Getting Started with Your Cisco SIP IP Phone

This chapter explains the Cisco SIP IP phone initialization and the process that you should follow to install and connect the Cisco SIP IP phone. It provides the following sections:

Overview of the Initialization Process

The initialization process of the Cisco SIP IP phone establishes network connectivity and makes the phone operational in your IP network. Once you connect your phone to the network and to an electrical supply, the phone begins its initialization process. During the initialization process, the following events take place:

1. The stored image is loaded.

   The Cisco SIP IP phone has nonvolatile Flash memory in which it stores the firmware images, user-defined preferences, and permanent factory information about the phone.

   During initialization, the phone runs a bootstrap loader that loads and executes the phone image stored in Flash memory.

2. The VLAN is configured.

   If the Cisco SIP IP phone is connected to a Catalyst switch, the switch notifies the phone of the voice VLAN defined on the switch. The phone needs to know its VLAN membership before it can proceed with the Dynamic Host Configuration Protocol (DHCP) request for its IP settings (if using DHCP).

3. An IP address is acquired.

   If the Cisco SIP IP phone is using DHCP to obtain the IP settings, the phone queries the DHCP server. If the phone is not using DHCP, the phone uses IP settings that are stored in Flash memory.

4. The TFTP server is contacted.

The TFTP server contains the latest Cisco SIP IP phone firmware image and the dual boot file (OS79XX.TXT) that enables the phone to automatically determine and initialize the VoIP environment in which it is being installed.

If the phone is using the TFTP server to obtain its SIP parameters, there should also be a configuration file or files on the TFTP server that the phone will request and download. In the configuration file or files, SIP parameters that are required by the phone to operate in a SIP VoIP environment are defined. If the phone is not obtaining its SIP parameters via the TFTP server, the phone uses SIP settings that are stored in Flash memory.

5. The firmware version is verified.

If the phone is obtaining its SIP parameters via a TFTP server, the configuration files are requested. If the phone determines that the image defined in a configuration file differs from the image it has stored in Flash memory, it performs a firmware upgrade. When performing a firmware upgrade, the phone downloads the firmware image from the TFTP server, programs the image into Flash memory, and reboots.

Installing the Cisco SIP IP Phone

This section contains information on how to install Cisco SIP IP phones in your IP network. Before getting started, read over the information in this section carefully.

Installation Task Summary

To successfully install the Cisco SIP IP phone, complete the following steps:

Step 1 Download the required files from Cisco.com to the TFTP server as described in the "Downloading Files to Your TFTP Server" section on page 2-3.

Step 2 If you are configuring SIP parameters via a TFTP server, create and store the configuration files as described in the "Configuring SIP Parameters Using a TFTP Server" section on page 2-4. If you are not configuring the SIP parameters via a TFTP server, manually configure the required parameters as described in the "Configuring the SIP Parameters Manually" section on page 2-7.

Step 3 If you are using DCHP to configure the phone network settings, configure the required network parameters on your DHCP server as described in the "Configuring Network Parameters Using a DHCP Server" section on page 2-10. If you are not using DHCP to configure network parameters, manually configure the required network parameters as described in the "Configuring the Network Parameters Manually" section on page 2-10.

Step 4 Connect the phone to the network and to a power supply as described in the "Connecting the Phone" section on page 2-11.

Downloading Files to Your TFTP Server

Before installing the Cisco SIP IP phones, copy the following files from Cisco.com to the root directory of your TFTP server.

File	Required or Optional	Description
OS79XX.TXT	Required	Enables the phone to automatically determine and initialize the VoIP environment in which it is being installed.
		After downloading this file, you must use an ASCII editor to open it and specify the filename (without the file extension) of the image version that you plan to run on your phones.
SIPDefault.cnf	Optional	File in which to configure SIP parameters intended for all phones.
		For more information on using the SIPDefault.cnf file, see the "Creating the Default SIP Configuration File" section on page 2-5.
SIPConfigGeneric.cnf	Required	File that can be used as a template to configure SIP parameters specific to a phone. When customized for a phone, this file must be renamed to the MAC address of the phone.
RINGLIST.DAT	Optional	Lists audio files that are the custom ring type options for the phones. The audio files listed in the RINGLIST.DAT file must also be in the root directory of the TFTP server.
		For more information on custom ring types, see the "Customizing the Cisco SIP IP Phone Ring Types" section on page 2-17.
P0S3*xxyy*.bin	Required	The Cisco SIP IP phone firmware image. The *xx* variable represents the version number, and the *yy* variable represents the subversion number.
		Note Applies to Cisco SIP IP Phone Release 2.3 and earlier.
P0S3-*xx-y-zz*.bin	Required	The Cisco SIP IP phone firmware image. The *xx* variable represents the major version number, the *y* variable represents the minor version number, and the *zz* variable represents the subversion number.
		Note Applies to Cisco SIP IP Phone Release 3.0 and later.
dialplan.xml	Optional	North American sample dial plan. The dialplan.xml file can be pushed down to the phones using a NOTIFY with a check-sync Event header.
syncinfo.xml	Optional	Controls the image version and associated synchronization value to be used for remote reboots.
SIPPhone Release Notes.4.2.pdf	Optional	Contains the 4.2 release notes.

Configuring SIP Parameters

Note   This section describes how to configure the basic SIP parameters that are required for the phone to operate in a SIP VoIP environment. For a complete list of the SIP parameters that you can configure, see the "Modifying the SIP Settings" section on page 3-6.

The SIP parameters are those parameters that a Cisco SIP IP phone needs in order to operate in a SIP VoIP environment. You can configure SIP parameters via a TFTP server, or you can manually configure the parameters on a phone-by-phone basis after connecting the phones.

When the phone initializes, it loads the parameters stored in Flash memory. After loading the parameters stored in Flash memory, the phone requests the default configuration file from the TFTP server. If the default configuration file has been configured and stored in the root directory of the TFTP server, the phone reads the parameters defined in the file and stores those parameters that differ in Flash memory. The phone then requests its phone-specific configuration file.

If the phone-specific configuration file has been configured and placed on the TFTP server (in the root directory or a subdirectory), the phone reads the parameters defined in the file and stores those parameters that differ in Flash memory.

Therefore, when configuring SIP parameters, remember the following:

- Parameters defined in the default configuration file override the values stored in Flash memory.
- Parameters defined in the phone-specific configuration file override the values specified in the default configuration file.
- Parameters entered locally are used by the phone until the next reboot (if a phone-specific configuration file exists).
- If you choose not to configure the phone via a TFTP server, you must manage the phone locally.

Configuring SIP Parameters Using a TFTP Server

If you are configuring SIP parameters using a TFTP server, you must use configuration files.

There are two configuration files that you can use to define the SIP parameters: the default configuration file (optional) and the phone-specific configuration file (required). If you choose to use a default configuration file, you must store the file in the root directory of your TFTP server. Phone-specific configuration files can be stored in the root directory or in a subdirectory in which all phone-specific configuration files are stored.

Except for parameters used to define the lines and users on a phone, all other SIP parameters can be defined in either the default configuration file or the phone-specific configuration file. However, for network control and maintenance purposes, we recommend that you define the parameters that you want to apply to all phones in the default configuration file (SIPDefault.cnf). Phone-specific parameters should only be defined via a phone-specific configuration file or be configured manually. Phone-specific parameters should not be defined in the default configuration file.

Configuration File Guidelines

When modifying the default configuration file and creating the phone-specific configuration files, adhere to the following guidelines and requirements:

- SIP parameters specified in the default configuration file (SIPDefault.cnf) override those parameters stored in Flash memory. Parameters specified in a phone-specific configuration file override those stored in Flash memory and those specified in the default configuration file.

- The name of each phone-specific configuration file is unique and is based on the MAC address of the phone.

  The format of the filename must be SIP*XXXXYYYYZZZZ*.cnf, where *XXXXYYYYZZZZ* is the MAC address of the phone. The MAC address must be in uppercase, and the cnf extension must be in lowercase (for example, SIP00503EFFD842.cnf).

  > ✎
  >
  > **Note**   The MAC address of a phone is identified on the middle sticker adhered to the base of the phone and can also be viewed on the Network Configuration menu.

- The default configuration file must be stored in the root directory of the TFTP server. The phone-specific configuration file can be stored in the root directory or in a subdirectory in which all phone-specific configuration files are located.

- Each line in the configuration files must use the following format:

  ```
  variable-name : value ; optional comments
  ```

- Use colons to separate variable names and values.

- Only one value can be associated with a variable.

- The variable and value can contain white space before or after them and can contain any characters. However, if white spaces are needed within the value, the value must be enclosed in single or double quotes. If the value is enclosed in quotes, the end quote must be the same as the start quote.

- After the value, you can include optional comments. Use the semicolon (;) and pound (#) delimiters to distinguish the comments.

- Blank lines are allowed.

- Comment lines are allowed.

- Variable names are not case sensitive.

- Only one variable can be set per line.

- Distinguish the end of a line using <lf> or <cr><lf>.

- The variable and value must be on the same line and cannot break the line.

- Except for parameters used to defined the lines and users on a phone, all other SIP parameters can be defined in either the default configuration file or the phone-specific configuration file. However, for network control and maintenance purposes, Cisco recommends that you define the parameters that you want to apply to all phones in the default configuration file (SIPDefault.cnf).

Creating the Default SIP Configuration File

In the default configuration file (SIPDefault.cnf), Cisco recommends that you define the SIP parameters that will be common to all of your phones such as the image_version parameter and call environment parameters (for example, you will want to consider if the phones are required to register with a proxy server, and which codec the phones will use when initiating a call).

By maintaining these parameters in the default configuration file, you can perform global changes, such as upgrading the image version, without having to modify the phone-specific configuration file for each phone.

Before You Begin

- Ensure that you have downloaded the SIPDefault.cnf file from Cisco.com to the root directory of your TFTP server.

- Review the guidelines documented in the "Configuration File Guidelines" section on page 2-5.

- For a complete list of the SIP parameters that you can configure, see the "Modifying the SIP Settings" section on page 3-6.

Procedure

Step 1   Using an ASCII editor, open the SIPDefault.cnf file and define values for the following SIP global parameters:

- image_version—(Required) Firmware version that the Cisco SIP IP phone should run.

    Enter the name of the image version (as it is released by Cisco). Do not enter the extension. You cannot change the image version by changing the filename, because the version is also built into the file header. Trying to change the image version by changing the filename causes the firmware to fail when it compares the version in the header against the filename.

- proxy1_address—(Required) IP address of the primary SIP proxy server that will be used by the phones.

- tftp_cfg_dir—(Required if phone-specific configuration files are located in a subdirectory) Path to the TFTP subdirectory in which phone-specific configuration files are stored.

Step 2   Save the file with the same filename, SIPDefault.cnf, to the root directory of your TFTP server.

The following is an example of a default SIP configuration file:

```
#Image Version
image_version:P0S3-xx-y-zz ;

#Proxy server address
proxy1_address: 192.168.1.1 ;

#Subdirectory config file location
tftp_cfg_dir: /tftpboot/configs/sipphone
```

Creating the Phone-Specific SIP Configuration File

In the phone-specific SIP configuration file, define the parameters that are specific to a phone, such as the lines configured on a phone and the users defined for those lines.

Before You Begin

- Review the guidelines documented in the "Configuration File Guidelines" section on page 2-5.

- Line parameters (those identified as line*x*) define a line on the phone. If you configure a line to use an e-mail address, that line can be called only by using an e-mail address. Similarly, if you configure a line to use a number, that line can be called only by using the number. Each line can have a different proxy configured.

- For a complete list of the SIP parameters that you can configure, see the "Modifying the SIP Settings" section on page 3-6.

Procedure

Step 1   Using an ASCII editor, create a phone-specific configuration file for each phone that you plan to install. In the phone-specific configuration file, define values for the following SIP parameters (where x is a number 1 through 6):

- line$x$_name—(Required) Number or e-mail address used when registering. When entering a number, enter the number without any dashes. For example, enter 555-1212 as 5551212. When entering an e-mail address, enter the e-mail ID without the host name.

- line$x$_authname—(Required when registration is enabled and the proxy server requires authentication) Name used by the phone for authentication if a registration is challenged by the proxy server during initialization. If a value is not configured for the line$x$_authname parameter when registration is enabled, the default name is used. The default name is UNPROVISIONED.

- line$x$_password—(Required when registration is enabled and the proxy server requires authentication) Password used by the phone for authentication if a registration is challenged by the proxy server during initialization. If a value is not configured for the line$x$_password parameter when registration is enabled, the default logical password is used. The default logical password is UNPROVISIONED.

Step 2   Save the file to your TFTP server (in the root directory or in a subdirectory that contains all the phone-specific configuration files). Name the file SIP*XXXXYYYYZZZZ*.cnf where *XXXXYYYYZZZZ* is the MAC address of the phone. The MAC address must be in uppercase and the extension, cnf, must be in lowercase (for example, SIP00503EFFD842.cnf).

The following is an example of a phone-specific SIP configuration file:

```
; Line 1 phone number
line1_name : 5551212

; Line 1 name for authentication with proxy server
line1_authname : 5551212

; Line 1 authentication name password
line1_password : password
```

Configuring the SIP Parameters Manually

If you did not configure the SIP parameters via a TFTP server, you must manually configure them after you have connected the phone as described in the "Connecting the Phone" section on page 2-11.

Before You Begin

- Connect your phone as described in the "Connecting the Phone" section on page 2-11.

- Unlock configuration mode as described in the "Locking Configuration Mode" section on page 3-3. By default, the SIP parameters are locked to ensure that end users cannot modify settings that might affect their call capabilities.

- Review the guidelines on using the Cisco SIP IP phone menus documented in the "Using the Cisco SIP IP Phone Menu Interface" section on page 2-15.

- When configuring the Preferred Codec and Out of Band DTMF parameters, press the **Change** soft key until the option that you desire is displayed and then press the **Save** soft key.
- After making your changes, relock configuration mode as described in the "Locking Configuration Mode" section on page 3-3.
- For a complete list of the SIP parameters that you can configure, see the "Modifying the SIP Settings" section on page 3-6.

Procedure

Step 1	Press the **settings** key. The Settings menu is displayed.
Step 2	Highlight **SIP Configuration**. The SIP Configuration menu is displayed.
Step 3	Highlight **Line 1 Settings**.
Step 4	Press the **Select** soft key. The Line 1 Configuration menu is displayed.
Step 5	Highlight and press the **Select** soft key to configure the parameters shown in Table 2-1:

Table 2-1   Manual SIP Configuration Parameters

Parameter	Required or Optional	Description
Name	Required	Number or e-mail address used when registering. When entering a number, enter the number without any dashes. For example, enter 555-1212 as 5551212. When entering an e-mail address, enter the e-mail ID without the host name.
Shortname	Optional	Name or number associated with the line*x*_name as you want it to display on the phone's LCD if the line*x*_name value exceeds the display area. For example, if the line*x*_name value is the phone number 111-222-333-4444, you can specify 34444 for this parameter to have 34444 display on the LCD instead. Alternately, if the value for the line*x*_name parameter is the e-mail address "username@company.com," you can specify the "username" to have just the username appear on the LCD instead. This parameter is used for display purposes only. If a value is not specified for this parameter, the value in the Name variable is displayed.
Authentication Name	Required when registration is enabled	Name used by the phone for authentication if a registration is challenged by the proxy server during initialization. If a value is not configured for the Authentication Name parameter when registration is enabled, the default name is used. The default name is UNPROVISIONED.
Authentication Password	Required when registration is enabled	Password used by the phone for authentication if a registration is challenged by the proxy server during initialization. If a value is not configured for the Authentication Password parameter when registration is enabled, the default logical password is used. The default password is UNPROVISIONED.

Table 2-1   Manual SIP Configuration Parameters (continued)

Parameter	Required or Optional	Description
Display Name	Optional	Identification as it should appear for caller-identification purposes. For example, instead of jdoe@company.com displaying on phones that have caller ID, you can specify John Doe in this parameter to have John Doe display on the called party end instead. If a value is not specified for this parameter, the Name value is used.
Proxy Address	Required for the first line configured on the phone	IP address of the primary SIP proxy server that will be used by the phone. Enter this address in IP dotted-decimal notation or as an FQDN.
Proxy Port	Required for the first line configured on the phone	Port of the primary SIP proxy server that is used by the phone.

Step 6   Press the **Back** soft key to exit the Line 1 Configuration menu.

Step 7   To configure additional lines on the phone, highlight the next **Line *x* Settings**, press the **Select** soft key, and repeat Step 5 and Step 6.

Step 8   When done, press the **Save** soft key to save your changes and exit the SIP Configuration menu.

Configuring Network Parameters

Note   This section describes how to configure the basic network parameters that are required for the phone to operate on the network. For a complete list of the network parameters that you can configure, see the "Modifying the SIP Settings" section on page 3-6.

The network parameters include those parameters that must be configured on a phone in order for the phone to operate in an IP network. You can configure the required network parameters using DHCP or configure the parameters manually after you have connected the phone to a power supply.

The following parameters must be defined in order for your phone to establish network connectivity:

- Phone IP address
- Subnet mask
- Default gateway for the subnet (use "0.0.0.0" if not required)
- Domain name
- DNS server IP address (use "0.0.0.0" if not required)
- TFTP server IP address
- Backup proxy server

When configuring the network parameters of an IP phone, adhere to the following guidelines:

- Use 0.0.0.0 for unused IP addresses.
- You can use 0.0.0.0 for the subnet mask only if the default gateway is also 0.0.0.0.
- The TFTP server must have a nonzero IP address.
- The default gateway must be on the same subnet as the phone.
- The default gateway can be 0.0.0.0 only if the TFTP or DNS server is on the same subnet as the phone.

Note   By default, DHCP is enabled on your phone. Before you can manually configure the network parameters, you must disable DHCP after connecting your phone to a power supply.

Configuring Network Parameters Using a DHCP Server

If you are using DHCP to configure the network parameters, configure the following DHCP options on your DHCP server before you connect your Cisco SIP IP phone:

- dhcp option #50 (IP address)
- dhcp option #1 (IP subnet mask)
- dhcp option #3 (Default IP gateway)
- dhcp option #15 (Domain name)
- dhcp option #6 (DNS server IP address)
- dhcp option #66 (TFTP server IP address)

Configuring the Network Parameters Manually

If you are not using DHCP to configure your network parameters, you must manually configure them.

Before You Begin

- Connect your phone as described in the "Connecting the Phone" section on page 2-11.
- Unlock configuration mode as described in the "Unlocking Configuration Mode" section on page 3-2. By default, the network parameters are locked to ensure that end users cannot modify settings that might affect their network connectivity.
- Review the guidelines on using the Cisco SIP IP phone menus documented in the "Using the Cisco SIP IP Phone Menu Interface" section on page 2-15.
- When configuring a domain name:
  – Press the **Number** soft key to enter a numerical ID or press the **Alpha** soft key to enter a name.
  – If entering letters, use the numbers on the dial pad associated with a particular letter. For example, the 2 key has the letters A, B, and C. For a lowercase *a*, press the 2 key once. To scroll through the available letters and numbers, press the key repeatedly.
  – Press the **<<** soft key to delete any mistakes.

- After making your changes, relock configuration mode as described in the "Locking Configuration Mode" section on page 3-3.

- For a complete list of the SIP parameters that you can configure, see the "Modifying the SIP Settings" section on page 3-6.

Procedure

Step 1   Press the **settings** key. The Settings menu is displayed.

Step 2   Highlight **Network Configuration**.

Step 3   Press the **Select** soft key. The Network Configuration menu is displayed.

Step 4   Highlight **DHCP Enabled**.

Step 5   Press the **No** soft key. DHCP is now disabled.

Step 6   Highlight and configure each of the following parameters:

- IP Address—IP address of the phone.

- Subnet Mask—IP subnet mask used by the phone.

- TFTP Server—IP address of the TFTP server from which the phone downloads its configuration files and firmware images.

- Default routers 1 through 5—IP address of the default gateway used by the phone. Default routers 2 through 5 are the IP addresses of the gateways that the phone attempts to use as an alternate gateway if the primary gateway is not available.

- Domain Name—Name of the DNS domain in which the phone resides.

- DNS servers 1 through 5—IP address of the DNS server used by the phone to resolve names to IP addresses. The phone attempts to use DNS servers 2 through 5 if DNS server 1 is unavailable.

Step 7   When done, press the **Save** soft key. The phone programs the new information into Flash memory and resets.

Connecting the Phone

You must connect the phone to the network and to a power source before using it.

Before You Begin

Refer to Figure 2-1 for a graphical overview of the procedures in this section.

Figure 2-1   Cisco SIP IP Phone Cable Connections

Procedure

Step 1 Connect a Category 3 or 5 straight-through Ethernet cable from the switch or hub to the *network* port on the phone.

See the "Connecting to the Network" section on page 1-13 for more information on the network port.

Step 2 Connect the handset and headset to their respective ports.

See the "Using a Headset" section on page 1-14 for more information on the headset port.

Step 3 Connect a Category 3 or 5 straight-through Ethernet cable from another network device, such as a desktop computer, to the *access* port on the phone (optional).

See the "Connecting to the Network" section on page 1-13 for more information on the access port.

Step 4 Connect the power plug to the Cisco AC adapter port (optional).

See the "Connecting to Power" section on page 1-13 for more information.

Adjusting the Placement of the Cisco SIP Phone

The Cisco SIP IP phone includes an adjustable footstand. When placing the phone on a desktop surface, you can adjust the tilt height to several different angles in 7.5 degree increments from flat to 60 degrees. Alternatively, you can mount the phone to the wall using the footstand or using the optional locking accessory.

Adjusting the Phone Placement on the Desktop

Adjust the footstand to the height that provides optimum view of the display and use of the buttons and keys as shown in Figure 2-2.

To adjust the phone placement on the desktop:

Step 1   Push in the footstand adjustment knob.

Step 2   Adjust the footstand to its desired height and release the knob.

Mounting the Phone to the Wall

You can mount the Cisco SIP IP phone on the wall using the footstand as a mounting bracket or using the optional locking bracket. Use the following procedure to mount the phone on the wall using the standard footstand. To use the optional locking bracket, refer to the *Installing the Wall Mount Kit for the Cisco IP Phone* document.

Before You Begin

Mounting the Cisco SIP IP phone on the wall requires some tools and equipment that are not provided as standard equipment.

Following are the tools and parts required for a typical Cisco SIP IP phone installation:

- Screwdriver
- Screws to secure the Cisco SIP IP phone to the wall

Procedure

Step 1   Push in the footstand adjustment knob.

Step 2   Adjust the footstand so that it is flat against the back of the phone, as shown in Figure 2-2.

Step 3   Modify the handset rest so that the handset remains on the ear-piece rest when the phone is vertically placed.

    a.   Remove the handset from the ear-piece rest.

    b.   Locate the tab (handset wall hook) at the base of the ear-piece rest.

    c.   Slide this tab out, rotate it 180 degrees, and reinsert it.

    d.   Place the handset on the ear-piece rest.

Step 4   Insert two screws into a wall stud, matching them to the two screw holes on the back of the footstand.

The keyholes fit standard phone jack mounts.

Step 5   Hang the phone on the wall.

Figure 2-2   Adjusting the Footstand

Verifying Startup

After the phone has power connected to it, the phone begins its startup process by cycling through these steps:

1. The following buttons flash on and off in sequence:

   – Headset

   – Mute

   – Speaker

2. The Cisco Systems, Inc. copyright appears on the LCD.

3. The following messages appear as the phone starts up:

   – Configuring VLAN—The phone is configuring the Ethernet connection.

   – Configuring IP—The phone is contacting the DHCP server to obtain network parameters and the IP address of the TFTP server.

   – Requesting Configuration—The phone is contacting the TFTP server to request its configuration files and compare firmware images.

   – Upgrading Software—The Upgrade Software message displays only if the phone has determined that an image upgrade is required. After upgrading the image, the phone automatically reboots to run the new image.

4. The main LCD displays the following:

   – Primary directory number

   – Soft keys

If the phone successfully passes through these stages, it has started up properly.

Using the Cisco SIP IP Phone Menu Interface

As you configure your phone's settings via the menu interface, follow these guidelines:

- Select a parameter by pressing the down arrow to scroll to and highlight the parameter or by pressing the number that represents the parameter (located to the left of the parameter on the LCD).

- During configuration, use * for dots (periods) or press the "." soft key when available on the LCD.

- Press **Cancel** during configuration to cancel all changes and exit a menu.

- When configuring a SIP IP address or ID parameter:

  – Press the **Number** soft key to enter a numerical value or press the **Alpha** soft key to enter a name.

  – Use the buttons on the dial pad to enter a new value.

  – If entering letters, use the numbers on the dial pad associated with a particular letter. For example, the 2 key has the letters A, B, and C. For a lowercase *a*, press the 2 key once. To scroll through the available letters and numbers, press the key repeatedly.

  – Press the **<<** soft key to delete any mistakes.

- When configuring a network IP address or ID parameter:

  – Use the buttons on the dial pad to enter a new value.

  – Press the **<<** soft key to delete any mistakes.

- After editing a parameter, press the **Validate** soft key to save the value that you have entered and exit the Edit panel.

Reading the Cisco SIP IP Phone Icons

When using the Cisco SIP IP phone, a variety of icons can display on the phone's LCD. Table 2-2 lists and describes each icon that you might see while using the Cisco SIP IP phone.

Table 2-2   Description of the Cisco SIP IP Phone User Interface Icons

Icon	Description
	The Cisco IP phone that you are using is running SIP.
	The line is configured for E.164 number dialing, and you can enter only numbers when placing the call. The character *x* displayed to the right of the icon indicates that registration has failed.

Table 2-2   Description of the Cisco SIP IP Phone User Interface Icons (continued)

Icon	Description
	The line is configured for E.164 number dialing and is ready for you to place the call. When a line is configured for E.164 number dialing, you can enter only numbers when placing the call. You can change to URL dialing at any time while dialing on a line by pressing the **URL** soft key. The character *x* displayed to the right of the icon indicates that registration has failed.
	The line is configured for URL dialing, and you can enter both numbers and letters when placing the call. The character *x* displayed to the right of the icon indicates that registration has failed.
	The line is configured for URL dialing and is ready for you to place the call. When a line is configured for URL dialing, you can enter both numbers and letters when placing the call. You can change to E.164 number dialing at any time while dialing on a line by pressing the **Number** soft key. The character *x* displayed to the right of the icon indicates that registration has failed.
	The Cisco SIP IP phone configuration mode is locked. When the phone is locked, the phone's network or SIP settings cannot be modified.
	The Cisco SIP IP phone configuration mode is unlocked. When the phone is unlocked, the phone's network or SIP settings can be modified.
	A normal two-way call is on hold (call display is blinking).
	A normal two-way call is connected and communicating.
	A three-way call is on hold (call display is blinking).
	A three-way call is connected and communicating.

Customizing the Cisco SIP IP Phone Ring Types

The Cisco SIP IP phone ships with two ring types: Chirp1 and Chirp2. By default, your ring type options will be those two choices. However, using the RINGLIST.DAT file, you can customize the ring types that are available to Cisco SIP IP phone users.

Step 1   Create a pulse code modulation (PCM) file of the desired ring types and store the PCM files in the root directory of your TFTP server. PCM files must contain no header information and must comply with the following format guidelines:

- 8000-Hz sampling rate

- 8 bits per sample

- u-law compression

Step 2   Using an ASCII editor, open the RINGLIST.DAT file, and for each of the ring types that you are adding, specify the name as you want it to appear on the Ring Type menu, press **Tab**, and then specify the filename of the ring type. For example, the format of a pointer in your RINGLIST.DAT file should appear similar to the following:

```
Ring Type 1ringer1.pcm
```

Step 3   After defining pointers for each of the ring types that you are adding, save your modifications and close the RINGLIST.DAT file.

Creating Dial Plans

Dial plans enable the Cisco SIP IP phone to support automatic dialing and automatic generation of a secondary dial tone. If a single dial plan is to be used for a system of phones, the dial plan is best specified in the default configuration file. However, you can create multiple dial plans and specify which phones are to use which dial plan by defining the dial_template parameter in the phone-specific configuration file. If one phone in a system of phones needs to use a different dial plan than the rest, you need to define the differing dial plan by specifying the dial_template parameter in that phone's phone-specific configuration file.

Note   We recommend that you define the dial_template parameter in the default configuration file for maintenance and control purposes. Specify the dial_template parameter in a phone-specific configuration file only if that phone needs to use a different dial plan than is being used by the other phones in the same system.

When creating a dial plan, remember the following:

- Dial plans must be in an .xml format and must be stored on your TFTP server.

- You must specify which dial plan a phone is to use by specifying the path to the dial plan in the dial_template parameter that you define in either the phone-specific configuration file or the default configuration. We recommend that the dial_template parameter be defined in the default configuration file unless a specific phone must use a dial plan that differs from the one being used by other phones in the same system.

- <DIALTEMPLATE> indicates the start of a template and </DIALTEMPLATE> indicates the end of a template.
- Rules are matched from start to finish with the longest matching rule taken as the one to use. Matches against a period are not counted for the length to be the longest.

To create your dial plan, perform the following steps:

Step 1   Using an ASCII editor, open a new file.

Step 2   Type **<DIALTEMPLATE>** to indicate the start of the dial-plan template.

Step 3   For each of the numbering schemes that you wish to define, add the following string to the template, each starting on a separate line:

`TEMPLATE MATCH="pattern" Timeout="sec" User="type" Rewrite="xxx" Route="route"`

Where:

- MATCH= "*pattern*" is the dial pattern to match. When entering the value of *pattern*, use a period (.) to match any character or use an asterisk (*) to match one or more characters. To have the phone generate a secondary dial tone when the part of the template matches, use a comma (,).
- Timeout= "*sec*" is the number of seconds before a timeout occurs, and the number is dialed as entered by the user. To have the number dial immediately, specify 0.
- User= "*type*" is the either IP or Phone. Enter **User=phone** or **User=IP** to have the tag automatically added to the dialed number. This entry is not case sensitive.
- Rewrite= "*xxx*" is the alternate string to be dialed instead of what the user enters.

  The rewrite rules are matched from start to finish with the longest matching rule taken as the one to use. Matches against a period are not counted for the length to be the longest. A complete rule is not matched unless it has more nonwildcard matches than an incomplete rule. You can put comments into the file with <!-- to start the comment and --> to end it.

  The rules allow for substitution of up to five replacement strings as well as picking off replaced digits one at a time. For example, with a match string of "ab..cd..ef*" and an input string of "ab12cd34ef5678."

  The following replacement strings work, as follows.

REWRITE	Output	Notes
%s	ab12cd34ef5678	
%0	ab12cd34ef5678	
%1	12	
%2	34	
%3	56	
%4	None	
%5	None	
XYZ....	XYZ1234	
X.Y.Z...	X1Y2Z345	
919%1%2%3	919123456	
AB...X%1X..	AB123X12X45	Note how "12" appears twice.
X%1X%1X%1	X12X12X12	You can reuse the string.

REWRITE	Output	Notes
X%s%%	Xab12cd34ef5678% %%	This produces a %.
919	919	No need to use the input.
....	12345678	Note that nothing goes in for the extra dots.

- Route= "*route*" is default, emergency, or FQDN. FQDN is treated the same as default proxy. Route indicates which proxy the call is to be routed to. This entry is not case sensitive.

Step 4   If desired, specify <!--*comment*--> at the end of each string, where *comment* defines the type of plan (for example, Long Distance or Corporate Dial Plan).

Step 5   When completed, enter </DIALTEMPLATE> to indicate the end of the dial-plan template.

Step 6   Give the file a unique name specific to the dial plan it defines, and save the file with an .xml extension to your TFTP server.

Step 7   If the dial plan applies to a specific phone, add the path to the dial plan (without specifying the file type of .xml) using the dial_template parameter in the phone-specific configuration file. If the dial plan applies to a system of phones, add the path to the dial plan using the dial_template parameter in the default configuration file. For more information on defining the dial_template parameter, see the "Modifying the SIP Settings" section on page 3-6.

The following is an example of a North American dial plan:

```
<DIALTEMPLATE>
    <TEMPLATE MATCH="0" Timeout="1" "User=phone"/> <!-- Local operator -->
    <TEMPLATE MATCH="9,011*" Timeout="6" "User=phone"/> <!-- International calls -->
    <TEMPLATE MATCH="9,0" Timeout="2" User="Phone"/> <!-- PSTN Operator-->
    <TEMPLATE MATCH="9,11" Timeout="0" User="Phone" Rewrite="9911"/> <!-- Emergency -->
    <TEMPLATE MATCH="w!" Timeout="1" User="PHONE" Rewrite="9911"/> <!-- 911 when entered
                     in Alpha mode -->
    <TEMPLATE MATCH="9,.11" Timeout="0" User="Phone"/> <!-- Service numbers -->
    <TEMPLATE MATCH="9,101..............." Timeout="0" User="Phone"/> <!-- Long Distance
                Service -->
    <TEMPLATE MATCH="9,10............." Timeout="0" User="Phone"/> <!-- Long Distance
                Service -->
    <TEMPLATE MATCH="9,10*" Timeout="6" User="Phone"/> <!-- Long Distance Service -->
    <TEMPLATE MATCH="9,1.........." Timeout="0" User="Phone"/> <!-- Long Distance -->
    <TEMPLATE MATCH="9,......." Timeout="0" User="Phone"/> <!-- Local numbers -->
    <TEMPLATE MATCH="*" Timeout="15"/> <!-- Anything else -->
</DIALTEMPLATE>
```

Updates to the Dial-Plan Template in Release 4.4 and Later

The dial-plan template has been updated so that you can specify the # and * as a dialed digits and the comma (,) can be specified a secondary dial tone.

Specifying the #

Pressing the # is still processed as a "dial now" event and remains the default behavior unless you specify the # in the dial-plan template. If you specify the # in the dial-plan template, the phone does not dial immediately when the # is pressed but does continue to match the dial-plan template that specifies the #. The # is not matched by the wildcard character * or the period (.).

Specifying the *

The * was used as a dial-plan wildcard character, so there was no way to specify it as a dialed digit. Beginning with Release 4.4, an escape sequence was added to indicate that the * should be processed as a dialed digit rather than as a wildcard. The escape sequence is a backward slash (\) and * so that the syntax is *. The phone automatically strips the \ so that it does not appear in the outgoing dial string. When * is received as a dialed digit, it is matched by the wildcard characters * and period (.).

Specifying the Comma

Prior to this release, specifying a comma (,) in the dial-plan template caused the phone to play the default secondary dial tone (Bellcore-Outside). With this release and later, support is added so that you can specify which tones are played when the comma (,) is specified.

A new XML token named TONE was added to the dial-plan template. If the comma (,) is specified and there is no TONE token present, the phone plays the default secondary dial tone. If the comma (,) is specified and there is a TONE token present, the phone plays the indicated tone instead of the secondary dial tone. If a tone is specified but there is no comma (,) in the match string, the tone is ignored.

Up to 3 different secondary dial tones can be specified in a single dial-plan template. The order in which the tones are listed determines the order in which the tones are played. If multiple commas are specified, they are condensed to a single comma for the purposes of processing the dial-plan template. For example, if the match string is entered as "9,,,234" the phone interprets the string as "9,234" and the 3 commas are treated as a single comma.

The tones are case insensitive and are defined as follows:

- Bellcore-Inside
- Bellcore-Outside
- Bellcore-Busy
- Bellcore-dr1
- Bellcore-Reorder
- Bellcore-CallWaiting
- Bellcore-Hold
- Bellcore-Reminder
- Cisco-ZipZip
- Cisco-Zip
- Cisco-BeepBonk
- Bellcore-None
- Bellcore-Confirmation

- Bellcore-Permanent

The following are examples of the changes to the dial-plan template:

Specifying the # in the Dial-Plan Template Example

The following is an example using the # as a dialed digit:

```
<DIALTEMPLATE>
    <TEMPLATE MATCH="123#45#6" TIMEOUT="0" User="Phone"/> <!-- Match '#' -->
    <TEMPLATE MATCH="34#..."   TIMEOUT="0" User="Phone"/> <!-- Match '#' -->
    <TEMPLATE MATCH="*"        TIMEOUT="15" User="Phone"/>
<DIALTEMPLATE/>
```

In the example above, the "123#45#6" string is matched if the user dials "123#45#6". Pressing the # does not cause the phone to dial immediately because the # is specified. However, dialing "1#" or "123#4#" would cause the phone to dial immediately.

Specifying the * in the Dial-Plan Template Example

The following is an example specifying the * as a dialed digit:

```
<DIALTEMPLATE>
    <TEMPLATE MATCH="12\*345" TIMEOUT="0" User="Phone"/> <!-- Match * Char -->
    <TEMPLATE MATCH="*"       TIMEOUT="10" User="Phone"/> <!-- Wildcard -->
</DIALTEMPLATE>
```

If the user specifies the \ with a digit other than the *, the \ is ignored and \\ is matched, for example, specifying \7 will match the digit 7. The \ is not sent out as part of the dialed digit string because the phone removes it before sending the dial string.

Specifying a Secondary Dial Tone in the Dial-Plan Template Example

The following is an example specifying two different tones:

```
<DIALTEMPLATE>
    <TEMPLATE MATCH="7,..." TIMEOUT="0" /> <!-- Default Secondary Dial tone -->
    <TEMPLATE MATCH="9,..." TIMEOUT="0" Tone="Cisco-Zip" /> <!-- Play Zip -->
    <TEMPLATE MATCH="8,...." TIMEOUT="0" Tone="Bellcore-Hold" /> <!-- Play Hold -->
    <TEMPLATE MATCH="8,123,...." TIMEOUT="0" Tone="Bellcore-Hold" Tone="Cisco-Zip" />
                                  <!--Play Hold after 8, Play Zip Tone after 123-->
</DIALTEMPLATE>
```