Monitoring Phone Systems

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Monitoring Phone Systems Overview

You can view a variety of information about the phone using the phone status menu on the phone and the phone web pages. This information includes:

- Device information
- Network setup information
- Network statistics
- Device logs
- Streaming statistics

This chapter describes the information that you can obtain from the phone web page. You can use this information to remotely monitor the operation of a phone and to assist with troubleshooting.

Related Topics

Troubleshooting

Cisco IP Phone Status

The following sections describes how to view model information, status messages, and network statistics on the Cisco IP Phone.

- Model Information: Displays hardware and software information about the phone.
- Status menu: Provides access to screens that display the status messages, network statistics, and statistics for the current call.

You can use the information that displays on these screens to monitor the operation of a phone and to assist with troubleshooting.

You can also obtain much of this information, and obtain other related information, remotely through the phone web page.

Display the Phone Information Window

Procedure

Step 1 Press **Settings** softkey.

Step 2 Select **Status** > **Product Information**.
When a user password is set, a corresponding icon (lock or certificate) displays at the top-right corner of the phone screen.

Step 3 To exit the Model Information screen, press **Exit**.

View the Phone Status

Procedure

Step 1 Press **Settings** softkey.

Step 2 Select **Status** > **Phone Status**.
You can view the following information:

- **Elapsed time—**Total time elapsed since the last reboot of the system
- **Tx (Packets)—**Transmitted packets from the phone.
- **Rx (Packets)—**Received packets from the phone.

View the Status Messages on the Phone

Procedure

Step 1 Press **Settings** softkey.

Step 2 Select **Information and settings** > **Status** > **Status messages**.
You can view a log of the various phone statuses since provisioning was last done.

Note Status messages reflect UTC time and are not affected by the timezone settings on the phone.

Step 3 Press **Back**.

View the Network Status

Procedure

Step 1 Press **Settings** softkey.

Step 2 Select **Status** > **Network Status**.
You can view the following information:

- **Network type—**Indicates the type of Local Area Netwrok (LAN) connection that the phone uses.
- **Network status—**Indicates if the phone is connected to a network.
- **IP address—**IP address of the phone.
- **VLAN ID—**VLAN ID of the phone.
- **Addressing type—**Indicates if the phone has DHCP or Static IP enabled.
- **IP status—**Status of IP that the phone uses.
- **Subnet mask—**Subnet mask used by the phone.
- **Default router—**Default router used by the phone.
- **DNS 1—**Primary Domain Name System (DNS) server that the phone uses.
- **DNS 2—**Optional Backup DNS server that the phone uses.
- **MAC address—**Unique Media Access Control (MAC) address of the phone.
- **Host name—**Displays the current host name assigned to the phone.
- **Domain—**Displays the network domain name of the phone. Default: cisco.com
- **Switch port link—**Status of the switch port.
- **Switch port config—**Indicates speed and duplex of the network port.

Display Call Statistics Window

You can access the Call Statistics screen on the phone to display counters, statistics, and voice-quality metrics of the most recent call.

Note You can also remotely view the call statistics information by using a web browser to access the Streaming Statistics web page. This web page contains additional RTCP statistics that are not available on the phone.

A single call can use multiple voice streams, but data is captured for only the last voice stream. A voice stream is a packet stream between two endpoints. If one endpoint is put on hold, the voice stream stops even though the call is still connected. When the call resumes, a new voice packet stream begins, and the new call data overwrites the former call data.

To display the Call Statistics screen for information about the latest voice stream, follow these steps:

Procedure

Step 1 Press **Settings** softkey.

Step 2 Select **Status** > **Phone Status** > **Call Statistics**.

Step 3 To exit the Status menu, press **Back**.

Call Statistics Fields

The following table describes the items on the Call Statistics screen.

Table 1: Call Statistics Items for the Cisco IP Phone

Item	Description
Receiver Codec	Type of received voice stream (RTP streaming audio from codec): G.729, G.722, G.711 mu-law, G.711 A-law, OPUS, and iLBC.
Sender Codec	Type of transmitted voice stream (RTP streaming audio from codec): G.729, G.722, G.711 mu-law, G.711 A-law, OPUS, and iLBC.
Receiver Size	Size of voice packets, in milliseconds, in the receiving voice stream (RTP streaming audio).
Sender Size	Size of voice packets, in milliseconds, in the transmitting voice stream.
Rcvr Packets	Number of RTP voice packets that were received since voice stream opened. **Note** This number is not necessarily identical to the number of RTP voice packets that were received since the call began because the call might have been placed on hold.
Sender Packets	Number of RTP voice packets that were transmitted since voice stream opened. **Note** This number is not necessarily identical to the number of RTP voice packets that were transmitted since the call began because the call might have been placed on hold.

Item	Description
Avg Jitter	Estimated average RTP packet jitter (dynamic delay that a packet encounters when going through the network), in milliseconds, that was observed since the receiving voice stream opened.
Max Jitter	Maximum jitter, in milliseconds, that was observed since the receiving voice stream opened.
Receiver Discarded	Number of RTP packets in the receiving voice stream that were discarded (bad packets, too late, and so on). **Note** The phone discards payload type 19 comfort noise packets that Cisco Gateways generate, because they increment this counter.
Rcvr Lost Packets	Missing RTP packets (lost in transit).
Voice-Quality Metrics	
Cumulative Conceal Ratio	Total number of concealment frames divided by total number of speech frames that were received from start of the voice stream.
Interval Conceal Ratio	Ratio of concealment frames to speech frames in preceding 3-second interval of active speech. If using voice activity detection (VAD), a longer interval might be required to accumulate 3 seconds of active speech.
Max Conceal Ratio	Highest interval concealment ratio from start of the voice stream.
Conceal Seconds	Number of seconds that have concealment events (lost frames) from the start of the voice stream (includes severely concealed seconds).
Severely Conceal Seconds	Number of seconds that have more than 5 percent concealment events (lost frames) from the start of the voice stream.
Latency	Estimate of the network latency, expressed in milliseconds. Represents a running average of the round-trip delay, measured when RTCP receiver report blocks are received.

Cisco IP Phone Web Page

This section describes the information that you can obtain from the phone web page. You can use this information to remotely monitor the operation of a phone and to assist with troubleshooting.

Related Topics

Access the Web-Based Configuration Utility
Determine the IP Address of the Phone
Allow Web Access to the Cisco IP Phone

Info

The fields on this tab are read-only and cannot be edited.

Status

System Information

Parameter	Description
Host Name	Displays the current host name assigned to the phone.
Domain	Displays the network domain name of the phone. Default: cisco.com
Primary NTP Server	Displays the primary NTP server assigned to the phone.
Secondary NTP Serve	Displays the secondary NTP server assigned to the phone.

IPv4 Information

Parameter	Description
IP Status	Indicates that the connection is established.
Connection Type	Indicates the type of internet connection for the phone: • DHCP • Static IP
Current IP	Displays the current IP address assigned to the IP phone.
Current Netmask	Displays the network mask assigned to the phone.
Current Gateway	Displays the default router assigned to the phone.
Primary DNS	Displays the primary DNS server assigned to the phone.
Secondary DNS	Displays the secondary DNS server assigned to the phone.

Reboot History

For information about Reboot History, see the Reboot Reasons.

Product Information

Parameter	Description
Product Name	Model number of the Cisco IP Phone.
Software Version	Version number of the Cisco IP Phone firmware.
MAC Address	Hardware address of the Cisco IP Phone.
Customization	For an RC unit, this field indicates whether the unit has been customized or not. Pending indicates a new RC unit that is ready for provisioning. If the unit has already retrieved its customized profile, this field displays the name of the company that provisioned the unit.
Serial Number	Serial number of the Cisco IP Phone.
Hardware Version	Version number of the Cisco IP Phone hardware.
Client Certificate	Status of the client certificate, which authenticates the Cisco IP Phone for use in the ITSP network. This field indicates if the client certificate is properly installed in the phone.

Downloaded Locale Package

Parameter	Description
Download Status	Displays the downloaded locale package status.
Download URL	Displays the location from where the local package is downloaded.

Phone Status

Parameter	Description
Current Time	Current date and time of the system; for example, 08/06/14 1:42:56 a.m.
Elapsed Time	Total time elapsed since the last reboot of the system; for example, 7 days, 02:13:02.
SIP Messages Sent	Total number of SIP messages sent (including retransmissions).
SIP Bytes Sent	Total number of SIP messages received (including retransmissions).
SIP Messages Recv	Total number of bytes of SIP messages sent which includes retransmissions.

Parameter	Description
SIP Bytes Recv	Total number of bytes of SIP messages received (including retransmissions).
Network Packets Sent	Total number of network packets sent.
Network Packets Recv	Total number of network packets received.
External IP	External IP of the phone.
Operational VLAN ID	ID of the VLAN currently in use if applicable.
SW Port	Displays the type of Ethernet connection from the IP phone to the switch.
Upgrade Status	Displays status of the last phone upgrade.
SW Port Config	Displays the type of SW port configuration.
Last Successful Login	Displays the time when the phone has last successful log in.
Last Failed Login	Displays the time when the phone has last failed log in.

Dot1x Authentication

Parameter	Description
Transaction status	Indicates if the phone is authenticated.
Protocol	Displays the protocol of the registered phone.

Ext Status

Parameter	Description
Registration State	Shows “Registered” if the phone is registered, or “Not Registered” if the phone is not registered to the ITSP.
Last Registration At	Last date and time the line was registered.
Next Registration In Seconds	Number of seconds before the next registration renewal.
Message Waiting	Indicates whether message waiting is enabled or disabled.
Mapped SIP Port	Port number of the SIP port mapped by NAT.
Hoteling State	Indicates whether Hoteling is enabled or disabled.

Parameter	Description
Extended Function Status	Indicates whether extended function is enabled.

Line Call Status

Parameter	Description
Call State	Status of the call.
Tone	Type of tone that the call uses.
Encoder	Codec used for encoding.
Decoder	Codec used for decoding.
Type	Direction of the call.
Remote Hold	Indicates whether the far end placed the call on hold.
Callback	Indicates whether the call was triggered by a call back request.
Mapped RTP Port	The port mapped for Real Time Protocol traffic for the call.
Peer Name	Name of the internal phone.
Peer Phone	Phone number of the internal phone.
Duration	Duration of the call.
Packets Sent	Number of packets sent.
Packets Recv	Number of packets received.
Bytes Sent	Number of bytes sent.
Bytes Recv	Number of bytes received.
Decode Latency	Number of milliseconds for decoder latency.
Jitter	Number of milliseconds for receiver jitter.
Round Trip Delay	Number of milliseconds for delay in the RTP-to-RTP interface round trip.
Packets Lost	Number of packets lost.
Loss Rate	The fraction of RTP data packets from the source lost since the beginning of reception. Defined in RFC-3611—RTP Control Protocol Extended Reports (RTCP XR).

Parameter	Description
Packet Discarded	The fraction of RTP data packets from the source lost since the beginning of reception. Defined in RFC-3611—RTP Control Protocol Extended Reports (RTCP XR).
Discard Rate	The fraction of RTP data packets from the source that have been discarded since the beginning of reception, due to late or early arrival, under-run or overflow at the receiving jitter buffer. Defined in RFC-3611—RTP Control Protocol Extended Reports (RTCP XR).
Burst Duration	The mean duration, expressed in milliseconds, of the burst periods that have occurred since the beginning of reception. Defined in RFC-3611—RTP Control Protocol Extended Reports (RTCP XR).
Gap Duration	The mean duration, expressed in milliseconds, of the gap periods that have occurred since the beginning of reception. Defined in RFC-3611—RTP Control Protocol Extended Reports (RTCP XR).
R-Factor	Voice quality metric that describes the segment of the call that is carried over this RTP session. Defined in RFC-3611—RTP Control Protocol Extended Reports (RTCP XR).
MOS-LQ	The estimated mean opinion score for listening quality (MOS-LQ) is a voice quality metric on a scale from 1 to 5, in which 5 represents excellent and 1 represents unacceptable. Defined in RFC-3611—RTP Control Protocol Extended Reports (RTCP XR).
MOS-CQ	The estimated mean opinion score for conversational quality (MOS-CQ) is defined as including the effects of delay and other effects that affect conversational quality. Defined in RFC-3611—RTP Control Protocol Extended Reports (RTCP XR).

TR-069 Status

Parameter	Description
TR-069 Feature	Indicates if TR-069 function is enabled or disabled.
Periodic Inform Time	Displays the inform time interval from CPE to ACS.
Last Inform Time	Indicates the last inform time.
Last Transaction Status	Displays the success or the failure status.
Last Session	Indicates the start and end time of the session.

Parameter	Description
ParameterKey	Displays the key for reference checkpoint for parameter set configured.

Custom CA Status

These fields display the status of provisioning using a custom Certificate Authority (CA).

Parameter	Description
Custom CA Provisioning Status	Indicates whether provisioning using a custom CA succeeded or failed: • Last provisioning succeeded on mm/dd/yyyy HH:MM:SS; or • Last provisioning failed on mm/dd/yyyy HH:MM:SS
Custom CA Info	Displays information about the custom CA: • Installed—Displays the “CN Value,” where “CN Value” is the value of the CN parameter for the Subject field in the first certificate. • Not Installed—Displays if no custom CA certificate is installed.

Custom CA certificates are configured in the Provisioning tab. For more information about custom CA certificates, see the *Cisco IP Phone 7800 Series and Cisco IP Phone 8800 Series Multiplatform Phones Provisioning Guide*.

Provisioning Status

Parameter	Description
Provisioning Profile	Displays the profile file name of the phone.
Provisioning Status 1	Displays the provisioning status (resync) of the phone.
Provisioning Status 2	
Provisioning Status 3	
Provisioning Failure Reason	Displays the reason for the failure of provisioning of the phone.

Note The Upgrade and Provisioning Status are displayed in reverse chronological order (like reboot history) displaying status with time and reason.

Debug Info

Console Logs

Displays the syslog output of the phone in the reverse order, where messages is the latest one. The display includes hyperlinks to individual log files. The console log files include debug and error messages received on the phone and the time stamp reflects UTC time, regardless of time zone settings.

Parameter	Description
Debug Message 1	messages
Debug Message 2	messages.1
Debug Message 3	messages.2
Debug Message 4	messages.3
Debug Message 5	messages.4
Debug Message 6	messages.5
Debug Message 7	messages.6
Debug Message 8	messages.7

Problem Reports

Parameter	Description
Report Problem	Displays the tab Generate PRT.
Prt file	Displays the file name of the PRT logs.

Attendant Console Status

Attendant Console Status

Parameter	Description
Console Subscribe Expires	Displays the time when subscription of the key expansion module that is added to the phone will expire.
Subscribe Retry Interval	Displays the time when subscription of the key expansion module that is added to the phone will try to subscribe again.

Unit

Enter the programming information for each line key for the Attendant Console unit.

Parameter	Description
Unit Enable	Indicates whether the key expansion module that is added to the phone is enabled.
Unit Online	Indicates whether the key expansion module that is added to the phone is active.
HW Version	Displays the hardware version of the key expansion module that is added to the phone..
SW Version	Displays the software version of the key expansion module that is added to the phone.

Network Statistics

Ethernet Information

Parameter	Description
TxFrames	Total number of packets that the phone transmitted.
TxBroadcasts	Total number of broadcast packets that the phone transmitted.
TxMulticasts	Total number of multicast packets that the phone transmitted.
TxUnicasts	Total number of unicast packets that the phone transmitted.
RxFrames	Total number of packets received by the phone.

Parameter	Description
RxBroadcasts	Total number of broadcast packets that the phone received.
RxMulticasts	Total number of multicast packets that the phone received.
RxUnicasts	Total number of unicast packets that the phone received.

Network Port Information

Parameter	Description
RxtotalPkt	Total number of packets that the phone received.
Rxunicast	Total number of unicast packets that the phone received.
Rxbroadcast	Total number of broadcast packets that the phone received.
Rxmulticast	Total number of multicast packets that the phone received.
RxDropPkts	Total number of packets dropped.
RxUndersizePkts	The total number of packets received that are less than 64 octets long, which excludes framing bits, but includes FCS octets, and are otherwise well formed.
RxOversizePkts	The total number of packets received that are longer than 1518 octets, which excludes framing bits, but includes FCS octets, and are otherwise well formed.
RxJabbers	The total number of packets received that are longer than 1518 octets, which excludes framing bits, but incudes FCS octets, and do not end with an even number of octets (alignment error), or had an FCS error.
RxAlignErr	Total number of packets between 64 and 1522 bytes in length that were received and that had a bad Frame Check Sequence (FCS).
Rxsize64	Total number of received packets, including bad packets, that were between 0 and 64 bytes in size.
Rxsize65to127	Total number of received packets, including bad packets, that were between 65 and 127 bytes in size.
Rxsize128to255	Total number of received packets, including bad packets, that were between 128 and 255 bytes in size.
Rxsize256to511	Total number of received packets, including bad packets, that were between 256 and 511 bytes in size.

Parameter	Description
Rxsize512to1023	Total number of received packets, including bad packets, that were between 512 and 1023 bytes in size.
Rxsize1024to1518	Total number of received packets, including bad packets, that were between 1024 and 1518 bytes in size.
TxtotalGoodPkt	Total number of good packets (multicast, broadcast, and unicast) that the phone received.
lldpFramesOutTotal	Total number of LLDP frames that the phone sent out.
lldpAgeoutsTotal	Total number of LLDP frames that timed out in the cache.
lldpFramesDiscardedTotal	Total number of LLDP frames that were discarded when any of the mandatory TLVs is missing, out of order, or contains out of range string length.
lldpFramesInErrorsTotal	Total number of LLDP frames that were received with one or more detectable errors.
lldpFramesInTotal	Total number of LLDP frames that the phone received.
lldpTLVDiscardedTotal	Total number of LLDP TLVs that were discarded.
lldpTLVUnrecognizedTotal	Total number of LLDP TLVs that were not recognized on the phone.
CDPNeighborDeviceId	Identifier of a device connected to this port that CDP discovered.
CDPNeighborIP	IP address of the neighbor device discovered that CDP discovered.
CDPNeighborPort	Neighbor device port to which the phone is connected discovered by CDP.
LLDPNeighborDeviceId	Identifier of a device connected to this port discovered by LLDP discovered.
LLDPNeighborIP	IP address of the neighbor device that LLDP discovered.
LLDPNeighborPort	Neighbor device port to which the phone connects that LLDP discovered.
PortSpeed	Speed and duplex information.

Access Port Information

Parameter	Description
RxtotalPkt	Total number of packets that the phone received.

Parameter	Description
Rxunicast	Total number of unicast packets that the phone received.
Rxbroadcast	Total number of broadcast packets that the phone received.
Rxmulticast	Total number of multicast packets that the phone received.
RxDropPkts	Total number of packets dropped.
RxUndersizePkts	The total number of packets received that are less than 64 octets long, which excludes framing bits, but includes FCS octets, and are otherwise well formed.
RxOversizePkts	The total number of packets received that are longer than 1518 octets, which excludes framing bits, but includes FCS octets, and are otherwise well formed.
RxJabbers	The total number of packets received that are longer than 1518 octets, which excludes framing bits, but incudes FCS octets, and do not end with an even number of octets (alignment error), or had an FCS error.
RxAlignErr	Total number of packets between 64 and 1522 bytes in length that were received and that had a bad Frame Check Sequence (FCS).
Rxsize64	Total number of received packets, including bad packets, that were between 0 and 64 bytes in size.
Rxsize65to127	Total number of received packets, including bad packets, that were between 65 and 127 bytes in size.
Rxsize128to255	Total number of received packets, including bad packets, that were between 128 and 255 bytes in size.
Rxsize256to511	Total number of received packets, including bad packets, that were between 256 and 511 bytes in size.
Rxsize512to1023	Total number of received packets, including bad packets, that were between 512 and 1023 bytes in size.
Rxsize1024to1518	Total number of received packets, including bad packets, that were between 1024 and 1518 bytes in size.
TxtotalGoodPkt	Total number of good packets (multicast, broadcast, and unicast) that the phone received.
lldpFramesOutTotal	Total number of LLDP frames that the phone sent out.
lldpAgeoutsTotal	Total number of LLDP frames that timed out in the cache.

Parameter	Description
lldpFramesDiscardedTotal	Total number of LLDP frames that were discarded when any of the mandatory TLVs is missing, out of order, or contains out of range string length.
lldpFramesInErrorsTotal	Total number of LLDP frames that were received with one or more detectable errors.
lldpFramesInTotal	Total number of LLDP frames that the phone received.
lldpTLVDiscardedTotal	Total number of LLDP TLVs that were discarded.
lldpTLVUnrecognizedTotal	Total number of LLDP TLVs that were not recognized on the phone.
CDPNeighborDeviceId	Identifier of a device connected to this port that CDP discovered.
CDPNeighborIP	IP address of the neighbor device discovered that CDP discovered.
CDPNeighborPort	Neighbor device port to which the phone is connected discovered by CDP.
LLDPNeighborDeviceId	Identifier of a device connected to this port discovered by LLDP discovered.
LLDPNeighborIP	IP address of the neighbor device that LLDP discovered.
LLDPNeighborPort	Neighbor device port to which the phone connects that LLDP discovered.
PortSpeed	Speed and duplex information.

Voice

System

System Configuration

Parameter	Description
Restricted Access Domains	This feature is used when implementing software customization.
Enable Web Server	Enable/disable web server of the IP phone. Default: Yes

Parameter	Description
Enable Protocol	Choose the type of protocol: • Http • Https If you specify the HTTPS protocol, you must include https: in the URL.
Enable Direct Action Url	Enables the direct action of the URL. Default: Yes
Session Max Timeout	Allows you to enter maximum timeout of the session. Default: 3600
Session Idle Timeout	Allows you to enter idle timeout of the session. Default: 3600
Web Server Port	Allows you to enter port number of the phone web user interface. Default: • 80 for protocol HTTP. • 443 for protocol HTTPS. If you specify a port number other than the default value for that protocol, you must include the nondefault port number in the server URL. Example: https://192.0.2.1:999/admin/advanced
Enable Web Admin Access	Allows you to enable or disable local access to the phone web user interface. Select Yes or No from the drop-down menu. Default: Yes
Admin Password	Allows you to enter password for the administrator. Default: No password
User Password	Allows you to enter password for the user. Default: Blank
Phone-UI-readonly	Allows you to make the phone menus and options that the phone users see as read-only fields.

Parameter	Description
Phone-UI-User-Mode	Allows you to restrict the menus and options that phone users see when they use the phone interface. Choose yes to enable this parameter and restrict access. Default: No Specific parameters are then designated as “na” or “ro” using provisioning files. Parameters designated as “na” will not appear on the phone interface. Parameters designated as “ro” will not be editable by the user.

IPv4 Settings

Parameter	Description
Connection Type	Internet connection type that is configured for the phone. Options are DHCP and Static IP. Default: DHCP
NetMask	Subnet mask of the phone.
Static IP	IP address of the phone.
Gateway	IP address of the gateway.
Primary DNS	Primary Domain Name Server (DNS) assigned to the phone.
Secondary DNS	Secondary Domain Name Server (DNS) if assigned to the phone.

802.1X Authentication

Parameter	Description
Enable 802.1X Authentication	Enables/disables 802.1X Default: No

Optional Network Configuration

Parameter	Description
Host Name	The hostname of the Cisco IP Phone.
Domain	The network domain of the Cisco IP Phone. If you are using LDAP, see the LDAP Configuration.

Parameter	Description
DNS Query Mode	Specified mode of DNS query. • Paraller • Sequential
DNS Caching Enable	When set to Yes, the DNS query results are not cached. Default: Yes
Switch Port Config	Allows you to select speed and duplex of the network port. Values are: • Auto 10MB half 10MB full 100 MB half 100MB full 100 half 1000 full
Syslog Server	Specify the syslog server name and port. This feature specifies the server for logging IP phone system information and critical events. If both Debug Server and Syslog Server are specified, Syslog messages are also logged to the Debug Server.
Debug Level	The debug level from 0 to 2. The higher the level, the more debug information is generated. Zero (0) means that no debug information is generated. To log SIP messages, you must set the Debug Level to at least 2. Default: 0
Primary NTP Server	IP address or name of the primary NTP server used to synchronize its time. Default: Blank
Secondary NTP Server	IP address or name of the secondary NTP server used to synchronize its time. Default: Blank
DNS Server Order	Specifies the method for selecting the DNS server: • Manual-Dhcp • Manual • Dhcp-Manual

Parameter	Description
Enable SSLv3	Choose Yes to enable SSLv3. Choose No to disable. Default: No

VLAN Settings

Parameter	Description
Enable VLAN	Choose Yes to enable VLAN. Choose No to disable.
Enable CDP	Enable CDP only if you are using a switch that has Cisco Discovery Protocol. CDP is negotiation based and determines which VLAN the IP phone resides in.
Enable LLDP-MED	Choose Yes to enable LLDP-MED for the phone to advertise itself to devices that use that discovery protocol. When the LLDP-MED feature is enabled, after the phone has initialized and Layer 2 connectivity is established, the phone sends out LLDP-MED PDU frames. If the phone receives no acknowledgment, the manually configured VLAN or default VLAN will be used if applicable. If the CDP is used concurrently, the waiting period of 6 seconds is used. The waiting period will increase the overall startup time for the phone.
Network Startup Delay	Setting this value causes a delay for the switch to get to the forwarding state before the phone will send out the first LLDP-MED packet. The default delay is 3 seconds. For configuration of some switches, you might need to increase this value to a higher value for LLDP-MED to work. Configuring a delay can be important for networks that use Spanning Tree Protocol.
VLAN ID	If you use a VLAN without CDP (VLAN enabled and CDP disabled), enter a VLAN ID for the IP phone. Note that only voice packets are tagged with the VLAN ID. Do not use 1 for the VLAN ID.

Inventory Settings

Parameter	Description
Asset ID	Provides the ability to enter an asset ID for inventory management when using LLDP-MED. The default value for Asset ID is empty. Enter a string of less than 32 characters if you are using this field. The Asset ID can be provisioned only by using the web management interface or remote provisioning. The Asset ID is not displayed on the phone screen. Changing the Asset ID field causes the phone to reboot.

SIP

SIP Parameters

Parameter	Description
Max Forward	SIP Max Forward value, which can range from 1 to 255. Default: 70
Max Redirection	Number of times an invite can be redirected to avoid an infinite loop. Default: 5
Max Auth	Maximum number of times (from 0 to 255) a request can be challenged. Default: 2
SIP User Agent Name	Used in outbound REGISTER requests. Default: $VERSION If empty, the header is not included. Macro expansion of $A to $D corresponding to GPP_A to GPP_D allowed
SIP Server Name	Server header used in responses to inbound responses. Default: $VERSION
SIP Reg User Agent Name	User-Agent name to be used in a REGISTER request. If this is not specified, the <SIP User Agent Name> is also used for the REGISTER request. Default: Blank
SIP Accept Language	Accept-Language header used. To access, click the SIP tab, and fill in the SIP Accept Language field. There is no default. If empty, the header is not included.

Parameter	Description
DTMF Relay MIME Type	MIME Type used in a SIP INFO message to signal a DTMF event. This field must match that of the Service Provider. Default: application/dtmf-relay
Hook Flash MIME Type	MIME Type used in a SIPINFO message to signal a hook flash event.
Remove Last Reg	Enables you to remove the last registration before registering a new one if the value is different. Select yes or no from the drop-down menu.
Use Compact Header	If set to yes, the phone uses compact SIP headers in outbound SIP messages. If inbound SIP requests contain normal headers, the phone substitutes incoming headers with compact headers. If set to no, the phones use normal SIP headers. If inbound SIP requests contain compact headers, the phones reuse the same compact headers when generating the response, regardless of this setting. Default: No
Escape Display Name	Enables you to keep the Display Name private. Select Yes if you want the IP phone to enclose the string (configured in the Display Name) in a pair of double quotes for outbound SIP messages. Default: Yes.
Talk Package	Enables support for the BroadSoft Talk Package that lets users answer or resume a call by clicking a button in an external application. Default: No
Hold Package	Enables support for the BroadSoft Hold Package, which lets users place a call on hold by clicking a button in an external application. Default: No
Conference Package	Enables support for the BroadSoft Conference Package that enables users to start a conference call by clicking a button in an external application. Default: No
RFC 2543 Call Hold	If set to yes, unit includes c=0.0.0.0 syntax in SDP when sending a SIP re-INVITE to the peer to hold the call. If set to no, unit will not include the c=0.0.0.0 syntax in the SDP. The unit will always include a=sendonly syntax in the SDP in either case. Default: Yes

Parameter	Description
Random REG CID on Reboot	If set to yes, the phone uses a different random call-ID for registration after the next software reboot. If set to no, the Cisco IP phone tries to use the same call-ID for registration after the next software reboot. The Cisco IP phone always uses a new random Call-ID for registration after a power-cycle, regardless of this setting. Default: No.
SIP TCP Port Min	Specifies the lowest TCP port number that can be used for SIP sessions. Default: 5060
SIP TCP Port Max	Specifies the highest TCP port number that can be used for SIP sessions. Default: 5080
Caller ID Header	Provides the option to take the caller ID from PAID-RPID-FROM, PAID-FROM, RPID-PAID-FROM, RPID-FROM, or FROM header. Default: PAID-RPID-FROM
Hold Target Before Refer	Controls whether to hold call leg with transfer target before sending REFER to the transferee when initiating a fully-attended call transfer (where the transfer target has answered). Default: No
Dialog SDP Enable	When enabled and the Notify message body is too big causing fragmentation, the Notify message xml dialog is simplified; Session Description Protocol (SDP) is not included in the dialog xml content.
Keep Referee When Refer Failed	If set to yes, it configures the phone to immediately handle NOTIFY sipfrag messages.
Display Diversion Info	Display the Diversion info included in SIP message on LCD or not.
Display Anonymous From Header	Show the caller ID from the SIP INVITE message “From” header when set to Yes, even if the call is an anonymous call. When the parameter is set to no, the phone displays "Anonymous Caller" as the caller ID.
Sip Accept Encoding	Supports the content-encoding gzip feature. The options are none and gzip. If gzip is selected, the SIP message header contains the string “Accept-Encoding: gzip”, and the phone is able to process the SIP message body, which is encoded with the gzip format.
Disable Local Name To Header	The options are No and Yes. If No is selected, no changes are made. The default value is No. If Yes is selected, it disables the display name in “Directory”, “Call History”, and in the “To” header during an outgoing call.

SIP Timer Values

Parameter	Description
SIP T1	RFC 3261 T1 value (RTT estimate) that can range from 0 to 64 seconds. Default: 0.5 seconds
SIP T2	RFC 3261 T2 value (maximum retransmit interval for non-INVITE requests and INVITE responses) that can range from 0 to 64 seconds. Default: 4 seconds
SIP T4	RFC 3261 T4 value (maximum duration a message remains in the network), which can range from 0 to 64 seconds. Default: 5 seconds.
SIP Timer B	INVITE time-out value, which can range from 0 to 64 seconds. Default: 16 seconds.
SIP Timer F	Non-INVITE time-out value, which can range from 0 to 64 seconds. Default: 16 seconds.
SIP Timer H	INVITE final response, time-out value, which can from 0 to 64 seconds. Default: 16 seconds.
SIP Timer D	ACK hang-around time, which can range from 0 to 64 seconds. Default: 16 seconds.
SIP Timer J	Non-INVITE response hang-around time, which can range from 0 to 64 seconds. Default: 16 seconds.
INVITE Expires	INVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Ranges from 0 to 2000000. Default: 240 seconds
ReINVITE Expires	ReINVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Ranges from 0 to 2000000. Default: 30
Reg Min Expires	Minimum registration expiration time allowed from the proxy in the Expires header or as a Contact header parameter. If the proxy returns a value less than this setting, the minimum value is used.

Parameter	Description
Reg Max Expires	Maximum registration expiration time allowed from the proxy in the Min-Expires header. If the value is larger than this setting, the maximum value is used.
Reg Retry Intv	Interval to wait before the Cisco IP Phone retries registration after failing during the last registration.The range is from 1 to 2147483647 Default: 30 See the note below for additional details.
Reg Retry Long Intvl	When registration fails with a SIP response code that does not match<Retry Reg RSC>, the Cisco IP Phone waits for the specified length of time before retrying. If this interval is 0, the phone stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. Default: 1200 See the note below for additional details.
Reg Retry Random Delay	Random delay range (in seconds) to add to <Register Retry Intvl> when retrying REGISTER after a failure. Minimum and maximum random delay to be added to the short timer. The range is from 0 to 2147483647. Default: 0
Reg Retry Long Random Delay	Random delay range (in seconds) to add to <Register Retry Long Intvl> when retrying REGISTER after a failure. Default: 0
Reg Retry Intvl Cap	Maximum value of the exponential delay. The maximum value to cap the exponential backoff retry delay (which starts at the Register Retry Intvl and doubles every retry). Defaults to 0, which disables the exponential backoff (that is, the error retry interval is always at the Register Retry Intvl). When this feature is enabled, the Reg Retry Random Delay is added to the exponential backoff delay value. The range is from 0 to 2147483647. Default: 0
Sub Min Expires	Sets the lower limit of the REGISTER expires value returned from the Proxy server.
Sub Max Expires	Sets the upper limit of the REGISTER minexpires value returned from the Proxy server in the Min-Expires header. Default: 7200.
Sub Retry Intvl	This value (in seconds) determines the retry interval when the last Subscribe request fails. Default: 10.

Note The phone can use a RETRY-AFTER value when it is received from a SIP proxy server that is too busy to process a request (503 Service Unavailable message). If the response message includes a RETRY-AFTER header, the phone waits for the specified length of time before to REGISTER again. If a RETRY-AFTER header is not present, the phone waits for the value specified in the Reg Retry Interval or the Reg Retry Long Interval.

Response Status Code Handling

Parameter	Description
Try Backup RSC	This parameter may be set to invoke failover upon receiving specified response codes. Default: Blank For example, you can enter numeric values 500 or a combination of numeric values plus wild cards if multiple values are possible. For the later, you can use 5?? to represent all SIP Response messages within the 500 range. If you want to use multiple ranges, you can add a comma "," to delimit values of 5?? and 6??
Retry Reg RSC	Interval to wait before the phone retries registration after failing during the last registration. Default: Blank For example, you can enter numeric values 500 or a combination of numeric values plus wild cards if multiple values are possible. For the later, you can use 5?? to represent all SIP Response messages within the 500 range. If you want to use multiple ranges, you can add a comma "," to delimit values of 5?? and 6??

RTP Parameters

Parameter	Description
RTP Port Min	Minimum port number for RTP transmission and reception. Minimum port number for RTP transmission and reception. Should define a range that contains at least 10 even number ports (twice the number of lines); for example, configure RTP port min to 16384 and RTP port max to 16538. Default: 16384

Parameter	Description
RTP Port Max	Maximum port number for RTP transmission and reception. Should define a range that contains at least 10 even number ports (twice the number of lines); for example, configure RTP port min to 16384 and RTP port max to 16538. Default: 16538
RTP Packet Size	Packet size in seconds, which can range from 0.01 to 0.13. Valid values must be a multiple of 0.01 seconds. Default: 0.02
Max RTP ICMP Err	Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the phone terminates the call. If value is set to 0, the phone ignores the limit on ICMP errors.
RTCP Tx Interval	Interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. Default: 0

SDP Payload Types

Parameter	Description
G722.2 Dynamic Payload	G722 Dynamic Payload type. Default: 96
iLBC Dynamic Payload	iLBC Dynamic Payload type. Default: 97
iSAC Dynamic Payload	iSAC Dynamic Payload type. Default: 98
OPUS Dynamic Payload	OPUS Dynamic Payload type. Default: 99
AVT Dynamic Payload	AVT dynamic payload type. Ranges from 96-127. Default: 101
INFOREQ Dynamic Payload	INFOREQ Dynamic Payload type.
G711u Codec Name	G711u codec name used in SDP. Default: PCMU

Parameter	Description
G711a Codec Name	G711a codec name used in SDP. Default: PCMA
G729a Codec Name	G729a codec name used in SDP. Default: G729a
G729b Codec Name	G729b codec name used in SDP. Default: G729b
G722 Codec Name	G722 codec name used in SDP. Default: G722
G722.2 Codec Name	G722.2 codec name used in SDP. Default: G722.2
iLBC Codec Name	iLBC codec name used in SDP. Default: iLBC
iSAC Codec Name	iSAC codec name used in SDP. Default: iSAC
OPUS Codec Name	OPUS codec name used in SDP. Default: OPUS
AVT Codec Name	AVT codec name used in SDP. Default: telephone-event

NAT Support Parameters

Parameter	Description
Handle VIA received	Enables the phone to process the received parameter in the VIA header. Default: No
Handle VIA rport	Enables the phone to process the rport parameter in the VIA header. Default: No
Insert VIA received	Enables to insert the received parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Default: No

Parameter	Description
Insert VIA rport	Enables to insert the rport parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Default: No
Substitute VIA Addr	Enables the user to use NAT-mapped IP:port values in the VIA header. Default: No
Send Resp To Src Port	Enables to send responses to the request source port instead of the VIA sent-by port. Default: No
STUN Enable	Enables the use of STUN to discover NAT mapping. Default: No
STUN Test Enable	If the STUN Enable feature is enabled and a valid STUN server is available, the phone can perform a NAT-type discovery operation when it powers on. It contacts the configured STUN server, and the result of the discovery is reported in a Warning header in all subsequent REGISTER requests. If the phone detects symmetric NAT or a symmetric firewall, NAT mapping is disabled. Default: No
STUN Server	IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery. You can use a public STUN server or set up your own STUN server. Default: Blank
EXT IP	External IP address to substitute for the actual IP address of phone in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed. If this parameter is specified, phone assumes this IP address when generating SIP messages and SDP (if NAT Mapping is enabled for that line). Default: Blank
EXT RTP Port Min	External port mapping number of the RTP Port Minimum number. If this value is not zero, the RTP port number in all outgoing SIP messages is substituted for the corresponding port value in the external RTP port range. Default: 0
NAT Keep Alive Intvl	Interval between NAT-mapping keep alive messages. Default: 15

Parameter	Description
Redirect Keep Alive	If enabled, the IP phone redirects the keepalive message when SIP_301_MOVED_PERMANENTLY is received as the registration response.

Provisioning

Configuration Profile

Parameter	Description
Provision Enable	Allows or denies resync actions. Default: 160,159,66,150
Resync On Reset	The device performs a resync operation after power-up and after each upgrade attempt when set to **Yes**. Default: Yes
Resync Random Delay	A random delay following the boot-up sequence before performing the reset, specified in seconds. In a pool of IP Telephony devices that are scheduled to simultaneously powered up, this introduces a spread in the times at which each unit sends a resync request to the provisioning server. This feature can be useful in a large residential deployment, in the case of a regional power failures. Default: 2
Resync At (HHmm)	Time in 24-hour format (hhmm) to resync the device. When this parameter is provisioned, the Resync Periodic parameter is ignored. Default: Blank
Resync At Random Delay	To avoid flooding the server with simultaneously resync requests from multiple phones set to resync at the same time, the phone triggers the resync up to ten minutes after the specified time. The input value (in seconds) is converted to minutes. The default value is 600 seconds (10 minutes). If the parameter value is set to less than 600, the default value is used. Default: 600
Resync Periodic	Time in seconds between periodic resyncs. If this value is empty or zero, the device does not resync periodically. Default: 3600

Parameter	Description
Resync Error Retry Delay	If a resync operation fails because the IP Telephony device was unable to retrieve a profile from the server, if the downloaded file is corrupt, or an internal error occurs, the device tries to resync again after a time specified in seconds. If the delay is set to 0, the device does not try to resync again following a failed resync attempt. Default: 3600
Forced Resync Delay	Forced resync delay typically takes place when it is time to a resync and you are in an active call. For example, if you set 5 minute for Periodic Resync and you place a call right after the resync, the resync happens while you are 6 minutes into the call (normal time of Resync + Forced Resync Delay). Default: 14400
Resync From SIP	Controls requests for resync operations via a SIP NOTIFY event sent from the service provider proxy server to the IP Telephony device. If enabled, the proxy can request a resync by sending a SIP NOTIFY message containing the Event: resync header to the device. Default: Yes
Resync After Upgrade Attempt	Enables or disables the resync operation after any upgrade occurs. If Yes is selected, sync is triggered. Default: Yes
Resync Trigger 1 Resync Trigger 2	If the logical equation in these parameters evaluates to FALSE, Resync is not triggered even when Resync On Reset is set to TRUE. Only Resync via direct action URL and SIP notify ignores these Resync Trigger. Default: Blank
Resync Fails On FNF	A resync is considered unsuccessful if a requested profile is not received from the server. This can be overridden by this parameter. When it is set to **No**, the device accepts a `file-not-found` response from the server as a successful resync. Default: Yes
Profile Rule Profile Rule B Profile Rule C Profile Rule D	Remote configuration profile rules evaluated in sequence. Each resync operation can retrieve multiple files, potentially managed by different servers. Default: /$PSN.xml
DHCP Option To Use	DHCP options, delimited by commas, used to retrieve firmware and profiles. Default: 66,160,159,150,60,43,125

Parameter	Description
Log Request Msg	The message sent to the syslog server at the start of a resync attempt. Default: `$PN $MAC –Requesting % $SCHEME://$SERVIP:$PORT$PATH`
Log Success Msg	The syslog message issued upon successful completion of a resync attempt. Default: `$PN $MAC -Successful Resync % $SCHEME://$SERVIP:$PORT$PATH`
Log Failure Msg	The syslog message that is issued after a failed download attempt. Default: `$PN $MAC -- Resync failed: $ERR`
HTTP Report Method	Allows to select HTTP options. Options are POST and PUT.
User Configurable Resync	Allows a user to resync the phone from the phone screen. Default: Yes

Firmware Upgrade

Parameter	Description
Upgrade Enable	Allows firmware update operations independent of resync actions. Default: Yes
Upgrade Error Retry Delay	The interval applied in the event of an upgrade failure. The firmware upgrade error timer activates after a failed firmware upgrade attempt and is initialized with this value. The next firmware upgrade attempt occurs when this timer counts down to zero. Default: 3600 seconds

Parameter	Description

Upgrade Rule	A firmware upgrade script that defines upgrade conditions and associated firmware URLs. It uses the same syntax as Profile Rule.
	Use the following format to enter the upgrade rule:
	`protocol://server[:port]/profile_pathname`
	For example:
	`tftp://192.168.1.5/image/sip88xx.10-3-1-9-3PCC.loads`
	If no protocol is specified, TFTP is assumed. If no server-name is specified, the host that requests the URL is used as the server name. If no port is specified, the default port is used (69 for TFTP, 80 for HTTP, or 443 for HTTPS).
	Default: Blank

Log Upgrade Request Msg	Syslog message issued at the start of a firmware upgrade attempt. Default: $PN $MAC -- Requesting upgrade $SCHEME://$SERVIP:$PORT$PATH
Log Upgrade Success Msg	Syslog message issued after a firmware upgrade attempt completes successfully. Default: $PN $MAC -- Successful upgrade $SCHEME://$SERVIP:$PORT$PATH -- $ERR
Log Upgrade Failure Msg	Syslog message issued after a failed firmware upgrade attempt. Default: $PN $MAC -- Upgrade failed: $ERR

For information about the Provisioning page, see the *Cisco IP Phone 7800 Series and Cisco IP Phone 8800 Series Multiplatform Phones Provisioning Guide*.

CA Settings

Parameter	Description
Custom CA Rule	The URL to download Custom CA. Default: Blank

HTTP Settings

Parameter	Description
HTTP User Agent Name	Allows you to enter a name for HTTP user. Default: Blank

Problem Report Tool

Parameter	Description
PRT Upload Rule	Path to the PRT upload script.
PRT Upload Method	Method used to upload PRT logs to the remote server. Can be either HTTP POST or PUT. Default: POST

General Purpose Parameters

Parameter	Description
GPP A - GPP P	The general purpose parameters GPP_* are used as free string, registers when configuring the Cisco IP phones to interact with a particular provisioning server solution. They can be configured to contain diverse values, including the following: • Encryption keys. • URLs. • Multistage provisioning status information. • Post request templates. • Parameter name alias maps. • Partial string values, eventually combined into complete parameter values. Default: Blank

Regional

Call Progress Tones

Parameter	Description
Dial Tone	Prompts the user to enter a phone number.
Outside Dial Tone	Alternative to the Dial Tone. It prompts the user to enter an external phone number, as opposed to an internal extension. It is triggered by a, (comma) character encountered in the dial plan.

Parameter	Description
Prompt Tone	Prompts the user to enter a call forwarding phone number.
Busy Tone	Played when a 486 RSC is received for an outbound call.
Reorder Tone	Played when an outbound call has failed or after the far end hangs up during an established call. Reorder Tone is played automatically when <Dial Tone> or any of its alternatives times out.
Ring Back Tone	Played during an outbound call when the far end is ringing.
Call Waiting Tone	Played when a call is waiting.
Confirm Tone	Brief tone to notify the user that the last input value has been accepted.
MWI Dial Tone	Played instead of the Dial Tone when there are unheard messages in the caller's mailbox.
Cfwd Dial Tone	Played when all calls are forwarded.
Holding Tone	Informs the local caller that the far end has placed the call on hold.
Conference Tone	Played to all parties when a three-way conference call is in progress.
Secure Call Indication Tone	Played when a call has been successfully switched tosecure mode. It should be played only for a short while (less than 30 seconds) and at a reduced level (less than -19 dBm) so it does not interfere with the conversation.
Page Tone	Specifies the tone transmitted when the paging feature is enabled.
Alert Tone	Played when an alert occurs.
System Beep	Audible notification tone played when a system error occurs.
Call Pickup Tone	Provides the ability to configure an audio indication for call pickup.

Distinctive Ring Patterns

Parameter	Description
Cadence 1	Cadence script for distinctive ring 1. Defaults to 60(2/4).
Cadence 2	Cadence script for distinctive ring 2. Defaults to 60(.3/.2, 1/.2,.3/4).

Parameter	Description
Cadence 3	Cadence script for distinctive ring 3. Defaults to 60(.8/.4,.8/4).
Cadence 4	Cadence script for distinctive ring 4. Defaults to 60(.4/.2,.3/.2,.8/4).
Cadence 5	Cadence script for distinctive ring 5. Defaults to 60(.2/.2,.2/.2,.2/.2,1/4).
Cadence 6	Cadence script for distinctive ring 6. Defaults to 60(.2/.4,.2/.4,.2/4).
Cadence 7	Cadence script for distinctive ring 7. Defaults to 60(4.5/4).
Cadence 8	Cadence script for distinctive ring 8. Defaults to 60(0.25/9.75)
Cadence 9	Cadence script for distinctive ring 9. Defaults to 60(.4/.2,.4/2).

Control Timer Values (sec)

Parameter	Description
Reorder Delay	Delay after far end hangs up before reorder (busy) tone is played. 0 = plays immediately, inf = never plays. Range: 0–255 seconds. Set to 255 to return the phone immediately to on-hook status and to not play the tone.
Interdigit Long Timer	Long timeout between entering digits when dialing. The interdigit timer values are used as defaults when dialing. The Interdigit_Long_Timer is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. Range: 0–64 seconds. Default: 10
Interdigit Short Timer	Short timeout between entering digits when dialing. The Interdigit_Short_Timer is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences. Range: 0–64 seconds. Default: 3

Vertical Service Activation Codes

Parameter	Description
Call Return Code	This code calls the last caller. Defaults to *69.
Blind Transfer Code	Begins a blind transfer of the current call to the extension specified after the activation code. Defaults to *88.
Cfwd All Act Code	Forwards all calls to the extension specified after the activation code. Defaults to *72.
Cfwd All Deact Code	Cancels call forwarding of all calls. Defaults to *73.
Cfwd Busy Act Code	Forwards busy calls to the extension specified after the activation code. Defaults to *90.
Cfwd Busy Deact Code	Cancels call forwarding of busy calls. Defaults to *91.
Cfwd No Ans Act Code	Forwards no-answer calls to the extension specified after the activation code. Defaults to *92.
Cfwd No Ans Deact Code	Cancels call forwarding of no-answer calls. Defaults to *93.
CW Act Code	Enables call waiting on all calls. Defaults to *56.
CW Deact Code	Disables call waiting on all calls. Defaults to *57.
CW Per Call Act Code	Enables call waiting for the next call. Defaults to *71.
CW Per Call Deact Code	Disables call waiting for the next call. Defaults to *70.
Block CID Act Code	Blocks caller ID on all outbound calls. Defaults to *67.

Parameter	Description
Block CID Deact Code	Removes caller ID blocking on all outbound calls. Defaults to *68.
Block CID Per Call Act Code	Removes caller ID blocking on the next inbound call. Defaults to *81.
Block CID Per Call Deact Code	Removes caller ID blocking on the next inbound call. Defaults to *82.
Block ANC Act Code	Blocks all anonymous calls. Defaults to *77.
Block ANC Deact Code	Removes blocking of all anonymous calls. Defaults to *87.
DND Act Code	Enables the do not disturb feature. Defaults to *78.
DND Deact Code	Disables the do not disturb feature. Defaults to *79.
Secure All Call Act Code	Makes all outbound calls secure. Defaults to *16.
Secure No Call Act Code	Makes all outbound calls not secure. Defaults to *17.
Paging Code	The star code used for paging the other clients in the group. Defaults to *96.
Call Park Code	The star code used for parking the current call. Defaults to *38.
Call Pickup Code	The star code used for picking up a ringing call. Defaults to *36.
Call Unpark Code	The star code used for picking up a call from the call park. Defaults to *39.
Group Call Pickup Code	The star code used for picking up a group call. Defaults to *37.

Parameter	Description
Referral Services Codes	These codes tell the IP phone what to do when the user places the current call on hold and is listening to the second dial tone. One or more *code can be configured into this parameter, such as *98, or *97|*98|*123, and so on. Max total length is 79 chars. This parameter applies when the user places the current call on hold (by Hook Flash) and is listening to second dial tone. Each *code (and the following valid target number according to current dial plan) entered on the second dial-tone triggers the phone to perform a blind transfer to a target number that is prepended by the service *code. For example, after the user dials *98, the IP phone plays a special dial tone called the Prompt Tone while waiting for the user the enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the phone sends a blind REFER to the holding party with the Refer-To target equals to *98<target_number>. This feature allows the phone to hand off a call to an application server to perform further processing, such as call park. The *codes should not conflict with any of the other vertical service codes internally processed by the IP phone. You can empty the corresponding *code that you do not want to the phone to process.

Parameter	Description
Feature Dial Services Codes	These codes tell the phone what to do when the user is listening to the first or second dial tone. One or more *code can be configured into this parameter, such as *72, or *72|*74|*67|*82, and so forth. The maximum total length is 79 characters. This parameter applies when the user has a dial tone (first or second dial tone). Enter *code (and the following target number according to current dial plan) entered at the dial tone triggers the phone to call the target number prepended by the *code. For example, after user dials *72, the phone plays a prompt tone awaiting the user to enter a valid target number. When a complete number is entered, the phone sends a INVITE to *72<target_number> as in a normal call. This feature allows the proxy to process features like call forward (*72) or BLock Caller ID (*67). The *codes should not conflict with any of the other vertical service codes internally processed by the phone. You can empty the corresponding *code that you do not want to the phone to process. You can add a parameter to each *code in Features Dial Services Codes to indicate what tone to play after the *code is entered, such as *72‘c‘|*67‘p‘. Below are a list of allowed tone parameters (note the use of back quotes surrounding the parameter without spaces) • c = Cfwd Dial Tone • d = Dial Tone • m = MWI Dial Tone • o = Outside Dial Tone • p = Prompt Dial Tone • s = Second Dial Tone • x = No tones are place, x is any digit not used above If no tone parameter is specified, the phone plays Prompt tone by default. If the *code is not to be followed by a phone number, such as *73 to cancel call forwarding, do not include it in this parameter. In that case, simple add that *code in the dial plan and the phone sends INVITE *73@..... as usual when user dials *73.

Vertical Service Announcement Codes

Parameter	Description
Service Annc Base Number	Defaults to blank.
Service Annc Extension Codes	Defaults to blank.

Outbound Call Codec Selection Codes

Parameter	Description
Prefer G711u Code	Makes this codec the preferred codec for the associated call. Defaults to *017110.
Force G711u Code	Makes this codec the only codec that can be used for the associated call. Defaults to *027110.
Prefer G711a Code	Makes this codec the preferred codec for the associated call. Defaults to *017111
Force G711a Code	Makes this codec the only codec that can be used for the associated call. Defaults to *027111.
Prefer G722 Code	Makes this codec the preferred codec for the associated call. Defaults to *01722. Only one G.722 call at a time is allowed. If a conference call is placed, a SIP re-invite message is sent to switch the calls to narrowband audio.
Force G722 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *02722. Only one G.722 call at a time is allowed. If a conference call is placed, a SIP re-invite message is sent to switch the calls to narrowband audio.
Prefer G722.2 Code	Makes this codec the preferred codec for the associated call.
Force G722.2 Code	Makes this codec the only codec that can be used for the associated call.
Prefer G729a Code	Makes this codec the preferred codec for the associated call. Defaults to *01729.
Force G729a Code	Makes this codec the only codec that can be used for the associated call. Defaults to *02729.
Prefer iLBC Code	Makes this codec the preferred codec for the associated call.

Parameter	Description
Force iLBC Code	Makes this codec the only codec that can be used for the associated call.
Prefer ISAC Code	Makes this codec the preferred codec for the associated call.
Force ISAC Code	Makes this codec the only codec that can be used for the associated call.
Prefer OPUS Code	Makes this codec the preferred codec for the associated call.
Force OPUS Code	Makes this codec the only codec that can be used for the associated call.

Time

Parameter	Description
Set Local Date (mm/dd/yyyy)	Sets the local date (mm represents the month and dd represents the day). The year is optional and uses two or four digits. Default: Blank
Set Local Time (HH/mm)	Sets the local time (hh represents hours and mm represents minutes). Seconds are optional. Default: Blank
Time Zone	Selects the number of hours to add to GMT to generate the local time for caller ID generation. Choices are GMT-12:00, GMT-11:00,…, GMT, GMT+01:00, GMT+02:00, …, GMT+13:00. Default: GMT-08:00
Time Offset (HH/mm)	This specifies the offset from GMT to use for the local system time. Default: 00/00
Ignore DHCP Time Offset	When used with some routers that have DHCP with time offset values configured, the IP phone uses the router settings and ignores the IP phone time zone and offset settings. To ignore the router DHCP time offset value, and use the local time zone and offset settings, choose yes for this option. Choosing no causes the IP phone to use the router's DHCP time offset value. Default: Yes.

Parameter	Description
Daylight Saving Time Rule	Enter the rule for calculating daylight saving time; it should include the start, end, and save values. This rule is comprised of three fields. Each field is separated by ; (a semicolon) as shown below. Optional values inside [ ] (the brackets) are assumed to be 0 if they are not specified. Midnight is represented by 0:0:0 of the given date. This is the format of the rule: Start = <start-time>; end=<end-time>; save = <save-time>. The <start-time> and <end-time> values specify the start and end dates and times of daylight saving time. Each value is in this format: <month> /<day> / <weekday>[/HH:[mm[:ss]]] The <save-time> value is the number of hours, minutes, and/or seconds to add to the current time during daylight saving time. The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition. The <save-time> value is in this format: [/[+|-]HH:[mm[:ss]]] The <month> value equals any value in the range 1-12 (January-December). The <day> value equals [+|-] any value in the range 1-31. If <day> is 1, it means the <weekday> on or before the end of the month (in other words the last occurrence of < weekday> in that month).
Daylight Saving Time Rule (continued)	The <weekday> value equals any value in the range 1-7 (Monday-Sunday). It can also equal 0. If the <weekday> value is 0, this means that the date to start or end daylight saving is exactly the date given. In that case, the <day> value must not be negative. If the <weekday> value is not 0 and the <day> value is positive, then daylight saving starts or ends on the <weekday> value on or after the date given. If the <weekday> value is not 0 and the <day> value is negative, then daylight saving starts or ends on the <weekday> value on or before the date given. Where: • HH stands for hours (0-23). • mm stands for minutes (0-59). • ss stands for seconds (0-59). Default: 3/-1/7/2;end=10/-1/7/2;save=1.
Daylight Saving Time Enable	Enables Daylight Saving Time. Default: Yes

Language

Parameter	Description
Dictionary Server Script	Defines the location of the dictionary server, the languages available, and the associated dictionary. See the Dictionary Server Script. Default: Blank

Language Selection	Specifies the default language. The value must match one of the languages supported by the dictionary server. The script (dx value) is:
	`<Language_Selection ua="na"> </Language_Selection>`
	Default: Blank
	The maximum number of characters is 512. For example:
	`<Language_Selection ua="na"> Spanish </Language_Selection>`

Locale	Choose the locale that should be set in the HTTP Accept-Language header Default: en-US

Phone

General

Parameter	Description
Station Name	Name of the phone.
Station Display Name	Name to identify the phone; appears on the phone screen. You can use spaces in this field and the name does not have to be unique.
Voice Mail Number	A phone number or URL to check voice mail. Default: None
Select Logo	Select from None, PNG Picture, or Text Logo. Default: None

Handsfree

Parameter	Description
Bluetooth Mode	Shows the method of Bluetooth connection. • Phone—Pairs with a Bluetooth headset only. • Handsfree—Operates as a handsfree device with a Bluetooth-enabled mobile phone. • Both—Uses a Bluetooth headset, or operates with a Bluetooth-enabled mobile phone.
Line	Specifies the line number for which the Bluetooth is enabled.

Line Key

Parameter	Description
Extension	Specifies the n extensions to be assigned to Line Key n. Default: Line Key n
Short Name	Specifies the user name for Line Key. Default: $USER
Share Call Appearance	Specifies whether the incoming call appearance is shared with other phones or it is private.
Extended Function	Use to assign Busy Lamp Field, Call Pickup, and Speed Dial Functions to Idle Lines on the IP phone.

Miscellaneous Line Key Settings

Parameter	Description
Line ID Mapping	Specifies the shared call appearance line ID mapping. If Vertical First is set, the first call makes the LED flash. If Horizontal first is set, the second call makes the same LED flash. **Note** 7811 Cisco IP Phone does not support Line ID Mapping. Default: Vertical First
SCA Barge-In-Enable	Enables the SCA Barge-In. Default: No

Parameter	Description
SCA Sticky Auto Line Seize	If enabled, restricts to automatically pick up an incoming call on a shared line when you take the phone off-hook.
Call Appearances Per Line	This parameter allows you to choose the number of calls per line button. You can choose a value from 2 to 10. Default: 2

Supplementary Services

Parameter	Description
Conference Serv	Enable/disable three-way conference service. Default: Yes
Attn Transfer Serv	Enable/disable attended-call-transfer service. Default: Yes
Blind Transfer Serv	Enable/disable blind-call-transfer service. Default: Yes
DND Serv	Enable/disable do not disturb service. Default: Yes
Block ANC Serv	Enable/disable block-anonymous-call service. Default: Yes
Block CID Serv	Enable/disable blocking outbound Caller-ID service. Default: Yes
Cfwd All Serv	Enable/disable call-forward-all service. Default: Yes
Cfwd Busy Serv	Enable/disable call-forward-on-busy service. Default: Yes
Cfwd No Ans Serv	Enable/disable call-forward-no-answer service. Default: Yes

Ringtone

Parameter	Description
Ring	Ring tone scripts for different rings.
Silent Ring Duration	Controls the duration of the silent ring. For example, if the parameter is set to 20 seconds, the phone plays the silent ring for 20 seconds then sends 480 response to INVITE message.

Extension Mobility

Parameter	Description
EM Enable	Options to enable or to disable the extension mobility support for the phone. Default: No
EM User Domain	Name of the domain for the phone or the authentication server. Default: Blank
Inactivity Timer(m)	Specifies the duration for which the extension mobility remains inactive.
Countdown Timer(s)	Specifies the duration for which it waits before it logs out". Default is 10

BroadSoft Settings

Parameter	Description
Directory Enable	Set to Yes to enable BroadSoft directory for the phone user. Default: No
XSI Host Server	Enter the name of the server; for example, xsi.iop1.broadworks.net. Default: Blank
Directory Name	Name of the directory. Displays on the phone as a directory choice. Default: Blank

Parameter	Description
Directory Type	Select the type of BroadSoft directory: Enterprise: Allows users to search on last name, first name, user or group ID, phone number, extension, department, or email address. Group: Allows users to search on last name, first name, user ID, phone number, extension, department, or email address. Personal: Allows users to search on last name, first name, or telephone number. Default: Enterprise
Directory User ID	BroadSoft User ID of the phone user; for example, johndoe@xdp.broadsoft.com. Default: Blank
Directory Password	Alphanumeric password associated with the User ID. Default: Blank

XML Service

Parameter	Description
XML Directory Service Name:	Name of the XML Directory. Displays on the user's phone as a directory choice Default: Blank
XML Directory Service URL	URL where the XML Directory is located. Default: Blank
XML User Name	XML service username for authentication purposes Default: Blank
XML Password	XML service password for authentication purposes Default: Blank

LDAP

Parameter	Description
LDAP Dir Enable	Choose Yes to enable LDAP. Default: No

Parameter	Description
Corp Dir Name	Enter a free-form text name, such as “Corporate Directory.” Default: Blank
Server	Enter a fully qualified domain name or IP address of an LDAP server in the following format: nnn.nnn.nnn.nnn Enter the host name of the LDAP server if the MD5 authentication method is used. Default: Blank
Search Base	Specify a starting point in the directory tree from which to search. Separate domain components [dc] with a comma. For example: dc=cv2bu,dc=com Default: Blank
Client DN	Enter the distinguished name domain components [dc]; for example: dc=cv2bu,dc=com If you are using the default Active Directory schema (Name(cn)->Users->Domain), an example of the client DN follows: cn=”David Lee”,dc=users,dc=cv2bu,dc=com Default: Blank
User Name	Enter the username for a credentialed user on the LDAP server. Default: Blank
Password	Enter the password for the LDAP username. Default: Blank
Auth Method	Select the authentication method that the LDAP server requires. Choices are: None—No authentication is used between the client and the server. Simple—The client sends its fully-qualified domain name and password to the LDAP server. Might present security issues. Digest-MD5—The LDAP server sends authentication options and a token to the client. The client returns an encrypted response that is decrypted and verified by the server. Default: None
Last Name Filter	This defines the search for surnames [sn], known as last name in some locations. For example, sn:(sn=*$VALUE*). This search allows the provided text to appear anywhere in a name: beginning, middle, or end. Default: Blank

Parameter	Description
First Name Filter	This defines the search for the common name [cn]. For example, cn:(cn=*$VALUE*). This search allows the provided text to appear anywhere in a name: beginning, middle, or end. Default: Blank
Search Item 3	Additional customized search item. Can be blank if not needed. Default: Blank
Search Item 3 Filter	Customized filter for the searched item. Can be blank if not needed. Default: Blank
Search Item 4	Additional customized search item. Can be blank if not needed. Default: Blank
Search Item 4 Filter	Customized filter for the searched item. Can be blank if not needed. Default: Blank

Parameter	Description
Display Attrs	Format of LDAP results displayed on phone, where: • a—Attribute name • cn—Common name • sn—Surname (last name) • telephoneNumber—Phone number • n—Display name For example, n=Phone causes “Phone:” to be displayed in front of the phone number of an LDAP query result when the detail soft button is pressed. • t—type When t=p, that is, t is of type phone number, the retrieved number can be dialed. Only one number can be made dialable. If two numbers are defined as dialable, only the first number is used. For example, a=ipPhone, t=p; a=mobile, t=p; This example results in only the IP Phone number being dialable and the mobile number is ignored. • p—phone number When p is assigned to a type attribute, example t=p, the retrieved number is dialable by the phone. For example, a=givenName,n=firstname;a=sn,n=lastname;a=cn,n=cn;a=telephoneNumber,n=tele,t=p Default: Blank
Number Mapping	Can be blank if not needed. **Note** With the LDAP number mapping, you can manipulate the number that was retrieved from the LDAP server. For example, you can append 9 to the number if your dial plan requires a user to enter 9 before dialing. Add the 9 prefix by adding (<:9xx.>) to the LDAP Number Mapping field. For example, 555 1212 would become 9555 1212. If you do not manipulate the number in this fashion, a user can use the Edit Dial feature to edit the number before dialing out. Default: Blank

Programmable Softkeys

Parameter	Description
Programmable Softkey Enable	Enables programmable softkeys.
Idle Key List	Softkeys that display when the phone is idle.
Off Hook Key List	Softkeys that display when the phone is off-hook.
Dialing Input Key List	Softkeys that display when the user must enter dialing data.
Progressing Key List	Softkeys that display when a call is attempting to connect.
Connected Key List	Softkeys that display when a call is connected.
Start-Xfer Key List	Softkeys that display when a call transfer has been initiated.
Start-Conf Key List	Softkeys that display when a conference call has been initiated.
Conferencing Key List	Softkeys that display when a conference call is in progress.
Releasing Key List	Softkeys that display when a call is released.
Hold Key List	Softkeys that display when one or more calls are on hold.
Ringing Key List	Softkeys that display when a call is incoming.
Shared Active Key List	Softkeys that display when a call is active on a shared line.
Shared Held Key List	Softkeys that display when a call is on hold on a shared line.
PSK 1 through PSK 16	Programmable softkey fields. Enter a string in these fields to configure softkeys that display on the phone screen. You can create softkeys for speed dials to numbers or extensions, vertical service activation codes (* codes), or XML scripts.

User

Hold Reminder

Parameter	Description
Hold Reminder Timer	Specifies the time delay (in seconds), that a ring splash is heard on an active call when another call was placed on hold. Default: 0

Parameter	Description
Hold Reminder Ringtone	Specifies the volume of the timer ringtone.

Call Forward

Parameter	Description
Cfwd Setting	Select **Yes** to enable call forwarding.
Cfwd All Dest	Enter the extensions to which the call is forwarded.
Cfwd Busy Dest	Enter the extensions to forward calls to when the line is busy. Default: voicemail
Cfwd No Ans Dest	Enter the extension to forward calls to when the call is not answered. Default: voicemail
Cfwd No Ans Delay	Enter the delay in time (in seconds) to wait before forwarding a call that is unanswered. Default: 20 seconds

Speed Dial

You can configure speed dials on the Cisco IP Phone from the LCD GUI or the web GUI.

Speed Dial 2 to 9: Target phone number (or URL) assigned to speed dial 2, 3, 4, 5, 6, 7, 8, or 9. Press the digit key (2-9) to dial out the assigned number.

Default: Blank

Supplementary Services

Parameter	Description
CW Setting	Enables or disables the Call Waiting service. Default: Yes
Block CID Setting	Enables or disables the Block CID service. Default: No
Block ANC Setting	Enables or disables the Block ANC service. Default: No
DND Setting	Enables or disables the DND settings options for a user.

Parameter	Description
Handset LED Alert	Enables or disables LED alert on the handset. Options are: Voicemail and Voicemail, Missed Call. Default: Voicemail
Secure Call Setting	Enables or disables Secure Call. Default: No
Auto Answer Page	Enables or disables automatic answering of paged calls. Default: Yes
Preferred Audio Device	Choose the type of audio that the phone will use. Options are: Speaker and Headset. Default: None
Time Format	Choose the time format for the phone (12 or 24 hour). Default: 12hr
Date Format	Choose the date format for the phone (month/day or day/month). Default: month/day
Miss Call Shortcut	Enables or disables the option for creating a missed call shortcut.
Alert Tone Off	Enables or disables the alert tone.
Log Missed Calls for EXT (n)	Enables or disables the missed calls logs for a specific extension.
Shared Line DND Cfwd Enable	Enable/disable the Shared Line DND Call Forward.

Audio

Parameter	Description
Ringer Volume	Sets the default volume for the ringer. Default: 9
Speaker Volume	Sets the default volume for the speakerphone. Default: 8
Handset Volume	Sets the default volume for the handset. Default: 10
Headset Volume	Sets the default volume for the headset. Default: 10

LCD

Parameter	Description
Back Light Timer (minutes)	Select the number of minutes before the back light should turn off (1m, 5m, or 30m) or Always On. Default: 5m
Brightness	Enter a number value from 1 to 15. The higher the number, the greater the brightness on the IP phone screen. Default: 10

Extension

Extension

In a configuration profile, the Line parameters must be appended with the appropriate numeral to indicate the line to which the setting applies. For example:

```
[1] to specify line one
[2] to specify line two
```

General

Parameter	Description
Line Enable	To enable this line for service, select yes. Otherwise, select No. Default: Yes

Share Line Appearance

Parameter	Description
Share Ext	Indicates whether this extension is to be shared with other Cisco IP phones or private. Default: Yes
Shared User ID	The user identified assigned to the shared line appearance. Default: Blank
Subscription Expires	Number of seconds before the SIP subscription expires. Before the subscription expiration, the phone gets NOTIFY messages from the SIP server on the status of the shared phone extension. Default: 3600

Parameter	Description
Restrict MWI	When enabled, the message waiting indicator lights only for messages on private lines. Default: No

NAT Settings

Parameter	Description
NAT Mapping Enable	To use externally mapped IP addresses and SIP/ RTP ports in SIP messages, select yes. Otherwise, select no. Default: No
NAT Keep Alive Enable	To send the configured NAT keep alive message periodically, select yes. Otherwise, select no. Default: No
NAT Keep Alive Msg	Enter the keep alive message that should be sent periodically to maintain the current NAT mapping. If the value is $NOTIFY, a NOTIFY message is sent. If the value is $REGISTER, a REGISTER message without contact is sent. Default: $NOTIFY
NAT Keep Alive Dest	Destination that should receive NAT keep alive messages. If the value is $PROXY, the messages are sent to the current or outbound proxy.

Network Settings

Parameter	Description
SIP TOS/DiffServ Value	Time of service (ToS)/differentiated services (DiffServ) field value in UDP IP packets carrying a SIP message. Defaults to 0x68.
RTP ToS/DiffServ Value	ToS/DiffServ field value in UDP IP packets carrying RTP data. Defaults to 0xb8.

SIP Settings

Parameter	Description
SIP Transport	Select from **UDP**, **TCP**, or **TLS**. Default: UDP
SIP Port	Port number of the SIP message listening and transmission port. Default: 5060

Parameter	Description
SIP 100REL Enable	Support of 100REL SIP extension for reliable transmission of provisional responses (18x) and use of PRACK requests. Select **Yes** to enable. Default: No
EXT SIP Port	The external SIP port number.
Auth Resync-Reboot	The Cisco IP Phone authenticates the sender when it receives a NOTIFY message with the following requests: • resync • reboot • report • restart • XML-service Select **Yes** to enable. Default: Yes
SIP Proxy-Require	The SIP proxy can support a specific extension or behavior when it sees this header from the user agent. If this field is configured and the proxy does not support it, it responds with the message, unsupported. Enter the appropriate header in the field provided.
SIP Remote-Party-ID	The Remote-Party-ID header to use instead of the From header. Select **Yes** to enable. Default: Yes
Referor Bye Delay	Controls when the phone sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referor, Refer Target, Referee, and Refer-To Target) are configured on this screen. For the Referror Bye Delay, enter the appropriate period of time in seconds. Default: 4
Refer-To Target Contact	Indicates the refer-to target. Select **Yes** to send the **SIP Refer** to the contact. Default: No
Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. Default: 0

Parameter	Description
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. Default: 0
Sticky 183	When enabled, the IP telephony ignores further 180 SIP responses after receiving the first 183 SIP response for an outbound INVITE. To enable this feature, select **Yes**. Otherwise, select **No**. Default: No
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy. To enable this feature, select **Yes**. Default: No
Ntfy Refer On 1xx-To-Inv	If set to **Yes**, as a transferee, the phone will send a NOTIFY with Event:Refer to the transferor for any 1xx response returned by the transfer target, on the transfer call leg. If set to **No**, the phone will only send a NOTIFY for final responses (200 and higher).
Set G729 annexb	Configure G.729 Annex B settings.
Set iLBC mode	Select iLBC 20ms or 30ms frame size mode. Default: 20
User Equal Phone	When a tel URL is converted to a SIP URL and the phone number is represented by the user portion of the URL, the SIP URL includes the optional : user=phone parameter (RFC3261). For example: To: sip:+12325551234@example.com; user=phone To enable this optional parameter, select **Yes**. Default: No

Call Feature Settings

Parameter	Description
Blind Attn-Xfer Enable	Enables the phone to perform an attended transfer operation by ending the current call leg and performing a blind transfer of the other call leg. If this feature is disabled, the phone performs an attended transfer operation by referring the other call leg to the current call leg while maintaining both call legs. To use this feature, select Yes. Otherwise, select No. Default: No

Parameter	Description
Message Waiting	Indicates whether the Message Waiting Indicator on the phone is lit. This parameter toggles a message from the SIP proxy to indicate if a message is waiting.
Auth Page	Specifies whether to authenticate the invite before auto answering a page. Default: No
Default Ring	Type of ring heard. Choose from No Ring or 1 through 10. Ring options are Sunlight, Chirp 1, Chirp 2, Delight, Evolve, Mellow, Mischief, Reflections, Ringer, Ascent, Are you there, and Chime.
Auth Page Realm	Identifies the Realm part of the Auth that is accepted when the Auth Page parameter is set to Yes. This parameter accepts alphanumeric characters.
Conference Bridge URL	URL used to join into a conference call, generally in the form of the word conference or user@IPaddress:port.
Auth Page Password	Identifies the password used when the Auth Page parameter is set to Yes. This parameter accepts alphanumeric characters.
Mailbox ID	Identifies the voice mailbox number/ID for the phone.
Voice Mail Server	Identifies the SpecVM server for the phone, generally the IP address, and port number of the VM server.
Voice Mail Subscribe Interval	The expiration time, in seconds, of a subscription to a voice mail server.
Broadsoft ACD	Enables support for basic BroadSoft Automatic CallDistribution (ACD). The supported values for this option are Yes and No. Default: No
Auto Ans Page On Active Call	Determines the behavior of the phone when a page call arrives.
Feature Key Sync	Enable/disable the Feature Key synchronization. Applies to DND and Call Forward All features.
Call Park Monitor Enable	BroadSoft server only specific feature. If call park is enabled on the server or on any of the programmable line key, you need to enable this field for call park notification to work. Default: No
Enable Broadsoft Hoteling	When this parameter is set to yes, the phone sends out subscription message (without body) to the server. Default: No

Parameter	Description
Hoteling Subscription Expires	An expiration value that is added in the subscription message. Default value is 3600.

Proxy and Registration

Parameter	Description
Proxy	SIP proxy server and port number set by the service provider for all outbound requests. For example: 192.168.2.100:6060. The port number is optional. Default: 5060
Outbound Proxy	All outbound requests are sent as the first hop. Enter an IP address or domain name.
Alternate Proxy Alternate Outbound Proxy	This feature provides fast fall back when there is network partition at the Internet or when the primary proxy (or primary outbound proxy) is not responsive or available. The feature works well in a Verizon deployment environment as the alternate proxy is the Integrated Service Router (ISR) with analog outbound phone connection. Enter the proxy server addresses and port numbers in these fields. After the phone is registered to the primary proxy and the alternate proxy (or primary outbound proxy and alternate outbound proxy), the phone always sends out INVITE and Non-INVITE SIP messages (except registration) via the primary proxy. The phone always registers to both the primary and alternate proxies. If there is no response from the primary proxy after timeout (per the SIP RFC spec) for a new INVITE, the phone attempts to connect with the alternate proxy. The phone always tries the primary proxy first, and immediately tries the alternate proxy if the primary is unreachable. Active transactions (calls) never fall back between the primary and alternate proxies. If there is fall back for a new INVITE, the subscribe/notify transaction will fall back accordingly so that the phone's state can be maintained properly. You must also set Dual Registration in the Proxy and Registration section to Yes.
Use OB Proxy In Dialog	Determines whether to force SIP requests to be sent to the outbound proxy within a dialog. Ignored if the **Use Outbound Proxy** field is set to **No** or if the **Outbound Proxy** field is empty. Default: Yes
Register	Enables periodic registration with the proxy. This parameter is ignored if a proxy is not specified. To enable this feature, select **Yes**. Default: Yes

Parameter	Description
Make Call Without Reg	Enables making outbound calls without successful (dynamic) registration by the phone. If set to No, the dial tone plays only when registration is successful. To enable this feature, select **Yes**. Default: No
Register Expires	Defines how often the phone renews registration with the proxy. If the proxy responds to a REGISTER with a lower expires value, the phone renews registration based on that lower value instead of the configured value. If registration fails with an "Expires too brief" error response, the phone retries with the value specified in the Min-Expires header of the error. The range is from 32 to 2000000. Default: 3600 seconds
Ans Call Without Reg	If enabled, the user does not have to be registered with the proxy to answer calls. Default: No
Use DNS SRV	Enables DNS SRV lookup for the proxy and outbound proxy. To enable this feature, select **Yes**. Otherwise, select **No**. Default: No
DNS SRV Auto Prefix	Enables the phone to automatically prepend the proxy or outbound proxy name with _sip._udp when performing a DNS SRV lookup on that name. Default: No
Proxy Fallback Intvl	Sets the delay after which the phone retries from the highest priority proxy (or outbound proxy) after it has failed over to a lower priority server. The phone should have the primary and backup proxy server list from a DNS SRV record lookup on the server name. It needs to know the proxy priority; otherwise, it does not retry. The range is from 0 to 65535. Default: 3600 seconds
Proxy Redundancy Method	Select **Normal** or **Based on SRV Port**. The phone creates an internal list of proxies returned in the DNS SRV records. If you select Normal, the list contains proxies ranked by weight and priority. If you select Based on SRV Port, the phone uses normal, then inspects the port number based on the first-listed proxy port. Default: Normal

Parameter	Description
Dual Registration	Set to **Yes** to enable the Dual registration/Fast Fall back feature. To enable the feature you must also configure the alternate proxy/alternate outbound proxy fields in the Proxy and Registration section.
Auto Register When Failover	If set to No, the fallback happens immediately and automatically. If the Proxy Fallback Intvl is exceeded, all the new SIP messages go to the primary proxy. If set to Yes, the fallback happens only when current registration expires, which means only a REGISTER message can trigger fallback. For example, when the value for Register Expires is 3600 seconds and Proxy Fallback Intvl is 600 seconds, the fallback is triggered 3600 seconds later and not 600 seconds later. When the value for Register Expires is 600 seconds and Proxy Fallback Intvl is 1000 seconds, the fallback is triggered at 1200 seconds. After successfully registering back to primary server, all the SIP messages go to primary server.

Subscriber Information

Parameter	Description
Display Name	Name displayed as the caller ID.
User ID	Extension number for this line.
Password	Password for this line. Default: Blank (no password required)
Auth ID	Authentication ID for SIP authentication. Default: Blank
SIP URI	The parameter by which the user agent will identify itself for this line. If this field is blank, the actual URI used in the SIP signaling should be automatically formed as: sip:UserName@Domain where UserName is the username given for this line in the User ID, and Domain is the domain given for this profile in the User Agent Domain. If the User Agent Domain is an empty string, then the IP address of the phone should be used for the domain. If the URI field is not empty, but if a SIP or SIPS URI contains no @ character, the actual URI used in the SIP signaling should be automatically formed by appending this parameter with an @ character followed by the IP address of the device.

Audio Configuration

Parameter	Description
Preferred Codec	Preferred codec for all calls. The actual codec used in a call still depends on the outcome of the codec negotiation protocol. Select one of the following: • G711u • G711a • G729a • G729ab • G722 • G722.2 • iLBC • OPUS • iSAC Default: G711u
Use Pref Codec Only	Select **No** to use any code. Select **Yes** to use only the preferred codes. When you select Yes, calls fail if the far end does not support the preferred codecs. Default: No
Second Preferred Codec	Codec to use if the first codec fails. Default: Unspecified
Third Preferred Codec	Codec to use if the second codec fails. Default: Unspecified
G711u Enable	Enables use of the G.711u codec. Default: Yes
G711a Enable	Enables use of the G.711a codec. Default: Yes
G729a Enable	To enable use of the G.729a codec at 8 kbps, select **Yes**. Otherwise, select **No**. Default: Yes
G722 Enable	Enables use of the G.722 codec. Default: Yes

Parameter	Description
G722.2 Enable	Enables use of the G.722.2 codec. Default: No
iLBC Enable	Enables use of the iLBC codec. Default: Yes
OPUS Enable	Enables the use of OPUS codec. Default: Yes
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select **Yes**. Otherwise, select **No**. Default: No
DTMF Tx Method	The method for transmitting DTMF signals to the far end. The options are: • AVT—Audio video transport. Sends DTMF as AVT events. • InBand—Sends DTMF by using the audio path. • Auto—Uses InBand or AVT based on the outcome of codec negotiation. • INFO—Uses the SIP INFO method.
Use Remote Pref Codec	Lists all codecs or it uses the default codecs supported. Default: Default.
Codec Negotiation	When set to Default, the Cisco IP phone responds to an Invite with a 200 OK response advertising the preferred codec only. When set to List All, the Cisco IP phone responds listing all the codecs that the phone supports. The default value is Default, or to respond with the preferred codec only.
Encryption Method	Encryption method to be used during secured call. Options are AES 128 and AES 256 GCM Default: 128.

Dial Plan

Parameter	Description
Dial Plan	Dial plan script for the selected extension. The dial plan syntax allows the designation of three parameters for use with a specific gateway: • uid – The authentication user-id • pwd – The authentication password • nat – If this parameter is present, use NAT mapping. Separate each parameter with a semi-colon (;).
Caller ID Map	Inbound caller ID numbers can be mapped to a different string. For example, a number that begins with +44xxxxxx can be mapped to 0xxxxxx. This feature has the same syntax as the Dial Plan parameter. With this parameter, you can specify how to map a caller ID number for display on screen and recorded into call logs.
Enable URI Dialing	Enables or disables URI dialing.
Emergency Number	Enter a comma-separated list of emergency numbers. When one of these numbers is dialed, the unit disables processing of CONF, HOLD, and other similar softkeys or buttons to avoid accidentally putting the current call on hold. The phone also disables hook flash event handling. Only the far end can terminate an emergency call. The phone is restored to normalcy after the call is terminated and the receiver is back on-hook. Maximum number length is 63 characters. Defaults to blank (no emergency number).

Att Console

General

Note The attendant console tab, labeled **Att Console**, is only available in **Admin Login** > **advanced** mode.

Parameter	Description
Subscribe Expires	Specifies how long the subscription remains valid. After the specified period of time elapses, the Cisco Attendant Console initiates a new subscription. Default: 1800

Parameter	Description
Subscribe Retry Interval	Specifies the length of time to wait to try again if the subscription fails. Default: 30
Subscribe Delay	Length of delay before attempting to subscribe. Default: 1
BLF List URL	Domain name or user name that is defined in the Broadsoft server for the phone. Default: Blank
Use Line Keys For BLF List	Options to enable or disable the line keys for BLF. Default: No
Call Pickup Audio Notification	By default, this parameter is set to **No**. If you set it to **Yes**, the phone plays the Call Pickup tone when there are incoming calls to any of the lines that the user is monitoring with the Call Pickup function. Default: No
BXfer to Starcode Enable	When set to **Yes**, the phone performs a blind transfer when the *code is defined in a speed dial extended function,. If set to **No**, the current call is held and a new call is started to the speed dial destination. Default: No
BXfer On Speed Dial Enable	When set to **Yes**, the phone performs a blind transfer when the speed dial function key is selected. When set to no, the current connected call is held and a new call to the speed dial destination is started. For example, when a user parks a call using the speed dial function, if the parameter is enabled, a blind transfer is performed to the parking lot. If the parameter is not enabled, an attended transfer is performed to the parking lot. Default: No
BLF Label Display Mode	Options to select a mode which displays on the phone screen for BLF. Default: Blank

Unit

Enter the programming information for each line key for the Attendant Console unit.

Parameter	Description
Unit Enable	Indicates whether the key expansion module that is added to the phone is enabled.

Parameter	Description
Unit Online	Indicates whether the key expansion module that is added to the phone is active.
HW Version	Displays the hardware version of the key expansion module that is added to the phone..
SW Version	Displays the software version of the key expansion module that is added to the phone.

TR-069

TR-069

Parameter	Description
Enable TR-069	Settings that enables or disables the TR-069 function.
ACS URL	URL of the ACS that uses the CPE WAN Management Protocol. This parameter must be in the form of a valid HTTP or HTTPS URL. The host portion of this URL is used by the CPE to validate the ACS certificate when it uses SSL or TLS.
ACS Username	Username that authenticates the CPE to the ACS when ACS uses the CPE WAN Management Protocol. This username is used only for HTTP-based authentication of the CPE. If the user name is not configured, admin is used as default.
ACS Password	Password to access to the ACS for a specific user. This password is used only for HTTP-based authentication of the CPE. If the password is not configured, admin is used as default.
ACS URL In Use	URL of the ACS that is currently in use. This is a read-only field.
Connection Request URL	URL of the ACS that makes the connection request to the CPE.

Parameter	Description
Connection Request Username	Username that authenticates the ACS that makes the connection request to the CPE.
Connection Request Password	Password used to authenticate the ACS that makes a connection request to the CPE.
Periodic Informal Interval	Duration in seconds of the interval between CPE attempts to connect to the ACS when Periodic Inform Enable is set to yes. Default value is 20 seconds.
Periodic Inform Enable	Settings that enables or disables the CPE connection requests. Default value is Yes.
TR-069 Traceability	Settings that enables or disables TR-069 transaction logs. The default value is No.
CWMP V1.2 Support	Settings that enables or disables CPE WAN Management Protocol (CWMP) support. If set to disable, the phone does not send any Inform messages to the ACS nor accept any connection requests from the ACS. Default value is Yes.
TR-069 VoiceObject Init	Settings to modify voice objects. Select Yes to initialize all voice objects to factory default values or select No to retain the current values.
TR-069 DHCPOption Init	Settings to modify DHCP settings. Select Yes to initialize the DHCP settings from the ACS or select No to retain the current DHCP settings.
TR-069 Fallback Support	Settings that enables or disables the TR-069 fallback support. If the phone attempts to discover the ACS with DHCP and is unsuccessful, the phone next uses DNS to resolve the ACS IP address.
BACKUP ACS URL	Backup URL of the ACS that uses the CPE WAN Management Protocol. This parameter must be in the form of a valid HTTP or HTTPS URL. The host portion of this URL is used by the CPE to validate the ACS certificate when it uses SSL or TLS.

Parameter	Description
BACKUP ACS User	Backup username that authenticates the CPE to the ACS when ACS uses the CPE WAN Management Protocol. This username is used only for HTTP-based authentication of the CPE.
BACKUP ACS Password	Backup password to access to the ACS for a specific user. This password is used only for HTTP-based authentication of the CPE.
Note If you do not configure the above parameters, you can also fetch them through DHCP options 60,43, and 125.	

Call History

Displays the call history for the phone. To change the information displayed, select the type of call history from the following tabs:

- All Calls
- Missed
- Received
- Placed

Select **Add to Directory** to add the call information to your Personal Directory.

Personal Directory

The Personal Directory allows a user to store a set of personal numbers. Directory entries can include the following contact information:

- No. (the directory number)
- Name
- Work
- Mobile
- Home
- Speed Dials

To edit contact information, click **Edit Contacts**.