Managing Cisco SIP IP Phones

This chapter provides information on the following:

Entering Configuration Mode

When you access the network configuration information on your Cisco SIP IP phone, you will notice that there is a padlock symbol located in the upper right corner of your LCD. By default, the network configuration information is locked. Before you can modify any of the network configuration parameters, you must unlock the phone.

Unlocking Configuration Mode

To unlock the Cisco SIP IP phone, complete the following steps:

Step 1 Press ****#**.

Note You have activated the configuration mode for your phone. There is no indication an action has taken place.

Step 2 Press the **settings** key.

Step 3 Highlight **Network Configuration**.

Step 4 Press the **Select** soft key. The Network Configuration menu is displayed. The lock symbol in the upper right corner of your LCD should be in an unlocked state. The unlocked symbol indicates that you can modify the network settings.

Locking Configuration Mode

To lock the Cisco SIP IP phone when you are done modifying the settings, complete the following steps:

Step 1 Press ****#**.

Step 2 Press the **settings** key.

Step 3 Highlight **Network Configuration**.

Step 4 Press the **Select** soft key. The Network Configuration menu is displayed. The lock symbol in the upper right corner of your LCD should be in a locked state. The locked symbol indicates that the network settings cannot be modified.

Modifying the Phone's Network Settings

You can display and configure the network settings of a Cisco SIP IP phone. The network settings include information such as the phone's DHCP server, MAC address, IP address, and domain name.

Before You Begin

When configuring network settings, remember the following:

- 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 2-21.
- After making your changes, relock configuration mode as described in the "Locking Configuration Mode" section on page 3-2.

Procedure

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

Step 2 Highlight **Network Configuration**.

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

The following network parameters are available on the Network Configuration menu:

- DHCP Server—IP address of the DHCP server from which the phone received its IP address and additional network settings. You cannot change the information in this field.
- MAC Address—Factory-assigned unique 48-bit hexadecimal MAC address of the phone. You cannot change the information in this field.
- Host Name—Unique host name assigned to the phone. The value in this field is always SIP*mac* where *mac* is the MAC address of the phone. You cannot change the information in this field.
- Domain Name—Name of the DNS domain in which the phone resides.

- IP Address—IP address of the phone that was assigned by DHCP or locally configured. To edit this field, DHCP must be disabled.
- Subnet Mask—IP subnet mask used by the phone. A subnet mask partitions the IP address into a network and a host identifier. To edit this field, DHCP must be disabled.
- TFTP Server—IP address of the TFTP server from which the phone downloads its configuration files and firmware images. To edit this field, DHCP must be disabled.
- 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 will attempt to use as an alternate gateway if the primary gateway is not available. To edit this field, DHCP must be disabled.
- DNS Servers 1 through 5—IP address of the DNS server used by the phone to result names to IP addresses. The phone will attempt to use DNS Servers 2 through 5 if DNS Server 1 is unavailable. To edit this field, DHCP must be disabled.
- Operational VLAN Id—Unique identifier of the VLAN of which the phone is a member. This identifier is obtained through Cisco Discovery Protocol (CDP). You cannot change the information in this field.
- Admin. VLAN Id—Unique identifier of the VLAN to which the phone is attached. The value in this field is only used in non-Cisco switched networks. You can change the administrative VLAN used by the phone, however, if you have an administrative VLAN assigned on the Catalyst switch, that setting overrides any changes made on the phone.
- DHCP Enabled—Whether the phone will use DHCP to configure network settings (IP address, subnet mask, domain name, default router list, DNS server list, and TFTP address). Possible values for this field are Yes and No. By default, DHCP is enabled on the phone. To manually configure your IP settings, you must first disable DHCP.
- DHCP Address Released—Whether the IP address of the phone can be released for reuse in the network. When you set this field to **Yes**, the phone sends a DHCP release message to the DHCP server and goes into a release state. The release state provides enough time to remove the phone from the network before the phone attempts to acquire another IP address from the DHCP server. When moving the phone to a new network segment, you should first release the DHCP address.

- Alternate TFTP—Whether to use an alternate TFTP server. This field enables an administrator to specify the remote TFTP server rather than the local one. Possible values for this parameter are Yes and No. The default is No. When Yes is specified, the IP address in the TFTP Address parameter must be changed to the address of the alternate TFTP server.
- Erase Configuration—Whether to erase all of the locally-defined settings on the phone and reset the values to the defaults. Selecting Yes will re-enable DHCP. For more information on erasing the local configuration, see the "Erasing the Locally-Defined Settings" section on page 3-16.

Step 4 When done, press the **Save** soft key. The phone programs the new information into Flash memory and resets.

Caution When you have completed your changes, ensure that you lock the phone as described in the "Locking Configuration Mode" section on page 3-2.

Modifying the Phone's SIP Settings

You can modify the SIP parameters of a Cisco SIP IP phone.

When modifying SIP parameters, remember the following:

- Parameters defined in the default configuration file will override the values stored in Flash memory.
- Parameters defined in the phone-specific configuration file will override the values specified in the default configuration file.
- Parameters entered locally will be 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.

Table 3-1 lists each of the SIP parameters that you can configure. In the Configuration column, the name of a parameter as you would specify it in a configuration file is listed. In the SIP Configuration Menu, the name of the same parameter as it would appear on the Cisco SIP IP phone user interface is listed.

If a parameter name does not appear in the SIP Configuration menu column, it can only be defined via a configuration file.

Table 3-1 SIP Parameters

Configuration File	SIP Configuration Menu
image_version	
proxy1_address	Proxy Address
proxy1_port	Proxy Port
tos_media	
preferred_codec	Preferred Codec
dtmf_inband	
dtmf_db_level	
dtmf_outofband	Out of Band DTMF
timer_t1	
timer_t2	
timer_invite_expires	
sip_retx	
sip_invite_retx	
proxy_register	Register with Proxy
timer_register_expires	Register Expires
messages_uri	Messages URI
linex_name	Name
linex_shortname	Shortname
linex_authname	Authentication Name
linex_password	Authentication Password
userx_name	User Name

Table 3-1 SIP Parameters (continued)

Configuration File	SIP Configuration Menu
userx_authname	User Authentication Name
userx_password	User Password

Modifying SIP Parameters via a TFTP Server

If you have set up your phones to retrieve their SIP parameters via a TFTP server as described in the "Configuring SIP Parameters via a TFTP Server" section on page 2-6, you can also modify your SIP parameters using the configuration files.

As explained in the "Configuring SIP Parameters" section on page 2-5, there are two configuration files that you can use to define the SIP parameters; the default configuration file and the phone-specific configuration file. These configuration files must be stored in the root directory of your TFTP server.

We recommend that you use the default configuration file to define values for SIP parameters that are common to all the phones in your system. You can use the phone-specific configuration file to define parameters that are specific to a phone. Phone-specific parameters should only be defined via a phone-specific configuration file or manually configured. Phone-specific parameters should not be defined in the default configuration file.

Modifying the Default SIP Configuration File

In the default configuration file (SIPDefault.cnf), we recommend that you maintain the SIP parameters that are common to all of your phones.

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 CCO to the root directory of your TFTP server.
- Review the guidelines and restrictions documented in the "Configuration File Guidelines" section on page 2-6.

Procedure

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

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

  Enter the name of the image version (as it is release by Cisco). Do not enter the extension. You cannot change the image version by changing the file name because the version is also built into the file header. Trying to change the image version by changing the file name will cause the firmware to fail when it compares the version in the header against the file name.
- proxy1_address—(Required) IP address of the primary SIP proxy server that will be used by the phones. Enter this address in IP dotted-decimal notation.
- proxy1_port—(Optional) Port number of the primary SIP proxy server. This is the port on which the SIP client will listen for messages. The default is 5060.
- tos_media—(Optional) Type of Service (ToS) level for the media stream being used. Valid values are:
  - 0 (IP_ROUTINE)
  - 1 (IP_PRIORITY)
  - 2 (IP_IMMEDIATE)
  - 3 (IP_FLASH)
  - 4 (IP_OVERIDE)
  - 5 (IP_CRITIC)

  The default is 5.
- preferred_codec—(Optional) CODEC to use when initiating a call. Valid values are g711alaw, g711ulaw, and g729a. The default is g711ulaw.
- dtmf_inband—(Optional) Whether to detect and generate in-band signaling format. Valid values are 1 (generate DTMF digits in-band) and 0 (do not generate DTMF digits in-band). The default is 1.
- dtmf_db_level—(Optional) In-band DTMF digit tone level. Valid values are:
  - 1 (6 db below nominal)
  - 2 (3 db below nominal)

  - 3 (nominal)
  - 4 (3 db above nominal)
  - 5 (6 db above nominal)

  The default is 3.

- dtmf_outofband—(Optional) Whether to detect and generate the out-of-band signaling (for tone detection on the IP side of a gateway) and if so, when. The Cisco SIP IP phone supports out-of-bound signaling via the AVT tone method. Valid values are:
  - none—Do not generate DTMF digits out-of-band.
  - avt—If requested by the remote side, generate DTMF digits out-of-band (and disable in-band DTMF signaling), otherwise, do not generate DTMF digits out-of-band.
  - avt_always—Always generate DTMF digits out-of-band. This option disables in-band DTMF signaling.

  The default is avt.
- timer_t1—(Optional) Lowest value (in milliseconds) of the retransmission timer for BYE, CANCEL, OPTIONS, or REGISTER requests. The valid value is any positive integer. The default is 500.
- timer_t2—(Optional) Highest value (in milliseconds) of the retransmission timer for BYE, CANCEL, OPTIONS, or REGISTER requests. The valid value is any positive integer greater than time_t1. The default is 4000.
- timer_invite_expires—(Optional) The amount of time, in seconds, after which a SIP INVITE will expire. This value is used in the Expire header field. The valid value is any positive number, however, we recommend 180 seconds. The default is 180.
- sip_retx—(Optional) Maximum number of times a SIP message other than an INVITE request will be retransmitted. The valid value is any positive integer. The default is 11.
- sip_invite_retx—(Optional) Maximum number of times an INVITE request will be retransmitted. The valid value is any positive integer. The default is 7.

- proxy_register—(Optional) Whether the phone must register with a proxy server during initialization. Valid values are 0 and 1. Specify 0 to disable registration during initialization. Specify 1 to enable registration during initialization. The default is 1.

Note If you enable registration, and authentication is required, you must specify values for the line1_authname and line1_password parameters in the phone-specific configuration file. For information on configuring the phone-specific configuration file, see the "Modifying the Phone-Specific SIP Configuration File" section on page 3-11.

- timer_register_expires—(Optional) The amount of time, in seconds, after which a REGISTRATION request will expire. This value is inserted into the Expire header field. The valid value is any positive number, however, we recommend 3600 seconds. The default is 3600.
- messages_uri—(Optional) Number to call to check voicemail. This number will be called when the **Messages** key is pressed.

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

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

```
; sip default configuration file

#Image Version
image_version:P0S3xxyy ;

#Default Codec
preferred_codec :g711ulaw

#Enable Registration
proxy_register :1 ;

#Registration expiration
timer_register_expires :3600 ;

#Proxy address
proxy1_address: 192.168.1.1 ;
```

Modifying the Phone-Specific SIP Configuration File

In the phone-specific SIP configuration file, maintain those 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 and restrictions documented in the "Configuration File Guidelines" section on page 2-6.
- 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 using an e-mail address. Similarly, if you configure a line to use a number, that line can only be called using the number.
- User parameters (those identified as user*x*) define a user that is using the line for call placing purposes. If the user*x* parameters are not defined, the line*x* parameters will be used by default.

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 (in which *x* is 1 or 2):

- 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*_shortname—(Optional) Displays a name or number associated with the value specified in the line*x*_name parameter. This parameter is for display-only purposes. For example, you can display a short name on the LCD for a long line*x*_name value. If a value is not specified for this parameter, the value in the line*x*_name variable is displayed.
- line*x*_authname—(Required when registration is enabled) Name used by the phone for authentication if a registration is challenged by the proxy server during initialization.
- line*x*_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 line*x*_password

parameter when registration is enabled, the default logical password is used. The default logical password is SIP*macaddress* where *macaddress* is the MAC address of the phone.

- user*x*_name—(Optional) Name of the user of the line. This information is used when a call is being placed. The value defined in this parameter will be used in the From: header field in the SIP INVITE request. If no value is specified for this parameter, the value for the line*x*_name parameter will be used.
- user*x*_authname—(Optional) Name used to authenticate the user if the call is challenged. If a value is not specified for this parameter, the value for the line*x*_authname parameter is used.
- user*x*_password—(Optional) Password used to authenticate the user if the call is challenged. If a value is not specified for this parameter, the value for the line*x*_password parameter is used.

Step 2 Save the file in the root directory of your TFTP server, naming it "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 lower case (for example, SIP00503EFFD842.cnf).

The following is an example of a configuration file:

```
; phone-specific configuration file sample
; 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
```

Modifying the SIP Parameters Manually

If you did not configure the SIP parameters via a TFTP server, you can configure them manually after you have connected the phone.

Before You Begin

- Unlock configuration mode as described in the "Unlocking Configuration Mode" section on page 3-2. 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 2-21.
- 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 using an e-mail address. Similarly, if you configure a line to use a number, that line can only be called using the number.
- User parameters (those identified as user*x*) define a user that is using the line for call placing purposes. If the user*x* parameters are not defined, the values defined for the line*x* parameters will be used by default.
- When configuring the Preferred Codec and Out of Band DTMF parameters, press the **Change** soft key until the option 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 3-2.

Procedure

Step 1 Press the s**ettings** 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 following parameters as necessary:

- 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.
- Short Name—(Optional) Displays a name or number associated with the value specified in the Name parameter. This parameter is for display-only purposes. For example, you can display a short name on the LCD for a long Name value. 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.
- 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 logical password is SIP*macaddress* where *macaddress* is the MAC address of the phone.
- User Name—(Optional) Name of the user of the line. This information is used when a call is being placed. The value defined in this parameter will be used in the From: header field in the SIP INVITE request. If no value is specified for this parameter, the value for the line*x*_name parameter is used.
- User Authentication Name—(Optional) Name used to authenticate the user if the call is challenged. If a value is not specified for this parameter, the value for the Authentication parameter is used.
- User Password—(Optional) Password used to authenticate the user if the call is challenged. If a value is not specified for this parameter, the value for the Authentication Password parameter will be used.
- Proxy Address—(Required) IP address of the primary SIP proxy server that will be used by the phone. Enter this address in IP dotted-decimal notation.
- Proxy Port—(Optional) Port number of the primary SIP proxy server. This is the port on which the SIP client will listen for messages. The default is 5060.

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

Step 7 To configure the second line on the phone, highlight **Line 2 Settings**.

Step 8 Press the **Select** soft key. The Line 2 Configuration menu is displayed.

Step 9 Repeat Step 5 and Step 6, then continue with Step 10.

Step 10 In addition to the line settings, you can highlight and press **Select** to configure the following parameters on the SIP Configuration menu:

- Message URI—Number to call to check voicemail. This number will be called when the **Messages** key is pressed.
- Preferred Codec—(Optional) CODEC to use when initiating a call. Valid values are g711alaw, g711ulaw, and g729a. The default is g711ulaw.
- Out of Band DTMF—(Optional) Whether to detect and generate the out-of-band signaling (for tone detection on the IP side of a gateway) and if so, when. The Cisco SIP IP phone supports out-of-bound signaling via the AVT tone method. Valid values are:
  - none—Do not generate DTMF digits out-of-band.
  - avt—If requested by the remote side, generate DTMF digits out-of-band (and disable in-band DTMF signaling), otherwise, do not generate DTMF digits out-of-band.
  - avt_always—Always generate DTMF digits out-of-band. This option disables in-band DTMF signaling.

  The default is avt.
- Register with Proxy—(Optional) Whether the phone must register with a proxy server during initialization. Valid values are Yes and No. Select the **No** soft key to disable registration during initialization. Select the **Yes** soft key to enable registration during initialization. The default is No.

  Note If you enable registration, and authentication is required, you must specify values for the Authentication Name and Authentication Password parameters.

- Register Expires—(Optional) The amount of time, in seconds, after which a REGISTRATION request will expire. This value is used the Expire header field. The valid value is any positive number, however, we recommend 3600 seconds. The default is 3600.

Step 11 When done, press the **Save** soft key to save your changes and exit the SIP Configuration menu.

Caution When you have completed your changes, ensure that you lock the phone as described in the "Locking Configuration Mode" section on page 3-2.

Erasing the Locally-Defined Settings

You can erase the locally-defined network settings and the SIP settings that have been configured in the phone.

Erasing the Locally-Defined Network Settings

When you erase the locally-defined settings, the values are reset to the defaults.

Before You Begin

- Unlock configuration mode as described in the "Unlocking Configuration Mode" section on page 3-2.
- If DHCP has been disabled on a phone, clearing the phone's settings will reenable it.
- Select the Erase Config 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 name on the LCD).

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 settings are displayed.

Step 4 Highlight **Erase Configuration**.

Step 5 Press the **Yes** soft key.

Step 6 Press the **Save** soft key. The phone programs the new information into Flash memory and resets.

Erasing the Locally-Defined SIP Settings

When you erase the locally-defined SIP settings, the values are reset to the defaults.

Note If your system has been set up to have the phones retrieve their SIP parameters via a TFTP server, you will need to edit the configuration file in which a parameter is defined to delete the parameter. When deleting a parameter, leave the variable in the file, but remove its value. If both the variable and its value are removed, the phone will use the setting for that variable that it has stored in Flash memory.

Before You Begin

Unlock configuration mode as described in the "Unlocking Configuration Mode" section on page 3-2.

Procedure

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

Step 2 Highlight **SIP Configuration**.

Step 3 Press the **Select** soft key. The SIP Configuration settings are displayed.

Step 4 Highlight the parameter for which you wish to erase the setting.

Step 5 Press the **Edit** soft key.

Step 6 Press the **<<** soft key to delete the current value.

Step 7 Press the **Validate** soft key to save your change and exit the Edit panel.

Step 8 If modifying a line parameter, press the **Back** soft key to exit the Line Configuration panel.

Step 9 Press the **Save** soft key. The phone programs the new information into Flash memory and resets.

Accessing Status Information

There are several types of status information that you can access via the **settings** key. The information that you can obtain via the **settings** key can aid in system management.

To access status information, select **settings** and then select **Status** from the Settings menu. The Status menu is displayed from which the following three options are available:

- Status Messages—Displays diagnostic messages.
- Network Status—Displays performance messages.
- Firmware Version—Displays information about the current firmware version on the phone.

In addition to the status messages available via the Setting Status menu, you can also obtain status messages for a current call.

Viewing Status Messages

To view status messages that you can use to diagnose network problems, complete the following steps:

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

Step 2 Highlight **Status**.

Step 3 Press the **Select** soft key. The Setting Status menu is displayed.

Step 4 Highlight **Status Messages**.

Step 5 Press the **Select** soft key. The Status Messages panel is displayed.

Step 6 To exit the Status Messages panel, press the **Exit** soft key.

Viewing Network Statistics

To view statistical information about the phone and network performance, complete the following steps:

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

Step 2 Highlight **Status**.

Step 3 Press the **Select** soft key. The Setting Status menu is displayed.

Step 4 Highlight **Network Statistics**.

Step 5 Press the **Select** soft key. The Network Statistics panel is displayed.

The following information is displayed on this panel:

- Rcv—Number of packets received by the phone; not through the switch.
- Xmit—Number of packets sent by the phone; not through the switch.
- REr—Number of packets received by the phone that contained errors.
- BCast—Number of broadcast packets received by the phone.
- Phone State Message—TCP messages indicating the state of the phone. Possible messages are:
  - Phone Initialized—TCP connection has not gone down since the phone was powered on.
  - Phone Closed TCP—TCP connection was closed by the phone.
  - TCP Timeout—TCP connection was closed because of a retry timeout.
  - Error Code—Error messages indicating unusual reasons the TCP connection was closed.

- Elapsed Time—Length of time (in days, hours, minutes, and seconds) since the last power cycle.
- Port 0 Full, 100—Indicates that the network is in a linked state and has auto-negotiated a full-duplex 100Mbps connection.
- Port 0 Half, 100—Indicates that the network is in a linked state and has auto-negotiated a half-duplex 100Mbps connection.
- Port 0 Full, 10—Indicates that the network is in a linked state and has auto-negotiated a full-duplex 10Mbps connection.
- Port 0 Half, 10—Indicates that the network is in a linked state and has auto-negotiated a half-duplex 10Mbps connection.
- Port 1 Full, 100—Indicates that the network is in a linked state and has auto-negotiated a full-duplex 100Mbps connection.
- Port 1 Half, 100—Indicates that the network is in a linked state and has auto-negotiated a half-duplex 100Mbps connection.
- Port 1 Full, 10—Indicates that the network is in a linked state and has auto-negotiated a full-duplex 10Mbps connection.
- Port 1 Half, 10—Indicates that the network is in a linked state and has auto-negotiated a half-duplex 10Mbps connection.

Step 6 To exit the Network Statistics panel, press the **Exit** soft key.

Note To reset the values displayed on Network Statistics panel, power off and power on the phone.

Viewing the Firmware Version

To view network statistics, complete the following steps:

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

Step 2 Highlight **Status**.

Step 3 Press the **Select** soft key. The Setting Status menu is displayed.

Step 4 Highlight **Firmware Versions**.

Step 5 Press the **Select** soft key. The Firmware Versions panel is displayed.

The following information is displayed on this panel:

- Application Load ID—Current software image on the phone.
- Boot Load ID—Bootstrap loader image version that is manufactured on the phone. This image name does not change.

Step 6 To exit the Firmware Versions panel, press the **Exit** soft key.

Updating the Cisco SIP IP Phone Firmware

There two methods that you can use to update the firmware on your Cisco SIP IP phones. You can update the firmware on one phone at a time via the phone-specific configuration or you can update the firmware on a system of phones using the default configuration file.

Before You Begin

- To update the firmware on just one phone at a time, you update the image_version in the phone-specific configuration file. To update the firmware on a system of phones, specify the image_version in the default configuration file and do not define the image_version in the phone-specific configuration files.
- Ensure that the latest version of the Cisco SIP IP phone firmware has been copied from CCO to the root directory of your TFTP server.

Procedure

Step 1 Copy the binary file P0S3*xxyy*.bin (where *xx* is the version number and *yy* is the subversion number) from CCO to the root directory of the TFTP server.

Step 2 Using a text editor, open the configuration file and update the image version specified in the image_version variable. The version name in image_version variable should match the version name (without the .bin extension) of the latest firmware that you downloaded.

Step 3 Reset each phone.

The phone contacts the TFTP server and requests its configuration files. The phone compares the image defined in the file to the image that it has stored in Flash memory. If the phone determines that the image defined in the file differs from the image in Flash memory, it downloads the image defined in the configuration file (which is stored in the root directory on the TFTP server). Once the new image has been downloaded, the phone programs that image into Flash memory and then reboots.

Note If you do not define the image_version parameter in the default configuration file, only phones for which you have updated their phone-specific configuration file with the new image version and restarted will use the latest firmware image. All other phones will use the older version until their configuration files have been updated with the new image version.