
# Reference List of Parameters

In general, configuring the Cisco Unified IP Phones 7905G and 7912G can be accomplished through Cisco Unified CallManager and through the network configuration options on a phone. These processes are described in detail in this manual.

This appendix describes, for your reference, all the configuration parameters that can be viewed from the Phone Configuration Web Page.

**Note** The parameters on the Phone Configuration Web Page are read-only parameters, with the exception of an the encryption key parameter. For more information, see the "Configuring the Encryption Key" section on page 4-7.

This section includes the following topics:

# Accessing the Web Page for a Phone

You can display device and network information for a Cisco Unified IP Phone by accessing the Phone Configuration Web Page. You can access the web page using any graphically capable web browser.

To access the web page for your Cisco Unified IP Phone, perform the following steps.

**Procedure**

**Step 1** Display the main Phone Configuration Web Page.

For instructions, see the "Viewing Network Settings through a Phone Configuration Web Page" section on page 4-22.

**Step 2** From the main Phone Configuration Web Page, choose the hyperlink for the information that you want to display:

- **Device Information**—For more information, see the "Device Information" section on page A-3.
- **Network Configuration**—For more information, see the "Network Configuration" section on page A-4.
- **Network Statistics**—For more information, see the "Network Statistics" section on page A-6.
- **Device Logs**—For more information, see the "Device Logs" section on page A-7.
- **Network Parameters**—For more information, see the "Network Parameters" section on page A-8.
- **SIP Parameters**—For more information, see the "SIP Parameters" section on page A-15.
- **Call Preference Parameters**—For more information, see the "Call Preference Parameters" section on page A-22.
- **Tone Parameters**—For more information, see the "Tone Parameters" section on page A-28.
- **Audio Parameters**—For more information, see the "Audio Parameters" section on page A-32.

**Step 3** Close your web browser.

**Related Topics**

- Device Information, page A-3
- Network Configuration, page A-4
- Network Statistics, page A-6
- Device Logs, page A-7
- Network Parameters, page A-8
- Tone Parameters, page A-28
- Audio Parameters, page A-32

# Device Information

The Device Information area of the Cisco Unified IP Phone Configuration Web Page displays device settings and related information for the phone. Table A-1 describes these items.

To display the Device Information area, perform either of these steps:

- Access the web page for the phone as described in the "Viewing Network Settings through a Phone Configuration Web Page" section on page 4-22 and then click the **Device Information** hyperlink.
- In a web browser, enter this URL: *ip-address*/**DeviceInformation**, where *ip-address* is the IP address of the phone.

***Table A-1 Device Information Area Items***

| Item | Description |
|---|---|
| MAC address | Unique Media Access Control (MAC) address of the phone |
| Software version | Version of the software running on the phone |
| Hardware revision | Version of the phone hardware |
| Serial number | Serial number of the phone |

*Table A-1 Device Information Area Items (continued)*

| Item | Description |
|---|---|
| Product ID | Product identifier of the phone |
| H/W features | Reserved for future use |
| BTXML cards version | Version of the graphics card in the phone |
| Configuration version stamp | Version of the phone configuration file |

# Network Configuration

The Network Configuration area of the Cisco Unified IP Phone Configuration Web Page displays network configuration information and information about other phone settings. Table A-2 describes these items.

**Note** You can obtain and set many of these items from phone itself. For more information, see Chapter 4, "Configuring Settings on the Cisco Unified IP Phone."

To display the Network Configuration area, perform either of these steps:

- Access the web page for the phone as described in the "Viewing Network Settings through a Phone Configuration Web Page" section on page 4-22 and then click the **Network Configuration** hyperlink.
- In a web browser, enter this URL: *ip-address*/**NetworkConfiguration**, where *ip-address* is the IP address of the phone.

*Table A-2 Network Configuration Area Items*

| Item | Description |
|---|---|
| DHCP server | If DHCP is enabled, the DHCP server that the pone contacts |
| BOOTP server | Not used |
| MAC address | Unique MAC address of the phone |
| Host name | Unique host name assigned to the phone |

*Table A-2 Network Configuration Area Items (continued)*

| Item | Description |
|---|---|
| Domain name | If DHCP is enabled, Domain Name System (DNS) in which the phone resides |
| IP address | If DHCP is enabled, Internet protocol (IP) address of the phone |
| Default router | If DHCP is enabled, default router used by the phone |
| Subnet mask | If DHCP is enabled, subnet mask used by the phone |
| TFTP server 1 | If DHCP is enabled, IP address of the primary TFTP server used by the phone |
| TFTP server 2 | If DHCP is enabled, IP address of the alternate TFTP server used by the phone |
| NTP Server 1 | Primary Network Time Protocol Server that the phone uses |
| NTP Server 2 | Secondary Network Time Protocol Server that the phone uses |
| DNS server 1 | If DHCP is enabled, IP address of the primary DNS server used by the phone |
| DNS server 2 | If DHCP is enabled, IP address of the alternate DNS server used by the phone |
| Alt NTP Server 1 | Not used. |
| Alt NTP Server 2 | Not used. |

*Table A-2 Network Configuration Area Items (continued)*

| Item | Description |
|---|---|
| CallManager 1 – 4 | Prioritized list of Cisco Unified CallManager systems that are available for processing calls from this phone. Possible states include:<br>• Active—Cisco Unified CallManager server from which the phone is currently receiving call-processing services.<br>• Standby—Cisco Unified CallManager server to which the phone switches if the current server goes down.<br>• Blank—No TCP connection to this Cisco Unified CallManager server.<br>This field might also include the Survivable Remote Site Telephony (SRST) designation, indicating an SRST router that assumes control of call processing if all other Cisco Unified CallManager servers are unreachable. |
| DHCP enabled | 1 if DHCP is enabled. 0 if not |

# Network Statistics

The Network Statistics area of the Cisco Unified IP Phone Configuration Web Page displays provide information about network traffic on the phone. Table A-3 describes the items on this menu.

To display the Network Statistics area, perform either of these steps:

- Access the web page for the phone as described in "Viewing Network Settings through a Phone Configuration Web Page" section on page 4-22 and then click the **Network Statistics** hyperlink.
- In a web browser, enter this URL: *ip-address*/**EthernetInformation**, where *ip-address* is the IP address of the phone.

*Table A-3 Network Statistics Area Items*

| Item | Description |
|---|---|
| Elapsed time | Time that has elapsed since the phone or Cisco Unified CallManager was last reset |
| Receive packets | Number of packets that the phone has received during the elapsed time |
| Transmit packets | Number of packets that the phone has transmitted during the elapsed time |
| Broadcast | Number of packets that the network has broadcast during the elapsed time |
| Multicast | Number of multicast packets on the network during the elapsed time |
| Receive errors | Number receive errors that the phone has experienced during the elapsed time |
| Transmit errors | Number transmit errors that the phone has experienced during the elapsed time |
| Receive overflow | Number packet overflows that the phone has experienced during the elapsed time |

# Device Logs

The Device Logs area of the Cisco Unified IP Phone Configuration Web Page is reserved for future use.

**Note** You can access the Device Logs area directly from a web browser by entering this URL: *ip-address*/**DeviceLog**, where *ip-address* is the IP address of your phone.

# Configuration Parameters

This section provides information on the parameters that are used configure the Cisco Unified IP Phones 7905G and 7912G through a profile file or through the Phone Configuration Web Page.

**Note** The configuration parameters are display only on the Phone Configuration Web Page and in the profile configuration file. One exception is encryption, which, under certain circumstances, can be configured on the Phone Configuration Web Page. For more information about encryption, see the "Configuring the Encryption Key" section on page 4-7.

These parameters are organized into the following categories:

- Network parameters—Control various network-related activities of the phone. See the "Network Parameters" section on page A-8.
- SIP Parameters—Control protocol-specific items. See the "SIP Parameters" section on page A-15.
- Call Preference Parameters—Control user features. See the "Call Preference Parameters" section on page A-22.
- Tone parameters—Control how the phone handles the various tones that it plays, and related options. See the "Tone Parameters" section on page A-28.
- Audio parameters—Control how the phone handles various audio-related activities. See the "Audio Parameters" section on page A-32.

# Network Parameters

Table A-4 describes the network parameters that you can configure through a phone's profile file or through its Network Parameters web page.

**Tip** To access the Network Parameters area directly from a web browser, enter *IP_address*/**NetCfg**, where *IP_address* is the IP address of your Cisco Unified IP Phone.

*Table A-4 Network Parameters*

| Parameter | Description | Usage |
| --- | --- | --- |
| UseTftp | Enables or disables downloading of a profile from a TFTP server. If you set this parameter to 1 and subsequently make changes using the web interface, you must set this parameter to 0 before saving those changes; otherwise, the changes will be overwritten by the profile obtained from a TFTP server. | • 0=Do not use a TFTP server.<br>• 1=Use a TFTP server:<br>– If TftpURL is set to 0, use the TFTP IP address or URL obtained from the DHCP server to contact the TFTP server.<br>– If TftpURL is set to a value other than 0, use the specified IP address or URL to contact the TFTP server.<br>**Note** Do not specify a port. The Cisco Unified IP Phone always contacts the TFTP server at port 69.<br>**Value type:** Boolean.<br>**Default value:** 1 |
| TftpURL | IP address or URL of the TFTP server from which a phone obtains a profile. Required if the DHCP server does not provide the TFTP server address. | Optionally, you can include the path prefix to the profile to download. For example, if the TFTP server IP address is 192.168.2.170 or www.cisco.com, and the path to the file is /ip7905, you can specify the URL as 192.168.2.170/ip7905 or www.cisco.com/ip7905.<br>Do not specify a port. The Cisco Unified IP Phone always contacts the TFTP server at port 69.<br>If DHCP is used, a nonzero values of TftpURL overwrites any DHCP-supplied addresses and values of 0 tell the phone to use DHCP-supplied addresses.<br>This parameter is ignored if UseTftp is 0.<br>**Value type:** IP address, up to 31 characters.<br>**Default value:** 0.0.0.0 |

*Table A-4 Network Parameters (continued)*

| Parameter | Description | Usage |
|---|---|---|
| CfgInterval | Number of seconds between automatic profile refreshes from the TFTP server. At the earliest idle time following each interval expiration, the phone retrieves its profile using the **tftp get** command. | Intervals should vary among phones on the network to prevent simultaneous contact of the TFTP server by many phones. This value can range from 60 to 4294967295.<br>This parameter is ignored if UseTftp is 0.<br>**Value type:** Integer.<br>**Default value:** 3600 |
| Symmetric Key | Key for decrypting the configuration profile that is downloaded from the TFTP server. | • 0=The configuration profile is not encrypted.<br>• *string*=The profile is encrypted with this key.<br>This parameter is ignored if UseTftp is 0.<br>**Value type:** Alphanumeric string, up to 8 characters.<br>**Default value:** 0 |
| Dhcp | Specifies whether the phone contacts the DHCP server to obtain values for various network parameters, including IP address, router IP address, and subnet mask. | Set to 0 (disable) if you are not using a DHCP server.<br>Set to 1 (enable) if you are using a DHCP server.<br>The default setting is 1. |
| StaticIP | Static IP address of the phone when DHCP is not used. | Enter the assigned IP address for the phone.<br>This value is ignored when the Dhcp parameter is enabled. |
| StaticRoute | Static IP address of the network router when DHCP is not used. | Enter the assigned IP address of the network router.<br>This value is ignored when the Dhcp parameter is enabled. |
| StaticNetMask | Static subnet mask of the phone when DHCP is not used. | Enter the subnet mask for the phone.<br>This value is ignored when the Dhcp parameter is enabled. |

*Table A-4 Network Parameters (continued)*

| Parameter | Description | Usage |
|---|---|---|
| Domain | Domain name on the network in which the phone operates, used in the creation of the fully qualified domain name (FQDN) for DNS queries. | Alphanumeric string up to 31 characters.<br>• 0—Use the DHCP-provided domain name.<br>• *string*—Use the specified domain name string and overwrite any DHCP-provided domain name.<br>The default setting is 0. |
| DNS1IP | IP address of the primary DNS server. | The address specified overwrites the primary DNS server address supplied by DHCP. Set 0 or 0.0.0.0 if you want the phone to use the address supplied by DHCP.<br>The default setting is 0.<br>**Note** Do not specify a port number. The Cisco Unified IP Phone uses the default DNS port. |
| DNS2IP | IP address of the secondary DNS server. | The address specified overwrites the secondary DNS server address supplied by DHCP. Set 0 or 0.0.0.0 if you want the phone to use the address supplied by DHCP.<br>The default setting is 0.<br>**Note** Do not specify a port number. The Cisco Unified IP Phone uses the default DNS port. |
| NTPIP | IP address of the primary NTP server. DHCP may also supply a NTP server but NTPIP, if specified, overwrites that value. | Do not specify a port; the phone uses the default NTP port only.<br>If DHCP is used, a nonzero values of NTPIP overwrites any DHCP-supplied addresses and values of 0 tell the phone to use DHCP-supplied addresses.<br>**Value type:** IP address.<br>**Default value:** 0 (use DHCP) |
| AltNTPIP | IP address of a secondary NTP server if redundancy is desired. | If only one NTP server exists, set this to 0 or to the address of the primary server. Do not specify a port; the phone uses the default NTP port only.<br>**Value type:** IP address.<br>**Default value:** 0 (use DHCP) |

*Table A-4 Network Parameters (continued)*

| Parameter | Description | Usage |
|---|---|---|
| TimeZone | Offset to apply to the Greenwich Mean Time (GMT) returned by an NTP server to determine local time (to use for Call ID display, for example). | Local time is generated as follows:<br>• Local time = GMT + TimeZone, if TimeZone <= 12.<br>• Local time = GMT + TimeZone – 25, if TimeZone > 12.<br>For example, TimeZone = 17 for Pacific Standard Time.<br>**Note** You must update this parameter manually when daylight savings time goes into effect and ends.<br>**Value type:** Integer<br>**Default value:** 17 |

*Table A-4 Network Parameters (continued)*

| Parameter | Description | Usage |
|---|---|---|
| OpFlags | Enables or disables various features on the phone. | Bit map, as follows:<br>• Bit 0—If 1, ignore the DHCP-assigned TFTP file name and use the following file name when using TFTP for configuring: ld*xxxxxxxxxxxx* (for Cisco Unified IP Phone 7905G), and gk*xxxxxxxxxxxx* (for Cisco Unified IP Phone 7912G), where each *xx* is the two-digit lowercase hexadecimal representation of each integer in the phone's MAC address.<br>• Bit 1—If the phone is configured to use static IP (that is, the router and gateway addresses are statically assigned), set to 1 to prevent probing at boot time. Otherwise, the phone probes the route/gateway address to determine if there is network connectivity.<br>• Bit 2—Reserved for future use.<br>• Bit 3—If 1, do not request DHCP option 150 in the DHCP DISCOVERY message.<br>• Bit 4—If 1, assume operation under VLAN. (The VLAN ID is specified in the VLANSetting parameter.)<br>• Bit 5—If 1, turn off VLAN IP encapsulation.<br>• Bit 6—If 1, do not perform CDP[1] discovery.<br>• Bit 7—If 1, do not allow web configuration of the phone.<br>• Bit 8—If 0, allow refreshing the phone through the web. If 1, do not allow refreshing the phone through the web.<br>• Bit 9—If 0, allow resetting the phone through the web. If 1, do not allow resetting the phone through the web.<br>The default value is 0x00000002. |

*Table A-4 Network Parameters (continued)*

| Parameter | Description | Usage |
|---|---|---|
| VLANSetting | Specifies various VLAN settings. | Bit map, as follows:<br>• Bits 0–2—Designate 802.1Q priority for signalling IP packets.<br>• Bits 3–5—Designates 802.1Q priority for audio voice IP packets.<br>• Bits 6–17—Reserved for future use.<br>• Bits 18–29—User-specified 802.1Q VLAN id.<br>• Bits 30–31—Reserved for future use.<br>The default setting is 0x0000002b. |
| TOS | ToS (Type of Service) bits. Specifies the precedence and delay of Audio and Signaling IP packets. Higher values increase routing priority for data packets, resulting in less latency. | • Bits 0-7: ToS value for audio data packets. Range: 0-255; Default: 184<br>• Bits 8-15: ToS value for signaling data packets. Range: 0-255; Default: 104<br>• Bits 16-31: Reserved<br>**Value type:** Bitmap.<br>**Default value:**0x000068b8 |

*Table A-4 Network Parameters (continued)*

| Parameter | Description | Usage |
|---|---|---|
| Nprintf | IP address and port of a server to which Cisco Unified IP Phone troubleshooting messages are sent. | Enter the extended IP address of the server to which troubleshooting messages are sent and collected in a log file.<br><br>The default value is 0, which specifies that no messages are sent.<br><br>**Note** To collect troubleshooting messages, the prserv.exe tool must be running on the server specified with this parameter.<br><br>For more information about using Nprintf and preserv.exe to collect troubleshooting information, see the "Logging Information for Troubleshooting" section on page 7-12. |
| TraceFlags | Enables specific trace features when Nprintf is set to a valid host address and port. | • 0=Disable debug messages.<br>• 1=Enable debug messages.<br><br>**Value type:** Bitmap.<br><br>**Default value:** 0x00000000 |

1. CDP = Cisco Discovery Protocol

# SIP Parameters

Table A-5 describes the SIP parameters that you can view through a phone's profile or through the SIP Parameters area on the phone's web page.

**Tip** To access the SIP Parameters area directly from a web browser, enter *IP_address*/**SIPConfiguration**, where *IP_address* is the IP address of your Cisco Unified IP Phone.

*Table A-5 SIP Parameters*

| Parameter | Description |
|---|---|
| UID, PWD | User ID and password for registration and authentication. Authentication can be performed on the (UID, PWD) pair or on the (LoginID, PWD) pair depending on the UseLoginID flag. However, the phone identifies itself to the outside world with the UID only and can be reached only with the corresponding UID but not the LoginID. If UID is set to "." or to "0" (zero), the phone is disabled (does not register with the proxy server).<br>**Value type:** Alphanumeric string.<br>**Default value:** UID=123, PWD=0 |
| Proxy 1-4 | SIP proxy server.<br>SIP Proxy 1 and SIP Proxy 2 accept either the IP address or URL of the SIP proxy server used by the phone.<br>SIP Proxy 3 and SIP Proxy 4 accept only the IP address used by the phone.<br>**Value type:** Alphanumeric string.<br>**Default value:** 0 |
| UseLoginID | Specifies which value to use for authentication:<br>• 0=Use UID for authentication.<br>• 1=Use LoginID for authentication.<br>**Value type:** Boolean.<br>**Default value:** 0 |
| LoginID | Alternate user name used for authentication.<br>**Value type:** Alphanumeric string.<br>**Default value:** 0 |
| SIPRegInterval | SIP registration interval, in seconds, between each registration renewal to the SIP proxy server. If set to 0, the phone uses the default value.<br>The value can range from 0 to 86400.<br>**Value type:** Integer.<br>**Default value:** 3600 |

*Table A-5 SIP Parameters (continued)*

| Parameter | Description |
|---|---|
| SIPRegInterval2 | SIP registration interval, in seconds, between each registration renewal to the backup SIP proxy server. If set to 0, the phone uses the default value.<br>The value can range from 0 to 86400.<br>**Value type:** Integer.<br>**Default value:** 3600 |
| MAXRedirect | Maximum number of redirections the phone attempts to reach a callee.<br>The value can range from 0 to 10.<br>**Value type:** Integer.<br>**Default value:** 5 |
| | |
| SIPRegOn | Whether to enable SIP registration. This parameter is ignored if a SIP proxy server is not specified. If set to 1, the phone periodically registers to the SIP proxy server at an interval given by the SIPRegInterval parameter.<br>Valid values are 0 or 1.<br>**Value type:** Boolean.<br>**Default value:** 0 |
| NATIP (not used in Cisco Unified CallManager release 5.0) | Network Address Translation (NAT) WAN IP address where other SIP user agents can communicate with the phone.<br>**Value type:** IP address.<br>**Default value:** 0.0.0.0 |
| SIPPort | Port where the phone listens for incoming requests and sends outgoing requests.<br>Valid values are 5060 to 65535.<br>**Value type:** Integer.<br>**Default value:** 5060 |

*Table A-5 SIP Parameters (continued)*

| Parameter | Description |
|---|---|
| MediaPort | Port from which the phone transmits and receives media streams. This value must be an even number; each connection uses the next available even-numbered port for RTP. 0=Use the default value.<br>This value can range from 0 to 65535.<br>**Value type:** Even integer.<br>**Default value:** 16384 |
| OutBoundProxy(not used in Cisco Unified CallManager release 5.0) | IP address or URL of the outbound proxy server, with or without a port parameter; for example, 209.165.201.30, 209.165.201.30.5060, 209.165.201.30:5061, sip.cisco.com, sip.xyz.cisco.com:5061.<br>• For IP address, the port, if included, can be preceded by a period (.) or a colon (:).<br>• For URL, the port must be preceded by a colon (:).<br>• If no port is specified, the default port 5060 is used.<br>**Value type:** Alphanumeric string, up to 31 characters.<br>**Default value:** 0 |

*Table A-5 SIP Parameters (continued)*

| Parameter | Description |
|---|---|
| MsgRetryLimits | Lets you configure the number of times that a phone retransmits various SIP requests to the current proxy and the number of times that a phone sends responses to specific requests from the SIP user agent.<br><br>When the phone sends a SIP message to the remote SIP user agent, the message does not always reach its destination for various reasons. In this event, the phone retries sending the same message a specified number of times before timing out.<br><br>The number of retries is configurable for the following SIP requests:<br>• REGISTER<br>• INVITE<br>• BYE<br>• CANCEL<br>• REFER<br>• NOTIFY<br>• PRACK<br><br>The number of retries also is configurable for the phone's final response to an INVITE request from the SIP user agent.<br><br>Table A-6 describes how to configure the MsgRetryLimits parameter. For this parameter:<br>• Value type—Bitmap<br>• Default value—0x00000000 |
| NatServer(not used in Cisco Unified CallManager release 5.0) | Specifies a server to which a dummy, single-byte UDP packed is sent to maintain a Network Address Translation (NAT) during a session.<br><br>This parameter can contain up to 47 characters in fully qualified domain name (FQDN) or IP format with an optional port parameter (separated from the address by a colon); for example, xyz.cisco.com;1234. If no port is specified, the phone uses port 5060.<br><br>**Value type:** IP address, up to 47 characters.<br><br>**Default value:** 0.0.0.0 (port 5060 will be used) |

*Table A-5 SIP Parameters (continued)*

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>NatTimer (not used in Cisco Unified CallManager release 5.0)</td><td>Retransmission interval (in seconds) for sending a dummy packet to the server specified with the NatServer parameter, specified in bits 0-11 the parameter. The upper 20 bits are reserved and should be set to 0.<br><br>Value type: Bitmap.<br><br>Default value: 0x00000000 (no dummy packets will be sent to NatServer)</td></tr>
<tr><td>DialPlan</td><td>Dial plan rules. No syntax check is performed by the implementation. The administrator must ensure that the dial plan is syntactically valid.<br><br>Value type: Alphanumeric string, up to 199 characters.<br><br>Default value: *St4-|#St4-|911|1>#t8.r9t2-|0>#t811.rat4-|^1t4>#.-</td></tr>
<tr><td>IPDialPlan (not used in Cisco Unified CallManager release 5.0)</td><td>Allows for detection of IP-like destination addresses in the dial plan:<br>• 1=If two periods (.) are detected, assume that this is an IP address.<br>• 2=If three periods (.) are detected, assume that this is an IP address.<br><br>Value type: Integer.<br><br>Default value: 1</td></tr>
</table>

*Table A-6 MsgRetryLimits Parameter*

| Bit Number | Definition |
|---|---|
| 0 - 3 | Number of times to retransmit the following SIP requests or responses to the SIP user agent:<br>• NOTIFY request<br>• PRACK request<br>• Final response to an INVITE request<br>Range: 0 - 15<br>Default value: 0<br>With the default setting, the following retransmission attempts are used:<br>• NOTIFY request: 6<br>• PRACK request: 5<br>• INVITE final response: 7 |
| 4 - 7 | Number of times to retransmit REGISTER request.<br>Default value: 0 (number of attempts is 10). |
| 8 - 11 | Number of times to retransmit INVITE request.<br>Default value: 0 (number of attempts is 2). |
| 12 - 15 | Number of times to retransmit BYE request.<br>Default value: 0 (number of attempts is 4). |
| 16 - 19 | Number of times to retransmit CANCEL request.<br>Default value: 0 (number of attempts is 4). |
| 20 - 23 | Number of times to retransmit REFER request.<br>Default value: 0 (number of attempts is 5). |
| 24 - 31 | Reserved. The value of these bits must be 0. |

# Call Preference Parameters

Table A-7 describes the call preference parameters that you can view through a phone's profile or through the Call Preferences area on the phone's web page. These parameters specify various user features on the phone.

**Tip** To access the Call Preferences area directly from a web browser, enter *IP_address*/**CallPrefConfiguration**, where *IP_address* is the IP address of your Cisco Unified IP Phone.

***Table A-7 Call Preference Parameters***

| Parameter | Description |
|---|---|
| Allow Call Waiting (on web page)<br><br>CallWaiting (in profile) | Whether the call waiting feature is enabled on a phone.<br>• 0=Disable the call waiting feature.<br>• 1=Enable the call waiting feature.<br>**Value type:** Boolean.<br>**Default value:** 1 |
| Allow Call Transfer (on web page)<br><br>AttendedTransfer (in profile) | Whether the call transfer feature is enabled on a phone. If this feature is enabled, the **Trnsfer** softkey will appear when appropriate, allowing a user to transfer a call. If this feature is disabled, this softkey will never be available.<br>• 0=Disable the call transfer feature.<br>• 1=Enable the call transfer feature.<br>**Value type:** Boolean.<br>**Default value:** 1 |
| Allow Blind Transfer (on web page)<br><br>BlindTransfer (in profile) | Whether the blind call transfer feature is enabled on a phone. If this feature is enabled, the **BlndXfr** softkey will appear when appropriate, allowing a blind transfer of a call. If this feature is disabled, this softkey will never be available.<br>• 0=Disable the blind call transfer feature.<br>• 1=Enable the blind call transfer feature.<br>**Value type:** Boolean.<br>**Default value:** 1 |

*Table A-7 Call Preference Parameters (continued)*

| Parameter | Description |
|---|---|
| Allow Conference (on web page)<br><br>Conference (in profile) | Whether the conference feature is enabled on a phone. If this feature is enabled, the **Confrn** softkey will appear when appropriate, allowing a three-way conference to be established. If this feature is disabled, this softkey will never be available.<br>• 0=Disable the conference feature.<br>• 1=Enable the conference feature.<br>**Value type:** Boolean.<br>**Default value:** 1 |
| Block Caller ID (on web page)<br><br>BlockCallerId (in profile) | Whether a phone's caller ID information, which includes the value specified by the Display Name parameter and the phone number, is sent with an outgoing call. If this parameter is enabled, outgoing calls will use "Anonymous" as the phone's caller ID.<br>• 0=Do not block outgoing caller ID information.<br>• 1=Block outgoing caller ID information.<br>**Value type:** Boolean.<br>**Default value:** 0 |
| Block Anonymous Calls (on web page)<br><br>BlockAnonymous (in profile) | Whether a phone rejects (blocks) an incoming call with "Anonymous" caller ID will be rejected.<br>• 0=Do not block anonymous incoming calls.<br>• 1=Block anonymous incoming calls.<br>**Value type:** Boolean.<br>**Default value:** 0 |
| Do Not Disturb (on web page)<br><br>DoNotDisturb (in profile) | Whether the Do Not Disturb feature is enabled on a phone. If this feature is enabled, calls to the phone will receive a busy signal.<br>• 0=Disable the do not disturb feature.<br>• 1=Enable the do not disturb feature.<br>**Value type:** Boolean.<br>**Default value:** 0 |

*Table A-7 Call Preference Parameters (continued)*

| Parameter | Description |
|---|---|
| Voice Mail Number (on web page)<br><br>VoiceMailNumber (in profile) | Specifies a telephone number where voice messages are stored and retrieved. If a number (other than 0) is specified:<br>• The **Message** softkey will appear on the phone when appropriate.<br>• Incoming calls forward to this number if the phone is busy or if the calls are not answered within the number of seconds specified with the Forward to VMail Delay(s) parameter.<br><br>**Value type:** Alphanumeric string, up to 31 characters.<br><br>**Default value:** 0 (disables this feature) |
| Call Forward Number (on web page)<br><br>CallForwardNumber (in profile) | Specifies a telephone number to which all calls to the phone are forwarded. If a number (other than 0) is specified, the Call Forward All feature is activated.<br><br>**Value type:** Alphanumeric string, up to 31 characters.<br><br>**Default value:** 0 (disables call forwarding) |
| Call Forward on Busy Number (on web page)<br><br>CallForwardOnBusy Number (in profile) | Specifies a telephone number to which calls to the phone are forwarded if the phone is busy. If a number (other than 0) is specified, the Call Forward On Busy feature is activated.<br><br>**Value type:** Alphanumeric string, up to 31 characters.<br><br>**Default value:** 0 (disables call forwarding on busy) |
| Display Name (on web page)<br><br>DisplayName (in profile) | Specifies a name to be used as part of the phone's caller ID and that will be displayed on the phone's LCD screen. A 0 (zero) or a blank value disables this feature.<br><br>**Note** If the Block Caller ID parameter is set to Yes, outgoing calls will use "Anonymous" as the phone's caller ID regardless of the Display Name setting.<br><br>**Value type:** Alphanumeric string, up to 31 characters.<br><br>**Default value:** blank (no character string) |

*Table A-7 Call Preference Parameters (continued)*

| Parameter | Description |
|---|---|
| Short Name (on web page)<br><br>ShortName (in profile) | Specifies a name to be displayed on the phone's LCD screen in place of the Display Name value. Short Name will not affect the Display Name value that is used as part of the phone's caller ID.<br><br>**Value type:** Alphanumeric string, up to 31 characters.<br><br>**Default value:** 0 (causes the Display Name value to be displayed) |
| Time Format (on web page)<br><br>TimeFormat (in profile) | Specifies the format for the time that appears on the phone's LCD screen. This format is specified by one or more of the following characters:<br><br>h—designates the hour in 12-hour format.<br><br>H—designates the hour in 24-hour format.<br><br>i or I—designates minutes.<br><br>a or A—for 12-hour format, include "p" with times from noon until one minute before midnight.<br><br>: (colon)—displays a colon that blinks every second.<br><br>For example, when the current time is 1:30 P.M.,<br><br>• h:ia causes the time to appear as 1:30 p.<br><br>• H:I causes the time to appear as 13:30.<br><br>**Value type:** Alphanumeric string, up to 15 characters.<br><br>**Default value:** h:ia |

*Table A-7 Call Preference Parameters (continued)*

| Parameter | Description |
|---|---|
| Date Format (on web page)<br><br>DateFormat (in profile) | Specifies the format for the date that appears on the phone's LCD screen. This format is specified by one or more of the following characters:<br><br>m—designates the month as a number 1 through 12.<br><br>M—designates the month as a three-letter abbreviation, Jan through Dec.<br><br>d or D—designates the day of the month.<br><br>y—designates a two-digit year (such as 03 for 2003).<br><br>Y—designates a four-digit year.<br><br>Other characters—appear as entered.<br><br>For example, when the current date is March 10, 2003:<br>• m-d-y causes the date to appear as 3-10-03.<br>• M d, Y causes the date to appear as Mar 20, 2003.<br>• Y/m/d causes the date to appear as 2003/3/10.<br><br>**Value type:** Alphanumeric string, up to 15 characters.<br><br>**Default value:** m-d-y |
| Forward to VMail Delay (on web page)<br><br>ForwardToVMDelay (in profile) | Specifies number of seconds after which an incoming call ringing at the phone will be forwarded to the telephone number specified by the Voice Mail Number parameter.<br><br>Valid values are from 0 to 2147483647.<br><br>**Value type:** Integer<br><br>**Default value:** 20 (seconds) |

*Table A-7 Call Preference Parameters (continued)*

| Parameter | Description |
|---|---|
| GUI Show Mask (on web page)<br><br>CallPrefGuiShow (in profile) | Specifies which parameters, if any, appear on the Call Preferences menu on the phone.<br><br>Set a bit to 0 if the corresponding parameter should not appear. Set the bit to 1 if the parameter should appear.<br>• Bit 0: Do Not Disturb.<br>• Bit 1: Allow Call Waiting.<br>• Bit 2: Block Caller ID.<br>• Bit 3: Call Forward Number.<br>• Bits 4–5: Reserved.<br>• Bit 6: Display Name.<br>• Bit 7: Time Format.<br>• Bit 8: Date Format.<br>• Bit 9: Voice Mail Number.<br>• Bit 10: Allow Call Transfer.<br>• Bit 11: Allow Attended Transfer.<br>• Bit 12: Allow Conference.<br>• Bit 13: Short Name.<br>• Bits 14–23: Reserved.<br>• Bit 24: Block Anonymous Call.<br>• Bit 25: Reserved.<br>• Bit 26: Forward to Voice Mail Delay.<br>• Bit 27 and bits 29–31: Reserved.<br><br>In addition, bit 28, if set to 1, causes a registration status icon to appear on the phone’s LCD screen. This icon indicates whether the phone is registered to the SIP proxy server.<br><br>**Value type:** Bitmap.<br><br>**Default value:** 0xFFFFFFFF |

*Table A-7 Call Preference Parameters (continued)*

| Parameter | Description |
|---|---|
| GUI Set Mask (on web page)<br>CallPrefGuiSet (in profile) | Specifies which parameters on the Call Preferences menu on the phone can be changed by an end user.<br>Set a bit to 0 if an end user should not be able to change the corresponding parameter. Set the bit to 1 if an end user should be able to change the parameter.<br>• Bit 0: Do Not Disturb.<br>• Bit 1: Allow Call Waiting.<br>• Bit 2: Block Caller ID.<br>• Bit 3: Call Forward Number.<br>• Bits 4–5: Reserved.<br>• Bit 6: Display Name.<br>• Bit 7: Time Format.<br>• Bit 8: Date Format.<br>• Bit 9: Voice Mail Number.<br>• Bit 10: Allow Call Transfer.<br>• Bit 11: Allow Blind Transfer.<br>• Bit 12: Allow Conference.<br>• Bit 13: Short Name.<br>• Bit 24: Block Anonymous Call.<br>• Bit 25: Reserved.<br>• Bit 26: Forward to Voice Mail Delay.<br>**Value type:** Bitmap.<br>**Default value:** 0xFFFFFFFFparameters:network<$startrange> |

# Tone Parameters

Table A-8 describes the tone parameters that you can configure through a phone's profile file or through its Tone Parameters web page.

**Tip** To access the Call Preferences area directly from a web browser, enter *IP_address*/**ToneConfiguration**, where *IP_address* is the IP address of your Cisco Unified IP Phone.

*Table A-8 Tone Parameters*

| Parameter | Description |
|---|---|
| SigTimer | Timeout values for signal events.<br>• Bits 0–7: CWT Period, which the number of 0.1-second intervals to wait between each burst of call-waiting tone. The range is 0 to 255. The default value is 100 (0x64, 10 seconds). 0=Use default.<br>• Bits 8–13: Reserved.<br>• Bits 14–19: Ring Timeout, which is the number of 10-second intervals to wait between when the phone starts ringing and the phone rejects the incoming call. Range: 0 to 63. 0=Never time out. Default value: 6 (0x6, 60 seconds).<br>• Bits 20–25: NoAns Timeout, which is the number of seconds to wait between when the phone starts ringing and the phone initiates call forwarding on no answer. Range: 0 to 63. Default value: 20 (0x14, 20 seconds).<br>• Bits 28–29: First Key Repeat Interval, which specifies how long a user must hold the **Volume** or the **Navigation** button up or down before the desired change begins to repeat. Valid values: 0 (1 second), 1 (disable repeat), 2 (2 seconds), and 3 (3 seconds). Default value: 0 (1 second).<br>• Bits 30–31: Subsequent Key Repeat Interval, which specifies the intervals at which a change continues to repeat (after the First Key Repeat Interval) when a user continues to hold the **Volume** or the **Navigation** button up or down. Valid values: 0 (0.25 second), 1 (0.5 second), 2 (0.75 second), and 3 (1 second). Default value: 0 (0.25 second).<br>**Value type:** Bitmap.<br>**Default value:** 0x01418064 |

*Table A-8 Tone Parameters (continued)*

| Parameter | Description |
|---|---|
| RingOnOffTime | Specifies the ringer cadence pattern, expressed as three comma-separated integers *a*,*b*,*c*, where:<br>• *a*—Number of seconds to wait before turning the ring on.<br>• *b*—Number of seconds to wait before turning the ring off.<br>• *c*—Ring frequency.<br>**Value type:** Three comma-separated integers.<br>**Default value:** 2,4,25 |
| DialTone | Tone that plays when the phone is ready to accept the first digit of a telephone number or an IP address.<br>**Value type:** Array of 11 short integers.<br>**Default value:** 2,31538,814,30831,2032,0,0,0,0,0,0 |
| DialTone2 | Secondary dial tone. For example, this tone might play after you dial a number to obtain an outside line.<br>**Value type:** Array of 11 short integers.<br>**Default value:** 2,30743,1384,29864,1252,0,0,0,0,0,0 |
| BusyTone | Tone that plays when the called party's line is busy.<br>**Value type:** Array of 11 short integers.<br>**Default value:** 2,30467,1104,28959,1404,1,4000,4000,0,0,0 |
| ReorderTone | Tone that plays when the called number does not exist or when the external network circuit is busy.<br>**Value type:** Array of 17 short integers.<br>**Default value:** 0,2,30467,1104,28959,1404,0,0,1,2000,2000,0,0,0,0,0,0 |

*Table A-8 Tone Parameters (continued)*

| Parameter | Description |
|---|---|
| RingBackTone | Tone that plays when the called party's line is ringing.<br>**Value type:** Array of 11 short integers.<br>**Default value:** 2,30831,2032,30467,1104,1,16000,32000,0,0,0 |
| CallWaitTone | Tone that plays to indicate that you have a call waiting.<br>**Value type:** Array of 11 short integers.<br>**Default value:** 1,30831,2412,0,0,1,2400,2400,0,0,4800 |

# Audio Parameters

Table A-9 describes the network parameters that you can configure through a phone's profile file or through its Audio Parameters web page.

**Tip** To access the Call Preferences area directly from a web browser, enter *IP_address*/**AudioConfiguration**, where *IP_address* is the IP address of your Cisco Unified IP Phone.

***Table A-9 Audio Parameters***

| Parameter | Description |
|---|---|
| RxCodec | Preferred audio decoder (receiving codec):<br>• 1: G.711A-law<br>• 2: G.711u-law<br>• 3: G.729a<br>**Value type:** Integer.<br>**Default value:** 2 |
| TxCodec | Preferred audio encoder (transmitting codec):<br>• 1=G.711A-law<br>• 2=G.711u-law<br>• 3=G.729a<br>**Value type:** Integer.<br>**Default value:** 2 |

*Table A-9 Audio Parameters (continued)*

| Parameter | Description |
|---|---|
| AudioMode | Audio operating mode.<br>• Bit 0: G.711 silence suppression:<br>– 0=Disable.<br>– 1=Enable.<br>• Bits 1–3: Reserved.<br>• Bits 4–5: DTMF transmission method:<br>– 0=Always inband.<br>– 1=Negotiated via SDP.<br>– 2=Always out-of-band.<br>• Bits 6–31: Reserved.<br>**Value type:** Bitmap.<br>**Default value:** 0x00000011 |
| ConnectMode | Connection mode for the selected call-signaling protocol.<br>See Table A-10 on page A-34 for syntax and usage details.<br>**Value type:** Bitmap.<br>**Default value:** 0x00000000 |
| NumTxFrames | Number of frames per outbound audio RTP packet. For G.711 and G.729 codecs, a frame is 10 ms. Cisco recommends that you use the default value.<br>Valid values are 1, 2, 3, 4, 5, or 6.<br>**Value type:** Integer.<br>**Default value:** 2 |

*Table A-10 ConnectMode Parameter*

| Bit | Description |
|---|---|
| 0–15 | Reserved. Must be set to 0. |
| 16 | Registration removal prior to re-registration.<br>• 0=Disable (default).<br>• 1=When the phone powers up, "Contact: *" is used to remove all registrations. On subsequent registration cycles, "Contact: *current_SIP_URL*;expires=0" is used. |
| 17–18 | Reserved. Must be set to 0. |
| 19 | IP ringback and early media.<br>• 0=Do not send a ringback tone to the caller (default).<br>• 1=Send a ring back tone to the caller. |
| 20 | Include "action=proxy" in REGISTER request. Do not enable if bit 21 is enabled.<br>• 0=Disable (default).<br>• 1=Enable. |
| 21 | Include "action=redirect" in REGISTER request.<br>• 0=Disable (default).<br>• 1=Enable. |

*Table A-10 ConnectMode Parameter (continued)*

| Bit | Description |
|---|---|
| 22 | Process a *received* = parameter in the VIA header to extract the external IP addresses used by the Network Address Translation (NAT) router.<br>• 0=Disable (default).<br>• 1=Enable.<br>When a Cisco Unified IP Phone is operating behind a NAT, the NATIP parameter must be set to the external IP address of the NAT router. This setting allows the correct IP address to be placed in the Contact and SDP headers.<br>You may leave the NATIP address set to the default value of 0 (or 0.0.0.0) and let the phone automatically scan the VIA header for a *received* = parameter. The parameter, if present, would indicate that the phone is operating behind a firewall.<br>The phone proceeds as follows:<br>1. If the *received* = parameter is an INVITE response, the current INVITE is canceled and a new INVITE is sent with the new IP address extracted from the received =*NAT_IP_Address* parameter in the Contact header. This step is performed only if registration is currently in an idle state.<br>2. If the *received* = parameter is in a REGISTER response as a result of a REGISTER command, the phone will cancel all previous registrations and re-register with the new IP address extracted from the received = *NAT_IP_Address* parameter in the Contact header.<br>**Note** For the phone to automatically detect its presence behind a NAT, the SIP proxy server or remote user agent server must include the *received* = parameter in the VIA header in the responses to the phone if the proxy detects that the source address and port do not match those in the VIA header. |
| 23 | Reserved. Must be set to 0. |

*Table A-10 ConnectMode Parameter (continued)*

| Bit | Description |
|---|---|
| 24 | Include RTP statistics in BYE request and response.<br>• 0=Disable (default).<br>• 1=Enable.<br>If this bit is enabled, the phone will insert the headers RxStat and TxStat as follows:<br>RxStat: *Dur=a,Pkt=b,Oct=c,LatePkt=d,LostPkt=e,AvgJit=f*<br>TxStat: *Dur=g,Pkt=h,Oct=i*<br>where:<br>• *Dur* is the total number of seconds since the beginning of reception or transmission<br>• *Pkt* is the total number of RTP packets received or transmitted<br>• *Oct* is the total number of RTP payload octets received or transmitted (not including RTP header)<br>• *LatePkt* is the total number of late RTP packets received<br>• *LostPkt* is the total number of lost RTP packets received (not including late RTP packets)<br>• *AvgJit* is the average jitter, which is an estimate of the statistical variance of the RTP packet inter-arrival time, measured in timestamp unit and calculated according to RFC 1889.<br>• *a*, *b*, *c*, *d*, *e*, *f*, *g*, *h*, and *i* are integers |
| 25–31 | Reserved. Must be set to 0. |