# Cisco 7940 and 7960 IP Phones

## 1 Warnings

Check the *SIP 3rd Party Validation website* for certification status. It can be located at:

http://testlab.inin.com

Vendor Documentation

Cisco SIP products:

http://www.cisco.com/warp/public/cc/techno/tyvdve/sip/

Cisco IP Phone Comparison Sheet:

http://www.cisco.com/en/US/products/hw/phones/ps379/products_qanda_item09186a00801739d1.shtml

Cisco Phone Data Sheets (not all Cisco phone support SIP yet):
http://www.cisco.com/en/US/products/hw/phones/ps379/products_data_sheets_list.html

Cisco IP Phone Documentation for Session Initiation Protocol:
http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/

Managing Cisco IP Phones (describes all the phones parameters):

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/sipadm30/maintain.htm#xtocid23

Localization (describes the languages supported):

http://www.cisco.com/en/US/products/sw/voicesw/ps2156/products_administration_guide_chapter09186a0080187870.html

## 2 Versions Verified

Firmware for **All** Cisco's SIP Phones (requires Cisco login):

http://www.cisco.com/kobayashi/sw-center/sw-voice.shtml

Firmware for **7940 and 7960** Cisco's SIP Phones (requires Cisco login):

http://www.cisco.com/cgi-bin/tablebuild.pl/sip-ip-phone7960

Supported Versions: 7.4

# 3 PreInstall

Each Cisco phone requests its 4 files via TFTP. You will need a TFTP server. See section **Error! Reference source not found.** "**Error! Reference source not found.**" for a list of free TFTP servers.

# 4 Install

The Cisco phone will requests the following files (in this order):

| File Name | Description |
|---|---|
| OS79XX.TXT | This file contains the name of the protocol and firmware, i.e. POS30202 (without the .bin extension). This file should be contain POS30202 (even if 3.1 is being used). The phone will get this file, download POS30202.bin if necessary (2.2 is sometimes necessary to download before going to 3.1), and then get it's SIP config files, and then download the firmware for that specific phone from the parameter *image_version* in the SIP config files. |
| [firmware filename].bin | This file contains the SIP firmware. Its name (without the .bin extension) is specified with the parameter *image_version*. An example file name is POS30202.bin for 2.2 and POS3-03-2-00.bin for 3.2. |
| SIPDefault.cnf | This is the Phone-Common file. It contains parameters common to all phones. The sample file supplied by Cisco is *SIPDefaultGeneric.cnf*. |
| SIP[*MAC address*].cnf | This is the Phone-Specific file. It contains the parameters specific to the individual phone with the specific MAC address. An example file name is SIP1234567890AB.cnf. "SIP" must be uppercase, letters in the MAC address must be uppercase, and ".cnf" must be lowercase. The sample file supplied by Cisco is *SIPConfigGeneric.cnf.* |
| RingList.dat | This file contains the ring files. |
| DialPlan.xml | This file contains the dial plan used by the phones |

# 5 Required Post Installation Steps

Check with your Cisco representative about phone licensing. The SIP firmware on the Cisco website is downloadable – but might need to be licensed.

# 6 Configuration

Line Appearances: Up to 2 (7940) and up to 6 (7960). Each line appearance is equivalent to a station in the Interaction Administrator.

Call Appearances: Each line appearance can handle up to 2 call appearances.

Using *persistent* connections allows the phone to handle more call appearances than the phone is physically capable. This is done by using the *persistent* connections and the Interaction Center Client. To manage more calls than the phone is capable (for instance an operator want to handle up to 20 simultaneous calls), check the *Persistent* checkbox in the Station configuration in Interaction Administrator. The Interaction Client can be used to manage a large number of calls while the phone will be the audio device for the calls. The phone will show one call (from the Interaction Center) while the Interaction Client will be used to manipulate the calls.

Put parameters for all phones in the SIPDefault.cnf file. Put parameters that are specific to a certain phone in its Phone-Specific file. For example, *proxy1_address* parameter can be in either file.

**Important:**

**Confusion Alert:** Make sure you update the image_version (in the SIPDefault.cnf file). See below.

**Bug Alert:** If 4.1 firmware does not download, it is because the phone can not handle the size of the SIPDefault.cnf file. An TFTP error will occur (check this by hitting the phone's *Settings* button, choose *Status*, then choose *Status Messages*). To fix, you must delete some unused lines out of the SIPDefault.cnf (such as the proxy 3-6 entries).

*Confusion Alert:* "Which parameter is used" logic is confusing, since the same parameter can appear in the Phone-Common file, the Phone-Specific file, and manually entered on the phone itself. Here is the rule:

1. Parameters defined in the Phone-Common file will override values in the flash memory.
2. Parameters defined in the Phone-Specific file will override values in the Phone-Common file.
3. Parameters entered locally on the phone will be used until the Phone-Common or Phone-Specific files are downloaded (done at reboot time). If the parameters don't exist in the Phone files, then the values entered locally will continue to be used.

**Simplest Example of a 7940/7960 Configuration**

<u>Phone-Common file SIPDefault.cnf (created from *SIPDefaultGeneric.cnf,* the sample file supplied by Cisco)</u>

*proxy1_address* – address of the Interaction Center

<u>Phone-Specific file SIP[*MAC address*].cnf. (created from *SIPConfigGeneric.cnf,* the sample file supplied by Cisco)</u>

*line1_name* – unique extension of the phone (i.e. 7101)

*line1_displayname* – name that is sent in SIP messages

*phone_label* – label displayed in upper right hand corner of phone

**In Interaction Administrator station configuration**

*Station Connection SIP address: – line1_name* @ [IP address of the phone]

| |
|---|
| *Station Identification SIP address: – line1_name* (preferred) OR *line1_name* @ *proxy1_address* |

Remote administration can be accomplished via telnet. See the telnet_level parameter in the Phone-Common file (SIPDefault.cnf) for more details.

Common parameters are in the Phone-Common file (SIPDefault.cnf) while parameters for specific phones are in the Phone-Specific file (SIP[*MAC address*].cnf). For complete documentation, see the Cisco instructions for their IP phone products (see section 0 "**Error! Reference source not found.**").

| **SIPDefault.cnf**: Phone-Common file (created from Cisco sample file *SIPDefaultGeneric.cnf*) | |
|---|---|
| Parameter | Description |
| image_version | The version of phone firmware being used.<br>*Values*: POS3-03-2-00<br>*Example*: image_version: POS3-03-2-00<br>**Confusion Alert**: This value is not correct in the 3.2 example file on the Cisco web site is P0S30202. This is done because you must install 2.2 firmware before upgrading to releases after 2.2. |
| XML URLS | |
| services_url | Phone Services URL. This is the URL for the initial services menu.<br>This file must be on a web server.<br>If using phone services, see the *SIP Application Note*, *Phone Services* section, on setting up the Cisco Phone services. |
| logo_url | Logo URL. Branding logo for the phone displays.<br>This file must be on a web server.<br>A sample file with the ININ logo can be found at<br>http://www.inin.com/support/sip/index.asp |
| Proxy Section | |
| outbound_proxy | Primary proxy. This is the IP address where the phone will send ALL its requests.<br>***WARNING***: Regardless of the dial plan, all requests from the phone will be sent to the outbound_proxy (if set).<br>*Recommended Setting*:<br>• For managed phones and using a proxy, set *outbound_proxy* to that proxy. The proxy will be configured to route calls originated from the managed phone<br>o to the Interaction Center by using the *proxy1_address* parameter,<br>o to emergency/local gateways since the phone's dial plan will put the gateway's address to be put in the "To" field<br>• For managed phones not using a proxy, do NOT set this parameter. Normal calls will go to the Interaction Center via the *proxy1_address* parameter, or to emergency/local gateways via the dial plan.<br>• For unmanaged phones, set *outbound_proxy* according |

| | |
|---|---|
| | to your site's specifications. |
| Proxy_backup | Backup Proxy. This is the IP address where the phone will send its requests if no system responses to the original SIP request. This proxy will be used in all cases, i.e. if any proxy (the outbound_proxy, the proxy1_address, or the proxy_emergency) fail to respond.<br>This value is used when one of the following are true:<br>• Using 2 or more proxies<br>• Using no proxies and using Interaction Center switchover<br>• Using no proxies and using a local gateway or emergency (911) gateway<br>The IP phone will set this address in the **host portion** of the FROM header (not the TO header) in the SIP messages. |
| Proxy_emergency | Emergency Proxy. This is the IP address where the phone will send its requests if the dialplan interprets the dialed number as an emergency number.<br>The IP phone will set this address in the **host portion** of the TO and FROM headers in the SIP messages. |
| Other… | Other gateways can be targeted by the dial plan. |
| Identification Section | |
| proxy1_address | This will also be the destination of the SIP message if outbound_proxy is NOT set (when calling from line appearance 1).<br>The IP phone will set this address in the **host portion** of the TO and FROM headers in the SIP messages.<br><br>Recommended Setting:<br>• For managed phones and using a proxy, *proxy1_address* should be set to the Interaction Center and the *outbound_proxy* should be set to the proxy's address.<br>• For managed phones and not using a proxy, *proxy1_address* should be set to the Interaction Center and the *outbound_proxy* should not be set.<br>• For unmanaged phones, set *proxy1_address* according to your site's specifications.<br>If using a mixture of managed and unmanaged phones, this parameter can be moved to each of the phone's Phone-Specific file.<br>*Interaction Center Note*: The *line1_name* and *proxy1_address* are the values configured in the station identification address in Interaction Administrator for managed phones. For example, if *line1_name* is "8105" and *proxy1_address* is "172.16.129.248", then the value of the identification address would be "8105" (preferred) or "sip:8105@172.16.129.248:5060". See the "*SIP Application Note"* for details.<br>*Important*: *line1_name* MUST be unique. Since *line1_name* and *proxy1_address* are used for identification with the Interaction Center, one of these fields must be unique (and be in the Phone-Specific file). If using a proxy, *proxy1_address* must be set to the Interaction Center for all the managed phones, thus line1_name must be unique. Therefore, *line1_name* should be unique, even if you are not using a proxy. |
| ialPlan Section | |
| dial_template | Dialplan template (.xml format file relative to the TFTP root |

| | |
|---|---|
| | directory)<br>*Example*: dial_template: dialplan |
| Call Appearances | |
| call_waiting | Call waiting – whether to allow a second call appearance on the line appearance.<br>Values:<br>• 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).<br>• [Default] 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).<br>• 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.<br>• 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) |
| Telnet | |
| telnet_level | Telnet level.<br>*Values*: [default] 0 (disabled), 1 (Enabled), 2 (Privileged)<br>*Note*: To remotely reboot the phone, set telnet_level to 2 (Privileged), telnet to the phone, and then use the telnet command "reset". |
| MWI (no configuration for MWI needed) | |
| Voicemail Retrieval | |
| messages_uri | The number to call to check voicemail. This is the number that is called when the "Messages" button is pressed.<br>*Recommended Setting*: Set this to the Interaction Center. The user portion of the address should match the Interaction Center server parameter *IP Message Button.* See the *SIP Application Note* about Configuring Message Buttons for Voicemail Retrieval.<br>*Example*: The *IP Message Button* server parameter is set to *messagebutton* on the Interaction Center Server *Fred*. Set this value to *messagebutton@Fred*. |
| DND/Forwarding | |
| dnd_control | Do Not Disturb feature.<br>Values:<br>• [Default] 0 (the Do Not Disturb feature is off by default, but can be turned on and off locally via the phone's user interface)<br>• 1 (the Do Not Disturb feature is on by default, but can be turned on and off locally via the phone's user interface).<br>• 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).<br>• 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). |
| Media Section | |
| preferred_codec | Codec for media stream.<br>*Values*: [default] g711ulaw, g711alaw, g729a<br>*Recommended Setting*: Network specific. Note this parameter can be moved to the individual Phone-Specific files if you want different phones to use different codecs. |
| enable_vad | VAD (Voice Activation/Deactivation)<br>*Values*: [default] 0 (disable), 1 (enable)<br>*Recommended Setting*: Network specific. |
| tos_media | TOS bits in media stream.<br>*Values*: 0 (IP_ROUTINE), 1 (IP_PRIORITY), 2 (IP_IMMEDIATE), 3 (IP_FLASH), 4 (IP_OVERIDE), [default] 5 (IP_CRITIC)<br>*Recommended Setting*: Network specific. |
| dtmf_inband<br><br>dtmf_outofband<br><br>dtmf_avt_payload<br><br>dtmf_db_level: 3 | Inband DTMF Settings<br>*Values*: 0 (disable), [default] 1 (enable)<br>Out of band DTMF Settings (done via AVT tone method)<br>*Values*: none (disable), [default] avt (avt will be used, and inband DTMF disabled, if avt is requested by remote side), avt_always (always use avt inband DTMF will be disabled)<br>DTMF AVT Payload (Dynamic payload range for AVT tones)<br>Values: 96-127 (default 101)<br><br>DTMF dB Level Settings<br>Values: 1 (6 db below nominal), 2 (3 db below nominal), [default] 3 (nominal), 4 (3 db above nominal), 5 (6 db above nominal) |

Common parameters are in the Phone-Common file (SIPDefault.cnf) while parameters for specific phones are in the Phone-Specific file (SIP[*MAC address*].cnf). For complete documentation, see the Cisco instructions for their IP phone products (see section 0 "**Error! Reference source not found.**").

| **SIP[*MAC address*].cnf**: Phone-Specific file (created from the Cisco sample file *SIPConfigGeneric.cnf*) | |
|---|---|
| Parameter | Description |
| Proxy Section | |
| proxy1_address | The IP phone will set this address in the **host portion** of the TO and FROM headers in the SIP messages.<br><br>Generally, this will be in the Phone-Common file. See the documentation above. |
| Line Appearance | |
| line1_name | Line1 Name. This value is sent as the **user portion** of the |

| | |
|---|---|
| | address in SIP FROM header. **This field should be unique**.<br>**BUG Alert**: This value specified can not contain spaces. If it does, the SIP messages sent by the phone do not properly escape the spaces thus sending illegal SIP syntax.<br>*Interaction Center Note*: The *line1_name* and *proxy1_address* are the values configured in the station identification address in Interaction Administrator for managed phones. For example, if *line1_name* is "8105" and *proxy1_address* is "172.16.129.248", then the value of the identification address would be "8105" (preferred) or "sip:8105@172.16.129.248:5060". See the "*SIP Application Note"* for details.<br>*Important*: *line1_name* MUST be unique. Since *line1_name* and *proxy1_address* are used for identification with the Interaction Center, one of these fields must be unique (and be in the Phone-Specific file). If using a proxy, *proxy1_address* must be set to the Interaction Center for all the managed phones, thus line1_name must be unique. Therefore, *line1_name* should be unique, even if you are not using a proxy. |
| Line1_displayname | Line1 Display name. This value is sent as the display name in the SIP FROM header and displays on the right hand side of the phone. If not configured then it will default to the value used in line1_name.<br>Recommended Setting: This parameter can be in the Phone-Common config file, so all phones will have the same display name (i.e. "Line1"). If you want each phone to have a unique display name, then the parameter can be moved here, to the Phone-Specific config file. |
| Line*X*_name | LineX Name (where X=2..6). This value is sent as the user portion of the address in SIP FROM header.<br>Recommended Setting: Generally, multiple line appearances are not needed. Each line appearance is represented as a station in Interaction Administrator. Each line appearance can handle 2 call appearances.<br>Using *persistent* connections allows the phone to handle more call appearances than the phone is physically capable. This is done by using the *persistent* connections and the Interaction Center Client. To manage more calls than the phone is capable (for instance an operator want to handle up to 20 simultaneous calls), check the *Persistent* checkbox in the Station configuration in Interaction Administrator. The Interaction Client can be used to manage a large number of calls while the phone will be the audio device for the calls. The phone will show one call (from the Interaction Center) while the Interaction Client will be used to manipulate the calls. |
| Line*X*_displayname | Line1 Display name (where X=2..6). This value is sent as the display name in the SIP FROM header and displays on the right hand side of the phone. If not configured then it will default to the value used in line1_name. |
| phone_label | shows on the upper right hand corner of phone, up to 11 characters |