



Managing Cisco SIP IP Phones

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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 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 will change 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 will change 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.

Modifying the Phone's Network Settings

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

Before You Begin

When configuring network settings, remember the following:

- Unlock configuration mode as described in the “[Unlocking Configuration Mode](#)” section on [page 3-1](#). 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 2-14](#).
- After making your changes, relock configuration mode as described in the “[Locking Configuration Mode](#)” section on [page 3-2](#).

Procedure

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

Step 2 Highlight **Network Configuration**.

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

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

- **DHCP Server**—IP address of the DHCP server from which the phone received its IP address and additional network settings. You cannot change the information in this field.
- **MAC Address**—Factory-assigned unique 48-bit hexadecimal MAC address of the phone. You cannot change the information in this field.
- **Host Name**—Unique host name assigned to the phone. The value in this field is always **SIPmac** where *mac* is the MAC address of the phone. You cannot change the information in this field.
- **Domain Name**—Name of the DNS domain in which the phone resides.
- **IP Address**—IP address of the phone that was assigned by DHCP or locally configured. To edit this field, DHCP must be disabled.
- **Subnet Mask**—IP subnet mask used by the phone. A subnet mask partitions the IP address into a network and a host identifier. To edit this field, DHCP must be disabled.
- **TFTP Server**—IP address of the TFTP server from which the phone downloads its configuration files and firmware images. To edit this field, DHCP must be disabled.

- Default Routers 1 through 5—IP address of the default gateway used by the phone. Default Routers 2 through 5 are the IP addresses of the gateways that the phone will attempt to use as an alternate gateway if the primary gateway is NA. To edit this field, DHCP must be disabled.
- DNS Servers 1 through 5—IP address of the DNS server used by the phone to result names to IP addresses. The phone will attempt to use DNS Servers 2 through 5 if DNS Server 1 is unavailable. To edit this field, DHCP must be disabled.
- Operational VLAN Id—Unique identifier of the VLAN of which the phone is a member. This identifier is obtained through Cisco Discovery Protocol (CDP). You cannot change the information in this field.
- Admin. VLAN Id—Unique identifier of the VLAN to which the phone is attached. The value in this field is only used in non-Cisco switched networks. You can change the administrative VLAN used by the phone, however, if you have an administrative VLAN assigned on the Catalyst switch, that setting overrides any changes made on the phone.
- Network Media Type—Ethernet port negotiation mode. Possible values are
 - Auto—Port is auto-negotiated.
 - Full-100—Port is configured to be a full-duplex, 100MB connection.
 - Half-100—Port is configured to be a half-duplex, 100MB connection.
 - Full-10—Port is configured to be a full-duplex, 10MB connection.
 - Half-10—Port is configured to be a half-duplex, 10MB connection.The default is Auto.
- Network Port 2 Device Type—Valid values are:
 - Hub/Switch (default)
 - PC



Note If the value is PC, port 2 can only be connected to a PC. If you are not sure about the connection, use the default value. If the value is PC and port 2 is connected to a switch, the result is spanning tree loops and wreaked havoc on the network.

- DHCP Enabled—Whether the phone will use DHCP to configure network settings (IP address, subnet mask, domain name, default router list, DNS server list, and TFTP address). Possible values for this field are Yes and No. By default, DHCP is enabled on the phone. To manually configure your IP settings, you must first disable DHCP.
- DHCP Address Released—Whether the IP address of the phone can be released for reuse in the network. When you set this field to **Yes**, the phone sends a DHCP release message to the DHCP server and goes into a release state. The release state provides enough time to remove the phone from the network before the phone attempts to acquire another IP address from the DHCP server. When moving the phone to a new network segment, you should first release the DHCP address.
- Alternate TFTP—Whether to use an alternate TFTP server. This field enables an administrator to specify the remote TFTP server rather than the local one. Possible values for this parameter are Yes and No. The default is No. When Yes is specified, the IP address in the TFTP Address parameter must be changed to the address of the alternate TFTP server.
- Erase Configuration—Whether to erase all of the locally-defined settings on the phone and reset the values to the defaults. Selecting Yes will re-enable DHCP. For more information on erasing the local configuration, see the “[Erasing the Locally-Defined Settings](#)” section on page 3-20.

■ Modifying the Phone's SIP Settings

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


Caution

When you have completed your changes, ensure that you lock the phone as described in the “[Locking Configuration Mode](#)” section on page 3-2.

Modifying the Phone's SIP Settings

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

When modifying SIP parameters, remember the following:

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

Table 3-1 SIP Parameters Summary

Table 3-1 lists each of the SIP parameters that you can configure. In the Configuration column, the name of a parameter as you would specify it in a configuration file is listed. In the 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 can cannot be defined via that menu.

Table 3-1 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
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
dst_start_time	NA	NA	NA
dst_start_week_of_month	NA	NA	NA

Table 3-1 SIP Parameters Summary (continued)

Configuration File	SIP Configuration Menu	Network Configuration Menu	Services Menu
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
dtmlf_avt_payload	NA	NA	NA
dtmlf_db_level	NA	NA	NA
dtmlf_inband	NA	NA	NA
dtmlf_outofband	Out of Band DTMF	NA	NA
enable_vad	Enable VAD		
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
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
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

Table 3-1 SIP Parameters Summary (continued)

Configuration File	SIP Configuration Menu	Network Configuration Menu	Services Menu
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
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 not required, we recommend 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 an easier task. You can then define only those parameters that are specific to a phone in the phone-specific configuration file. Phone-specific parameters should only be defined in a phone-specific configuration file or manually configured. Phone-specific parameters should not be defined in the default configuration file.

Modifying the Default SIP Configuration File

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

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

Before You Begin

- Ensure that you have downloaded the SIPDefault.cnf file from CCO to the root directory of your TFTP server.
- Review the guidelines and restrictions documented in the “[Configuration File Guidelines](#)” section on page 2-4.

Procedure

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

- **image_version**—(Required) Firmware version that the Cisco SIP IP phone should run.
Enter the name of the image version (as it is release by Cisco). Do not enter the extension. You cannot change the image version by changing the file name because the version is also built into the file header. Trying to change the image version by changing the file name will cause the firmware to fail when it compares the version in the header against the file name.
- **proxy1_address**—(Required) IP address of the primary SIP proxy server that will be used by the phones. Enter this address in IP dotted-decimal notation.
- **proxy1_port**—(Optional) Port number of the primary SIP proxy server. This is the port on which the SIP client will listen for messages. The default is 5060.



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 additional 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. If you want to use dotted ip, the proxyN_address parameters should be configured as dotted ip.
- **proxyN_port**—(Optional) Port number of the SIP proxy server that will be used by additional phone lines other than line 1.
- **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_OVERRIDE)
 - 5 (IP_CRITIC)The default is 5.
- **preferred_codec**—(Optional) CODEC to use when initiating a call. Valid values are g711alaw, g711ulaw, and g729a. The default is g711ulaw.

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- `dtmf_inband`—(Optional) Whether to detect and generate in-band signaling format. Valid values are 1 (generate DTMF digits in-band) and 0 (do not generate DTMF digits in-band). The default is 1.
- `dtmf_db_level`—(Optional) In-band DTMF digit tone level. Valid values are:

- 1 (6 db below nominal)
- 2 (3 db below nominal)
- 3 (nominal)
- 4 (3 db above nominal)
- 5 (6 db above nominal)

The default is 3.

- `dtmf_outofband`—(Optional) Whether to 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`.

- `dtmf_avt_payload`—(Optional) Payload type for AVT packets. Possible range is 96 to 127. If the value specified exceeds 127, the phone will default to 101.
- `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.
- `timer_invite_expires`—(Optional) The amount of time, in seconds, after which a SIP INVITE will expire. This value is used in the Expire header field. The valid value is any positive number, however, we recommend 180 seconds. The default is 180.
- `sip_retx`—(Optional) Maximum number of times a SIP message other than an INVITE request will be retransmitted. The valid value is any positive integer. The default is 10.
- `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.
- `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 will unregister 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 `linex_authname` and `linex_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 3-12.

- `timer_register_expires`—(Optional) The amount of time, in seconds, after which a REGISTRATION request will expire. This value is inserted into the Expire header field. The valid value is any positive number, however, we recommend 3600 seconds. The default is 3600.
- `messages_uri`—(Optional) Number to call to check voicemail. This number will be called when the **Messages** key is pressed.
- Date, Time, and Daylight Saving Time parameters. See the “[Setting the Date, Time, and Daylight Saving Time](#)” section on page 3-17 section for more information on setting the following parameters:
 - `sntp_mode`—(Optional) Mode in which the phone will listen for the SNTP server.
 - `sntp_server`—(Optional) IP address of the SNTP server from which the phone will obtain time data.
 - `time_zone`—(Optional) Time zone in which the phone is located.
 - `dst_offset`—(Optional) Offset from the phone’s time when DST is in effect.
 - `dst_start_month`—(Optional) Month in which DST starts.
 - `dst_start_day`—(Optional) Day of the month on which DST begins.
 - `dst_start_day_of_week`—(Optional) Day of the week on which DST begins.
 - `dst_start_week_of_month`—(Optional) Week of month in which DST begins.
 - `dst_start_time`—(Optional) Time of day on which DST begins.
 - `dst_stop_month`—(Optional) Month in which DST ends.
 - `dst_stop_day`—(Optional) Day of the month on which DST ends.
 - `dst_stop_day_of_week`—(Optional) Day of the week on which DST ends.
 - `dst_stop_week_of_month`—(Optional) Week of month in which DST ends.
 - `dst_stop_time`—(Optional) Time of day on which DST ends.
 - `dst_auto_adjust`—(Optional) Whether or not DST is automatically adjusted on the phones.
- `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 will block all calls placed to the phone and log those calls in the Missed Calls directory. Valid values are:
 - 0—The Do Not Disturb feature is off by default, but can be turn 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.
- `callerid_blocking`—(Optional) Whether the Caller ID Blocking feature is enabled or disabled by default on the phone. When enabled, the phone will block its number or email address from phones that have caller identification capabilities. Valid values are:

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- 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.
 - 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 will be received.
 - 1—The Anonymous Call Blocking features is enabled by default, but can be turned on and off via the phone’s user interface. When enabled, anonymous calls will be 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.
 - 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.
 - network_media_type—(Optional) Ethernet port negotiation mode. Valid values are:
 - Auto—Port is auto-negotiated.
 - Full100—Port is configured to be a full-duplex, 100MB connection.
 - Half100—Port is configured to be a half-duplex, 100MB connection.
 - Full10—Port is configured to be a full-duplex, 10MB connection.
 - Half10—Port is configured to be a half-duplex, 10MB connection.
- The default is Auto.
- 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.
 - sync—Value against which to compare the value in the syncinfo.xml before performing a remote reboot. Valid value is a character string up to 32 characters long.
 - time_format_24hr—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.
- proxy_emergency—(Optional) IP address of the emergency proxy/gateway. Enter this address in IP dotted-decimal notation.
- proxy_emergency_port—(Optional) Port number of the emergency proxy. Default is 5060.
- proxy_backup—(Optional) IP address of the backup proxy/gateway. Enter this address in IP dotted-decimal notation.
- proxy_backup_port—(Optional) Port number of the backup proxy. Default is 5060.
- enable_vad—(Optional) Use 0 to disable vad and 1 to enable vad. Default is 0.
- nat_enable—(Optional) Use 0 to disable network address translation (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 SDP message uses the nat_address and a 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.

- nat_address—The WAN IP address of the NAT or firewall server. You can use either a dotted IP address or a DNS name.
- voip_control_port—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.
- start_media_port—The start RTP range for media. Default is 16384. Valid values are 16384 to 32766.
- media_port—The end RTP range for media. Default is 32766. Valid values are 16384 to 32766.
- nat_received_processing—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.

- outbound_proxy—The IP address of the outbound proxy server. You can use either a dotted IP address or a DNS name.
- outbound_proxy_port—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.

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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 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.

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

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

```
; sip default configuration file

#Image Version
image_version:P0S3xxyy ;

#Default Codec
preferred_codec :g711ulaw

#Enable Registration
proxy_register :1 ;

#Registration expiration
timer_register_expires :3600 ;

#Proxy address
proxy1_address: 192.168.1.1 ;
```

Modifying the Phone-Specific SIP Configuration File

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

Before You Begin

- Review the guidelines and restrictions documented in the “[Configuration File Guidelines](#)” section on page 2-4.
- 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 using an e-mail address. Similarly, if you configure a line to use a number, that line can only be called using the number. Each line can have a different proxy configured.

Procedure

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

- 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_shortname**—(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 3444 display on the LCD instead. Alternately, if the value for the **linex_name** parameter is the email 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 purposes. 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** displaying on phones that have caller ID, you can specify John Doe in this parameter to have John Doe display on the callee end instead. If a value is not specified for this parameter, nothing is used.
- **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 will block all calls placed to the phone and log those calls in the Missed Calls directory. Valid values are:
 - 0—The Do Not Disturb feature is off by default, but can be turn 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**—Label to display on the top status line of the LCD. This field is for end-user display only purposes. For example, a phone’s label can display “John Doe’s phone.” Approximately up to 11 characters can be used when specifying the phone label.

■ Modifying the Phone's SIP Settings

- Step 2** 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, SIP00503EFFD842.cnf).
-

The following is an example of a configuration file:

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

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

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

Modifying the SIP Parameters Manually

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

Before You Begin

- Unlock configuration mode as described in the “[Unlocking Configuration Mode](#)” section on [page 3-1](#). 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 2-14](#).
- 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 using an e-mail address. Similarly, if you configure a line to use a number, that line can only be called 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](#).

Procedure

-
- Step 1** Press the **settings** key. The Settings menu is displayed.
- Step 2** Highlight **SIP Configuration**. The SIP Configuration menu is displayed.
- Step 3** Highlight **Line 1 Settings**.
- Step 4** Press the **Select** soft key. The Line 1 Configuration menu is displayed.

Step 5 Highlight and press the **Select** soft key to configure the following parameters as necessary:

- Name—(Required) Number or e-mail address used when registering. When entering a number, enter the number without any dashes. For example, enter 555-1212 as 5551212. When entering an e-mail address, enter the e-mail ID without the host name.
- Short Name—(Optional) Name or number associated with the `linex_name` as you want it to display on the phone's LCD if the `linex_name` value exceeds 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. Alternately, if the value for the `linex_name` parameter is the email 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 purposes.

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 `SIPmacaddress` where `macaddress` is the MAC address of the phone.
- Display Name—(Optional) Identification as it should appear for caller-identification purposes. For example, instead of `jdoe@company.com` displaying on phones that have caller ID, you can specify John Doe in this parameter to have John Doe display on the 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 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 following parameters on the SIP Configuration menu:

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

The default is `avt`.

■ Modifying the Phone's SIP Settings

- 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 will unregister 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 will expire. This value is used the Expire header field. The valid value is any positive number, however, we recommend 3600 seconds. The default is 3600.
- 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 purposes. For example, a phone's label can display “John Doe's phone.”
- Enable VAD—(Optional) Specifies whether VAD is enabled or disabled.
- VoIP Control Port
- Start Media Range
- End Media Range
- Backup Proxy—(Optional) IP address of the backup proxy/gateway. Enter this address in IP dotted-decimal notation.
- Backup Proxy Port—(Optional) Port number of the backup proxy. Default is 5060.
- Emergency Proxy—(Optional) IP address of the emergency proxy/gateway. Enter this address in IP dotted-decimal notation.
- Outbound Proxy
- Outbound Proxy Port
- NAT Enabled
- NAT WAN Address
- Emergency Proxy Port—(Optional) Port number of the emergency proxy. Default is 5060.

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.

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).

We recommend 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 3-2](#) before configuring the SNTP parameters:

[Table 3-2](#) lists the actions that take place when a null value (0.0.0.0) is specified in the sntp_server parameter.

Table 3-2 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 3-3](#) lists the actions that take place when a valid IP address is specified in the sntp_server parameter.

Table 3-3 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.

Procedure

- 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 will listen for the SNTP server. Valid values are unicast, multicast, anycast, or directedbroadcast.
See [Table 3-2](#) and [Table 3-3](#) 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 3-2](#) and [Table 3-3](#) 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.

■ Erasing the Locally-Defined Settings

The following is an example of the 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 an example of the 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 3-1.
- If DHCP has been disabled on a phone, clearing the phone’s settings will reenable it.
- Select the Erase Config parameter by pressing the down arrow to scroll to and highlight the parameter or by pressing the number that represents the parameter (located to the left of the parameter name on the LCD).

Procedure

-
- Step 1** Press the **settings** key. The Settings menu is displayed.
- Step 2** Highlight **Network Configuration**.
- Step 3** Press the **Select** soft key. The Network Configuration settings are displayed.
- Step 4** Highlight **Erase Configuration**.
- Step 5** Press the **Yes** soft key.
- Step 6** Press the **Save** soft key. The phone programs the new information into Flash memory and resets.
-

Erasing the Locally-Defined SIP Settings

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

**Note**

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

Before You Begin

Unlock configuration mode as described in the “[Unlocking Configuration Mode](#)” section on page 3-1.

Procedure

-
- Step 1** Press the **settings** key. The Settings menu is displayed.
- Step 2** Highlight **SIP Configuration**.
- Step 3** Press the **Select** soft key. The SIP Configuration settings are displayed.
- Step 4** Highlight the parameter for which you wish to erase the setting.
- Step 5** Press the **Edit** soft key.
- Step 6** Press the << soft key to delete the current value.
- Step 7** Press the **Validate** soft key to save your change and exit the Edit panel.
- Step 8** If modifying a line parameter, press the **Back** soft key to exit the Line Configuration panel.
- Step 9** Press the **Save** soft key. The phone programs the new information into Flash memory and resets.
-

Accessing Status Information

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

■ Accessing Status Information

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

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

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

Viewing Status Messages

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

-
- Step 1** Press the **Settings** key. The Settings menu is displayed.
 - Step 2** Highlight **Status**.
 - Step 3** Press the **Select** soft key. The Setting Status menu is displayed.
 - Step 4** Highlight **Status Messages**.
 - Step 5** Press the **Select** soft key. The Status Messages panel is displayed.
 - Step 6** To exit the Status Messages panel, press the **Exit** soft key.
-

Viewing Network Statistics

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

-
- Step 1** Press the **settings** key. The Settings menu is displayed.
 - Step 2** Highlight **Status**.
 - Step 3** Press the **Select** soft key. The Setting Status menu is displayed.
 - Step 4** Highlight **Network Statistics**.
 - Step 5** Press the **Select** soft key. The Network Statistics panel is displayed.

The following information is displayed on this panel:

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

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

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



Note

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

Viewing the Firmware Version

To view network statistics, complete the following steps:

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

Step 2 Highlight **Status**.

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

Step 4 Highlight **Firmware Versions**.

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

The following information is displayed on this panel:

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

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

Upgrading the Cisco SIP IP Phone Firmware

There are two methods that you can use to upgrade the firmware on your Cisco SIP IP phones. You can upgrade the firmware on one phone at a time via 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 CCO to the root directory of your TFTP server.

Procedure

-
- Step 1** Copy the binary file P0S3xxyy.bin (where xx is the version number and yy is the subversion number) from CCO to the root directory of the TFTP server.
- Step 2** Using a text editor, open the configuration file and update the image version specified in the image_version variable. The version name in image_version variable should match the version name (without the .bin extension) of the latest firmware that you downloaded.
- Step 3** Reset each phone.

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



Note

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

Performing an Image Upgrade and Remote Reboot

With Version 2.0 of the Cisco SIP IP Phone 7960, 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:

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.
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 sync level for the phone is 1. Valid values is a character-string up to 32 characters.
3. Send a NOTIFY message to the phone. In the Notify message, ensure that the an Event header equal to “check-sync” is included.

The following is an example of a 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:webadmin@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
```

Once 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 will wait 20 seconds and then contact the TFTP server for the syncinfo.xml file. If the phone is not in an idle state, the phone will wait until it is in an idle state for 20 seconds and then contact 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 c. If not, the phone proceeds to b.
 - b. Determines whether there is a wildcard entry (*) in the image version parameter. If so, the phone proceeds to c. If not, the phone proceeds to d.
 - c. Determines if the sync value is different than what is stored on the phone. If so, the phone proceeds to e. If not, the phone proceeds to d.
 - d. The phone does nothing.
 - e. The phone reboots.

The phone performs a normal reboot process as described in [“Initialization Process Overview” section on page 2-1](#), sees the new image, and upgrades to the new image with a sync value of what is specified in the syncinfo.xml file.

■ Performing an Image Upgrade and Remote Reboot