Managing Cisco SIP IP Phones

This chapter provides information on the following:

Changing Your Configuration

You can change your Cisco SIP IP phone configuration by any of the following methods:

- Using your phone's buttons and softkeys. You must first follow the instructions in the "Entering Configuration Mode" section on page 3-2.
- Edit the default and phone-specific configuration files on the TFTP server. See the "Modifying SIP Parameters via a TFTP Server" section on page 3-8.
- Use Telnet or a console to connect to your Cisco SIP IP phone and use the command-line interface (CLI). You will need to know your phone's IP address. Press **Settings**, select **Network Configuration**, and scroll down to IP Address to find this address. The default Telnet password is "cisco."

Note Use the CLI only to debug and troubleshoot your Cisco SIP IP phone.

You can change the following parameters:

- Network settings. See the "Modifying the Phone's Network Settings" section on page 3-2.
- SIP settings. See the "Modifying the Phone's SIP Settings" section on page 3-5.
- Call preferences settings. See the "Modifying the Phone's SIP Settings" section on page 3-5.

- XML URL settings. See the "Modifying the Phone's SIP Settings" section on page 3-5.
- Date, time, and Daylight Saving Time settings. See the "Setting the Date, Time, and Daylight Saving Time" section on page 3-36

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 Dynamic Host Configuration Protocol (DHCP) server, MAC address, IP address, and domain name.

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, press ****#**.

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

If the Network Configuration or SIP Configuration panel is displayed, the lock icon in the upper-right corner of your LCD changes to an unlocked state. If you are located elsewhere in the Cisco SIP IP phone menus, the next time you access the Network Configuration or the SIP Configuration panels, the lock icon will be displayed in an unlocked state.

The unlocked symbol indicates that you can modify the network and SIP configuration settings.

Locking Configuration Mode

To lock the Cisco SIP IP phone when you are done modifying the settings, press ****#**.

If the Network Configuration or SIP Configuration panel is displayed, the lock icon in the upper-right corner of your LCD changes to a locked state. If you are located elsewhere in the Cisco SIP IP phone menus, the next time you access the Network Configuration or the SIP Configuration panels, the lock icon will be displayed in a locked state.

The unlocked symbol indicates that you can modify the network and SIP configuration settings.

Changing the Network Settings

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 2-15.
- After making your changes, relock configuration mode as described in the "Locking Configuration Mode" section on page 3-2.

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.

Table 3-1 lists the network parameters available in the Network Configuration menu.

Table 3-1 Network Configuration Parameters

Parameter	Can Edit?	Description
Admin. VLAN Id	Yes, but if you have an administrative VLAN assigned on the Catalyst switch, that setting overrides any changes made on the phone.	Unique identifier of the VLAN to which the phone is attached. The value in this field is used only in switched networks that are not Cisco networks.
Alternate TFTP	Yes	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.
Default Routers 1 through 5	Yes, but DHCP must be disabled.	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 unavailable.
DHCP Address Released	Yes	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.
DHCP Enabled	Yes	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). Valid 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 Server	No	IP address of the DHCP server from which the phone received its IP address and additional network settings.

Table 3-1 Network Configuration Parameters (continued)

Parameter	Can Edit?	Description
DNS Servers 1 through 5	Yes, but DHCP must be disabled.	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.
Domain Name	Yes	Name of the DNS domain in which the phone resides.
Dynamic DNS Server 1 and 2	No	You can specify the IP address of a new dynamic DNS server. If a new DNS server is specified, it is used for any further DNS requests after the phone uses the initial DNS address upon bootup. The DNS addresses are used in the following order: **1.** dyn_dns_addr_1 (If present) **2.** dyn_dns_add_2 (If present) **3.** DNS Server 1 **4.** DNS Server 2 **5.** DNS Server 3 **6.** DNS Server 4 **7.** DNS Server 5 The dynamic DNS address is not stored in Flash memory.
Dynamic TFTP Server	No	You can specify the IP address of a new dynamic TFTP server. After initially querying the default TFTP server, the phone will re-request the default and MAC-specific configuration files from the new TFTP server. The dynamic TFTP server is not stored in Flash memory.
Erase Configuration	Yes	Whether to erase all of the locally defined network settings on the phone and reset the values to the defaults. Selecting Yes reenables DHCP. For more information on erasing the local configuration, see the "Erasing the Locally Defined Settings" section on page 3-41.
Host Name	No	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.
HTTP Proxy Address	Yes	The IP address of the HTTP proxy server. You can use either a dotted IP address or a DNS name (A record only).
HTTP Proxy Port	Yes	The port number of the outbound proxy port. The default is 80.
IP Address	Yes, but DHCP must be disabled.	IP address of the phone that either was assigned by DHCP or was locally configured.
MAC Address	No	Factory-assigned unique 48-bit hexadecimal MAC address of the phone.

Table 3-1 Network Configuration Parameters (continued)

Parameter	Can Edit?	Description
Network Media Type	Yes	Ethernet port negotiation mode. Valid values are: • Auto—Port is autonegotiated. (This is the default value.) • Full-100—Port is configured to be a full-duplex, 100-MB connection. • Half-100—Port is configured to be a half-duplex, 100-MB connection. • Full-10—Port is configured to be a full-duplex, 10-MB connection. • Half-10—Port is configured to be a half-duplex, 10-MB connection.
Network Port 2 Device Type	Yes	The device type that is connected to port 2 of the phone. Valid values are: • Hub/Switch (default) • PC **Note** If the value is PC, port 2 can be connected only to a PC. If you are not sure about the connection, use the default value. Using a value of "PC" and connecting port 2 to a switch results in spanning tree loops and network confusion.
Operational VLAN Id	No	Unique identifier of the VLAN of which the phone is a member. This identifier is obtained through Cisco Discovery Protocol (CDP).
Subnet Mask	Yes, but DHCP must be disabled.	IP subnet mask used by the phone. A subnet mask partitions the IP address into a network and a host identifier.
TFTP Server	Yes, but DHCP must be disabled.	IP address of the TFTP server from which the phone downloads its configuration files and firmware images.

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 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.

Table 3-2 lists each of the SIP parameters that you can configure. In the Configuration File column, the name of a parameter as you would specify it in a configuration file is listed. In the menu column (SIP Configuration, Network Configuration, Call Preferences, and Time/Date), the name of the same parameter as it would appear on the user interface is listed. If NA appears for a parameter name in a menu column, it cannot be defined using that menu.

Table 3-2 SIP Parameters Summary

Configuration File	SIP Configuration Menu	Network Configuration Menu	Call Preferences	Time/Date
anonymous_call_block	NA	NA	Anonymous Call Block	NA
autocomplete	NA	NA	Auto-Complete Numbers	NA
callerid_blocking	NA	NA	Caller ID Blocking	NA
call_waiting	NA	NA	Call Waiting	NA
cnf_join_enable	NA	NA	NA	NA
date_format	NA	NA	NA	Date Format
dial_template	NA	NA	NA	NA
dnd_control	NA	NA	Do Not Disturb	NA
dst_auto_adjust	NA	NA	NA	NA
dst_offset	NA	NA	NA	NA
dst_start_day	NA	NA	NA	NA
dst_start_day_of_week	NA	NA	NA	NA
dst_start_month	NA	NA	NA	NA
dst_start_time	NA	NA	NA	NA
dst_start_week_of_month	NA	NA	NA	NA
dst_stop_day	NA	NA	NA	NA
dst_stop_day_of_week	NA	NA	NA	NA
dst_stop_month	NA	NA	NA	NA
dst_stop_time	NA	NA	NA	NA
dst_stop_week_of_month	NA	NA	NA	NA
dtmf_avt_payload	NA	NA	NA	NA
dtmf_db_level	NA	NA	NA	NA
dtmf_inband	NA	NA	NA	NA
dtmf_outofband	Out of Band DTMF	NA	NA	NA
enable_vad	Enable VAD	NA	NA	NA
end_media_port	End Media Port	NA	NA	NA

Table 3-2 SIP Parameters Summary (continued)

Configuration File	SIP Configuration Menu	Network Configuration Menu	Call Preferences	Time/Date
image_version	NA	NA	NA	NA
line*x*_authname (line1 to line6)	Authentication Name	NA	NA	NA
line*x*_displayname (line1 to line6)	Display Name	NA	NA	NA
line*x*_name (line1 to line6)	Name	NA	NA	NA
line*x*_password (line1 to line6)	Authentication Password	NA	NA	NA
line*x*_shortname (line1 to line6)	Shortname	NA	NA	NA
messages_uri	Messages URI	NA	NA	NA
nat_address	NAT WAN Address	NA	NA	NA
nat_enable	NAT Enabled	NA	NA	NA
nat_received_processing	NA	NA	NA	NA
network_media_type	NA	Network Media Type	NA	NA
network_port2_type	NA	Network Port 2 Device Type	NA	NA
outbound_proxy	Outbound Proxy	NA	NA	NA
outbound_proxy_port	Outbound Proxy Port	NA	NA	NA
phone_label	Phone Label	NA	NA	NA
phone_password	NA	NA	NA	NA
phone_prompt	NA	NA	NA	NA
preferred_codec	Preferred Codec	NA	NA	NA
proxy_backup	Backup Proxy	NA	NA	NA
proxy_backup_port	Backup Proxy Port	NA	NA	NA
proxy_emergency	Emergency Proxy	NA	NA	NA
proxy_emergency_port	Emergency Proxy Port	NA	NA	NA
proxy_register	Register with Proxy	NA	NA	NA
proxyN_address (N=1 to 6)	Proxy Address	NA	NA	NA
proxyN_port (N=1 to 6)	Proxy Port	NA	NA	NA
remote_party_id	NA	NA	NA	NA
sip_invite_retx	NA	NA	NA	NA
sip_retx	NA	NA	NA	NA
sntp_mode	NA	NA	NA	NA
sntp_server	NA	NA	NA	NA
start_media_port	Start Media Port	NA	NA	NA
sync	NA	NA	NA	NA

Table 3-2 SIP Parameters Summary (continued)

Configuration File	SIP Configuration Menu	Network Configuration Menu	Call Preferences	Time/Date
tftp_cfg_dir	TFTP Directory	NA	NA	NA
time_format_24hr	NA	NA	NA	Time format 24-hr
time_zone	NA	NA	NA	Time Zone
timer_invite_expires	NA	NA	NA	NA
timer_register_expires	Register Expires	NA	NA	NA
timer_t1	NA	NA	NA	NA
timer_t2	NA	NA	NA	NA
tos_media	NA	NA	NA	NA
user_info	NA	NA	NA	NA
voip_control_port	VoIP Control Port	NA	NA	NA

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 "Modifying SIP Parameters via a TFTP Server" section on page 3-8, you can also modify your SIP parameters using the configuration files.

As explained in the "Configuring SIP Parameters" section on page 2-3, there are two configuration files that you can use to define the SIP parameters; the default configuration file and the phone-specific configuration file. If used, the default configuration file must be stored in the root directory of your TFTP server. The phone-specific configuration file can be stored in the root directory of the TFTP server or a subdirectory in which phone-specific configuration files are stored.

While it is not required, Cisco recommends that you use the default configuration file to define values for SIP parameters that are common to all phones. Doing so will make controlling and maintaining your network easier. You can then define only those parameters that are specific to a phone in the phone-specific configuration file. Phone-specific parameters should be defined only in a phone-specific configuration file or should be 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), Cisco recommends that you maintain the SIP parameters that are common to all 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 Cisco.com to the root directory of your TFTP server.

- Review the guidelines and restrictions documented in the "Configuration File Guidelines" section on page 2-4.

Note See the "Setting the Date, Time, and Daylight Saving Time" section on page 3-36 section for more information on those parameters.

Step 1 Using an ASCII editor, open the SIPDefault.cnf file and define or modify values for the SIP parameters shown in Table 3-3, as necessary.

Table 3-3 Default SIP Configuration File Parameters

Parameter	Required or Optional	Description
anonymous_call_block	Optional	Whether the Anonymous Call Block feature is enabled or disabled by default on the phone. Valid values are: • 0—The Anonymous Call Blocking feature is disabled by default, but can be turned on and off via the phone's user interface. When disabled, anonymous calls are received. • 1—The Anonymous Call Blocking feature is enabled by default, but can be turned on and off via the phone's user interface. When enabled, anonymous calls are rejected • 2—The Anonymous Call Blocking feature is disabled permanently and cannot be turned on and off locally via the phone's user interface. If specifying this value, specify this parameter in the phone-specific configuration file. • 3—The Anonymous Call Blocking feature is enabled permanently and cannot be turned on and off locally via the phone's user interface. If specifying this value, specify this parameter in the phone-specific configuration file. The default value is 0.
autocomplete	Optional	Whether to have numbers automatically completed when dialing. Valid values are 0 (disable auto completion) or 1 (enable auto completion). The default is 1.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
call_waiting	Optional	Whether the call waiting feature is enabled or disabled by default on the phone. Valid values are: • 0—The call waiting feature is disabled by default, but can be turned on and off via the phone's user interface. When disabled, call waiting calls are not received. • 1—The call waiting feature is enabled by default, but can be turned on and off via the phone's user interface. When enabled, call waiting calls are accepted. • 2—The call waiting feature is disabled permanently and cannot be turned on and off locally via the phone's user interface. If specifying this value, specify this parameter in the phone-specific configuration file. • 3—The call waiting feature is enabled permanently and cannot be turned on and off locally via the phone's user interface. If specifying this value, specify this parameter in the phone-specific configuration file. The default value is 1.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
callerid_blocking	Optional	Whether the Caller ID Blocking feature is enabled or disabled by default on the phone. When enabled, the phone blocks its number or e-mail address from phones that have caller identification capabilities. Valid values are: • 0—The Caller ID Blocking feature is disabled by default, but can be turned on and off via the phone's user interface. When disabled, the caller identification is included in the Request-URI header field. • 1—The Caller ID Blocking feature is enabled by default, but can be turned on and off via the phone's user interface. When enabled, "Anonymous" is included in place of the user identification in the Request-URI header field. • 2—The Caller ID Blocking feature is disabled permanently and cannot be turned on and off locally via the phone's user interface. If specifying this value, specify this parameter in the phone-specific configuration file. • 3—The Caller ID Blocking feature is enabled permanently and cannot be turned on and off locally via the phone's user interface. If specifying this value, specify this parameter in the phone-specific configuration file. The default value is 0.
cnf_join_enable	Optional	Specifies when the conference bridge hangs up whether or not it should attempt to join the two leaf nodes. Valid values are: • 0—Do not join two leaf nodes. • 1—Join two leaf nodes. The default value is 1, or join two leaf nodes.
date_format	Optional	The format to use for dates. Valid values are: • M/D/Y—Month/day/year • D/M/Y—Day/ month/year • Y/M/D—Year/month/day • Y/D/M—Year/day/month • Y-M-D—Year-month-day • YY-M-D—4-digit year-month-day The default is M/D/Y.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
directory_url	Optional	URL of the external directory server . This URL is accessed when the Directory key is pressed and the External Directory option is selected. For example, use directory_url: "http://10.10.10.10/CiscoServices/Directory.asp".
dnd_control	Optional	Whether the Do Not Disturb feature is enabled or disabled by default on the phone or whether the feature is permanently enabled. When the feature is permanently enabled, a phone is a "call out" phone only. When the Do Not Disturb feature is turned on, the phone blocks all calls placed to the phone and logs those calls in the Missed Calls directory. Valid values are: • 0—The Do Not Disturb feature is off by default, but can be turned on and off locally via the phone's user interface. • 1—The Do Not Disturb feature is on by default, but can be turned on and off locally via the phone's user interface. • 2—The Do Not Disturb feature is off permanently and cannot be turned on and off locally via the phone's user interface. If specifying this value, specify this parameter in the phone-specific configuration file. • 3—The Do Not Disturb feature is on permanently and cannot be turned on and off locally via the phone's user interface. This setting sets the phone to be a "call out" phone only. If specifying this value, specify this parameter in the phone-specific configuration file. The default value is 0.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
dst_auto_adjust dst_offset dst_start_day dst_start_day_of_week dst_start_month dst_start_time dst_start_week_of_month dst_stop_day dst_stop_day_of_week dst_stop_month dst_stop_time dst_stop_week_of_month	Optional	See the "Setting the Date, Time, and Daylight Saving Time" section on page 3-36 section for more information.
dtmf_avt_payload	Optional	Payload type for Audio/Video Transport (AVT) packets. Possible range is 96 to 127. If the value specified exceeds 127, the phone defaults to 101.
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_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_outofband	Optional	Whether to 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.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
dyn_dns_addr_1	Optional	You can specify the IP address of a new dynamic DNS server. If a new DNS server is specified, it is used for any further DNS requests after the phone uses the initial DNS address upon bootup. The DNS addresses are used in the following order: 1. dyn_dns_addr_1 (If present) 2. dyn_dns_add_2 (If present) 3. DNS Server 1 4. DNS Server 2 5. DNS Server 3 6. DNS Server 4 7. DNS Server 5 The dynamic DNS address is not stored in Flash memory. Only dotted IP addresses are accepted. This value can be cleared by removing it from the config file or changing its value to a null value "" "" or "UNPROVISIONED".
dyn_dns_addr_2	Optional	You can specify a second dynamic DNS server to be used for DNS requests.
dyn_tftp_addr	Optional	You can specify the IP address of a new dynamic TFTP server. After initially querying the default TFTP server, the phone will re-request the default and MAC-specific configuration files from the new TFTP server. The dynamic TFTP server is not stored in Flash memory. The number of dyn_tftp_addr values supported by the phone is limited to prevent the phone from bouncing between two TFTP servers. Only dotted IP addresses are accepted. This value can be cleared by removing it from the config file or changing its value to a null value "" "" or "UNPROVISIONED".
enable_vad	Optional	Use 0 to disable VAD and 1 to enable VAD. Default is 0.
end_media_port	Optional	The end Real-Time Transport Protocol (RTP) range for media. Default is 32,766. Valid values are 16,384 to 32,766.
http_proxy_addr	Optional	The IP address of the HTTP proxy server. You can use either a dotted IP address or a DNS name (a record only).
http_proxy_port	Optional	The port number of the HTTP proxy port. The default is 80.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
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 file name, because the version is also built into the file header. Trying to change the image version by changing the file name causes the firmware to fail when it compares the version in the header against the file name.
logo_url	Optional	Location of the company logo file. This logo appears on the phone display. The background space allocated for the image is 90 x 56 pixels. Images that are larger than this will automatically be scaled down to 90 x 56 pixels. The recommended file size for the image is 5 to 15k. For example, use logo_url: "http://10.10.10.10/companylogo.bmp". **Note** This parameter supports Windows 256 color bitmap format only. CMXML PhoneImage objects are not supported for this parameter. Using anything other than a Windows bitmap (.bmp) file can cause unpredictable results.
messages_uri	Optional	Number to call to check voice mail. This number is called when the **Messages** key is pressed.
nat_address	Optional	The WAN IP address of the Network Address Translation (NAT) or firewall server. You can use either a dotted IP address or a DNS name (A record only).

nat_enable	Optional	Use 0 to disable NAT and 1 to enable NAT. Default is 0. When NAT is enabled, the Contact header appears like this:
		`Contact: sip:lineN_name@nat_address:voip_control_port`
		If nat_address is invalid or UNPROVISIONED, then the Contact header appears like this:
		`Contact: sip:lineN_name@phone_ip_address:voip_control_port`
		and the Via header appears like this:
		`Via: SIP/2.0/UDP phone_ip_address:voip_control_port`
		If NAT is enabled, the Session Description Protocol (SDP) message uses the nat_address and an RTP port between the start_media_port and the end_media_port range in the C and M fields. All RTP traffic is sourced from the port advertised in the SDP.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
nat_received_processing	Optional	Use 0 to disable NAT received processing and 1 to enable NAT received processing. Default is 0. If nat_received_processing is enabled, and received= tag is in the Via header of the 200 OK response from a REGISTER, the IP address in the received= tag is used instead of the nat_address in the Contact header. If this switch occurs, the phone unregisters the old IP address and reregisters with the new IP address.
network_media_type	Optional	Ethernet port negotiation mode. Valid values are: • Auto—Port is autonegotiated. • Full100—Port is configured to be a full-duplex, 100-MB connection. • Half100—Port is configured to be a half-duplex, 100-MB connection. • Full10—Port is configured to be a full-duplex, 10-MB connection. • Half10—Port is configured to be a half-duplex, 10-MB connection. The default is Auto.
network_port2_type	Optional	The device type that is connected to port 2 of the phone. Valid values are: • Hub/Switch (default) • PC **Note** If the value is PC, port 2 can be connected only to a PC. If you are not sure about the connection, use the default value. Using a value of "PC" and connecting port 2 to a switch results in spanning tree loops and network confusion.
outbound_proxy	Optional	The IP address of the outbound proxy server. You can use either a dotted IP address or a DNS name.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
outbound_proxy_port	Optional	The port number of the outbound proxy server. The default is 5060. When outbound proxy is enabled, all SIP requests are sent to the outbound proxy server instead of the proxyN_address. All responses continue to follow the using the normal Via processing rules. The media stream is not routed through the outbound proxy. NAT and outbound proxy modes can be independently enabled or disabled. The received= tag is added to the Via header of all responses if there is no received= tag in the uppermost Via header and if the source IP address is different from the IP address in the uppermost Via header. Responses are sent back to the source under the following conditions: • If a received= tag is in the uppermost Via header, the response is sent back to the IP address contained in the received= tag. • If there is no received= tag and the IP address in the uppermost Via header is different than the source IP address, the response is sent back to the source IP. Otherwise, the response is sent back to the IP address in the uppermost Via header.
phone_password	Optional	Password to be used for console or Telnet access. The default password is "cisco."
phone_prompt	Optional	Prompt to be displayed when using Telnet or console access. The default phone prompt is "SIP Phone."
preferred_codec	Optional	Codec to use when initiating a call. Valid values are g711alaw, g711ulaw, and g729a. The default is g711ulaw.
proxy_backup	Optional	IP address of the backup proxy server or gateway. Enter this address in IP dotted-decimal notation.
proxy_backup_port	Optional	Port number of the backup proxy server. Default is 5060.
proxy_emergency	Optional	IP address of the emergency proxy server or gateway. Enter this address in IP dotted-decimal notation.
proxy_emergency_port	Optional	Port number of the emergency proxy server. Default is 5060.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
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 0. After a phone has initialized and registered with a proxy server, changing the value of this parameter to 0 unregister s the phone from the proxy server. To reinitiate a registration, change the value of this parameter back to 1. **Note** If you enable registration, and authentication is required, you must specify values for the line*x*_authname and line*x*_password parameters (where *x* is a number 1 through 6) 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 3-23.
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 listens for messages. The default is 5060. **Note** For additional phone lines, proxyN_address and proxyN_port parameters can be used to assign different proxy addresses to different phone lines. "N" in the parameters represents a phone line. The value of "N" can be from 2 to 6. If the value of "N" is not specified in the proxyN_address parameter, the phone uses the proxy1_address parameter as the default.
proxyN_address	Optional	IP address or DNS name of SIP proxy server that will be used by phone lines other than line 1. For IP address, use the IP dotted-decimal notation. If the proxyN_address parameter is provisioned with an FQDN, the phone sends REGISTER and INVITE messages by using the FQDN in the Req-URI, To, and From fields. If you want to use a dotted IP address, the proxyN_address parameters should be configured as dotted IP addresses.
proxyN_port	Optional	Port number of the SIP proxy server that will be used by phone lines other than line 1.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
remote_party_id	Optional	The Remote-Party-ID header supports network verification and screening of a call participant's identity (for example, name and number) as well as provides privacy for call participants. Valid values are: • 0 — Remote-Party-ID is disabled. The phone does not send or accept the Remote Party ID. • 1 — Remote-Party-ID is enabled. The phone sends the Remote Party ID, and can accept the Remote Party ID. The default value is 0.
semi_attended_transfer	Optional	Defines whether the caller can transfer the second leg of an attended transfer while the call is ringing. Valid values are: • 0 — Semi-attended transfer is disabled. • 1 — Semi-attended transfer is enabled. The default value is 1.
services_url	Optional	URL of the services BTXML files. This URL is accessed when the **Services** button is pressed. For example, use services_url:"http://10.10.10.10/CiscoServices/Services.asp"
sip_invite_retx	Optional	Maximum number of times an INVITE request will be retransmitted. The valid value is any positive integer. The default is 6.
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 10.
sntp_mode	Optional	See the "Setting the Date, Time, and Daylight Saving Time" section on page 3-36 section for more information.
sntp_server		
start_media_port	Optional	The start RTP range for media. Default is 16,384. Valid values are 16,384 to 32,766.
sync	Optional	Value against which to compare the value in the syncinfo.xml file before performing a remote reboot. Valid value is a character string up to 32 characters long.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
telnet_level	Optional	Enables Telnet for the phone. Valid values are: • 0 — Telnet is disabled • 1 — Telnet is enabled, no privileged commands • 2 — Telnet is enabled and privileged commands can be executed The default value is 0.
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.
time_format_24hr	Optional	Whether a 12- or 24-hour time format is displayed by default on the phones' user interface. Valid values are: • 0—The 12-hour format is displayed by default but can be changed to a 24-hour format via the phone's user interface. • 1—The 24-hour format is displayed by default but can be changed to a 12-hour format via the phone's user interface. • 2-The 12-hour format is displayed and cannot be changed to a 24-hour format via the phone's user interface. • 3—The 24-hour format is displayed and cannot be changed to a 12-hour format via the phone's user interface. The default value is 1.
time_zone	Optional	See the "Setting the Date, Time, and Daylight Saving Time" section on page 3-36 section for more information.
timer_invite_expires	Optional	The amount of time, in seconds, after which a SIP INVITE expires. This value is used in the Expire header field. The valid value is any positive number; however, Cisco recommends 180 seconds. The default is 180.
timer_register_expires	Optional	The amount of time, in seconds, after which a REGISTRATION request expires. This value is inserted into the Expire header field. The valid value is any positive number; however, Cisco recommends 3600 seconds. The default is 3600.
timer_t1	Optional	Lowest value (in milliseconds) of the retransmission timer for SIP messages. The valid value is any positive integer. The default is 500.

Table 3-3 Default SIP Configuration File Parameters (continued)

Parameter	Required or Optional	Description
timer_t2	Optional	Highest value (in milliseconds) of the retransmission timer for SIP messages. The valid value is any positive integer greater than timer_t1. The default is 4000.
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.
user_info	Optional	Configures the "user=" parameter in the REGISTER message. Valid values are: • none—No value is inserted. • phone—The value user=phone is inserted in the To, From, and Contact Headers for REGISTER. • ip—The value user=ip is inserted in the To, From, and Contact Headers for REGISTER. The default value is none.
voip_control_port	Optional	The UDP port used for SIP messages. Default is 5060. All SIP REQUESTS use voip_control_port as the UDP source port when nat_enable = 1. Valid values are 1025 to 65,535.

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

The following is a sample SIP default configuration file:

```
; sip default configuration file
# Image Version
image_version: "P0S3-xx-y-zz"

# Proxy Server
proxy1_address: "proxy.company.com"
proxy2_address: ""
proxy3_address: ""
proxy4_address: ""
proxy5_address: ""
proxy6_address: ""

# Proxy Server Port (default - 5060)
proxy1_port:"5060"
proxy2_port:""
```

```
proxy3_port:""
proxy4_port:""
proxy5_port:""
proxy6_port:""

# Emergency Proxy info
proxy_emergency: "1.2.3.4"
proxy_emergency_port: "5060"

# Backup Proxy info
proxy_backup: "1.2.3.4
proxy_backup_port: "5060"

# Proxy Registration (0-disable (default), 1-enable)
proxy_register: "1"

# Phone Registration Expiration [1-3932100 sec] (Default - 3600)
timer_register_expires: "180"

# Codec for media stream (g711ulaw (default), g711alaw, g729)
preferred_codec: "g711ulaw"

# TOS bits in media stream [0-5] (Default - 5)
tos_media: "5"

# Inband DTMF Settings (0-disable, 1-enable (default))
dtmf_inband: "1"

# Out of band DTMF Settings (none-disable, avt-avt enable (default), avt_always - always
avt )
dtmf_outofband: "avt"

# DTMF dB Level Settings (1-6dB down, 2-3db down, 3-nominal (default), 4-3db up, 5-6dB up)
dtmf_db_level: "3"

# SIP Timers
timer_t1: "500"                    ; Default 500 msec
timer_t2: "4000"                   ; Default 4 sec
sip_retx: "10"                      ; Default 11
sip_invite_retx: "6"                ; Default 7
timer_invite_expires: "180"         ; Default 180 sec

# Setting for Message speeddial to Voicemail
messages_uri: "9195551000"

#*********  Release 2 new config parameters **********

# TFTP Phone Specific Configuration File Directory
tftp_cfg_dir: "./"

# Time Server
sntp_mode: "directedbroadcast"
sntp_server: "172.16.10.150"
#sntp_server: "sntp.company.com"
time_zone: "EST"
dst_offset: "1"
dst_start_month: "April"
dst_start_day: ""
dst_start_day_of_week: "Sun"
dst_start_week_of_month: "1"
dst_start_time: "02"
dst_stop_month: "Oct"
dst_stop_day: ""
dst_stop_day_of_week: "Sunday"
```

```
dst_stop_week_of_month: "8"
dst_stop_time: "2"
dst_auto_adjust: "1"

# Do Not Disturb Control (0-off, 1-on, 2-off with no user control, 3-on with no user
control)
dnd_control: "0"                  ; Default 0 (Do Not Disturb feature is off)

# Caller ID Blocking (0-disabled, 1-enabled, 2-disabled no user control, 3-enabled no user
control)
callerid_blocking: "0"            ; Default 0 (Disable sending all calls as anonymous)

# Anonymous Call Blocking (0-disabled, 1-enabled, 2-disabled no user control, 3-enabled no
user control)
anonymous_call_block: "0"         ; Default 0 (Disable blocking of anonymous calls)

# DTMF AVT Payload (Dynamic payload range for AVT tones - 96-127)
dtmf_avt_payload: "101"           ; Default 101

# XML file that specifies the dialplan desired
dial_template: "dialplan"

# Network Media Type (auto, full100, full10, half100, half10)
network_media_type: "auto"

#Autocompletion During Dial (0-off, 1-on [default])
autocomplete: "1"

#Time Format (0-12hr, 1-24hr [default])
time_format_24hr: "1"

#Enable or Disbale VAD (0-disabled (default), 1-enabled)
enable_vad: 0

telnet_level: 0
phone_password: "cisco"

services_url: "http://www.company.com/phone/services.asp"
directory_url: "http://www.company.com/phone/companydirectory.asp"
logo_url: "http://www.company.com/phone/logo.bmp"
```

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 2-4.
- 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.

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 SIP parameters shown in Table 3-4.

Note For all variables, *x* is a number 1 through 6.

Table 3-4 Phone-Specific Configuration Parameters

Parameter	Required or Optional	Description
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	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 length exceeds the allowable space in 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 3444 display on the LCD instead. Alternatively, if the value for the linex_name parameter is the e-mail address "username@company.com", you can specify the "username" to have just the user name appear on the LCD instead. This parameter is used for display only. If a value is not specified for this parameter, the value in the line*x*_name variable is displayed.
line*x*_authname	Required for line 1 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 for a line when registration is enabled, the value defined for line 1 is used. If a value is not defined for line 1, the default line1_authname is UNPROVISIONED.
line*x*_password	Required for line 1 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 for a line when registration is enabled, the value defined for line 1 is used. If a value is not defined for line 1, the default line1_password is UNPROVISIONED.
line*x*_displayname	Optional	Identification as it should appear for caller identification purposes. For example, instead of jdoe@company.com appearing on phones that have caller ID, you can specify John Doe in this parameter to have John Doe appear on the callee end instead. If a value is not specified for this parameter, nothing is used.

Table 3-4 Phone-Specific Configuration Parameters (continued)

Parameter	Required or Optional	Description
dnd_control	Optional	Whether the Do Not Disturb feature is enabled or disabled by default on the phone or whether the feature is permanently enabled, making the phone a "call out" phone only. When the Do Not Disturb feature is turned on, the phone blocks all calls placed to the phone and logs those calls in the Missed Calls directory. Valid values are: • 0—The Do Not Disturb feature is off by default, but can be turned on and off locally via the phone's user interface. • 1—The Do Not Disturb feature is on by default, but can be turned on and off locally via the phone's user interface. • 2—The Do Not Disturb feature is off permanently and cannot be turned on and off locally via the phone's user interface. If specifying this value, specify this parameter in the phone-specific configuration file. • 3—The Do Not Disturb feature is on permanently and cannot be turned on and off locally via the phone's user interface. This setting sets the phone to be a "call out" phone only. If specifying this value, specify this parameter in the phone-specific configuration file. **Note** This parameter is best configured in the SIPDefault.dnf file unless configuring a phone to be a "call-out" phone only. When configuring a phone to be a "call-out" phone, define this parameter in the phone-specific configuration file.
phone_label	Optional	Label to display on the top status line of the LCD. This field is for end-user display only. For example, a phone's label can display "John Doe's phone." Up to 11 characters can be used when specifying the phone label. Save the file to your TFTP server (in the root directory or a subdirectory containing 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 lower case (for example, SIP00503EFFD842.cnf).

The following is a sample configuration file:

```
; phone-specific configuration file sample
line1_displayname:  "jdoe43"
line1_name:         "43"
line2_displayname:  "jdoe44"
line2_name:         "44"
line3_displayname:  "pgatour"
line3_name:         "duval"
line4_displayname:  "jdoe46"
line4_name:         "46"
line5_displayname:  "jdoe47"
line5_name:         "47"
line6_displayname:  "jdoe48"
line6_name:         "48"
phone_label:        "jdoe4X"
phone_prompt: "John-43"
```

```
proxy1_address: 1.2.3.4
proxy2_address: 1.2.3.4
proxy3_address: 1.2.3.4
proxy4_address: 1.2.3.4
proxy5_address: 1.2.3.4
proxy6_address: 1.2.3.4
proxy1_port: 5060
proxy2_port: 5060
proxy3_port: 5060
proxy4_port: 5060
proxy5_port: 5060
proxy6_port: 5060

callerid_blocking: 0
dtmf_outofband: avt
network_media_type: auto
tos_media: 5
dtmf_avt_payload: 101
time_zone: EST
call_waiting: 1
cnf_join_enable : 1
semi_attended_transfer : 1
```

Modifying the SIP Parameters Directly on Your Phone

If you did not configure the SIP parameters via a TFTP server, you can configure them directly on your phone 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 2-15.
- 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.
- 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.

Step 1 Press the **settings** key. The Settings menu appears.

Step 2 Highlight **SIP Configuration**. The SIP Configuration menu appears.

Step 3 Highlight **Line 1 Settings**.

Step 4 Press the **Select** soft key. The Line 1 Configuration menu appears.

Step 5 Highlight and press the **Select** soft key to configure the parameters shown in Table 3-5, as necessary:

Table 3-5 SIP Configuration Parameters

Parameter	Required or Optional	
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	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 3444 display on the LCD instead. Alternatively, if the value for the linex_name parameter is the e-mail address "username@company.com", you can specify the "username" to have just the user name appear on the LCD instead. This parameter is used for display 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.
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.
Display Name	Optional	Identification as it should appear for caller identification. For example, instead of jdoe@company.com appearing on phones that have caller ID, you can specify John Doe in this parameter to have John Doe appear on the callee end instead. If a value is not specified for this parameter, the Name value is 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 that the SIP client will use. The default is 5060.

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, and then continue with Step 8.

Step 8 In addition to the line settings, you can highlight and press **Select** to configure the parameters on the SIP Configuration menu shown in Table 3-6:

Table 3-6 Additional SIP Configuration Parameters

Parameter	Required or Optional	
Messages URI	Optional	Number to call to check voice mail. This number is 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. After a phone has initialized and registered with a proxy server, changing the value of this parameter to No unregisters the phone from the proxy server. To reinitiate a registration, change the value of this parameter back to Yes. **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 expires. This value is used the Expire header field. The valid value is any positive number; however, Cisco recommends 3600 seconds. The default is 3600.
TFTP Directory	Required if phone-specific configuration files are located in a subdirectory	Path to the TFTP subdirectory in which phone-specific configuration files are stored.
Phone Label	Optional	Label to display on the top status line of the LCD. This field is for end-user display only. For example, a phone's label can display "John Doe's phone." Up to 11 characters can be used when specifying the phone label.
Enable VAD	Optional	Specifies whether VAD is enabled or disabled.
VoIP Control Port	Optional	The UDP port used for SIP messages. Default is 5060. All SIP REQUESTS use voip_control_port as the UDP source port when nat_enable = 1. Valid values are 1025 to 65535.

Table 3-6 Additional SIP Configuration Parameters (continued)

Parameter	Required or Optional	
Start Media Port	Optional	The start RTP range for media. Default is 16,384. Valid values are 16,384 to 32,766.
End Media Port	Optional	The end RTP range for media. Default is 32,766. Valid values are 16,384 to 32,766.
Backup Proxy	Optional	IP address of the backup proxy server or gateway. Enter this address in IP dotted-decimal notation.
Backup Proxy Port	Optional	Port number of the backup proxy server. Default is 5060.
Emergency Proxy	Optional	IP address of the emergency proxy server or gateway. Enter this address in IP dotted-decimal notation.
Emergency Proxy Port	Optional	Port number of the emergency proxy. Default is 5060.
Outbound Proxy	Optional	The IP address of the outbound proxy server. You can use either a dotted IP address or a DNS name (A record only).
Outbound Proxy Port	Optional	The port number of the outbound proxy server. The default is 5060.
NAT Enabled	Optional	Choose No to disable NAT and Yes to enable NAT.
NAT Address	Optional	The WAN IP address of the NAT or firewall server. You can use either a dotted IP address or a DNS name (A record only).

Step 9 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.

Using the Command-Line Interface

You can use Telnet or a console to connect to your Cisco SIP IP phone to debug or troubleshoot the phone. Table 3-7 shows the available CLI commands:

Table 3-7 CLI Commands

Command	Purpose
`SIP Phone> clear {arp | malloc | tcp-stats}`	Clears the following, depending on keywords used: • **arp:** Clears the Address Resolution Protocol (ARP) cache. • **malloc:** Clears the memory allocation table. • **tcp-stats:** Clears the TCP statistics.
`SIP Phone> debug {arp | console-stall | strlib | malloc | malloc-table | sk-platform | flash | dsp | vcm | dtmf | task-socket | lsm | fsm | auth | fim | gsm | cc | cc-msg | error | sip-task | sip-state | sip-messages | sip-reg-state | dns | config | sntp | sntp-packet | http | arp-broadcast | xml-events | xml-deck | xml-vars | xml-post}`	Shows detailed debug output when used with the following keywords: • **arp:** Shows debug output for the ARP cache. • **console-stall**: Shows debug output for the console-stall driver output mode. • **strlib**: Shows debug output for the string library. • **malloc**: Shows debug output for memory allocation. • **malloc-table**: Enables the population of the memory allocation table. The table can be viewed with the **show malloc-table** command. • **sk-platform**: Shows debug output for the platform. • **flash**: Shows debug output for the Flash memory. • **dsp**: Shows debug output for DSP accesses. • **vcm**: Shows debug output for the voice channel manager (VCM), including tones, ringing, and volume. • **dtmf**: Shows debug output for DTMF relay. • **task-socket**: Shows socket task debug output. • **lsm:** Shows debug output for the Line State Manager. • **fsm**: Shows debug output for the Feature State Manager. • **auth**: Shows debug output for the SIP authorization state machine. • **fim**: Shows debug output for the Feature Interaction Manager. • **gsm**: Shows debug output for the Global State Manager. • **cc**: Shows debug output for call control. • **cc-msg**: Shows debug output for the call control messages.

Table 3-7 CLI Commands (continued)

Command	Purpose
debug command keywords (continued)	• **error**: Shows general error debug output. • **sip-task**: Shows debug output for the SIP task. • **sip-state**: Shows debug output for the SIP state machine. • **sip-messages**: Shows debug output for SIP messaging. • **sip-reg-state**: Shows debug output for the SIP registration state machine. • **dns:** Shows the DNS command-line interface (CLI) configuration; allows you to clear the cache and set servers). • **config**: Shows output for the **config system** command. • **sntp**: Shows debug output for Simple Network Time Protocol (SNTP). • **sntp-packet**: Displays full SNTP packet data. • **arp-broadcast**: Shows ARP broadcast messages. • **http**: Shows HTTP requests and responses. • **xml-events**: Shows XML events that are posted to the XML application chain. • **xml-deck**: Shows XML requests for XML cards and decks. • **xml-vars**: Shows XML content variables. • **xml-post**:Shows XML post strings. **Note** Do not use the **debug all** command, because it can cause the phone to become inoperable. This command is for use only by Cisco TAC personnel.
`SIP Phone>` **`dns`**	Manipulates the DNS system. The following arguments are used: • **-p**: Prints out the DNS cache table. • **-c**: Clears out the DNS cache table. • **-s** *ipaddress*: Sets the primary DNS server. • **-b** *ip address*: Sets the first backup server.
`SIP Phone>` **`erase protflash`**	Erases the protocol area of Flash memory. Forces the phone to reset its IP stack and request its configuration files again. This command can only be used if the telnet_level parameter is set to allow privileged commands to be executed.
`SIP Phone>` **`exit`**	Exits the Telnet or console session.

Table 3-7 CLI Commands (continued)

Command	Purpose
`SIP Phone>` **`ping`** *`ipaddress number packetsize timeout`*	Sends an Internet Control Message Protocol (ICMP) ping to a network address. You can use a dotted IP address or an alphanumeric address. The *number* value specifies how many pings to send; the default value is 5. The *packetsize* argument defines the size of the packet; you can send any size packet up to 1480 bytes and the default packet size is 100. The *timeout* value is measured in seconds and identifies how long to wait before the request times out; the default is 2.
`SIP Phone>` **`register`** `{`**`option`** `|` **`line`**`}`	Instructs the Cisco SIP IP phone to register with the proxy server. Option values are 0 and 1; 0 is unregister and 1 is register. These values are set for each line.
`SIP Phone>` **`reset`**	Resets the phone line. This command can be used only if the telnet_level parameter is set to allow privileged commands to be executed.

Table 3-7 CLI Commands (continued)

Command	Purpose
SIP Phone> **show** {**arp** | **debug** | **strpool** | **memorymap** | **dump** | **malloctable** | **stacks** | **status** | **abort_vector** | **flash** | **dspstate** | **rtp** | **tcp** | **lsm** | **fsm** | **fsmdef** | **fsmcnf** | **fsmxfr** | **fim** | **gsm** | **register** | **network** | **config** | **personaldir** | **dialplan** | **timers**}	Shows information about the SIP IP phone. The following keywords are used: • **arp**: Displays contents of the ARP cache. • **debug**: Shows which debug modes are activated. • **strpool**: Shows the string library pool of strings. This command can only be used if the telnet_level parameter is set to allow privileged commands to be executed. • **memorymap**: Shows memory mapping table, including free, used, and wasted blocks. • **dump**: Displays a dump of the memory contents. This command can only be used if the telnet_level parameter is set to allow privileged commands to be executed. • **malloctable**: Shows the memory allocation table. • **stacks**: Shows tasks and buffer lists. • **status**: Shows the current phone status, including errors. • **abort_vector:** Shows the address of the last recorded abort vector. • **flash**: Shows flash memory information. • **dspstate**: Shows the DSP status, including whether the DSP is ready, the audio mode, if keepalive pending is turned on, and the ringer state. • **rtp**: Shows packet statistics for the RTP streams. • **tcp**: Shows the status of TCP ports, including the state (listen or closed) and the port number. • **lsm**: Shows the current status of the Line Manager control blocks. • **fsm**: Shows the current status of the Feature State function control blocks. • **fsmdef**: Shows the current status of the default Feature State Manager data control blocks. • **fsmcnf**: Shows the current status of the Conference Feature State Manager call control blocks. • **fsmxfr**: Shows the current status of the Transfer Feature State Manager transfer control blocks. • **fim**: Shows the current status of the Feature Interaction Manager control blocks (interface control blocks and state control blocks). • **gsm**: Turns on debugging for vcm, lsm, fim, fsm, and gsm.

Table 3-7 CLI Commands (continued)

Command	Purpose
show command keywords (continued)	• **register**: Shows the current registration status of SIP lines. • **network**: Shows network information, such as phone platform, DHCP server, phone IP address and subnet mask, default GW, address of the TFTP server, phone MAC address, domain name, and phone name. • **config**: Shows the current Flash configuration, including network information, phone label and password, SNTP server address, DST information, time and date format, and input and output port numbers. • **personaldir**: Displays the current contents of the personal directory. This command can be used only if the telnet_level parameter is set to allow privileged commands to be executed. • **dialplan**: Shows the phone's dial plan. • **timers**: Shows the current status of the platform timers

Table 3-7 CLI Commands (continued)

Command	Purpose
`SIP Phone>` **`test`** `{`**`open`** | **`close`** | **`key`** | **`onhook`** | **`offhook`** | **`show`** | **`hide`**`}`	Accesses the remote call test interface, allowing you to control the phone from a remote site. To use this feature, enter the **test open** command. To prevent use of this feature, enter the **test close** command. This command can only be used only if the telnet_level parameter is set to allow privileged commands to be executed. The following commands are available: • **test key**: When a test session is open, you can simulate key presses using the **test key** *k1 k2 k3...k13* command, where k1 through k13 represent the following key names: – voldn—Volume down – volup—Volume up – headset—Headset – spkr—Speaker – mute—Mute – info—Info – msgs—Messages – serv—Services – dir—Directories – set—Settings – navup—Navigate up – navdn—Navigate down The keys 0 through 9, #, and * may be entered in continuous strings to better express typical dialing strings. A typical command would be **test ky 23234**. • **test onhook**: Simulates a handset onhook event. • **test offhook**: Simulates a handset offhook event. • **test show**: Shows test feedback. • **test hide**: Hides test feedback.
`SIP Phone>` **`tty`** `{`**`echo`** `{`**`on`** | **`off`**`}` | **`mon`** | **`timeout`** *`value`* | **`kill`** *`session`* | **`msg`**`}`	Controls the Telnet system. The **echo** keyword controls local echo. The **mon** keyword sends all debug output to both the console and Telnet sessions. The **timeout** *value* keyword sets the Telnet session timeout period based on the value. The *value* range is 0 through 65,535. The **kill** *session* keyword tears down the Telnet session specified by the *session* argument. The **msg** keyword allows you to send a message to another terminal logged into the phone; for example, you can send a message telling everyone else that is logged in to log off.

Table 3-7 CLI Commands (continued)

Command	Purpose
SIP Phone> **traceroute** *ip-address* [*ttl*]	Initiates a traceroute session from the console or from a Telnet session. Traceroute shows the route that IP datagrams follow from the SIP IP phone to the specified IP address. Use the following two arguments: • *ip-address*: The dotted IP address or alphanumeric address (host name) of the host to which you are sending the traceroute. • *ttl*: The time-to-live value, or the number of routers (hops) through which the datagram can pass. The default value is 30.
SIP Phone> **undebug** {**arp** | **console-stall** | **strlib** | **malloc** | **malloc-table** | **sk-platform** | **flash** | **vcm** | **dtmf** | **task-socket** | **lsm** | **fsm** | **auth** | **fim** | gsm | **cc** | **cc-msg** | **softkeys** | **error** | **sip-task** | **sip-state** | **sip-messages** | **sip-reg-state** | **dns** | **config** | **sntp** | **sntp-packet**}	Turns off debugging.

Setting the Date, Time, and Daylight Saving Time

The current date and time is supported on the Cisco SIP IP phone via SNTP and is displayed on the phone's LCD. In addition to supporting the current date and time, Daylight Saving Time (DST) and time zone settings are also supported. DST can be configured to be obtained via an absolute (for example, starts on April 1 and ends on October 1) or relative (for example, starts the first Sunday in April and ends on the last day of October) configuration.

The format for the date can be set using the date_format parameter.

International time zone abbreviations are supported and are case sensitive (must be in all capital letters).

Cisco recommends that date and time-related parameters be defined in the SIPDefault.cnf file. The time zone parameter can be set manually on the phone or in the configuration file.

Before You Begin

When configuring the date, time, time zone, and DST settings, remember the following:

- Review the guidelines and restrictions documented in the "Configuration File Guidelines" section on page 2-4.
- Determine whether you want to configure absolute DST or relative DST.
- The SNTP parameters specify how the phone will obtain the current time from an SNTP server. Review the guidelines in Table 3-8 and Table 3-9 before configuring the SNTP parameters.

  Table 3-8 lists the actions that take place when a null value (0.0.0.0) is specified in the sntp_server parameter.

Table 3-8 Actions Based on sntp_mode When the sntp_server Parameter Is Set to a Null Value

sntp_server =0.0.0.0	sntp_mode= unicast	sntp_mode= multicast	sntp_mode= anycast	sntp mode= directedbroadcast
Sends	Nothing. No known server with which to communicate.	Nothing. When in multicast mode, SNTP requests are not sent.	SNTP packet to the local network broadcast address. After the first SNTP response is received, the phone switches to unicast mode with the server being set as the one who first responded.	SNTP packet to the local network broadcast address. After the first SNTP response is received, the phone switches to multicast mode.
Receives	Nothing. No known server with which to communicate.	SNTP data via the SNTP/NTP multicast address from the local network broadcast address from any server on the network.	Unicast SNTP data from the SNTP server that first responded to the network broadcast request.	SNTP data from the SNTP/NTP multicast address and the local network broadcast address from any server on the network.

Table 3-9 lists the actions that take place when a valid IP address is specified in the sntp_server parameter.

Table 3-9 Actions Based on sntp_mode When the sntp_server Parameter Is Set to an IP Address

sntp_server = 192.168.1.9	sntp_mode= unicast	sntp_mode= multicast	sntp_mode= anycast	sntp_mode= directedbroadcast
Sends	SNTP request to the SNTP server.	Nothing. When in multicast mode, SNTP requests are not sent.	SNTP request to the SNTP server.	SNTP packet to the SNTP server. After the first SNTP response is received, the phone switches to multicast mode.
Receives	SNTP response from the SNTP server and ignores responses from other SNTP servers.	SNTP data via the SNTP/NTP multicast address from the local network broadcast address.	SNTP response from the SNTP server and ignores responses from other SNTP servers.	SNTP data from the SNTP/NTP multicast address and the local network broadcast address and ignores responses from other SNTP servers.

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

- sntp_mode—(Required) Mode in which the phone listens for the SNTP server. Valid values are unicast, multicast, anycast, or directedbroadcast.

  See Table 3-8 and Table 3-9 for an explanation on how these values work, depending on the sntp_server parameter value.
- sntp_server—(Required) IP address of the SNTP server from which the phone will obtain time data.

See Table 3-8 and Table 3-9 for an explanation on how these values work, depending on the sntp_server parameter value.

- time_zone—(Required) Time zone in which the phone is located. Valid values are the time zone abbreviations shown in Table 3-10. These abbreviations are case sensitive and must be in all capital letters.

Table 3-10 Time Zone Abbreviations

Abbreviation	GMT Offset	Cities	Time Zone Names
IDL	GMT-12:00	Eniwetok	IDL (International Date Line), IDLW (International Date Line West)
NT	GMT-11:00	Midway	BT (Bering Time), NT (Nome Time)
AHST	GMT-10:00	Hawaii	AHST (Alaska-Hawaii Standard Time), HST (Hawaiian Standard Time), CAT (Central Alaska Time)
IMT	GMT-09:30	Isle Marquises	Isle Marquises
YST	GMT-09:00	Yukon	YST (Yukon Standard Time)
PST	GMT-08:00	Los Angeles	PST (Pacific Standard Time),
MST	GMT-07:00	Phoenix	MST (Mountain Standard Time), PDT (Pacific Daylight Time)
CST	GMT-06:00	Dallas, Mexico City	CST (Central Standard Time), MDT (Mountain Daylight Time), Chicago
EST	GMT-05:00	New York	EST (Eastan Standard Time), CDT (Central Daylight Time), NYC
AST	GMT-04:00	La Paz	AST (Atlantic Standard Time), EDT (Eastan Daylight Time)
NST	GMT-03:30	Newfoundland	NST (Newfoundland Standard Time)
BST	GMT-03:00	Buenos Aires	BST (Brazil Standard Time), ADT (Atlantic Daylight Time), GST (Greenland Standard Time)
AT	GMT-02:00	Mid-Atlantic	AT (Azores Time)
WAT	GMT-01:00	Azores	WAT (West Africa Time)
GMT	GMT 00:00	London	GMT (Greenwich Mean Time), WET (Western European Time), UT (Universal Time)
CET	GMT+01:00	Paris	CET (Central European Time), MET (Middle European Time), BST (British Summer Time), MEWT(Middle European Winter Time), SWT (Swedish Winter Time), FWT (French Winter Time)
EET	GMT+02:00	Athens, Rome	EET (Eastean European Time), USSR-zone1, MEST (Middle European Summer Time), FST (French Summer Time)

Table 3-10 Time Zone Abbreviations

Abbreviation	GMT Offset	Cities	Time Zone Names
BT	GMT+03:00	Baghdad, Moscow	BT (Baghdad Time), USSR-zone2
IT	GMT+03:30	Tehran	IT (Iran Time)
ZP4	GMT+04:00	Abu Dhabi	USSR-zone3, ZP4 (GMT Plus 4 Hours)
AFG	GMT+04:30	Kabul	Afghanistan
ZP5	GMT+05:00	Islamabad	USSR-zone4, ZP5 (GMT Plus 5 Hours)
IST	GMT+05:30	Bombay, Delhi	IST (Indian Standard Time)
ZP6	GMT+06:00	Colombo	USSR-zone5, ZP6 (GMT Plus 6 Hours)
SUM	GMT+06:30	North Sumatra	NST (North Sumatra Time)
WAST	GMT+07:00	Bangkok, Hanoi	SST (South Sumatra Time), USSR-zone6, WAST (West Australian Standard Time)
HST	GMT+08:00	Beijing, Hong Kong	CCT (China Coast Time), HST (HongKong Standard Time), USSR-zone7, WADT (West Australian Daylight Time)
JST	GMT+09:00	Tokyo, Seoul	JST (Japan Standard Time/Tokyo), KST (Korean Standard Time), USSR-zone8
CAST	GMT+09:30	Darwin	SAST (South Australian Standard Time) , CAST (Central Australian Standard Time)
EAST	GMT+10:00	Brisbane, Guam	GST (Guam Standard Time),USSR-zone9, EAST (East Australian Standard Time)
EADT	GMT+11:00	Solomon Islands	USSR-zone10, EADT (East Australian Daylight Time)
NZST	GMT+12:00	Auckland	NZT (New Zealand Time/Auckland), NZST (New Zealand Standard Time), IDLE (International Date Line East)

Step 2 To configure common DST settings, specify values for the following parameters:

- dst_offset—Offset from the phone's time when DST is in effect. When DST is over, the specified offset is no longer applied to the phone's time. Valid values are hour/minute, -hour/minute, +hour/minute, hour, -hour, and +hour.
- dst_auto_adjust—Whether or not DST is automatically adjusted on the phones. Valid values are 0 (disable automatic DST adjustment) or 1 (enable automatic DST adjustment). The default is 1.
- dst_start_month—Month in which DST starts. Valid values are January, February, March, April, May, June, July, August, September, October, November, and December or 1 through 12 with January being 1 and December being 12. When specifying the name of a month, the value is not case sensitive. In the United States, the default value is April.
- dst_stop_month—Month in which DST ends. Valid values are January, February, March, April, May, June, July, August, September, October, November, and December or 1 through 12 with January being 1 and December being 12. When specifying the name of a month, the value is not case sensitive. In the United States, the default value is October.
- dst_start_time—Time of day on which DST begins. Valid values are hour/minute (02/00) or hour (02:00). In the United States, the default value is 02:00.

- dst_stop_time—Time of day on which DST ends. Valid values are hour/minute (02/00) or hour (02:00). In the United States, the default value is 02:00.

Step 3 To configure absolute DST, specify values for the following parameters or to configure relative DST, proceed to Step 4:

- dst_start_day—Day of the month on which DST begins.

  Valid values are 1 through 31 for the days of the month or 0 when specifying relative DST to specify that this field be ignored and that the value in the dst_start_day_of_week parameter be used instead.

- dst_stop_day—Day of the month on which DST ends.

  Valid values are 1 through 31 for the days of the month or 0 when specifying relative DST to specify that this field be ignored and that the value in the dst_stop_day_of_week parameter be used instead.

Step 4 To configure relative DST, specify values for the following parameters:

- dst_start_day_of_week—Day of the week on which DST begins.

  Valid values are Sunday or Sun, Monday or Mon, Tuesday or Tue, Wednesday or Wed, Thursday or Thu, Friday or Fri, Saturday or Sat, or Sunday or Sun or 1 through 7 with 1 being Sunday and 7 being Saturday. When specifying the name of the day, the value is not case sensitive. In the United States, the default value is Sunday.

- dst_start_week_of_month—Week of month in which DST begins.

  Valid values are 1 through 6 and 8, with 1 being the first week and each number thereafter being subsequent weeks and 8 specifying the last week in the month regardless of which week the last week is. In the United States, the default value is 1.

- dst_stop_day_of_week—Day of the week on which DST ends.

  Valid values are Sunday or Sun, Monday or Mon, Tuesday or Tue, Wednesday or Wed, Thursday or Thu, Friday or Fri, Saturday or Sat, or Sunday or Sun or 1 through 7, with 1 being Sunday and 7 being Saturday. When specifying the name of the day, the value is not case sensitive. In the United States, the default value is Sunday.

- dst_stop_week_of_month—Week of month in which DST ends.

  Valid values are 1 through 6 and 8, with 1 being the first week and each number thereafter being subsequent weeks and 8 specifying the last week in the month regardless of which week the last week is. In the United States, the default value is 8.

Step 5 Save the file with the same file name, SIPDefault.cnf, to the root directory of your TFTP server.

The following is a sample configuration for an absolute DST configuration:

```
; sip default configuration file
(additional configuration text omitted)

time_zone : PST
dst_offset : 01/00
dst_start_month : April
dst_start_day : 1
dst_start_time : 02/00
dst_stop_month : October
dst_stop_day : 1
dst_stop_time : 02/00
dst_stop_autoadjust : 1

(additional configuration text omitted)
```

The following is a sample configuration for a relative DST configuration:

```
; sip default configuration file
(additional configuration text omitted)

time_zone : PST
dst_offset : 01/00
dst_start_month : April
dst_start_day : 0
dst_start_day_of_week : Sunday
dst_start_week_of_month : 1
dst_start_time : 02/00
dst_stop_month : October
dst_stop_day : 0
dst_stop_day_of_week : Sunday
dst_stop_week_of_month : 8
dst_stop_time : 02/00
dst_stop_autoadjust : 1

(additional configuration text omitted)
```

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 reenables 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).

Step 1 Press the **settings** key. The Settings menu appears.

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 must edit the configuration file in which a parameter is defined to delete the parameter. When deleting a parameter, remove the variable in the file or change its value to a null value "" "" or "UNPROVISIONED". If both the variable and its value are removed, the phone uses the setting for that variable that it has stored in Flash memory.

Note If the telnet_level parameter is set to allow privileged commands to be executed, the entire SIP configuration can be erased. Use the **erase_protflash** command so that the phone can retrieve its config files.

Before You Begin

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

Step 1 Press the **settings** key. The Settings menu appears.

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 want 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. From the Status menu, 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 appears.

Step 2 Highlight **Status**.

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

Step 4 Highlight **Status Messages**.

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

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 appears.

Step 2 Highlight **Status**.

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

Step 4 Highlight **Network Statistics**.

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

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 autonegotiated a full-duplex 100-Mbps connection.
- Port 0 Half, 100—Indicates that the network is in a linked state and has autonegotiated a half-duplex 100-Mbps connection.
- Port 0 Full, 10—Indicates that the network is in a linked state and has autonegotiated a full-duplex 10-Mbps connection.
- Port 0 Half, 10—Indicates that the network is in a linked state and has autonegotiated a half-duplex 10-Mbps connection.

- Port 1 Full, 100—Indicates that the network is in a linked state and has autonegotiated a full-duplex 100-Mbps connection.
- Port 1 Half, 100—Indicates that the network is in a linked state and has autonegotiated a half-duplex 100-Mbps connection.
- Port 1 Full, 10—Indicates that the network is in a linked state and has autonegotiated a full-duplex 10-Mbps connection.
- Port 1 Half, 10—Indicates that the network is in a linked state and has autonegotiated a half-duplex 10-Mbps 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 appears.

Step 2 Highlight **Status**.

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

Step 4 Highlight **Firmware Versions**.

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

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.

Upgrading the Cisco SIP IP Phone Firmware

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

Before You Begin

- To upgrade the firmware on just one phone at a time, you upgrade the image_version in the phone-specific configuration file. To upgrade 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 Cisco.com to the root directory of your TFTP server.

See the upgrade scenarios in Table 3-11 to determine how to upgrade.

Table 3-11 Upgrade Scenarios

Image Name	Use Section
P0S30100, P0S30200, P0S30201, and P0S3Z*xxx*	Upgrading from Release 2.1 or Earlier Releases to Release 4.0, page 3-45
P003*xxxx* or P003*xxxxxxxx* (these images are loaded on the Cisco SIP IP phone when it is shipped)	Dual Booting from SCCP or MGCP to Release 4.0, page 3-46
P0M3*xx-y-zz*	Dual Booting from SCCP or MGCP to Release 4.0, page 3-46
P0S30202, P0S30203 and P0S3-03-y-xx	Upgrading from Release 2.2 or Later Releases to Release 4.0, page 3-45

Upgrading from Release 2.2 or Later Releases to Release 4.0

Step 1 Copy the new Release 4.0 binary image P0S3-*xx-y-zz*.bin, where *xx* is the release major version, *y* is the release minor version, and *zz* is the maintenance number, from Cisco.com 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 the image_version variable should match the version name (without the .bin extension) of the latest firmware that you downloaded (for example, P0S3-*xx-y-zz)*.

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 that have an updated phone-specific configuration file with the new image version and that have been restarted use the latest firmware image. All other phones use the older version until their configuration files have been updated with the new image version.

Upgrading from Release 2.1 or Earlier Releases to Release 4.0

Step 1 Copy the P0S30202.bin binary image from Cisco.com to the root directory of the TFTP server.

Step 2 If you are dual booting from a Cisco IP phone running the Skinny Client Control Protocol (SCCP) or MGCP protocol, open the OS79XX.TXT file with a text editor and change the file to include P0S30202.

Step 3 Open the phone configuration file with a text editor and edit the image_version variable to read P0S30202.

Step 4 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.

Step 5 Copy the new Release 4.0 binary image P0S-3*xx*-*y*-*zz*.bin, where *xx* is the release major version, *y* is the release minor version, and *zz* is the maintenance number, from Cisco.com to the root directory of the TFTP server.

Step 6 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 (for example, P0S3-*xx*-*y*-*zz)*.

Step 7 Reset each phone.

Dual Booting from SCCP or MGCP to Release 4.0

Step 1 Copy the P0S30202.bin binary image from Cisco.com to the root directory of the TFTP server.

Step 2 If you are dual booting from a Cisco IP phone running the SCCP or MGCP protocol, open the OS79XX.TXT file with a text editor and change the file to include P0S30202.

Step 3 Copy the new Release 4.0 binary image P0S3-*xx*-*y*-*zz*.bin, where *xx* is the release major version, *y* is the release minor version, and *zz* is the maintenance number, from Cisco.com to the root directory of the TFTP server.

Step 4 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 (for example, P0S3*xx*-*y*-*zz)*.

Step 5 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.

Performing an Image Upgrade and Remote Reboot

With Version 2.0 and newer of the Cisco SIP IP phone, you can perform an image upgrade and remote reboot using NOTIFY messages and the syncinfo.xml file. The dialplan.xml file can also be pushed down to the phones using a NOTIFY with a check-sync Event header.

Note To perform an image upgrade and remote reboot, a SIP proxy server and a TFTP server must exist in the phone network.

To upgrade the firmware image and perform a remote reboot, complete the following steps:

Step 1 Using an ASCII editor, open the SIPDefault.cnf file located in the root directory of your TFTP server and change the image_version parameter to the name of the latest image.

Step 2 Using an ASCII editor, open the syncinfo.xml file located in the root directory of your TFTP server and specify values for the image version and sync parameter as follows:

```
<IMAGE VERSION="image_version" SYNC="sync_number"/>
```

Where:

- *image_version* is the image version of the phone. The asterisk (*) can be used as a wildcard character.
- *sync_number* is the synchronization level of the phone. The default synchronization level for the phone is 1. A valid value is a character string of up to 32 characters.

Step 3 Send a NOTIFY message to the phone. In the NOTIFY message, ensure that the an Event header that is equal to "check-sync" is included.

The following is a sample NOTIFY message:

```
NOTIFY sip:lineX_name@ipaddress:5060 SIP/2.0
Via: SIP/2.0/UDP ipaddress:5060;branch=1
Via: SIP/2.0/UDP ipaddress
From: <sip:webadim@ipaddress>
To: <sip:lineX_name@ipaddress>
Event: check-sync
Date: Mon, 10 Jul 2000 16:28:53 -0700
Call-ID: 1349882@ipaddress
CSeq: 1300 NOTIFY
Contact: <sip:webadmin@ipaddress>
Content-Length: 0
```

After the remote reboot process is initiated on the phone via the NOTIFY message, the following actions take place:

1. If the phone is currently in an idle state, the phone waits 20 seconds and then contacts the TFTP server for the syncinfo.xml file. If the phone is not in an idle state, the phone waits until it is in an idle state for 20 seconds and then contacts the TFTP server for the syncinfo.xml file.
2. The phone reads the syncinfo.xml file and performs the following as appropriate:
   a. Determines whether the current image is specified. If so, the phone proceeds to Step c. If not, the phone proceeds to Step b.
   b. Determines whether there is a wildcard entry (*) in the image version parameter. If so, the phone proceeds to Step c. If not, the phone proceeds to Step d.
   c. Determines if the synchronization value is different than what is stored on the phone. If so, the phone proceeds to Step e. If not, the phone proceeds to Step d.
   d. The phone does nothing.
   e. The phone reboots.

      The phone the performs a normal reboot process as described in the "Initialization Process Overview" section on page 2-1, sees the new image, and upgrades to the new image with a synchronization value of what is specified in the syncinfo.xml file.