



Managing Cisco SIP IP Phones

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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 3-7.
- 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.” See the [“Setting the Date, Time, and Daylight Saving Time”](#) section on page 3-31.



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.
- Date, time, and Daylight Saving Time settings. See the [“Setting the Date, Time, and Daylight Saving Time”](#) section on page 3-31

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.
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.

Table 3-1 Network Configuration Parameters (continued)

Parameter	Can Edit?	Description
Erase Configuration	Yes	Whether to erase all of the locally defined 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 3-35.
Host Name	No	Unique host name assigned to the phone. The value in this field is always SIPmac where mac is the MAC address of the phone.
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.
Network Media Type	Yes	Ethernet port negotiation mode. Valid values are: <ul style="list-style-type: none"> • Auto—Port is auto-negotiated. (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: <ul style="list-style-type: none"> • Hub/Switch (default) • PC <p>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.</p>
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 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, and Services), 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 via that menu.

Table 3-2 SIP Parameters Summary

Configuration File	SIP Configuration Menu	Network Configuration Menu	Services Menu
anonymous_call_block	NA	NA	Anonymous Call Block
autocomplete	NA	NA	Auto-Complete Numbers
callerid_blocking	NA	NA	Caller ID Block
call_waiting	NA	NA	Call Waiting
cnf_join_enable	NA	NA	NA
dial_template	NA	NA	NA
dnd_control	NA	NA	Do Not Disturb
dst_auto_adjust	NA	NA	NA
dst_offset	NA	NA	NA
dst_start_day	NA	NA	NA
dst_start_day_of_week	NA	NA	NA
dst_start_month	NA	NA	NA

Table 3-2 SIP Parameters Summary (continued)

Configuration File	SIP Configuration Menu	Network Configuration Menu	Services Menu
dst_start_time	NA	NA	NA
dst_start_week_of_month	NA	NA	NA
dst_stop_day	NA	NA	NA
dst_stop_day_of_week	NA	NA	NA
dst_stop_month	NA	NA	NA
dst_stop_time	NA	NA	NA
dst_stop_week_of_month	NA	NA	NA
dtmf_avt_payload	NA	NA	NA
dtmf_db_level	NA	NA	NA
dtmf_inband	NA	NA	NA
dtmf_outofband	Out of Band DTMF	NA	NA
enable_vad	Enable VAD	NA	NA
end_media_port	End Media Port	NA	NA
image_version	NA	NA	NA
linex_authname (line1 to line6)	Authentication Name	NA	NA
linex_displayname (line1 to line6)	Display Name	NA	NA
linex_name (line1 to line6)	Name	NA	NA
linex_password (line1 to line6)	Authentication Password	NA	NA
linex_shortname (line1 to line6)	Shortname	NA	NA
messages_uri	Messages URI	NA	NA
nat_address	NAT WAN Address	NA	NA
nat_enable	NAT Enabled	NA	NA
nat_received_processing	NA	NA	NA
network_media_type	NA	Network Media Type	NA
network_port2_type	NA	Network Port 2 Device Type	NA
outbound_proxy	Outbound Proxy	NA	NA
outbound_proxy_port	Outbound Proxy Port	NA	NA
phone_label	Phone Label	NA	NA
phone_password	NA	NA	NA
phone_prompt	NA	NA	NA
preferred_codec	Preferred Codec	NA	NA
proxy_backup	Backup Proxy	NA	NA
proxy_backup_port	Backup Proxy Port	NA	NA
proxy_emergency	Emergency Proxy	NA	NA
proxy_emergency_port	Emergency Proxy Port	NA	NA

Table 3-2 SIP Parameters Summary (continued)

Configuration File	SIP Configuration Menu	Network Configuration Menu	Services Menu
proxy_register	Register with Proxy	NA	NA
proxyN_address (N=1 to 6)	Proxy Address	NA	NA
proxyN_port (N=1 to 6)	Proxy Port	NA	NA
sip_invite_retx	NA	NA	NA
sip_retx	NA	NA	NA
sntp_mode	NA	NA	NA
sntp_server	NA	NA	NA
start_media_port	Start Media Port	NA	NA
sync	NA	NA	NA
tftp_cfg_dir	TFTP Directory	NA	NA
time_format_24hr	NA	NA	Time format 24hr
time_zone	NA	NA	NA
timer_invite_expires	NA	NA	NA
timer_register_expires	Register Expires	NA	NA
timer_t1	NA	NA	NA
timer_t2	NA	NA	NA
tos_media	NA	NA	NA
user_info	NA	NA	NA
voip_control_port	VoIP Control Port	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 [“Configuring SIP Parameters via a TFTP Server”](#) section on page 2-4, 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 3-31](#) 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	<p>Whether the Anonymous Call Block feature is enabled or disabled by default on the phone. Valid values are:</p> <ul style="list-style-type: none"> • 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. <p>The default value is 0.</p>
autocomplete	Optional	<p>Whether to have numbers automatically completed when dialing. Valid values are 0 (disable auto completion) or 1 (enable auto completion). The default is 1.</p>

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

Parameter	Required or Optional	Description
call_waiting	Optional	<p>Whether the call waiting feature is enabled or disabled by default on the phone. Valid values are:</p> <ul style="list-style-type: none"> • 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. <p>The default value is 1.</p>

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

Parameter	Required or Optional	Description
callerid_blocking	Optional	<p>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:</p> <ul style="list-style-type: none"> • 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. <p>The default value is 0.</p>
cnf_join_enable	Optional	<p>Specifies when the conference bridge hangs up whether or not it should attempt to join the two leaf nodes. Valid values are:</p> <ul style="list-style-type: none"> • 0—Do not join two leaf nodes. • 1—Join two leaf nodes. <p>The default value is 1, or join two leaf nodes.</p>

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

Parameter	Required or Optional	Description
dnd_control	Optional	<p>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:</p> <ul style="list-style-type: none"> • 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. <p>The default value is 0.</p>
dst_auto_adjust	Optional	See the “Setting the Date, Time, and Daylight Saving Time” section on page 3-31 section for more information.
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		
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.

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

Parameter	Required or Optional	Description
dtmf_db_level	Optional	In-band DTMF digit tone level. Valid values are: <ul style="list-style-type: none"> • 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-band signaling via the AVT tone method. Valid values are: <ul style="list-style-type: none"> • 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.
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.
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.
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).

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

Parameter	Required or Optional	Description
nat_enable	Optional	<p>Use 0 to disable NAT and 1 to enable NAT. Default is 0. When NAT is enabled, the Contact header appears like this:</p> <pre>Contact: sip:lineN_name@nat_address:voip_control_port</pre> <p>If nat_address is invalid or UNPROVISIONED, then the Contact header appears like this:</p> <pre>Contact: sip:lineN_name@phone_ip_address:voip_control_port</pre> <p>and the Via header appears like this:</p> <pre>Via: SIP/2.0/UDP phone_ip_address:voip_control_port</pre> <p>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.</p>
nat_received_processing	Optional	<p>Use 0 to disable NAT received processing and 1 to enable NAT received processing. Default is 0.</p> <p>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.</p>
network_media_type	Optional	<p>Ethernet port negotiation mode. Valid values are:</p> <ul style="list-style-type: none"> • Auto—Port is auto-negotiated. • 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. <p>The default is Auto.</p>

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

Parameter	Required or Optional	Description
network_port2_type	Optional	The device type that is connected to port 2 of the phone. Valid values are: <ul style="list-style-type: none"> • Hub/Switch (default) • PC <p>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.</p>
outbound_proxy	Optional	The IP address of the outbound proxy server. You can use either a dotted IP address or a DNS name.
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. <p>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:</p> <ul style="list-style-type: none"> • 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.

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

Parameter	Required or Optional	Description
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.
proxy_register	Optional	<p>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.</p> <p>After a phone has initialized and registered with a proxy server, changing the value of this parameter to 0 unregisters the phone from the proxy server. To reinitiate a registration, change the value of this parameter back to 1.</p> <p>Note If you enable registration, and authentication is required, you must specify values for the <code>linex_authname</code> and <code>linex_password</code> parameters (where <i>x</i> 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 3-19.</p>
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	<p>Port number of the primary SIP proxy server. This is the port on which the SIP client listens for messages. The default is 5060.</p> <p>Note For additional phone lines, <code>proxyN_address</code> and <code>proxyN_port</code> 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 <code>proxyN_address</code> parameter, the phone uses the <code>proxy1_address</code> parameter as the default.</p>
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 <code>proxyN_address</code> 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 <code>proxyN_address</code> parameters should be configured as dotted IP addresses.

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

Parameter	Required or Optional	Description
proxyN_port	Optional	Port number of the SIP proxy server that will be used by phone lines other than line 1.
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 3-31 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.
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: <ul style="list-style-type: none"> • 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 3-31 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.

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

Parameter	Required or Optional	Description
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.
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: <ul style="list-style-type: none"> • 0 (IP_ROUTINE) • 1 (IP_PRIORITY) • 2 (IP_IMMEDIATE) • 3 (IP_FLASH) • 4 (IP_OVERRIDE) • 5 (IP_CRITIC) The default is 5.
user_info	Optional	Configures the “user=” parameter in the REGISTER message. Valid values are: <ul style="list-style-type: none"> • 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:POS3-xx-y-zz ;

#Default Codec
preferred_codec :g711ulaw

#Enable Registration
proxy_register :1 ;

#Registration expiration
timer_register_expires :3600 ;

#Proxy address
proxy1_address: 192.168.1.1 ;
```

Modifying the Phone-Specific SIP Configuration File

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

Before You Begin

- Review the guidelines and restrictions documented in the [“Configuration File Guidelines” section on page 2-4](#).
- 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
linex_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.
linex_shortcode	Optional	Name or number associated with the linex_name as you want it to display on the phone's LCD if the linex_name length exceeds the allowable space in the display area. For example, if the linex_name value is the phone number 111-222-333-4444, you can specify 34444 for this parameter to have 34444 display on the LCD instead. 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 linex_name variable is displayed.
linex_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 linex_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.
linex_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 linex_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.
linex_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

Parameter	Required or Optional	Description
dnd_control	Optional	<p>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:</p> <ul style="list-style-type: none"> • 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. <p>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.</p>
phone_label	Optional	<p>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.</p> <p>Save the file to your TFTP server (in the root directory or a subdirectory containing all the phone-specific configuration files). Name the file SIPXXXXYYYYZZZZ.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, SIP00503EFFF842.cnf).</p>

The following is a sample configuration file:

```
; phone-specific configuration file sample
; Line 1 phone number
line1_name : 5551212

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

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

Modifying the SIP Parameters 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 `linex`) 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 <code>linex_name</code> as you want it to display on the phone's LCD if the <code>linex_name</code> value exceeds the display area. For example, if the <code>linex_name</code> value is the phone number 111-222-333-4444, you can specify 3444 for this parameter to have 3444 display on the LCD instead. Alternatively, if the value for the <code>linex_name</code> 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.

Table 3-5 SIP Configuration Parameters (continued)

Parameter	Required or Optional	
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 <code>SIPmacaddress</code> , where <code>macaddress</code> is the MAC address of the phone.
Display Name	Optional	Identification as it should appear for caller identification. For example, instead of <code>jdoe@company.com</code> 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: <ul style="list-style-type: none"> • 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. <p>Note If you enable registration, and authentication is required, you must specify values for the Authentication Name and Authentication Password parameters.</p>
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
<pre>SIP Phone> debug {console-stall strlib malloc malloc-table sk-platform flash dsp 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}</pre>	<p>Shows detailed debug output when used with the following keywords:</p> <ul style="list-style-type: none"> • 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: Shows debug output for the memory allocation table. • 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. • softkeys: Displays the currently available soft key sets. • 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).

Table 3-7 CLI Commands (continued)

Command	Purpose
debug command keywords (continued)	<ul style="list-style-type: none"> • 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. <p>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.</p>
SIP Phone> dns	<p>Manipulates the DNS system. The following arguments are used:</p> <ul style="list-style-type: none"> • -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> exit	Exits the Telnet or console session.
SIP Phone> ping <i>ipaddress number packetsize timeout</i>	Sends an Internet Control Message Protocol (ICMP) ping to a network address. You can use a dotted IP address or an alphanumeric address. The <i>number</i> value specifies how many pings to send; the default value is 5. The <i>packetsize</i> 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 <i>timeout</i> value is measured in seconds and identifies how long to wait before the request times out; the default is 2.
SIP Phone> register { <i>option</i> <i>line</i> }	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.

Table 3-7 CLI Commands (continued)

Command	Purpose
<pre>SIP Phone> show {debug strpool memorymap dump malloctable stacks status abort_vector flash dspstate rtp tcp lsm fsm fsmdef fsmcnf fsmxfr fim gsm register network config personaldir}</pre>	<p>Shows information about the SIP IP phone. The following keywords are used:</p> <ul style="list-style-type: none"> • debug: Shows which debug modes are activated. • strpool: Shows the string library pool of strings. • memorymap: Shows memory mapping table, including free, used, and wasted blocks. • dump: Displays a dump of the memory contents. • 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)	<ul style="list-style-type: none"> • 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.
<pre>SIP Phone> test {open close key onhook offhook show hide}</pre>	<p>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.</p> <p>The following commands are available:</p> <ul style="list-style-type: none"> • 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: <ul style="list-style-type: none"> - 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 <p>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.</p> • test onhook: Simulates a handset onhook event. • test offhook: Simulates a handset offhook event. • test show: Shows test feedback. • test hide: Hides test feedback.

Table 3-7 CLI Commands (continued)

Command	Purpose
SIP Phone> <code>tty {echo {on off} mon timeout value kill session msg}</code>	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.
SIP Phone> <code>traceroute ip-address [ttl]</code>	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: <ul style="list-style-type: none"> <i>ip-address</i>: The dotted IP address or alphanumeric address (host name) of the host to which you are sending the traceroute. <i>ttl</i>: The time-to-live value, or the number of routers (hops) through which the datagram can pass. The default value is 30.
SIP Phone> <code>undebug {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}</code>	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.

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.

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

<code>sntp_server</code> =0.0.0.0	<code>sntp_mode</code> = unicast	<code>sntp_mode</code> = multicast	<code>sntp_mode</code> = anycast	<code>sntp mode</code> = 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

<code>sntp_server</code> = 192.168.1.9	<code>sntp_mode</code> = unicast	<code>sntp_mode</code> = multicast	<code>sntp_mode</code> = anycast	<code>sntp_mode</code> = 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 hour/minute, -hour/minute, +hour/minute, hour, -hour, +hour, PST, MST, CST, or EST. (Time zone abbreviations are case sensitive and must be in all capital letters.)
- 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 the same as for the `time_zone` parameter.
 - `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 : 03/00
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, leave the variable in the file, but 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.

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 auto-negotiated a full-duplex 100-Mbps connection.
- Port 0 Half, 100—Indicates that the network is in a linked state and has auto-negotiated a half-duplex 100-Mbps connection.
- Port 0 Full, 10—Indicates that the network is in a linked state and has auto-negotiated a full-duplex 10-Mbps connection.
- Port 0 Half, 10—Indicates that the network is in a linked state and has auto-negotiated a half-duplex 10-Mbps connection.
- Port 1 Full, 100—Indicates that the network is in a linked state and has auto-negotiated a full-duplex 100-Mbps connection.
- Port 1 Half, 100—Indicates that the network is in a linked state and has auto-negotiated a half-duplex 100-Mbps connection.
- Port 1 Full, 10—Indicates that the network is in a linked state and has auto-negotiated a full-duplex 10-Mbps connection.
- Port 1 Half, 10—Indicates that the network is in a linked state and has auto-negotiated 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 **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 **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 3-10](#) to determine how to upgrade.

Table 3-10 Upgrade Scenarios

Image Name	Use Section
POS30100, POS30200, POS30201, and POS3Zxxx	Upgrading from Release 2.1 or Earlier Releases to Release 3.0, page 3-39
P003xxxx or P003xxxxxxxx (these images are loaded on the Cisco SIP IP phone when it is shipped)	Dual Booting from SCCP or MGCP to Release 3.0, page 3-40
P0M3xxxx or P0M3xx-y-zz	Dual Booting from SCCP or MGCP to Release 3.0, page 3-40
POS30202 and POS30203	Upgrading from Release 2.2 or 2.3 to Release 3.0, page 3-39

Upgrading from Release 2.2 or 2.3 to Release 3.0

-
- Step 1** Copy the new Release 3.0 binary image `POS3xx-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, `POS3xx-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 3.0

-
- Step 1** Copy the `POS30202.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 `POS30202`.
- Step 3** Open the phone configuration file with a text editor and edit the `image_version` variable to read `POS30202`.
- 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 3.0 binary image `POS3xx-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, `POS3xx-y-zz`).
- Step 7** Reset each phone.
-

Dual Booting from SCCP or MGCP to Release 3.0

-
- Step 1** Copy the POS30202.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 POS30202.
 - Step 3** Copy the new Release 3.0 binary image POS3xx-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, POS3xx-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 of the Cisco SIP IP phone, you can perform an image upgrade and remote reboot using Notify messages and the `syncinfo.xml` file.



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 tasks:

-
- 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 then 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.

