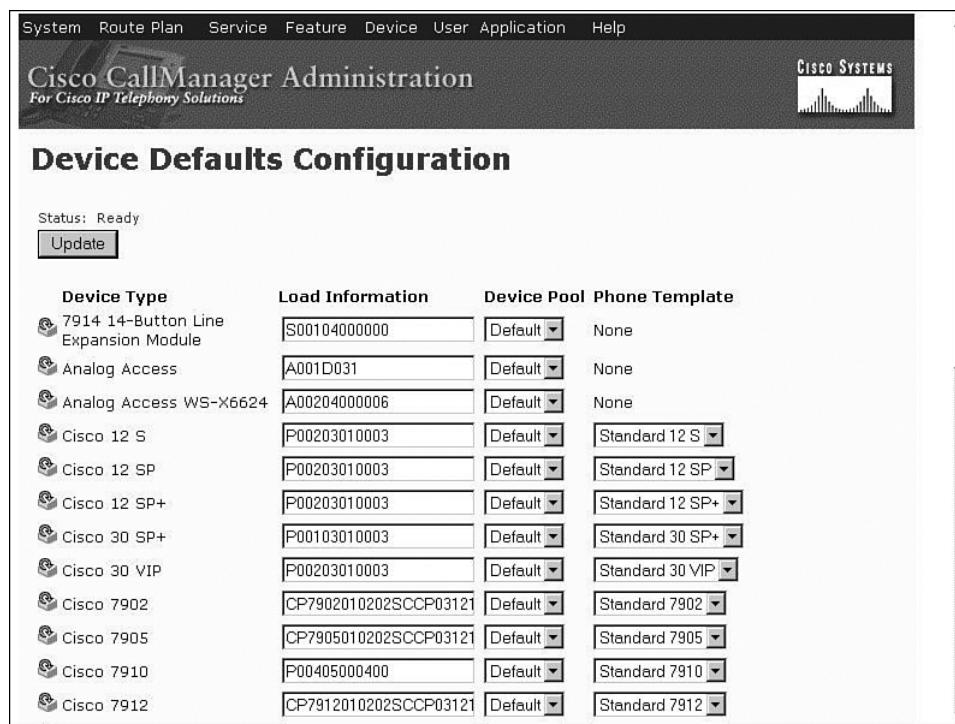
This field allows you to define what phone button template devices use when auto-registration takes place.

The following steps show how the device defaults are configured.

Step 1 From within CCMAdmin, select **System>Device Defaults**.

Step 2 A screen similar to that shown in Figure 3-3 displays. Most often the Load Information can remain at default. However, from time-to-time new loads need to be deployed. In this case, locate the name of the device for which you are updating the load and enter the new load ID in the Load Information field. Make sure you enter the ID correctly and that the ID is for the correct device. Entering an invalid or incorrect ID in this field can cause the device to fail.

Note When a new load is to be used, it must first be copied to the TFTP server so that the device will be able to download it.

Figure 3-3 Device Defaults Configuration

Step 3 Next, you edit the Device Pool. To change the default device pool for a device, simply click the **down arrow** in the Device Pool field associated with the device and select the desired device pool from the list that displays.

Step 4 The last item that can be selected on this page is the phone button template. To change this item for a device, click the **down arrow** in the Phone Template field associated with the device and select a template from the list that displays.

Adding Phones

When a phone is added to the system, information about the phone is entered into the SQL database. This information defines nearly every aspect of the device. Phones can be added to the system in a number of ways, but the net effect is the same.

Four methods for adding phones are explored in this chapter starting with the method known as auto-registration. Before delving too deeply into this method, let's take a quick look at all four.

- 1 Auto-Registration—allows phones to be plugged into the system and automatically register. An extension is assigned to the phone from a range defined by the administrator.
- 2 Manual—all information for the phones manually entered before the phone is plugged in.
- 3 Bulk Administration Tool (BAT)—information for a large number of phones inserted using BAT. This is done by entering the information into a Comma Separated Value (CSV) file and then using BAT to insert the information into the CallManager database.
- 4 Tools for Auto-Registration Phone Support (TAPS)—Similar to BAT except that the MAC address is not entered in the CSV file and additional steps are required when the phone is plugged into the network. However, even with the additional steps, it is quicker than just using BAT when performing a large installation.

Auto-Registration

As the name implies, using this method allows phones to register to the system by merely being plugged into the CallManager network. Although this method has the advantage of allowing you to quickly add phones, it has its disadvantages as well. When a phone is added this way, it is assigned an extension number from a range of numbers that you define. It assigns these extensions in a first-come-first-served fashion. For instance, if you defined the range to be from 2000 to 2100, the first phone to register would receive the extension 2000, the second 2001, and so on. You can see how this may not be desirable. If you are fortunate enough to be performing a deployment in which you can assign extensions as you wish, then this may not be a concern. However, in situations such as when you are replacing an existing PBX, you will most likely want to reuse the existing extensions, hence the auto assignment of extensions may not be desirable. Auto-registration may still be used in these environments; you will just need to modify the extensions that were automatically assigned.

Rogue phones are another anomaly that can arise when auto-registration is used. If improperly configured, it is possible for a phone to be added to the system without your knowledge. The addition would, of course, have to be made by someone who has a Cisco IP phone and is able to plug it into your network. Although this may seem less than likely, it is still important to ensure that the dialing capabilities be restricted for any rogue phones that may find their way onto your system. This is done by defining a Calling Search Space (CSS) for auto-registration in the device pool. CSS defines a device's calling privileges. This concept is discussed in greater detail in Chapter 5: Configuring Class of Service and Call Admission Control. Choose a CSS that has limited access, perhaps only internal or local access. This, however, also has a drawback. Limiting the CSS of auto-registered phones means that all phones that are added using auto-registration will have a limited CSS. This means for a phone that was added through auto-registration to have greater calling privileges, the CSS must be changed. Often auto-registration is used during the initial deployment and then turned off.

As you can see, using auto-registration has both benefits and drawbacks. The choice of whether to use it depends on the environment and is ultimately up to you.

If all the previously discussed predeployment tasks have been completed, very little additional configuration is required to implement auto-registration. All that needs to be done is to assign an extension range and enable auto-registration at the CallManager Group. If you choose to use auto-registration, the steps that follow help walk you through these tasks.

Step 1 From within CCMAdmin, select **System>Cisco CallManager**.

Step 2 Enter search criteria in the search field, and click **Find**. You may also leave the search field blank and click **Find**. This results in all CallManagers being displayed.

Step 3 From the list that displays, select the **CallManager** on which you want to enable auto-registration.

Step 4 A screen similar to that shown in Figure 3-4 displays. Enter the starting extension number in the **Starting Directory Number** field and the ending extension in the **Ending Directory Number** field.

Figure 3-4 CallManager Configuration for Auto-Registration

The screenshot shows the 'Cisco CallManager Configuration' page for a server named 'DTWA (10.3.3.3)'. The 'Auto-registration Information' section is highlighted, showing fields for 'Starting Directory Number*' (set to 1000) and 'Ending Directory Number*' (set to 1000). There is also a dropdown for 'Partition' and an 'External Phone Number Mask' field. A checkbox labeled 'Auto-registration Disabled on this Cisco CallManager' is checked. Below this, the 'Cisco CallManager TCP Port Settings for this Server' section is visible, listing port numbers for Ethernet Phone, Digital, Analog, MGCP Listen, and MGCP Keep-alive ports.

Setting	Value
Starting Directory Number*	1000
Ending Directory Number*	1000
Ethernet Phone Port*	2000
Digital Port*	2001
Analog Port*	2002
MGCP Listen Port*	2427
MGCP Keep-alive Port*	2428

Step 5 You should notice that in the Auto-Registration Information field the Auto-Registration Disabled checkbox is unchecked at this point. This checkbox automatically becomes unchecked when you enter a starting and ending extension number. DO NOT check this box. Checking this box resets the starting and ending extension numbers.

Step 6 Click the **Update** button to save settings.

NOTE

In the next section, you will be selecting a CallManager Group for auto-registration. If the CallManager Group that you select contains more than one CallManager, you should enter the starting and ending extension numbers for each of the CallManagers in the group.

After an extension range is defined, you need to enable auto-registration for the CallManager group. The steps that follow show how this is done.

- Step 1** From within CCMAdmin, select **System>Cisco CallManager Group**.
- Step 2** Enter search criteria in the search field and click **Find**. You may also leave the search field blank and click **Find**. This results in displaying all CallManager Groups.
- Step 3** From the list that displays, select the CallManager Group on which you want to enable auto-registration.
- Step 4** A screen similar to that shown in Figure 3-5 displays. Check the **Auto-registration Cisco CallManager Group** box.

Note

Only one CallManager Group may be selected as the Auto-registration CallManager Group. If you check this box and there is already an existing Auto-registration CallManager Group, a window displays stating, “You have selected this Cisco CallManager Group to be the Auto-registration Cisco CallManager Group. The old Auto-registration Cisco CallManager Group will be deselected.” Only check that box for the CallManager Group that is responsible for auto-registration.

Step 5 Click the **Update** button to save changes.

Figure 3-5 CallManager Group Configuration for Auto-registration

Manually Adding Phones

In addition to using auto-registration to add phones, you can manually add them. You often add phones manually for a small number of phones (such as 20). If all the predeployment tasks have been completed, the manual adding of phones is quite simple. This section takes you through the steps of the process of manually adding phones.

Manually adding phones has very few drawbacks, aside from the fact that it can be somewhat time consuming. You must enter the MAC address and other information, such as device pool and directory numbers, when you add each phone. Because CCMAdmin is a web-based interface, you must wait for the page to reload when you update information. Although a second or two may not seem too long, when you have to wait for multiple page reloads, it can start to add up. That is just the nature of this type of interface and little can be done about it.

The steps that follow take you through the process of manually adding a phone and explore each field that can be populated for each phone. A brief description is included for any field that has not yet been discussed in this book. The phone used in these steps is a 7960, but the process is very similar for most of the more popular Cisco IP Phone models. When you

add other phone models, a field may not display. This simply means that the model does not require or support that particular field. Because there are a number of steps to this process, section headings are used to mark the point at which each new set of parameters begins. These same headings display on the configuration screen as well, which should help you keep track of where you are.

Step 1 From within CCMAdmin, select **Device>Add a New Device**.

Step 2 Select **Phone** from the drop-down list and click **Next**.

Step 3 A new page displays. From the drop-down list, select the type of phone you want to add. Click Next.

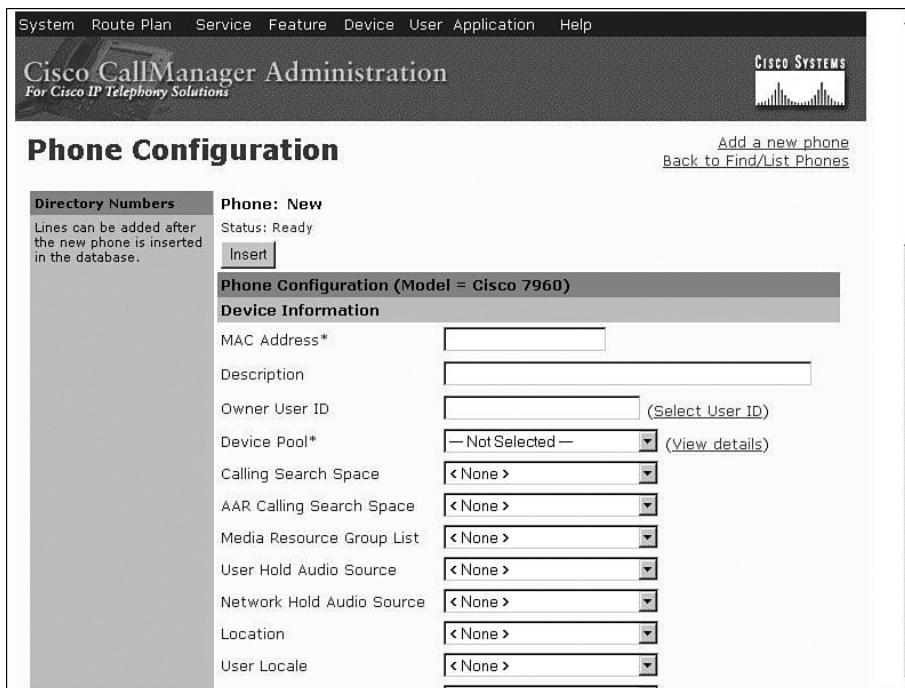
Device Information

Step 4 A screen similar to that shown in Figure 3-6 should display. The first field that must be entered is the MAC Address of the phone. The MAC address can be found on the back of the phone as well as the box in which the phone was packaged.

Note It seems that over the years the numbers on the back of the phone have been getting smaller. Then again it seems everything I am reading nowadays is getting smaller, so maybe it's not the phone. If you find the same to be true for you, you can also get the MAC address to display on the LCD of the phone. The method for this differs among the various models. On the 7960 and 7940, press the settings and then the 3 on the keypad. Of course, power must first be supplied to the phone.

Step 5 The next field is the Description field. Enter a description that will help you quickly identify the phone later.

Note If you do not enter anything in the Description field, a default description that is the MAC address preceded by SEP is entered automatically. SEP stands for Selsius Ethernet Phone. Selsius was the name of the company that made CallManager before Cisco bought the company and the product. A description I like to use follows the following format: Last Name, First Name Extension Number (for example, Smith, John 2012). This format allows me to search by name and easily see the extension number from the search page.

Figure 3-6 Phone Configuration

Step 6 In the **Owner User ID** field, enter the user ID of the primary user of the phone. The user must already exist in the directory that is used with CallManager, which by default is DC directories. Creating users is covered in Chapter 6: Configuring CallManager Features and Services. An easy way to assign a user is to click the **Select User ID** link to the left of the field and then search for the desired user. This information is included in Call Detail Records (CDRs). Leave this field blank if extension mobility is going to be used.

Step 7 From the Device Pool drop-down list, select the device pool the phone will use.

Note The device pool, MAC address, and phone button template are the only required fields, aside from product-specific information that can be found at the bottom of the configuration screen. Because the product-specific information is automatically set to default values, you could add the phone at this point without configuring any other parameters. This is useful if you need to quickly add phones and you are certain that the default system values, and those set in the device pool, are adequate for this phone.

- Step 8** The CSS determines the destinations that can be dialed from the phone. CSS are discussed in Chapter 5: Configuring Class of Service and Call Admission Control. Choose a CSS from the Calling Search Space drop-down list. If this field is left at None, the dial privileges of this phone could be limited.
- Step 9** The Automated Alternate Routing (AAR) is used to provide an alternate route if a call fails due to insufficient bandwidth. The AAR CSS can be used to limit the paths a call may use when it is rerouted. ARR is covered in Chapter 6: Configuring CallManager Features and Services. Select an AAR CSS from the AAR Calling Search Space drop-down list.
- Step 10** The next field is Media Resource Group List. This determines to which media resources this phone will have access. Media resources are discussed in further detail in Chapter 5: Configuring Class of Service and Call Admission Control. From the Media Resource Group List drop-down list, select the desired group. If no media resource group list is chosen, the list defined in the device pool is used.
- Step 11** The next two fields allow you to configure what audio source is heard when a call is placed on hold. The first of the two, which is labeled User Hold Audio Source, determines what is heard when the call is placed on hold by pressing the hold button. The second of the two, Network Hold Audio Source, determines what audio is heard when the call is placed on hold by pressing the transfer, call park, or conference button. Select the desired audio source from the drop-down list for each field. If no audio source is chosen, the source defined in the enterprise parameters is used.
- Step 12** Information entered in the Location field is used to prevent WAN links from becoming oversubscribed in centralized deployments. These locations are discussed more in Chapter 5, “Configuring Class of Service and Call Admission Control.” If you have defined locations, select the appropriate one for this phone from the drop-down list.
- Step 13** The User Locale field determines the language and fonts used for the phone. The default user locale, which is set in the enterprise parameters, is used if this field is left set to None. If this phone needs to use a different locale than is defined by its device pools, or the enterprise parameters, select the proper one from the drop-down list.
- Step 14** The Network Locale field determines what locale is used for this phone. This impacts the tones and cadences used. The default network locale was defined in the enterprise parameters. If a different value is selected here this value takes precedence. If this field is set to None, the enterprise parameter setting is used. If this phone needs to use a different locale than is defined by its device pools, or the enterprise parameters, select the proper locale from the drop-down list.

Step 15 The next field labeled Device Security Mode is used to determine if any security is used for calls placed from this phone and, if so, what type. Select the desired security mode or leave it set as Use System defaults.

Step 16 The next field is the Signal Packet Capture Mode. This field is for trouble shooting purposes only and should not be configured when adding a new phone.

Step 17 The Packet capture duration field is for trouble shooting purposes only and should not be configured when adding a new phone.

Step 18 The Built-In Bridge field is used to enable and disable the built-in bridge. This bridge can be used when the barge feature is invoked. Select the desired state of the built-in bridge from the drop-down list.

Note Barge is a feature that allows a phone to join an active call on another phone if the two phones have a shared line.

Step 19 If the Retry Video Call as Audio checkbox is checked, a video call will try to connect as an audio call if it cannot connect as a video call.

Step 20 If the Ignore Presentation Indicators (internal calls only) check box is checked, internal caller ID restrictions are ignored. This means that if an internal call is configured to block caller ID, the caller ID will still show up on this device.

Step 21 The Privacy field is used to determine if the phone can enable privacy for calls on a shared line. Select the desired state for this field from the drop-down list.

Phone Button Template Information

Step 22 In the next field, a phone button template is selected for this phone. From the drop-down list in the Phone Button Template field, select the desired template.

Note Not all the templates will display in this field, only those will that apply to the model of phone you are adding.

Softkey Template Information

Step 23 From the drop-down list in the Softkey Template field, select a softkey template for this phone.

Expansion Module Information

Step 24 The next two fields labeled Module 1 and Module 2 are used when expansion modules (7914) are used with the phone being added. If the phone has expansion modules, select the modules from the drop-down list.

Firmware Load Information

Step 25 The next three fields are used to define which firmware load ID the phone and its expansion modules use. In most cases, these fields should be left blank. When left blank, the Load ID specified on the Device Defaults Configuration page is used for this phone. If the need ever arises to set a specific phone to a specific load ID, the load ID should be entered in this field.

Cisco IP Phone—External Data Locations

Step 26 The next set of fields is used to define data locations for the phone. This information is used to determine where the phone should search for certain data, such as help screens and phone services. In most cases these fields can be left blank and the system defaults will be used. Table 3-2 lists these fields and a brief description of each. If values other than the system defaults need to be used by this phone, enter them in the appropriate fields. Any values entered in these fields will be used for this device and will override the values found in the enterprise parameters page.

Table 3-2 Cisco IP Phone—External Data Locations Parameters

Parameter	Description
Information	The URL the phone uses when the (i) is pressed on the phone
Directory	The location of the directory the phone uses
Messages	The URL that is used when the Messages button is pressed. Because you normally want a number dialed when this button is pressed, this field should be left blank.
Services	The URL where services can be found
Authentication Server	URL of the authentication server for requests made to the phone web server
Proxy Server	The proxy server used by the phone
Idle	The URL that is displayed on the phone after the Idle Timer expires
Idle Timer	The amount of time in seconds that the phone must remain idle before the Idle URL is displayed

Certificate Authority Proxy Function (CAPF) Information

Step 27 The next set of parameters deals with Certificate Authority Proxy Function (CAPF). These settings are used to configure certificate specific information. Certificates are used to help prevent the tampering of call signaling and media streams. For more information on CAPF refer to the “Cisco IP Phone Authentication and Encryption for Cisco CallManager” guide at Cisco.com. This guide can be found by searching **Cisco IP Phone Authentication and Encryption** at Cisco.com. If CAPF is not being used, these fields do not need to be configured. The Certificate Operation field is used to install, upgrade, or delete a certificate. The available options are:

- No Pending Action—Displays when no certificate operation is currently active.
- Install/Upgrade—Select this when you want to install or upgrade a certificate.
- Delete—When selected the current certificate will be deleted.
- Troubleshoot—Allows certificate information to be viewed in a CAPF trace file.

Step 28 The Authentication Mode field determines the method that will be used by the phone to authenticate with CAPF. Choose one of the following:

- By Authentication String
- By Null String
- By Existing Certificate (Precedence to LSC)
- By Existing Certificate (Precedence to MIC)

Step 29 The Authentication String is used only when the By Authentication String is chosen in the previous step. Enter the string you wish to use. It must be between 4 to 10 digits.

Step 30 The Key String field determines the key size for the certificate. The valid choices are 512, 1024 and 2048. Select the desired value.

Step 31 The Operation Completes By field specifies when the install, upgrade, or delete must be complete. Enter the desired date and time in this field.

Step 32 The Certificate Operation Status field displays the progress of the certificate operation. Nothing is entered in this field; it is read only.

Multilevel Precedence and Preemption Information

Step 33 The next three fields define Multilevel Precedence and Preemption (MLPP) characteristics of the phone. If these fields are left blank or set to default, the values set in the device pool are used. If MLPP is not being used, these fields may be left blank. The first MLPP field is the MLPP Domain. MLPP grants only higher priority from calls within the same MLPP domain. For this reason a MLPP domain is needed.

Step 34 The second field in this category, which is called MLPP Indication, determines whether tones and indications are presented when a precedence call is made. The precedence indication may be a special ring back or a display if the caller's phone supports it, and a special ringer on the called parties' side.

Step 35 The third MLPP field is MLPP Preemption. This parameter determines whether a higher precedence call preempts a lower precedence call. The value of Disabled does not allow this to happen. To cause a lower precedence call to be terminated if a higher precedence call requires the resources, set this parameter to Forceful.

Product Specific Information

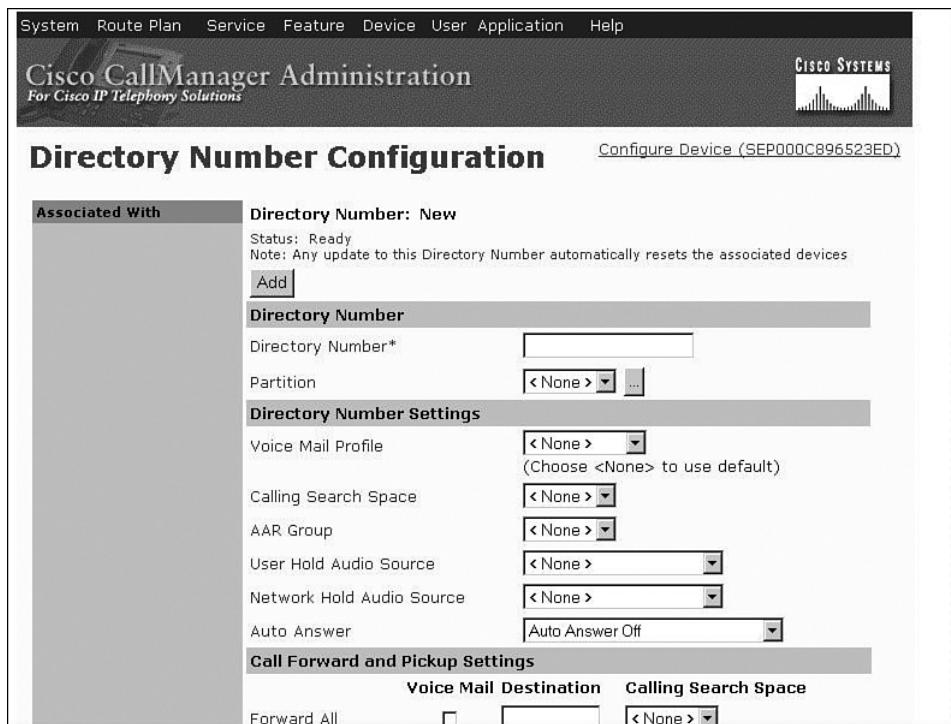
Step 36 The last set of fields labeled Product Specific Configuration are specific to the model of phone you are configuring. To see an explanation of each of these fields click the [i] icon located to the right of the category title.

Step 37 After all settings have been defined, click the **Insert** button at the top of the screen. A message displays informing you that the phone has been added and asking you if you would like to add extensions. Press **OK** and continue using the steps outlined in the following sections to add and configure a line on a phone.

Add a Line to a Phone

After a phone is added, a line must be configured for it. The following steps show how to add a line to a phone. Because there are a number of steps to this process, section headings are used to mark the point at which a new set of parameters begins. These same headings also display on the configuration screen, which should help you keep track of where you are.

Step 1 If you are adding a new phone and have used the steps in the preceding section, you should see a screen similar to that shown in Figure 3-7. To add a line to an existing phone, follow steps 2 though 4 to reach this screen. If you are already at this screen, skip to step 5.

Figure 3-7 Directory Number Configuration

Step 2 From within CCMAdmin, select **Device>Phone**.

Step 3 Enter search criteria in the search field to limit the results, and click the **Find** button.

Step 4 From the list that is generated, select the phone to which you want to add a line. On the left side of the Phone Configuration screen the available lines are listed. Choose a line that has the label **Add new DN**.

NOTE

If all lines have an extension number already assigned, no more lines can be added to this device. If the phone button template assigned to this phone has some of the buttons defined as speed dial, you may be able to add more lines by changing the phone button template.

Directory Number

Step 5 The first and only required field is the Directory Number. Enter the extension number in this field.

- Step 6** The Partition field defines the partition to which this directory number is assigned. The partition is used to determine what devices may call this extension. Partitions are discussed in more detail in Chapter 5: Configuring Class of Service and Call Admission Control.

Directory Number Settings

- Step 7** The Voice Mail Profile field determines which voice mail profile the directory number uses. The voice mail profile defines the number that is dialed when the messages button on the phone is pressed. Voice mail profiles are discussed in further detail in Chapter 5: Configuring Class of Service and Call Admission Control. Select the voice mail profile from the drop-down list.
- Step 8** The next field allows a CSS to be assigned as the line level. This determines what destinations can be reached when calling from this line. Select the Calling Search Space from the drop-down list.

Note	It is important to understand what happens when a CSS is assigned to the line and the device. In short, the two CSSs are combined, however there is a little more to it than this. For a detailed explanation, refer to Chapter 5: Configuring Class of Service and Call Admission Control.
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- Step 9** The AAR Group field determines the AAR group with which the line is associated. An AAR group defines the prefix that is assigned when a call fails due to insufficient bandwidth. AAR is discussed in further detail in Chapter 6: Configuring CallManager Features and Services. Select an AAR group if AAR is being used. If this field is set to None, AAR is, in effect, disabled on the line.
- Step 10** The next two fields allow you to configure what audio source is heard when a call is placed on hold. The first of the two, which is labeled User Hold Audio Source, determines what is heard when the call is placed on hold by pressing the hold button. The second of the two, Network Hold Audio Source, determines what audio is heard when the call is placed on hold by pressing the transfer, call park, or conference button. Select the desired audio source from the drop-down list for each field. If no audio source is chosen, the source defined at the device level is used, and if None is chosen there, the source set in the device pool is used.
- Step 11** The Auto Answer field determines if the line automatically answers incoming calls without the handset being lifted. This parameter can be set

to auto answer using the speakerphone or headset. If you want the line to auto answer, select either **Auto Answer with Headset** or **Auto Answer with Speakerphone** from the drop-down list. If auto answer is not desired on the line, leave this field set to **Auto Answer Off**.

Call Forward and Pickup Settings

Step 12 The next seven fields deal with call forwarding. These fields determine the forwarding destination, which depends on the reason for the forward. The seven types of forwards are:

- Forward All—forwards all incoming calls
- Forward Busy Internal—forwards calls from internal callers when the line is busy
- Forward Busy External—forwards calls from external callers when the line is busy
- Forward No Answer Internal—forwards calls from internal callers that are not answered
- Forward No Answer External—forwards calls from external callers that are not answered
- Forward No Coverage Internal—forwards calls from internal callers when a CTI route point has no coverage
- Forward No Coverage External—forwards calls from external callers when a CTI route point has no coverage

Note The no coverage forwards are needed only when configuring a line for a computer telephony integration (CTI) route point.

You may configure each type of “forward” to forward calls to voicemail or a specific extension. To forward to voice mail, check the Voice Mail box. For this to work, a voice mail profile must be defined for the line. To forward calls to another extension, enter the extension number in the Destination field. When a destination is entered into any of the internal forwards, the number is automatically entered into the corresponding external forward. If you wish the external calls to be forwarded to a different destination, simply enter the desired destination in the appropriate external forward field. A calling search space can be applied to each forward type, which limits the destinations to which a call can be forwarded. This is useful when you want to restrict a line from forwarding calls to numbers that are long distance, but still want long distance calls to be placed from the line.

Step 13 Enter the appropriate destinations and calling search spaces for each forward type.

Step 14 In the No Answer Ring Duration field, enter the number of seconds that the line will ring before forwarding to the Forward No Answer destination. If this field is left blank, the value configured in CallManager service parameter is used.

Step 15 The Call Pickup Group field determines which call pickup group this directory number belongs to. Call pickup groups allow a user to redirect an incoming call on another phone to the user's phone. Select the desired call pickup group from the drop-down list. Call pickup groups are covered in more detail in Chapter 6: Configuring CallManager Features and Services.

MLPP Alternate Party Settings

Step 16 The next set of parameters deals with MLPP alternate party settings. These settings allow you to configure an alternate destination for precedence calls that are not answered on this line or the forwarded number assigned to this line. If MLPP is not being used, these parameters can be left empty. In the first field, which is labeled Target (Destination), enter the number to which unanswered precedence calls should be forwarded.

Step 17 In the MLPP alternate party Calling Search Space field, select the appropriate search space from the drop-down list. This calling search space limits the destinations to which precedence calls can be forwarded.

Step 18 In the MLPP alternate party No Answer Ring Duration field, enter the number of seconds that the phone will ring when it receives a precedence call before forwarding to the Forward No Answer destination if unanswered.

Line Settings for all Devices

Step 19 In the Alerting Name field, enter the name that should be displayed on the caller's phone.

Line Settings for these Devices

Step 20 The set of parameters under Line Settings for this Device define Caller ID, Message Waiting Indicator (MWI) and ring settings. The settings configured here only affect the line on this phone even if it is a shared line

appearance. If you want these settings to apply to all phones that have this directory number as a shared line appearance, check the **Update Shared Device Settings** button next to the setting. The Update Shared Device Settings button is displayed only if the line is shared. The first of these fields, labeled Display (Internal Call ID), is used to configure which caller ID is displayed when calls are placed to other internal callers. Enter up to 30 characters in this field. Both letters and numbers are allowed. If this field is left blank, the lines directory number will be used.

Step 21 The next field, which is labeled Line Text Label, is used to define how the line displays on the phone. If you want the extension number to display next to the line button, leave this field blank. To display a label other than the directory number next to the line button, enter that label in this field.

Step 22 The External Phone Number Mask field may be used to modify the external caller ID for calls placed from this line. An example mask might be 408370XXXX. The extension number is used to fill in the XXXX portion. In this example, if the directory number is 1401, the external phone mask would cause the external caller ID number to be 4083701401. The External Phone Mask on the first line creates the fully qualified directory number that is displayed above the first extension on certain IP Phones. Masks are explored in further detail in Chapter 4: Implementing a Dial Plan.

Step 23 The Message Waiting Lamp Policy field which determines if the light on the phone is turned on when a new message is left for this extension. In most cases, this value should be left at Use System Policy. The available choices are as follows:

- Light and Prompt—The light turns on and the envelope icon next to the line displays.
- Prompt Only—Only the envelope icon next to the line displays.
- Light Only—Only the light is turned on.
- None—No indication is used.
- Use System Policy—Uses the setting selected in CCM services parameters.

Step 24 The next two settings determine if the phone rings when incoming calls are being received on this directory number. In most cases this value should be left at Use System Default. The available choices are as follows:

- Disable—Phone does not ring.

- Flash Only—The light flashes—No ring.
- Ring Once—Rings once and then stops.
- Ring—Normal Ringing.
- Beep Only—A beep is played (only valid for the Phone Active setting).
- Use System Policy—Uses the setting selected under services parameters.

Step 25 Select the desired value for Ring Setting (Phone Idle) and Ring Setting (Phone Active).

Multiple Call / Call Waiting Settings

Step 26 The next field, which is labeled Maximum Number of Calls, determines how many active calls can be on the line. The maximum is 200 active calls per phone. Enter the maximum number of calls in this field. The default of four should be adequate for most phones.

Step 27 The field labeled Busy Trigger determines how many active calls are required before the line is considered busy. The default is two. This means that if the maximum number of calls on the line is four and the busy trigger is two, the third call will receive a busy indication. However, two more calls could be placed from this phone because the maximum number of calls is four.

Forwarded Call Information Display

Step 28 The Forwarded Call Information Display section determines what information is sent when a call is forwarded. Select the information to be sent by checking the box next to each desired field.

Step 29 The last field on this page is the Character Set. Select the character set that is to be used on the display setting for the line from the drop-down list.

Step 30 If this is a new directory number that you have added to the phone, click the **Add** button at the top of the screen. If you are modifying an existing directory number, click the **Update** button at the top of the screen.

That's all there is to it! It really is a pretty simple task once you are familiar with all of the parameters that need to be configured. However, if you are new to CallManager, a number of these parameters may seem confusing. Rest assured that these parameters will be explained in greater detail throughout the remainder of this book.

Using BAT to Add Devices

Imagine having to add more than a hundred phones and you must choose between adding them manually or adding them using auto-registration. At first glance you would most likely choose to add them using auto-registration. Normally, this would be a good choice; however, you discover that each phone needs to have a specific directory number. Then, to make things more complicated, a number of the phones need to use different phone button templates. You could still use auto-registration, but after the phones were added you would have to change the directory numbers and phone button templates of each phone. It would be more efficient to add these phones and not have to go back and touch each one. This is where a utility called Bulk Administration Tool (BAT) comes in. BAT allows you to pre-populate CallManager with all the information for the phones before they are connected to the system. By adding the phones' information, a configuration file is created that the phones download from the TFPT server when it boots up.

BAT adds devices to CallManager by importing a Comma Separated Value (CSV) file that contains the required information about the phones. The CSV contains a number of fields that must be populated. The easiest way to create this CSV file is to use an Excel template that is included with BAT. The template offers an easy, well-formatted interface to add all the required and optional information. However, before this template can be used to create a CSV file, BAT must first be installed. BAT is included with CallManager, but is not installed by default. BAT must be installed on the publisher and will cause Internet Information Services (IIS), World Wide Web (WWW) publishing, and File Transfer Protocol (FTP) publishing services to stop during the installation. It is recommended that you install BAT during nonproduction hours, due to the impact that this can have on the server.

Installing BAT

The following steps guide you through the installation process.

- Step 1** From within CCMAdmin, select **Applications>Install Plugins**.
- Step 2** From the list of plug-ins on the Install Plug-ins page, click the icon to the left of Cisco Bulk Administration Tool.
- Step 3** A window displays asking, “Would you like to open the file or save it to your computer?” Select **Open**. Depending on the version of Internet Explorer you are using, this message may vary. It may give you the choice, “Run this program from its current location.” If it does, select this option and click **OK**.
- Step 4** The install preparation process begins automatically. During this time you may receive a security warning. If you do, select **Yes**.
- Step 5** A Welcome window displays. Click **Next** to start the installation.
- Step 6** When the installation is complete, a window displays informing you that it is complete. Click the **Finish** button.

After BAT is installed, the CSV files need to be created. To do this, use the Excel template that was mentioned previously in this section. The number of available fields in the template depends on the model of phone you are importing. For example, the 7960 can contain more than 30 fields. The top row displays what information should be entered into the field and whether it is a required field. The fields include, but are not limited to, information such as the user's name, MAC address of the phone, directory numbers, and speed dial numbers. When you first open the template, it displays only a few fields. The template then lets you add other fields, as you desire. This feature enables you to avoid dealing with unneeded fields. After the information is added to the template, a CSV file is created and must be placed in the C:\BATFiles\Phones\Insert\directory on the publisher. The necessary steps are presented later in this section.

NOTE Although Excel may be used as part of the overall BAT process, it should never be installed on the CallManager. Excel should be loaded on another PC and the file copied to the CallManager after the export to a CSV is completed.

NOTE On a large deployment, a Universal Serial Bus (USB) thumb drive may prove useful to transport these files.

It may appear that the hardest part of creating the CSV file is the entry of all the information. However, often what is harder is acquiring the information that must be entered into the template. Make sure you take the proper amount of time beforehand to ensure you have obtained all the required information. The best way to get this information is to perform station reviews. A station review is a formal process that is used to record information about each phone that is to be added to the system. The information acquired during this process includes the number of lines (and directory numbers assigned to each), number of speeds, required features (conference calls, voicemail, and so on) and the type of calls that may be placed by the phone (local, national, international, and so on). This, of course, is only a sample of the information that is gathered during the station review process. It is important that you gather as much information as possible during station reviews. After this information is collected, the creation of the CSV files can begin.

In addition to the CSV file that is created, templates within the BAT application must also be created. These templates define the characteristics of the phones that you are going to add. These templates contain parameters for things such as phone type, phone button template, softkey templates, calling search spaces, number of lines, and partitions just to name a few. After these templates are configured, they are used in concert with the CSV file that has been created to insert phones into the system. Because these templates contain many parameters, it may be necessary to create a number of templates depending upon your

environment. For each set of phones that require different settings on any parameter defined within a template, a new template must be created. Let's look at a simple example so you can see how easily the number of these templates can grow.

Let's say you have 70 7940s and 50 7960s to deploy. Right off the bat (no pun intended), you know you need at least two templates because there are two types of phones. For this example, let's assume half the 7940s are going to be two-line phones and the other half are going to be one-line phones. This means that at least two templates are needed for the 7940s alone. As for the 7960s, let's assume that through the station review process, you determine that you need two-, three-, and four-line phones, which means that at least three templates are needed for the 7960s. If all the phones with the same number of lines do not differ in any other way, five templates in all are needed. However, just to make things interesting, let's say that half the three-line phones and half of the four-line phones require different calling search spaces. This adds the need for two more templates, bringing the total to seven. You can see how minor changes can cause the number of required templates to grow quite quickly. It is important to understand that because each template is used with a CSV file, you need to create the same number of CSV files as there are templates.

NOTE

Make sure you take time to determine the number of templates that will be needed.

Sometimes after determining how many templates are needed, you may feel it is not worth the effort. I can remember an installation of 3000 phones that required more than 230 templates. Although this was a large number of templates, days if not weeks of time was saved on that deployment by creating the templates and CSV files. Keep in mind that this is an extreme example. Another deployment of 300 phones required only one template.

Before you can determine how many templates you need, you must be familiar with all of the parameters that are set in the BAT templates. Each model of phone has unique parameters, but there are also a number of parameters that are common among all models. The following is a list of parameters that can be defined in the template for most models of phones.

- Device Pool
- Calling Search Space
- AAR Calling Search Space
- User and Network Hold Audio Source
- Location
- User Locale
- Network Locale
- Phone Button Template
- Phone Load Name

- External Data Locations
- Multilevel Precedence and Preemption Information
- Number of Lines

By taking the number of different models of phone that will be deployed and determining how many of the parameters defined in the template will be different among like models, you can determine approximately how many templates and CSV files that must be created.

NOTE After you have set up a template, you can often reuse it over and over with only minor modifications.

Now that you have a good overview of what is required for BAT to work, let's take a look at the steps required to add phones to the system using BAT.

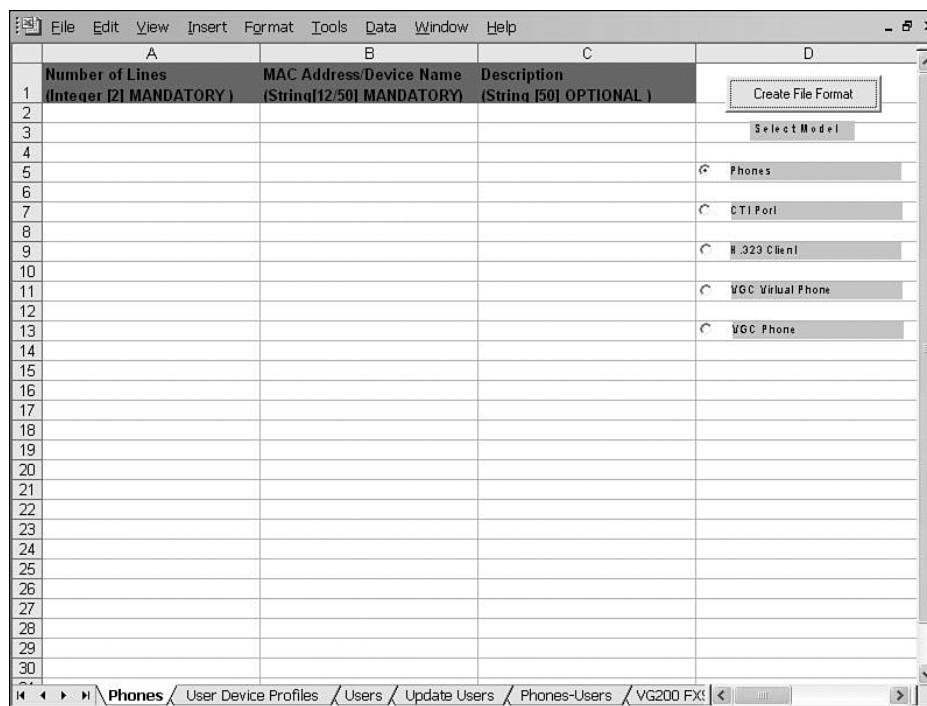
After BAT is installed, you can begin the process of creating the CSV files and the templates. Before creating the templates and CSV files you should have conducted a detailed analysis of the phones that are to be deployed. Based on the disparity of the phones in your system, you should have a good idea of how many templates and CSV files you need. The analysis should also have supplied the information that will be needed to create the templates and files.

You can create either the template or the CSV first. In the steps that follow the creation of the CSV is described first.

NOTE It's a good idea to use same naming convention for both the template and the CSV file that will be used together. This way, when you are selecting the CSV file, it is easy to determine which file goes with which template.

Creating a CSV File for BAT

- Step 1** Copy the Excel BAT template to a PC on which Excel is installed. The Excel BAT template can be found in C:\CiscoWebs\BATExcelTemplate on the publisher. The name of this file is bat.xlt.
- Step 2** Open the BAT template in Excel. You may get a security warning about macros. Macros must be enabled for this template to function properly, so you may need to change Excel's settings to allow this.
- Step 3** A template like that shown in Figure 3-8 displays. As you can see, by default, only three fields display in the template. Click the **Create File Format** button to add fields.

Figure 3-8 Excel BAT Template

Step 4 A screen similar to that shown in Figure 3-9 should display. Select the device and line fields you want added to the template by highlighting the field in the box on the left side and clicking the >> arrows.

Step 5 To remove device and line fields from the template, highlight the field in the box on the right side and click the << arrows.

Step 6 After you move all the desired fields to the box on the right side, click the **Create** button. When asked if you want to overwrite the existing CSV format, click **Yes**.

Step 7 Go to the far right of the template and in the appropriately labeled fields, enter the number of lines and speed dials these phones will have. When you enter information in these fields, the template adds the appropriate fields.

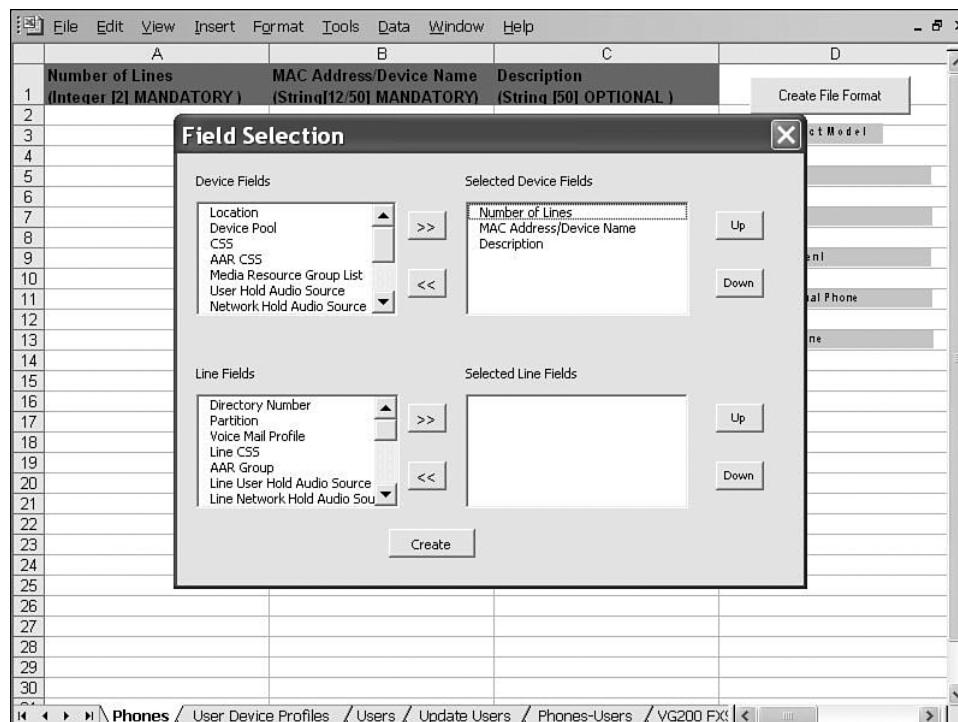
Step 8 At this point the template is ready for the data to be entered.

Step 9 After all the data is entered, click the **Export to BAT Format** button, which is located at the top far-right portion of the template.

Step 10 You are prompted for a place to store this file. BAT creates a default location and file name. An example is c:\XlsDataFiles\Phones#09072004193840. BAT always selects c:\XlsDataFiles as the default file location. The file

name is based in part on the tab you have selected to export, such as Phones, PhonesUsers, and UserDeviceProfiles, followed by the number symbol (#). The second part of the filename is the date and time that the file was exported. Note that in the preceding example, 09072004193840 represents September 7th, 2004 at 19:38 and 40 seconds. If you prefer a different file location or name, you can change it before clicking **OK**.

Figure 3-9 Excel BAT Template File Format



Step 11 Copy this file to the C:\BATFiles\Phones\Insert\ directory on the publisher.

Adding Phones Using BAT

The process of inserting phones using BAT is actually a four-stage process, and each stage has its own set of steps. The first stage of adding phones in BAT is creating the templates. The following steps walk you through this stage. A 7960 template is created in this example.

Step 1 At the console of the Publisher, navigate to **Start>Programs>CallManager 4.x>Bulk Admin Tool>BAT x.x** (x.x is the version number).

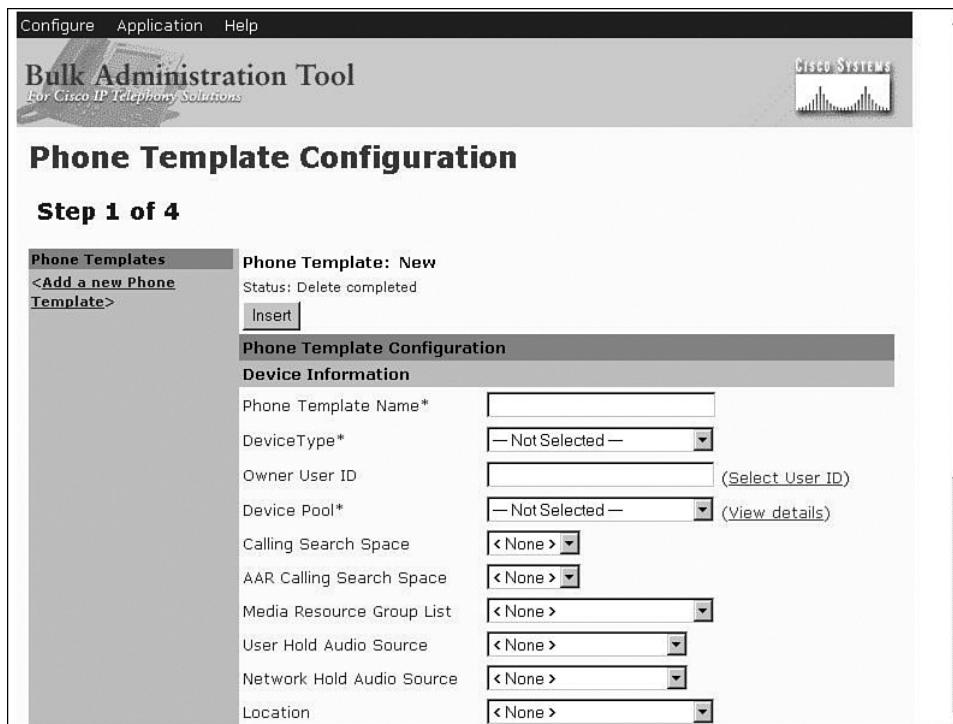
Step 2 From within the Bulk Administration Tool select **Configure>Phones**.

Step 3 From the Phone Options screen, make sure the **Insert Phones** radio button is selected and click the **Next** button at the bottom of the screen.

Step 4 The Steps to Insert Phones page appears and shows all the steps that must be completed to insert phones. In all, there are four steps, but each step includes additional substeps. Make sure the radio button labeled Step 1: Add, view, or modify phone templates is checked and click the **Next** button.

Step 5 A screen similar to that shown in Figure 3-10 displays. On this screen you define the parameters for this template. In the Phone Template Name field, enter a name for this template. Remember to try to keep the naming convention the same as the CSV file that will be used with this template.

Figure 3-10 BAT Phone Template Configuration

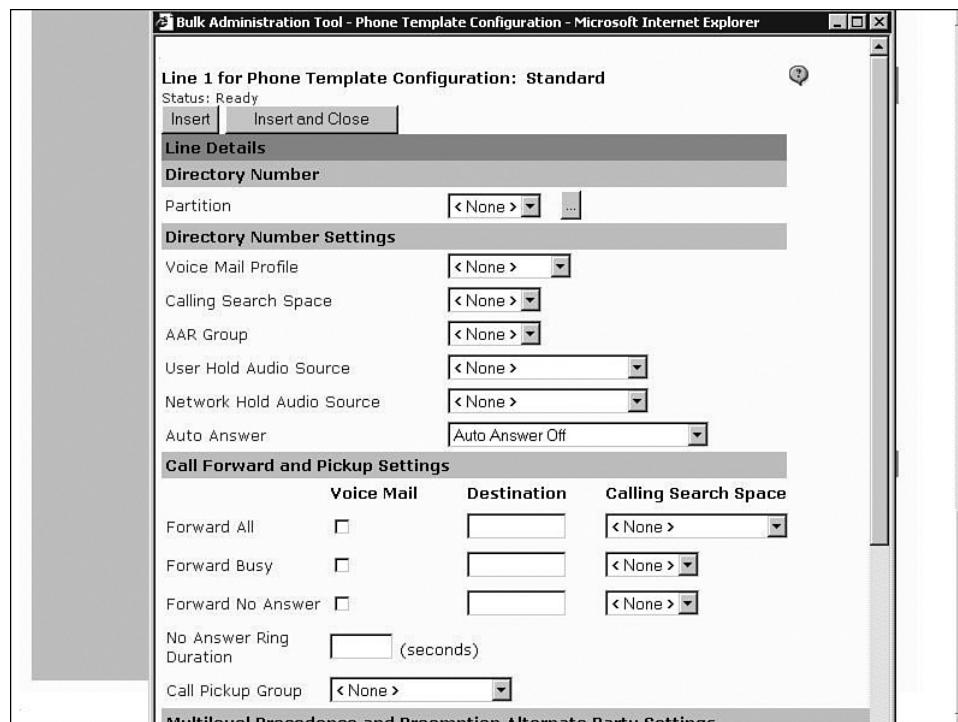


Step 6 From the drop-down list in the Device Type field, select the type of phone for this template. When a phone is selected, the page refreshes so that all applicable fields are displayed.

Step 7 The remainder of this page contains all the parameters that can be configured for this device. Because all these parameters are covered earlier in this chapter, they are not discussed in this section. If necessary, refer back to steps 6–26 in the *Manually Adding Phones* section in this chapter for details of each parameter. Enter the appropriate information in each field.

- Step 8** After the parameters have been configured, click the **Insert** button at the top of the screen. An alert window informs you if the operation is successful. Click the **OK** button in the alert window.
- Step 9** After the template is inserted, a Line Details section is added at bottom of the screen. Find this section and click **Add Line 1** to define the parameters for this line.
- Step 10** A screen similar to that shown in Figure 3-11 displays. This screen is used to define all the parameters for this line. Because all these parameters are covered earlier in this chapter, they are not discussed in this section. If necessary, refer back to steps 6–25 in the *Add a line to a Phone* section in this chapter for details on each parameter. Enter the appropriate information in each field.

Figure 3-11 BAT Phone Line Configuration



- Step 11** Enter the desired information in these fields and click the **Insert and Close** button. Repeat steps 9–10 for each additional line.
- Step 12** When configuring templates for certain models of phones, you may also add speed dial, and services, to the template. To add speed dials to the template, click the **Add/Update Speed Dials** link in the upper-right corner on the Phone Template Configuration page.

Step 13 A screen displays that allows you to add two types of speed dials. The first type is associated with a button on the phones. The user accesses the second type by pressing the two-digit speed dial number, which has the number they want to reach assigned to it, and by pressing the **AbbrDial** softkey. Enter the desired speed dials and click **Update and Close**.

Step 14 To add services to the template, click the **Subscribe/Unsubscribe Services** link in the upper-right corner on the Phone Template Configuration page. Services are covered in more detail in Chapter 6: Configuring CallManager Features and Services.

Step 15 A window displays that allows you to select the available services. Select the desired service and click **Continue**. Enter any additional information the service may require.

Step 16 After all services have been added, the template is complete. Click the **Back** button at the bottom of the page to return to the Steps to Insert Phones page.

The previous stage left you at the page labeled Steps to Insert Phones, on which you should see four options. Step 1: Add, view, or modify phone templates has just been completed, and if you created the CSV data file using the Excel template Step 2: Create the CSV file is also completed. This brings you to Step 3: Validate phone records. This part of the process checks the CSV file that you created to make sure it is formatted properly.

Step 1 On the Step to Insert Phones page, select the radio button that is labeled Step 3: Validate phone records and click the **Next** button.

Step 2 From the File Name drop-down list, select the CSV file that you created using the Excel template. Only CSV files that you placed in C:\BATFiles\Phones\Insert\ on the publisher display in this list.

Step 3 Select the corresponding template that you created from the Phone Template Name drop-down list.

Step 4 Click the **Validate** button.

Step 5 After the file has been validated, the status line should state Validate Completed. Click the **View Latest Log File** link next to the Validate button to view the log.

Step 6 If no errors are reported, close the log window, click the **Back** button, and move on to the next set of steps.

The last stage is to actually insert the phones into the CallManager database.

Step 1 On the Step to Insert Phones page, select the radio button labeled Step 4: Insert Phones, and click the **Next** button.

- Step 2** From the File Name drop-down list, select the CSV file that you created using the Excel template. Only CSV files that you placed in C:\BATFiles\PhonesInsert\ on the publisher display in this list.
- Step 3** Select the corresponding template that you created from the Phone Template Name drop-down list.
- Step 4** Click the **Insert** button.
- Step 5** A warning message displays informing you that you are about to insert records. Click **OK** to continue.
- Step 6** While the insertion is being made, a status window displays. After the insertion is completed, the status line should state “Insert Completed.” Click the **View Latest Log File** link next to the Insert button to view the log.
- Step 7** If no errors are reported, close the log window and exit BAT.

Adding Phones Using TAPS

A more advanced method of adding phones called Tool for Auto-registered Phone Support (TAPS) is also available. This tool allows you to pre-populate CallManager’s database with all the information about the phones, except for the MAC address. After the information is entered, the phones are plugged into the system and auto-registered. Because the MAC addresses have not been entered into the system, the TFTP server cannot deliver a configuration file to the phone based on the MAC address. This causes a default configuration to be issued and the phone to be registered with basic configuration. After the phone is registered, a predetermined number is dialed that routes the call to an Interactive Voice Response (IVR) server. The IVR server asks what language you would like to use and the extension number that should be assigned to the phone. After this is completed, the MAC address of the phone is associated with the configuration that matches the extension number entered, and a configuration file is created. The phone gets this file from the TFTP server when it reboots.

Integrators often use this method of adding phones during large deployments. Because this method allows the integrator to populate the system without having to add MAC addresses, the deployment of the phones is simpler. Integrators don’t have to make sure a certain phone with a certain MAC address is plugged into a certain jack. The integrator need only be concerned that the proper model is placed in each location. Previous to the introduction of this tool, the MAC addresses had to be entered before the phones were deployed, and this required the integrator to label the box of each phone with the extension number for which it was configured. This also meant that no information could be added to the system until the phones were in hand.

The detailed steps for TAPS are outside of the scope of this book, but further information can be found in Cisco’s Bulk Administration Tool documentation on the Cisco website. You can find the latest copy of this by searching “bulk administration tool guide” at Cisco.com.

Adding Gateways

After phones are configured, calls may be placed to other phones within the cluster. Although this is useful, it is also quite limiting. Imagine if the phone system were allowed to reach only other phones within the cluster. I have a feeling not too many of these systems would be sold. Therefore, you need to consider how you will go about allowing calls outside the cluster. This is accomplished by configuring gateways that connect the cluster to other systems. The function of a gateway is to allow connectivity between dissimilar systems. Basically, the job of the gateway is to allow your phone system to make calls to other phone systems.

In most cases, the first system to which you should connect your system is the Public Switched Telephone Network (PSTN). This allows callers on your system to call nearly any other phone in the world. In addition to connecting to the PSTN, you may also want to connect your system to other systems in your company and bypass the PSTN. In either case the steps are very similar.

This section explores four types of gateways. The main difference in these gateways is the protocol they use to communicate with the CallManager. Their functions are the same—to provide connectivity to another system.

The following is a list of the four types of gateways and a brief description of each:

- **H.323**—Typically used to connect to the PSTN and requires its own dial plan configuration. Requires a significant amount of configuration on the gateway.
- **MCG**—Typically used to connect to the PSTN and does not require its own dial plan configuration. Requires minimal configuration on the gateway.
- **Non-IOS MCGP**—Typically used to connect to the PSTN and does not require its own dial plan configuration. Requires almost no configuration on the gateway itself.
- **Inter-cluster Trunk**—Connects CallManager clusters together. Runs on the CallManager server and does not require additional hardware.

Adding H.323 Gateways

The first type of gateway to be discussed is an H.323 gateway. H.323 is best described as a suite of protocols. It is also often referred to as an umbrella under which a number of other protocols exist. The specifics of what H.323 is and how it works are far beyond the scope of this book. This section focuses on the process of configuring an H.323 gateway.

The configuration steps required for an H.323 gateway are somewhat more advanced than those of other types of gateways. This is because much of the configuration has to be done using the command-line interface (CLI) of the gateway itself. For example, when configuring a Media Gateway Control Protocol (MGCP) gateway, only a few commands must be entered in the CLI, and CallManager handles all the call routing. An H.323 gateway does not depend on the CallManager for call routing; it contains a dial plan of its own. This

means that the dial plan must be entered using the CLI, which increases the difficulty of configuring an H.323 gateway. In order to properly configure an H.323 gateway, you must have a high level of experience in configuring IOS devices for VoIP deployments. Because it is not possible to provide you with all the required knowledge to configure the CLI portion of an H.323 gateway in a book of this size, we focus on the steps required in CCMAdmin. The CLI configuration varies in each environment and can sometimes become quite complex. For more information on the CLI configuration for an H.323 gateway, refer to the Cisco IOS Voice, Video, and Fax Configuration Guide, which can be found at Cisco.com by searching “Cisco IOS Voice, Video, and Fax Configuration Guide.”

After the entire CLI configuration of the gateway is completed, it can be configured in CallManager. There are a number of Cisco devices that can be configured as H.323 gateways. Because the list of supported devices is always evolving, it is best to check at Cisco’s website to see what gateways currently support H.323. You can find a list of supported gateways by searching “Understanding Voice Gateways” at Cisco.com. You can further refine the search by including the CallManager version with this search. For example, by searching “Understanding Voice Gateways 4.1(2)” the very first result is a link to the correct page. Because there are a variety of devices that can be configured as an H.323 gateway, it is not possible to include a step-by-step guide on how to configure each of them. However, because the configurations are similar, you should be able to use the following steps, which show how to configure a Cisco 3620 as an H.323 gateway, as a guide to install the H.323 gateway you choose.

The following steps are required to configure a Cisco 3620 as an H.323 gateway. Because there are a number of steps to this process, section headings are used to mark the point at which each new set of parameters begins. These same headings display on the configuration screen as well, which should help you keep track of where you are.

- Step 1** From within CCMAdmin select **Device>Add a New Device**.
- Step 2** On the Add a New Device page, select **Gateway** from the Device type drop-down list and click **Next**.
- Step 3** On the next page, select the **H.323 Gateway** from the Gateway type drop-down list.
- Step 4** The Device Protocol field remains at H.225; this is the only option. Click the **Next** button.

Device Information

- Step 5** The Gateway Configuration screen, as shown in Figure 3-12, displays. Enter the DNS Name or IP address of the gateway in the Device Name field.
- Step 6** In the Description field, enter a description that will help make this device easily identifiable.
- Step 7** From the Device Pool drop-down list, select the desired CallManager group for this gateway.

Figure 3-12 H.323 Gateway Configuration

Gateway Configuration

Product : H.323 Gateway
Gateway : New
Device Protocol: H.225

Status: Ready

Device Information

Device Name*	<input type="text"/>
Description	<input type="text"/>
Device Pool*	<input type="button" value="—Not Selected—"/>
Media Resource Group List	<input type="button" value="<None>"/>
Location	<input type="button" value="<None>"/>
AAR Group	<input type="button" value="<None>"/>
Signaling Port*	<input type="text" value="1720"/>

Media Termination Point Required
 Retry Video Call as Audio
 Wait for Far End H.245 Terminal Capability Set

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain (e.g., "0000FF")

Step 8 The next field is Media Resource Group List. This determines to which media resources this phone will have access. Media resources are discussed in further detail in Chapter 6: Configuring CallManager Features and Services. From the Media Resource Group List drop-down list, select the desired group. If no Media Resource Group List is chosen, the one defined in the device pool is used.

Step 9 Information entered in the Location field is used to prevent WAN links from becoming oversubscribed in centralized deployments. These locations are discussed more in Chapter 5: Configuring Class of Service and Call Admission Control. If you have defined locations, select the appropriate one for this phone from the drop-down list.

Step 10 The AAR Group field determines the AAR group with which this device is associated. An AAR group provides the prefix that is assigned when a call fails due to insufficient bandwidth. AAR is discussed in further detail in Chapter 6: Configuring CallManager Features and Services. Select an AAR group if AAR is being used. If this field is set to None, AAR is, in effect, disabled on this device.

Step 11 The Signaling Port field defines the H.225 signaling the gateway uses. The default of 1720 can be used in most cases.

Step 12 The Media Termination Point Required check box needs to be checked if the H.323 device does not support features such as hold and transfers. This is the case with H.323 version 1 gateways. These gateways cannot support such features because H.323v1 does not have the ability to modify a channel, which means it cannot transfer a call or place a call on hold, because this would require modifying the channel.

Step 13 If the Retry Video Calls as Audio box is checked, CallManager sets up a voice call if a video calls fails to set up.

Step 14 When the Wait for Far End H.245 Terminal Capability Set box is checked, CallManager expects to receive the far ends capabilities before sending its own.

Step 15 The only MLPP setting that can be configured is the MLPP Domain. Enter the domain in this field. If left blank the settings found in enterprise parameters are used.

Call Routing Information—Inbound Calls

Step 16 The next set of fields deals with inbound calls. The Significant Digits field determines the number of digits of an incoming dialed number that CallManager uses. CallManager counts from right to left, so if the number entered in this field is 4 and the digits received are 8105559090, 810555 would be removed and only 9090 would be used to determine the destination of this call.

Step 17 A Calling Search Space (CSS) determines what destination inbound calls to this gateway will be able to reach. CSS is discussed in Chapter 5: Configuring Class of Service and Call Admission Control. Choose a CSS from the Calling Search Space drop-down list. If this field is left at None, the dial privileges of this device could be limited.

Step 18 The Automated Alternate Routing (AAR) is used to provide an alternate route if a call fails due to insufficient bandwidth. The AAR CSS can be used to limit the paths a call may use when it is rerouted. Select an AAR CSS from the AAR Calling Search Space drop-down list.

Step 19 The Prefix DN field defines what digits will be added to the front of the incoming destination number. This is applied to the number, after CallManager truncates the number based on the Significant Digits setting.

Step 20 If your voicemail system supports Redirecting Number IE, check the Redirecting Number IE Delivery—Inbound box. Otherwise, leave this box unchecked.

Step 21 Check the Enable Inbound Faststart box if you wish to support H.323 FastStart. H.323 FastStart requires only two message exchanges to open logical channels, whereas normal setup requires 12. However, if FastStart is selected, both ends must support and be configured for FastStart.

Call Routing Information—Outbound Calls

Step 22 The next set of fields deals with outbound calls. The Calling Party Selection field determines what number is sent to outbound calls. The choices are:

- Originator—the directory number of the device that placed the call
- First Redirect Number—the directory number of the first device to forward the call
- Last Redirect Number—the directory number of the last device to forward the call
- First Redirect Number (External)—the external directory number of the first device to forward the call
- Last Redirect Number (External)—the external directory number of the last device to forward the call

Select the desired value for this field.

Step 23 The Calling Party Presentation field determines if CallManager sends Caller ID information. To send caller ID information, select **Allowed** from the drop-down list. To block caller ID, select **Restricted** from the drop-down list.

Step 24 Cisco recommends that the next four fields remain set to the default of Cisco CallManager:

- Called party IE number type unknown
- Calling party IE number type unknown
- Called Numbering Plan
- Calling Numbering Plan

These fields deal with dial plan issues and should be changed only when advised to do so by Cisco or an experienced dial plan expert. The need to change these usually occurs when installing CallManager internationally.

Step 25 The Caller ID DN field is used to determine what caller ID is sent out this gateway. A mask or a complete number can be entered in this field. For example, if the mask 55536XX is entered in this field, CallManager sends 55536 and the last two digits of the directory number (DN) that is placing the call.

Step 26 If the Display IE Delivery check box is checked, then the calling and called party name information is included in messages.

Step 27 The Redirecting Number IE Delivery–Outbound check box should be checked when integrating with a voicemail system that supports Redirecting Number IE. Otherwise, leave it unchecked.

Adding MGCP Gateways

Cisco recommends that you use a Media Gateway Control Protocol (MGCP) gateway whenever possible. MGCP is a protocol that, as its name implies, is used to control gateways. Unlike H.323, MGCP does not handle the routing of calls; it depends on CallManager for this. Therefore, very little needs to be configured at the CLI.

MGCP gateways that are used in a Cisco CallManager solution are separated into two categories: Internetworking Operating System (IOS) and NON-IOS MGCP gateways. The difference is whether the device that is acting as the gateway is running IOS or not. For instance, a 3725, which runs IOS, is considered an IOS gateway; whereas, a Catalyst 6500 running native IOS is considered a NON-IOS gateway. This section focuses on ISO MGCP Gateways. NON-IOS gateways are discussed in the next section.

Adding ISO MGCP Gateways

As with an H.323 gateway, some configuration must be done from the CLI of the gateway. Figure 3-13 shows a sample of the commands that must be entered in the CLI in order to configure the gateway to communicate through MGCP. As you can see, the CLI configuration is quite simple. This is because it relies on CallManager for all call routing functions. For this reason, it is recommended that an MGCP gateway be used whenever possible.

For more information on the CLI configuration for a MGCP gateway, refer to the Cisco IOS Voice, Video, and Fax Configuration Guide, which can be found at Cisco.com by searching “Cisco IOS Voice, Video, and Fax Configuration Guide.”

After the CLI configuration is completed, the MGCP gateway must be configured in CCMAdmin. There are a number of Cisco routers that can act as MGCP gateways, and it is not possible to include a step-by-step guide for each type in this section. However, because their configuration is similar, you should be able to use the following steps that show how

to configure a Cisco 3620 as a MGCP gateway. You can use these steps as a guide to help you install the MGCP gateway you are using.

Figure 3-13 MGCP Gateway CLI Configuration Example

```

mgcp
mgcp call-agent 172.10.10.1
mgcp dtmf-relay codec all mode out-of-band
mgcp sdp simple
!
ccm-manager switchback graceful
ccm-manager redundant-host 172.10.10.10
ccm-manager mgcp
!
voice-port 1/1/1
!
dial-peer voice 4 pots
  application MGCPAPP
  port 1/1/1

```

The following steps are required to configure a Cisco 3620 as a MGCP gateway. Because there are a number of steps to this process, section headings are used to mark the point at which each new set of parameters begins. These same headings display on the configuration screen as well, which should help you keep track of where you are.

Step 1 From within CCMAdmin, select **Device>Add a New Device**.

Step 2 On the Add a New Device page, select **Gateway** from the **Device Type** drop-down list and click **Next**.

Step 3 On the next page, select the desired gateway from the **Gateway Type** drop-down list. In this example, the **362X** is selected. Click **Next**.

Step 4 A page similar to that shown in Figure 3-14 displays. In the first field, enter the DNS name of the gateway in the **Domain Name** field, if DNS is configured to resolve this name. If DNS is not used, enter the host name of the gateway. It is important to note that if an IP Domain Name is configured on the router, it must be included in the **Domain Name** field. For instance, if the hostname is **MGCPGateway** and the IP Domain Name is configured as **cisco.com**, the **Domain Name** field should read **MGCPGateway.cisco.com**. The name is case sensitive. The gateway will not register to CallManager if the name is not entered correctly.

Step 5 In the **Description** field, enter a description that will help make this device easily identifiable.

Figure 3-14 MGCP Gateway Configuration

Gateway Configuration

Product: Cisco 362X
Gateway: New

Status: Ready

Domain Name*

Description

Cisco CallManager Group*

Installed Voice Interface Cards	Endpoint Identifiers
Module in Slot 0 <input type="button" value="None"/>	<input type="button" value="None"/>
Module in Slot 1 <input type="button" value="None"/>	<input type="button" value="None"/>

Product Specific Configuration	
Switchback Timing*	<input type="button" value="Graceful"/>
Switchback uptime-delay (min)	<input type="text" value="10"/>
Switchback schedule (hh:mm)	<input type="text" value="12:00"/>

* indicates required item

Step 6 From the Cisco CallManager Group drop-down list, select the desired CallManager group for this gateway.

Installed Voice Cards

Step 7 The next set of fields defines what type of voice ports are installed in the gateway. This example assumes two FXS and two FXO ports are installed. Select the voice module(s) from the drop-down list next to each slot. For this example the NW-1V is selected.

Step 8 Click the **Insert** button. New fields display next to each configured slot. From the Subunit drop-down list, select the device that is installed in this slot. In this example VIC-2FXS was selected for subunit 0 and a VIC-2FXO was selected for subunit 1.

Step 9 Click the **Update** button. Endpoint Identifiers display next to each configured subunit as seen in Figure 3-15.

Figure 3-15 MCGP Gateway Configuration with Endpoints

Gateway Information

Step 10 The endpoint identifiers must be configured. Select an endpoint to display the Endpoint Configuration page. In this example an FXS port is used. Depending on the type of port selected, the fields you will work with may differ. A description for this endpoint is pre-populated in the Description field. You may modify it if you like.

Step 11 From the Device Pool drop-down list, select the device pool this endpoint will use.

Step 12 The Calling Search Space (CSS) determines where the device (in this case the gateway) will be able to dial. The CSS affects only incoming calls through the gateway. CSSs are discussed in Chapter 5: Configuring Class of Service and Call Admission Control. Choose a CSS from the Calling Search Space drop-down list. If this field is left at None, the dial privileges of this device could be limited.

Step 13 The Automated Alternate Routing (AAR) is used to provide an alternate route if a call fails due to insufficient bandwidth. The AAR CSS can be used to limit the paths a call may use when it is rerouted. Select an AAR CSS from the AAR Calling Search Space drop-down list.

Step 14 The next field is the Media Resource Group List, that determines to which media resources this phone will have access. Media resources are discussed in further detail in Chapter 6: Configuring CallManager Features and Services. From the Media Resource Group List drop-down list, select the desired group. If no Media Resource Group List is chosen, the one defined in the device pool is used.

Step 15 The next field allows you to configure what audio source is heard when a call is placed on hold. Select the desired audio source from the Network Hold drop-down list. If no audio source is chosen, the one defined in the device pool is used.

Step 16 In centralized deployments, locations are used to prevent WAN links from becoming oversubscribed. These are discussed more in Chapter 5: Configuring Class of Service and Call Admission Control. If you have defined locations, select the appropriate one for this phone from the Location drop-down list.

Step 17 The AAR Group field determines the AAR group with which this device is associated. An AAR group provides the prefix that is assigned when a call fails due to insufficient bandwidth. AAR is discussed in further detail in Chapter 6: Configuring CallManager Features and Services. If AAR is being used, select an AAR group. If this field is set to None, AAR is, in effect, disabled on this endpoint.

Step 18 The Network Locale field determines what locale is used for this endpoint. This impacts the tones and cadences used. The default Network Locale was defined in the enterprise parameters. If a different value is selected here, this value takes precedence. If this field is set to None, the enterprise parameter setting is used. If this phone needs to use a different locale than is defined by its device pools, or the enterprise parameters, select the proper one from the drop-down list.

Multilevel Precedence and Preemption (MLPP) Information

Step 19 The next three fields define the Multilevel Precedence and Preemption (MLPP) characteristics of the endpoint. The only field that applies to the device in this example is the MLPP Domain. MLPP grants higher priority only from calls with the same MLPP domain. For this reason a MLPP domain is needed if MLPP is being used.

Port Information

Step 20 The Prefix DN field defines what digits will be added to the front of an incoming destination number. This is applied to the number after CallManager truncates the number based on the Num Digits setting.

Step 21 The Num Digits field determines the number of digits of an incoming dialed number that CallManager will use. CallManager counts from right to left, so if the number entered in this field is four and the digits received are 8105559090, 810555 would be removed and only 9090 would be used to determine the destination of this call.

Step 22 The Expected Digits field defines how many digits are expected on inbound calls. This field is rarely used and can be left at the default of 0 in most cases.

Step 23 The last field, which is the SMDI Port Number, is used only when integrating with a traditional voicemail system. The port number should be set to the same port number as on the voicemail system.

Step 24 Click **Insert**. The endpoints now display on the left side of the page. After you have configured all the endpoints, directory numbers may be assigned.

Step 25 To add a directory number to an endpoint, click on the **Add DN** link next to the endpoint on the left side of the screen.

Step 26 The Directory Number Configuration page displays, which is just like the page used when adding a line to a phone. For detailed information on how to configure this page, refer back to steps 6-25 in the *Add a Line to a Phone* section in this chapter.

Step 27 Click the **Back to Main Gateway Configuration Link** and repeat steps 10 through 25 to configure all other endpoints.

Step 28 After all information has been entered click the **Add** button.

Step 29 After all configuration is completed, click the **Reset Gateway** button.

Adding Non-IOS MGCP Gateways

As mentioned in the previous section, some non-IOS devices can be configured as MGCP Gateways. Just as with the IOS MGCP gateway, these devices do not handle call routing tasks. They depend on CallManager to handle all call routing needs. Unlike the IOS MGCP gateway, there is virtually no configuration required on the gateway itself.

The most widely used non-IOS gateways are the Catalyst 6000 T1/E1 blades. The T1 blades are Cisco 6608s and have eight T1 ports. These ports must be configured in the Catalyst switch with the appropriate IP configuration, including the TFTP server address, and VLAN information. To complete the configuration of the T1 port in CCMAdmin, you need to know the MAC address of the port.

The following steps take you through the configuration of a single T1 port of a 6608 as a non-IOS MGCP Gateway. Because there are a number of steps to this process, section headings are used to mark the point at which each new set of parameters begins. These same headings also display on the configuration screen, which should help you keep track of where you are.

- Step 1** From within CCMAdmin, select **Device>Add a New Device**.
- Step 2** On the Add a New Device page, select **Gateway** from the Device Type drop-down list and click **Next**.
- Step 3** On the next page, select the desired gateway from the Gateway Type drop-down list. In this example the Cisco Catalyst 6000 T1 VoIP Gateway is selected.

Device Information

- Step 4** From the Device Protocol drop-down list, select the appropriate protocol. For this example Digital Access PRI is used. Click **Next**.
- Step 5** The Gateway Configuration page displays. Enter the MAC address of the T1 port in the field labeled MAC Address. The MAC address can be found by using the show port command at the CLI of the Catalyst 6000.
- Step 6** In the description field, enter a description that will help make this device easily identifiable.
- Step 7** From the Device Pool drop-down list, select the device pool this gateway or device will use.
- Step 8** The Device Destination field determines if calls on this trunk will be considered OnNet (on network) or OffNet (off network) calls. Different alerting tones are used for OnNet and OffNet calls. By default this is set to OnNet. If you are unsure of the value to assign, leave it at the default.
- Step 9** From the Network Locale drop-down list, select the locale for this gateway. If it remains set to None, the locale selected in the device pool is used.
- Step 10** The next field is Media Resource Group List. This determines to which media resources this gateway will have access. Media resources are discussed in further detail in Chapter 6: Configuring CallManager Features and Services. From the Media Resource Group List drop-down list, select the desired group. If no Media Resource Group List is chosen, the one defined in the device pool is used.
- Step 11** Locations are used to prevent WAN links from becoming oversubscribed in centralized deployments. These are discussed more in Chapter 5: Configuring Class of Service and Call Admission Control. If you have defined locations, select the appropriate one for this device from the drop-down list.

Step 12 The AAR Group field determines with which AAR group this gateway is associated. An AAR group defines the prefix that is assigned when a call fails due to insufficient bandwidth. AAR is discussed in further detail in Chapter 6: Configuring CallManager Features and Services. If AAR is being used, select an AAR group. If this field is set to None, AAR is, in effect, disabled on this gateway.

Step 13 The next field is used to define which firmware load ID the gateway uses. In most cases this field should be left blank. When left blank, the Load ID specified on the device defaults configuration page is used for this gateway. If the need ever arises to set a specific gateway to a specific load ID, the load ID should be entered in this field.

Multilevel Precedence and Preemption (MLPP) Information

Step 14 The next three fields define the Multilevel Precedence and Preemption (MLPP) characteristics of this gateway. If these fields are left blank or set to default, the values set in the device pool are used. The first MLPP field is the MLPP Domain. MLPP grants higher priority only from calls with the same MLPP domain. For this reason a MLPP domain is needed.

Step 15 The second field in this category, which is called MLPP Indication, determines whether tones and indications will be presented when a precedence call is made. If the is field set to Off, no precedence indication is presented. If this field is set to On, indication is used for a precedence call.

Step 16 The third MLPP field is MLPP Preemption. This parameter determines whether a higher precedence call preempts a lower precedence call. The value of Disabled does not allow this to happen. To cause a lower precedence call to be terminated if a higher precedence call requires the resources, set this parameter to Forceful.

Interface Information

Step 17 The next set of fields defines characteristics of the interface. The first of these fields, labeled PRI Protocol Type, defines the protocol used. The value placed in this field depends on the type of equipment the T1 is connected to on the other side. Your T1 provider should be able to supply this information. Select the appropriate protocol from the drop-down list.

Step 18 The field labeled Protocol Side, determines if the T1 connects to a Network Device or a User device. The simplest way to look at this is that one side must be User and the other side Network. If connecting to the phone company's Central Office (CO) you will most likely choose User. Select the appropriate protocol side from the drop-down list.

Step 19 The Channel Selection Order field determines in what order the ports are used, either starting at the first, which is referred to as TOP_DOWN, or starting at the last, which is referred to as BOTTOM_UP. In most cases, TOP_DOWN is used. Select the desired value from the drop-down list.

Step 20 The next field, which is labeled Channel IE Type, determines the channel selection method. The selection in this field depends on the connection on the other side.

Step 21 The PCM Type determines the type encoding format that is being used. For the United States, Japan, Taiwan, and Hong Kong select “ μ -law” (mu-law). In the rest of the world, select “a-law.”

Step 22 The Delay for first restart (1/8 sec ticks) field determines how long this port waits before restarting when instructed. This allows you to stagger the restart times when a large number of PRIs are installed.

Step 23 The next field, labeled Delay between restarts (1/8 sec ticks), determines the length of time between restarts of the PRI when PRI RESTART is sent.

Step 24 The Inhibit restarts at PRI initialization check box determines if a RESTART message is sent when the D-channel successfully connects. The default is to leave this box checked, which does not cause a RESTART message to be sent.

Step 25 When the Enable status poll check box is checked, the Change B-Channel Maintenance Status is enabled, which allows individual B-Channels to be taken out of service. The default is to leave this box unchecked.

Call Routing Information—Inbound Calls

Step 26 The next set of fields deals with inbound calls. The Significant Digits field determines the number of digits, of an incoming dialed number, that will be used by CallManager. CallManager counts from right to left, so if the number entered in this field is four and the digits received are 8105559090, 810555 will be removed and only 9090 will be used to determine the destination of this call.

Step 27 A Calling Search Space (CSS) determines what destination inbound calls to this gateway will be able reach. CSS is discussed in Chapter 5: Configuring Class of Service and Call Admission Control. Choose a CSS from the Calling Search Space drop-down list. If this field is left at None, the dial privileges of the calls coming in to the gateway could be limited.

Step 28 The Automated Alternate Routing (AAR) is used to provide an alternate route if a call fails due to insufficient bandwidth. The AAR CSS can be used to limit the paths a call may use when it is rerouted. Select an AAR CSS from the AAR Calling Search Space drop-down list.

Step 29 The Prefix DN field defines what digits are added to the front of an incoming destination number. This is applied to the number, after CallManager truncates the number, based on the Significant Digits setting.

Call Routing Information—Outbound Calls

Step 30 The next set of fields deals with outbound calls. The Calling Line ID Presentation field determines if CallManager sends Caller ID information. To send caller ID information, select **Allowed** from the drop-down list. To block caller ID, select **Restricted** from the drop-down list.

Step 31 The Calling Party Selection field determines what number is sent to outbound calls. The choices are:

- Originator—the directory number of the device that placed the call
- First Redirect Number—the directory number of the first device to forward the call
- Last Redirect Number—the directory number of the last device to forward the call
- First Redirect Number (External)—the number of the first device to forward the call using the external phone mask
- Last Redirect Number (External)—the number of the last device to forward the call using the external phone mask

Select the desired value for this field.

Step 32 Cisco recommends that the following four fields remain set to the default of Cisco CallManager:

- Called party IE number type unknown
- Calling party IE number type unknown
- Called Numbering Plan
- Calling Numbering Plan

These fields deal with dial plan issues and should be changed only when advised to do so by Cisco or an experienced dial plan expert. The need to change these usually occurs when installing CallManager internationally.

Step 33 The next field, which is labeled Number of digits to strip, determines how many digits should be stripped for outbound calls. For example, if this value is set to two and the number 995551001 is dialed, the 99 is stripped before the call is sent.

Step 34 The Caller ID DN field is used to determine what caller ID is sent out this gateway. A mask or a complete number can be entered in this field. For example, if the mask 55536XX is entered in this field, CallManager sends 55536 and the last two digits of the calling number.

Step 35 The last field, which is the SMDI Base Port, is used only when integrating with a traditional voicemail system. Enter the first T1 port being used for voicemail functions.

PRI Protocol Type Specific Information

Step 36 The PRI Protocol Type Specific Information fields determine what Information Elements are sent in messages. For most installations this field can remain at default.

UUIE Configuration

Step 37 The Passing Precedence Level Through UUIE check box determines if the MLPP precedence is sent through the use of the User-to-User Information Element (UUIE). To enable this check the box.

Product Specific Configuration

Step 38 The last set of fields, which is labeled Product Specific Configuration, is specific to the gateway you are configuring. To see an explanation of each of these fields, click the [i] icon located to the right of the category title. The information needed to determine the settings of these fields comes from the T1 carrier. Enter the information provided by your carrier in the appropriate fields. Table 3-3 lists these fields and supplies a brief description of each field.

Step 39 After the required and desired parameters have been entered, click the **Insert** button at the top of the page. After the gateway is added, an alert window informs you that the gateway needs to be reset. Press the **OK** button.

Table 3-3 *Product Specific Configuration Detail*

Parameter	Description
Clock Reference	This parameter specifies the location from which the master clock is derived. Required field. Default: Network.
TX-Level CSU	This parameter specifies the transmit level based on the distance between the gateway and the nearest repeater. The default is full power (0dB). Required field. Default: 0dB.

Table 3-3 Product Specific Configuration Detail (Continued)

Parameter	Description
FDL Channel	This parameter specifies what kind of facility data link (FDL), if any, is supported by the span. The FDL is a maintenance channel that allows remote troubleshooting of link-layer problems and remote monitoring of performance statistics of the link. Required field. Default: ATT 54016.
Framing	This parameter specifies the multiframe format of the span. Required field. Default: ESF.
Audio Signal Adjustment into IP Network	This parameter specifies the gain or loss applied to the received audio signal relative to the port application type. Required field. Default: NoDbPadding.
Audio Signal Adjustment from IP Network	This parameter specifies the gain or loss applied to the transmitted audio signal relative to the port application type. Required field. Default: NoDbPadding.
Yellow Alarm	This parameter specifies how a remote alarm indication is coded on a T1 span. A yellow alarm indicates that the other end of the link has lost frame synchronization on the signal being transmitted by this end. Required field. Default: Bit2.
Zero Suppression	This parameter specifies how the T1 or E1 span electrically codes binary 1s and 0s on the wire (line coding selection). Required field. Default: B8ZS.
Digit On Duration (50–500ms)	This parameter specifies the duration in milliseconds of Dual Tone Multi-Frequency (DTMF) digits generated by the gateway. Valid durations range from 50–500ms. Required field. Default: 100. Minimum: 50. Maximum: 500.
Interdigit Duration (50–500ms)	This parameter specifies the duration to pause between digits when sequences of DTMF digits are generated by the gateway. This parameter is in milliseconds. Valid durations range from 50–500ms. Required field. Default: 100. Minimum: 50. Maximum: 500.

continues

Table 3-3 Product Specific Configuration Detail (Continued)

Parameter	Description
SNMP Community String	This parameter specifies the community string to be used for accessing the Simple Network Management Protocol (SNMP) agent for this interface. Default: public.
Disable SNMP Set Operations	This parameter specifies whether all SNMP set operations are disabled. Required field. Default: false.
Debug Port Enable	This parameter specifies whether the developer (debug) port should be enabled. Required field. Default: true.
Hold Tone Silence Duration	This parameter specifies, in milliseconds, the inter-tone (off) duration of the hold tone. If this value is set to zero, the default duration of 10 seconds is used. Required field. Default: 0. Minimum: 0. Maximum: 65535.
Port Used for Voice Calls	This parameter should be selected if this port is used for voice calls. If this port is used only for fax or modem calls, do not select this option. Required field. Default: true.
Port Used for Modem Calls	This parameter should be selected if this port is used for modem calls. If this port is used only for fax or voice calls, do not select this option. Required field. Default: true.
Port Used for Fax Calls	This parameter should be selected if this port is used for fax calls. If this port is used only for voice or modem calls, do not select this option. Required field. Default: true.
Fax and Modem Parameters	
Parameter	Description
Fax Relay Enable	This parameter specifies whether Cisco fax relay encoding should be negotiated when a fax tone is detected. Required field. Default: true.
Fax Error Correction Mode Override	This parameter specifies whether error correction mode (ECM) should be disabled for fax transmissions using Cisco fax relay. Required field. Default: true.

Table 3-3 Product Specific Configuration Detail (Continued)

Parameter	Description
Maximum Fax Rate	This parameter specifies the maximum fax rate to negotiate. This parameter applies only to fax transmissions using Cisco fax relay. Required field. Default: 14400bps.
Fax Payload Size	This parameter specifies the fax payload size. The default is 20 bytes. Required field. Default: 20. Minimum: 20. Maximum: 48.
Non-Standard Facilities Country Code	This parameter specifies the non-standard facilities country code. A value of 65535 will leave this field unused. This parameter applies only to fax transmissions using Cisco fax relay. Required field. Default: 65535. Minimum: 0. Maximum: 65535.
Non-Standard Facilities Vendor Code	This parameter specifies the non-standard facilities vendor code. A value of 65535 will leave this field unused. This parameter applies only to fax transmissions using Cisco fax relay. Required field. Default: 65535. Minimum: 0. Maximum: 65535.
Fax/Modem Packet Redundancy	This parameter specifies whether packet redundancy (RFC2198) should be enabled for modem calls and fax calls not using Cisco fax relay. Required field. Default: false.
Named Service Event (NSE) Type	This parameter specifies the NSE type to be used for peer-to-peer messaging. Required field. Default: Non-IOS Gateways.
Playout Delay Parameters	
Parameter	Description
Initial Playout Delay	This parameter specifies the initial delay introduced by the jitter buffer in milliseconds. Required field. Default: 40. Minimum: 20. Maximum: 150.

continues

Table 3-3 Product Specific Configuration Detail (Continued)

Parameter	Description
Minimum Playout Delay	This parameter specifies the minimum delay introduced by the jitter buffer in milliseconds. Required field. Default: 20. Minimum: 20. Maximum: 150.
Maximum Playout Delay	This parameter specifies the maximum delay introduced by the jitter buffer in milliseconds. Required field. Default: 150. Minimum: 20. Maximum: 150.
Echo Canceller Configuration	
Parameter	Description
Echo TailLength (ms)	This parameter defines the tail length duration (in ms) to be used by the echo canceller. Supported values are 24, 32, 48 and 64ms. Required field. Default: 32ms.
Minimum Echo Return Loss (ERL) (db)	This parameter defines the minimum ERL value that can be handled by the Echo Canceller. Available options are 0db, 3db and 6db. Required field. Default: 6db.

Adding Inter-Cluster Trunks

The last type of gateway to be covered in this section is what is known an Inter-Cluster Trunk. This type of gateway is configured to allow calls to be placed between CallManager clusters across some type of IP connectivity. Inter-cluster trunks are normally used to connect clusters, within the same organization, together across an exiting WAN link, thus utilizing the WAN link for both data and voice.

The concept is very simple—in each cluster a gateway is configured that points to the IP address(es) of the CallManager in the other cluster. One thing that often causes confusion at first, is that there is no real gateway “device” that is used, unlike with other types of gateways. The easiest way to think of it is that an inter-cluster trunk is a process that runs on the CallManager cluster.

The following steps show how to configure an inter-cluster trunk and explain the various settings of one. Because there are a number of steps to this process, section headings are used to mark the point at which each new set of parameters begins. These same headings display on the configuration screen as well, which should help you keep track of where you are.

Step 1 From within CCMAdmin, select **Device>Add a New Device**.

Step 2 On the Add a New Device page, select **Trunk** from the Device Type drop-down list and click **Next**.

Step 3 On the next page, select the Inter-Cluster Trunk (Non-Gatekeeper Controlled) from the Trunk Type drop-down list.

Note Typically non-gatekeeper controlled inter-cluster trunks are used only when bandwidth is not an issue. Gatekeepers and gatekeeper-controlled inter-cluster trunks are discussed in Chapter 5: Configuring Class of Service and Call Admission Control. The configuration of inter-cluster trunks is very similar for both types.

Step 4 The Device Protocol field can be left at Inter-Cluster Trunk. No other option is available. Click the **Next** button.

Device Information

Step 5 The Trunk Configuration screen, as shown in Figure 3-16, displays. Enter a functional name for the gateway in the Device Name field.

Figure 3-16 *Trunk Configuration*

The screenshot shows the 'Trunk Configuration' page. At the top, it displays 'Product: Inter-Cluster Trunk (Non-Gatekeeper Controlled)' and 'Device Protocol: Inter-Cluster Trunk'. Below this, the 'Status' is listed as 'Ready' with an 'Insert' button. The main area is divided into two sections: 'Device Information' and 'Call Routing Information'.

Device Information:

- Device Name***: A required field with a red asterisk.
- Description**: An optional field.
- Device Pool***: A dropdown menu showing 'Not Selected'.
- Device Destination***: A dropdown menu showing 'OnNet'.
- Media Resource Group List**: A dropdown menu showing '<None>'.
- Location**: A dropdown menu showing '<None>'.
- AAR Group**: A dropdown menu showing '<None>'.
- Tunneled Protocol**: A dropdown menu showing '<None>'.
- Checkboxes:**
 - Media Termination Point Required
 - Retry Video Call as Audio
 - Path Replacement Support

Call Routing Information:

Inbound Calls:

- Significant Digits***: A dropdown menu showing 'All'.

- Step 6** In the Description field, enter a description that makes this device easily identifiable.
- Step 7** From the Device Pool drop-down list, select the desired device pool for this gateway.
- Step 8** The Device Destination is used to determine if calls on this trunk will be considered an OnNet (on network) or OffNet (off network) call. Different alerting tones are used for OnNet and OffNet calls. By default, this is set to OnNet. If you are unsure of the value to assign, leave it at the default.
- Step 9** The next field is the Media Resource Group List. This determines to which media resources the gateway and device have access. Media resources group list determines to which media resources a device has access. These are discussed further in Chapter 6: Configuring CallManager Features and Services.
- Step 10** Information entered in the Location field is used to prevent WAN links from becoming oversubscribed in centralized deployments. These are discussed more in Chapter 5: Configuring Class of Service and Call Admission Control. If you have defined locations, select the appropriate one for this device from the drop-down list.
- Step 11** The AAR Group field determines with which AAR group this device is associated. An AAR group provides the prefix that is assigned when a call fails due to insufficient bandwidth. AAR is discussed in further detail in Chapter 6: Configuring CallManager Features and Services. Select an AAR group if AAR is being used. If this field is set to None, AAR is, in effect, disabled on this device.
- Step 12** The Tunneled Protocol drop-down list allows you to select Q Signaling (QSIG), which enables inter-cluster trunk (ICT) to transport non-H.323 protocol information by tunneling it through H.323. Leave this set to None, unless you know that this type of tunneling is required.
- Step 13** The Media Termination Point Required check box needs to be checked if the H.323 device does not support features such as hold and transfers.
- Step 14** If the Retry Video Call as Audio box is checked, CallManager sets up a voice call if a video call fails to set up.
- Step 15** The Path Replacement Support is automatically checked if you select QSIG from the Tunneled Protocol drop-down list. Otherwise it is left unchecked.

Call Routing Information—Inbound Calls

- Step 16** The next set of fields deals with inbound calls. The Significant Digits field determines the number of digits of an incoming dialed number that CallManager uses. CallManager counts from right to left, so if the

number entered in this field is four and the digits received are 8105559090, 810555 would be removed and only 9090 would be used to determine the destination of this call.

Step 17 A Calling Search Space (CSS) determines what destinations inbound calls to this gateway will be able to reach. CSS is discussed in Chapter 5: Configuring Class of Service and Call Admission Control. Choose a CSS from the Calling Search Space drop-down list. If this field is left at None, the dialing privileges of this gateway could be limited.

Step 18 The Automated Alternate Routing (AAR) is used to provide an alternate route if a call fails due to insufficient bandwidth. The AAR CSS can be used to limit the paths a call may use when it is rerouted. Select an AAR CSS from the AAR Calling Search Space drop-down list.

Step 19 The Prefix DN field defines what digits are added to the front of an incoming destination number. This is applied to the number, after CallManager truncates the number, based on the Significant Digits setting.

Step 20 The Redirecting Number IE Delivery—Inbound should be used if your voicemail system supports Redirecting Number IE. Otherwise, leave this box unchecked.

Step 21 If the Enable Inbound Faststart check box is checked, FastStart will be used. H.323 FastStart requires only two message exchanges to open logical channels, whereas normal setup requires 12. However, if FastStart is selected, both ends must support and be configured for FastStart.

Call Routing Information—Outbound Calls

Step 22 The next set of fields deals with outbound calls. The Calling Party Selection field determines what number is sent for outbound calls. The choices are:

- Originator—the directory number of the device that placed the call
- First Redirect Number—the directory number of the first device to redirect the call
- Last Redirect Number—the directory number of the last device to redirect the call
- First Redirect Number (External)—the external directory number of the first device to redirect the call
- Last Redirect Number (External)—the external directory number of the last device to redirect the call

Select the desired value for this field.

Step 23 The Calling Line ID Presentation field determines if CallManager sends Caller ID information. To send caller ID information, select **Allowed** from the drop-down list. To block caller ID, select **Restricted** from the drop-down list.

Step 24 Cisco recommends that the next four fields remain set to the default of Cisco CallManager. The four fields are:

- Called party IE number type unknown
- Calling party IE number type unknown
- Called Numbering Plan
- Calling Numbering Plan

These fields deal with dial plan issues and should be changed only when advised to do so by Cisco or an experienced dial plan expert. The need to change these usually occurs when installing CallManager internationally.

Step 25 The Caller ID DN field is used to determine what caller ID is sent out this gateway. A mask or a complete number can be entered in this field. For example, if the mask 55536XX is entered in this field, CallManager sends 55536 and the last two digits of the calling number.

Step 26 If the Display IE Delivery check box is checked, the calling and called party name information is included in messages.

Step 27 The Redirecting Number IE Delivery–Outbound check box should be checked when integrating with a voicemail system that supports Redirecting Number IE. Otherwise, leave it unchecked.

Step 28 If the Enable Outbound Faststart check box is checked, FastStart will be used. H.323 FastStart requires only two message exchanges to open logical channels, whereas normal setup requires 12. However, if FastStart is selected, both ends must support and be configured for FastStart.

Step 29 If the Enable Outbound Faststart check box is checked, you must select the codec that is to be used. This is selected from the Codec For Outbound FastStart drop-down list.

Remote Cisco CallManager Information

Step 30 In the next three fields, the IP addresses of the CallManager in the destination cluster are entered. Enter the IP addresses of the remote CallManager group. Enter the remote primary CallManager IP address in the Server 1 field, the remote backup CallManager IP address in the Server 2 field and, if remote tertiary CallManager exists, enter its IP address in the Server 3 field. If there are more than three CallManagers in the remote cluster, additional inter-cluster trunks may need to be configured.

Multilevel Precedence and Preemption (MLPP) Information

Step 31 The next three fields define the Multilevel Precedence and Preemption (MLPP) characteristics of this trunk. If these fields are left blank or set to default, the values set in the device pool are used. The first MLPP field is the MLPP Domain. MLPP grants higher priority only from calls with the same MLPP domain. For this reason an MLPP domain is needed.

Step 32 The second field in this category, which is called MLPP Indication, determines whether tones and indications will be presented when a precedence call is made. If the is field set to Off, no precedence indication is presented. If this field is set to On, indication is used for a precedence call.

Step 33 The third MLPP field is MLPP Preemption. This parameter determines whether a higher precedence call preempts a lower precedence call. The value is not available for this device.

Step 34 After all the fields are configured, click the **Insert** button at the top of the screen to add this ICT.

Summary

This chapter covered all the required tasks needed to add phones and gateways to a CallManager system. Four methods to add phones to the system were discussed: manual registration, Auto-registration, BAT, and TAPS.

After phones were added, the addition of four types of gateways was discussed. A description and an installation example of H.323, MGCP, Non-MGCP, and Inter-Cluster Trunk gateways were provided.

The chapter should help you feel comfortable with the task of adding phones and gateways. Because it is not possible to include step-by-step instructions for every possible type of phone and gateway in this book, the examples given were based on widely deployed models. These instructions should serve as a good guide to help you deploy most other models as well.

Now that the phones and gateways are deployed, a dial plan is needed. A dial plan is used to determine how calls are routed. Let's move onto the next chapter and explore dial plans.