Monitoring Phone Systems

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

Include a Device Identifier in Uploaded Syslog Messages

You can choose to include a device identifier in syslog messages that are uploaded to the syslog server. While the IP address of a phone may change over time, the device identifier does not change. This can ease the process of identifying the source of each message in a stream of incoming messages from multiple phones. The device identifier appears after the timestamp in each message.

Before you begin

Configure a syslog sever for the phone to upload syslog messages. See **Syslog Server** in Optional Network Configuration, on page 24 for details.

Procedure

Step 1 On the phone administration web page, go to **Voice** > **System** > **Optional Network Configuration**.

Step 2 Configure the **Syslog Identifier** parameter as described in Optional Network Configuration, on page 24.

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 **Applications** .

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 **Back**.

View the Phone Status

Procedure

Step 1 Press **Applications** .

Step 2 Select **Status** > **Phone 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 **Applications** .

Step 2 Select **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 **Applications** .

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.
- **IPv4 status**—IP address of the phone. You can see information on IP address, Addressing type, IP status, Subnet mask, Default router, Domain Name Server (DNS) 1, DNS 2 of the phone.
- **IPv6 status** —IP address of the phone. You can see information on IP address, Addressing type, IP status, Subnet mask, Default router, Domain Name Server (DNS) 1, DNS 2 of the phone.
- **VLAN ID**—VLAN ID of the phone.
- **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.

- **PC port config**—Indicates speed and duplex of the PC port.
- **PC port link**—Indicates speed and duplex of the PC 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 **Applications** .

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

Step 3 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 • iLBC

Item	Description
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 • 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.
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.

Item	Description
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.

View the Customization State in the Configuration Utility

After the RC download from the EDOS server completes, you can view the customization state of a phone using the web interface.

Here are the descriptions of the remote customization states:

- Open—The phone has booted for the first time and is not configured.
- Aborted—Remote customization is aborted due to other Provisioning like DHCP options.
- Pending—The profile has been downloaded from the EDOS server.
- Custom-Pending—The phone has downloaded a redirect URL from the EDOS server.
- Acquired—In the profile downloaded from the EDOS server, there is a redirect URL for provision configuration. If the redirect URL download from the provisioning server is successful, this state is displayed.
- Unavailable—Remote customization has stopped because the EDOS server responded with an empty provisioning file and the HTTP response was `200 OK`.

Procedure

Step 1 On the Phone Web page, select **Admin Login** > **Info** > **Status**.

Step 2 In the **Product Information** section, you can view the customization state of the phone in the **Customization** field.

If any provisioning is failing, you can view the details in the **Provisioning Status** section on the same page.

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 Phone Web Page
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 Server	Displays the secondary NTP server assigned to the phone.

IPv4 Information

Parameter	Description
IP Status	Indicates that the connection is established.

Parameter	Description
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.

IPv6 Information

Parameter	Description
IP Status	Indicates that the connection is established.
Connection Type	Indicates the type of internet connection for the phone: • Static IP • DHCP
Current IP	Displays the current IPv6 address assigned to the IP phone.
Prefix Length	Identifies number of bits of a global unicast IPv6 address that are part of the network. For example, if the IPv6 address is 2001:0DB8:0000:000b::/64, the number 64 identifies that the first 64 bits are part of the network.
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 Reboot Reasons.

Product Information

Parameter	Description
Product Name	Model number of the phone.
Software Version	Version number of the phone firmware.
MAC Address	Hardware address of the 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 phone.
Hardware Version	Version number of the phone hardware.
Client Certificate	Status of the client certificate, which authenticates the 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
Locale download status	Displays the downloaded locale package status.
Locale download URL	Displays the location from where the local package is downloaded.
Font download status	Displays the downloaded font file status.
Font download URL	Displays the location from where the font file 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).

Parameter	Description
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.
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.
PC Port	Displays the type of Ethernet connection from PC Port.
Upgrade Status	Displays status of the last phone upgrade.
SW Port Config	Displays the type of SW port configuration.
PC Port Config	Displays the type of PC 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.

Parameter	Description
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.
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.

Parameter	Description
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).
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).

Paging Status

Parameter	Description
Multicast Rx Pkts	Indicates Rx packets during a multicast paging.
Multicast Tx Pkts	Indicates Tx packets during a multicast paging.

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

PRT Status

Parameter	Description
PRT Generation Status	The location of initiation and status of generation of the most recently initiated problem report. Problem reports may be initiated from the phone LCD user interface, from the phone administration web page, or remotely. See Report All Phone Issues from the Phone Web Page and Report a Phone Problem Remotely for details. XML tag in `status.xml`: `PRT_Generation_Status`
PRT Upload Status	The status of upload of the most recently initiated problem report. See Configure PRT Upload for information on configuring an upload rule for problem reports. XML tag in `status.xml`: `PRT_Upload_Status`

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; • 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 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). Each entry gives the status, 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 the time zone settings.

Parameter	Description
Debug Message	Displays debug messages when you click **messages** link.

Problem Reports

Parameter	Description
Report Problem	Displays the tab Generate PRT.
Prt file	Displays the file name of the PRT logs.
Packet Capture	Displays the tab **Start Packet Capture**. Click this tab to initiate capture packets. Click **All** to capture all packets that the phone receives or click **Host IP Address** to capture packets only when src/dest is the IP address of the phone. You can also stop the capture process after initiating it.
Capture File	Displays the file that contains the captured packets. Download the file to see the packet details.

Factory Reset

Parameter	Description
Factory Reset	Resets the phone when you click **Factory Reset** tab when the phone is idle.

Download Status

Firmware Upgrade Status

Parameter	Description
Firmware Upgrade Status 1	Displays the upgrade status (failed or succeeded) with reason for the same.
Firmware Upgrade Status 2	
Firmware Upgrade Status 3	

Provisioning Status

Parameter	Description
Provisioning Status 1	Displays the provisioning status (resync) of the phone.
Provisioning Status 2	
Provisioning Status 3	

Custom CA Status

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; • 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.

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.
RxBroadcasts	Total number of broadcast packets that the phone received.

Parameter	Description
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.
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.

Parameter	Description
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.

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
Enable Protocol	Choose the type of protocol: • Http • Https If you specify the HTTPS protocol, you must include `https:` in the URL. Default: Http
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 • 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

Parameter	Description
Admin Password	Allows you to enter password for the administrator. Default: Blank
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. Default: No
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”, “ro”, or “rw” using provisioning files. Parameters designated as “na” don't appear on the phone screen. Parameters designated as “ro” aren't editable by the user. Parameters designated as “rw” are editable by the user.
Block Nonproxy SIP	Enables or disables the phone receiving SIP messages from non-proxy server. If you choose **Yes**, the phone blocks any incoming non-proxy SIP messages except IN-dialog message. If you choose **No**, the phone does not block any incoming non-proxy SIP messages. Set **Block Nonproxy SIP** to No for phones that use TCP or TLS to transport SIP messages. Nonproxy SIP messages transported over TCP or TLS are blocked by default. Default: No

Network Settings

Parameter	Description
IP Mode	Allows you to select the internet protocol mode in which the phone operates. Options are: IPv4 Only, IPv6 Only, and Dual Mode. In dual mode, the phone can have both IPv4 and IPv6 addresses. Default: Dual Mode

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.

IPv6 Settings

Parameter	Description
Connection Type	Internet connection type that is configured for the phone. Options are DHCP and Static IP. Default: DHCP
Static IP	IPv6 address of the phone.
Prefix Length	Identifies number of bits of a global unicast IPv6 address that are part of the network. For example, if the IPv6 address is 2001:0DB8:0000:000b::/64, the number 64 identifies that the first 64 bits are part of the network.
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.
Broadcast Echo	Options are Disabled and Enabled. Default: Disabled
Auto Config	When enabled, phone generates an IPv6 address by default with the prefix length sent from the router. Options are Disabled and Enabled. Default: Enabled

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 LDAP Configuration.
DNS Server Order	Specifies the method for selecting the DNS server: • Manual, DHCP • Manual • DHCP,Manual
DNS Query Mode	Specified mode of DNS query. • Parallel • 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

Parameter	Description
PC Port Config	Allows you to select Speed and duplex of the Computer (access) port. • Auto • 10MB half • 10MB full • 100 MB half • 100MB full • 100 half • 1000 full
PC PORT Enable	Specifies if PC port is enabled. Options are Yes or No.
Enable PC Port Mirror	Adds the ability to port mirror on the PC port. When enabled, you can see the packets on the phone. Select **Yes** to enable PC port mirroring and select **No** to disable it.
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.

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>Syslog Identifier</td><td>Select the device identifier to include in syslog messages that are uploaded to the syslog server. The device identifier appears after the timestamp in each message.
None: No device identifier.
$MA: The MAC address of the phone, expressed as continuous lower case letters and digits. Example: <code>c4b9cd811e29</code>
$MAU: The MAC address of the phone, expressed as continuous upper case letters and digits. Example: <code>C4B9CD811E29</code>
$MAC: The MAC address of the phone in the standard colon-separated format. Example: <code>c4:b9:cd:81:1e:29</code>
$SN: The product serial number of the phone.
Default: None
Example XML configuration:
<code><Syslog_Identifier ua="na">$MAC</Syslog_Identifier></code></td></tr>
<tr><td>Debug Level</td><td>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</td></tr>
<tr><td>Primary NTP Server</td><td>IP address or name of the primary NTP server used to synchronize its time.
Default: Blank</td></tr>
<tr><td>Secondary NTP Server</td><td>IP address or name of the secondary NTP server used to synchronize its time.
Default: Blank</td></tr>
<tr><td>Enable SSLv3</td><td>Choose Yes to enable SSLv3. Choose No to disable.
Default: No</td></tr>
</table>

VLAN Settings

Parameter	Description
Enable VLAN	Choose **Yes** to enable VLAN. Choose **No** to disable.

Parameter	Description
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.
PC Port VLAN ID	VLAN ID for the PC port.
DHCP VLAN Option	A predefined DHCP VLAN option to learn the voice VLAN ID. You can use the feature only when no voice VLAN information is available by CDP/LLDP and manual VLAN methods. CDP/LLDP and manual VLAN are all disabled. Valid values are: • Null • 128 to 149 • 151 to 158 • 161 to 254 Set the value to Null to disable DHCP VLAN option. Cisco recommends that you use DHCP Option 132.

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

Parameter	Description
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.
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

Parameter	Description
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
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.

Parameter	Description
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 IP Preference	Sets if the phone uses IPv4 or IPv6. Default: IPv4.

SIP Timer Values (sec)

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.

Parameter	Description
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.
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.

Parameter	Description
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
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. The maximum value for the RTP port must be lesser than 49152. 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

Parameter	Description
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 IP Preferences	Select IPv4 or IPv6. Default: IPv4 If the phone is in dual-mode and has both ipv4 and ipv6 addresses, it will always include both addresses in SDP by attributes "a=altc … If IPv4 address is selected, then ipv4 address has higher priority than ipv6 address in SDP and indicates that phone prefers using ipv4 RTP address. If the phone has only ipv4 address or ipv6 address, SDP does not have ALTC attributes and RTP address is specified in “c=” line.

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.
H264 BP0 Dynamic Payload	H264 BPO Dynamic Payload type. Default: 110

Parameter	Description
H264 HP Dynamic Payload	H264 HP Dynamic Payload type. Default: 110
G711u Codec Name	G711u codec name used in SDP. Default: PCMU
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

Parameter	Description
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
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

Parameter	Description
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
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 power 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 failure. The value for this field must be an integer ranging between 0 and 65535. The default value is 2.

Parameter	Description
Resync At (HHmm)	The time (HHmm) that the device resynchronizes with the provisioning server. The value for this field must be a four-digit number ranging from 0000 to 2400 to indicate the time in HHmm format. For example, 0959 indicates 09:59. The default value is empty. If the value is invalid, the parameter is ignored. If this parameter is set with a valid value, the Resync Periodic parameter is ignored.
Resync At Random Delay	Prevents an overload of the provisioning server when a large number of devices power-on simultaneously. To avoid flooding resync requests to the server from multiple phones, the phone resynchronizes in the range between the hours and minutes, and the hours and minutes plus the random delay (hhmm, hhmm+random_delay). For example, if the random delay = (Resync At Random Delay + 30)/60 minutes, the input value in seconds is converted to minutes, rounding up to the next minute to calculate the final random_delay interval. The valid value ranges between 0 and 65535. This feature is disabled when this parameter is set to zero. The default value is 600 seconds (10 minutes).

Parameter	Description
Resync Periodic	The time interval between periodic resynchronizes with the provisioning server. The associated resync timer is active only after the first successful sync with the server. The valid formats are as follows: • An integer Example: An input of `3000` indicates that the next resync occurs in 3000 seconds. • Multiple integers Example: An input of `600,1200,300` indicates that the first resync occurs in 600 seconds, the second resync occurs in 1200 seconds after the first one, and the third resync occurs in 300 seconds after the second one. • A time range Example, an input of `2400+30` indicates that the next resync occurs in between 2400 and 2430 seconds after a successful resync. Set this parameter to zero to disable periodic resynchronization. The default value is 3600 seconds.

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, or the downloaded file is corrupt, or an internal error occurs, the device tries to resync again after a time specified in seconds. The valid formats are as follows: • An integer Example: An input of `300` indicates that the next retry for resync occurs in 300 seconds. • Multiple integers Example: An input of `600,1200,300` indicates that the first retry occurs in 600 seconds after the failure, the second retry occurs in 1200 seconds after the failure of the first retry, and the third retry occurs in 300 seconds after the failure of the second retry. • A time range Example, an input of `2400+30` indicates that the next retry occurs in between 2400 and 2430 seconds after a resync failure. If the delay is set to 0, the device does not try to resync again following a failed resync attempt.
Forced Resync Delay	Maximum delay (in seconds) the phone waits before performing a resynchronization. The device does not resync while one of its phone lines is active. Because a resync can take several seconds, it is desirable to wait until the device has been idle for an extended period before resynchronizing. This allows a user to make calls in succession without interruption. The device has a timer that begins counting down when all of its lines become idle. This parameter is the initial value of the counter. Resync events are delayed until this counter decrements to zero. The valid value ranges between 0 and 65535. The default value is 14,400 seconds.

Parameter	Description
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

Parameter	Description

Profile Authentication Type	Specifies the credentials to use for profile account authentication. The available options are:
	• **Disabled**: Disables the profile account feature. When this feature is disabled, the **Profile account setup** menu doesn't display on the phone screen.
	• **Basic HTTP Authentication**: The HTTP login credentials are used to authenticate the profile account.
	• **XSI Authentication**: XSI login credentials or XSI SIP credentials are used to authenticate the profile account. The authentication credentials depend on the **XSI Authentication Type** for the phone:
	    • When the **XSI Authentication Type** for the phone is set to **Login Credentials**, the XSI login credentials are used.
	    • When the **XSI Authentication Type** for the phone is set to **SIP Credentials**, the XSI SIP credentials are used.
	
	Default: Basic HTTP Authentication

Profile Rule Profile Rule B Profile Rule C Profile Rule D	Each profile rule informs the phone of a source from which to obtain a profile (configuration file). During every resync operation, the phone applies all the profiles in sequence. Default: **/$PSN.xml** If you are applying AES-256-CBC encryption to the configuration files, specify the encryption key with the **--key** keyword as follows: **[--key <encryption key>]** You can enclose the encryption key in double-quotes (") optionally.
DHCP Option To Use	DHCP options, delimited by commas, used to retrieve firmware and profiles. Default: 66,160,159,150,60,43,125
DHCPv6 Option To Use	DHCP options, delimited by commas, used to retrieve firmware and profiles. Default: 17,160,159

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`
User Configurable Resync	Allows a user to resync the phone from the phone screen. Default: Yes

Upload Configuration Options

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Report Rule</td><td>
Specifies how the phone reports its current internal configuration to the provisioning server. The URLs in this field specify the destination for a report and can include an encryption key.
You can use the following keywords, encryption key, and file locations and names to control how you store the phone configuration information:

No keywords and only an XML file reports the entire configuration data to server.
[--status] keyword reports the status data to server.
[--delta] keyword reports the changed configuration to server.
[--key <encryption key>] keyword tells the phone to apply AES-256-CBC encryption with the specified encryption key to the configuration report, before sending it to the server.
You can enclose the encryption key in double-quotes (") optionally.
Note If you have provisioned the phone with Input Keying Material (IKM) and want the phone to apply RFC 8188-based encryption to the file, do not specify a AES-256-CBC encryption key.
Two rules used together as:
[--delta]http://my_http_server/config-mpp-delta.xml
[--status]http://my_http_server/config-mpp-status.xml

Caution If you need to use the [--delta]xml-delta file rule and the [--status]xml-status file rule together, you must separate the two rules with a space.
</td></tr>
<tr><td>HTTP Report method:</td><td>
Specifies whether the HTTP Request that the phone sends should be an HTTP PUT or an HTTP POST.

PUT Method–To create a new report or overwrite an existing report at a known location on the server. For example, you may want to keep overwriting each report that you send and only store the most current configuration on the server.
POST Method–To send the report data to the server for processing, such as, by a PHP script. This approach provides more flexibility for storing the configuration information. For example, you may want to send a series of phone status reports and store all the reports on the server.

</td></tr>
</table>

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Report to Server:</td><td>Defines when the phone reports its configuration to the provisioning servers.
On Request: The phone reports its configuration only when an administrator sends a sip notify event, or the phone restarts.
On Local Change: The phone reports its configuration when any configuration parameter changes by an action on the phone or on the phone administration web page. The phone waits for a few seconds after a change is made, and then reports the configuration. This delay ensures that changes are reported to the web server in batches, rather than reporting a single change at a time.
Periodically: The phone reports its configuration at regular intervals. The interval is expressed in seconds.
Example XML configuration:
<Report_to_Server ua="na">
Periodically
</Report_to_Server></td></tr>
<tr><td>Periodic Upload to Server:</td><td>Defines the interval (in seconds) that the phone reports its configuration to the provisioning servers.
This field is used only when Report to Server is set to Periodically.
Default: 3600
Minimum: 600
Maximum: 2592000 (30 days)
Example XML configuration:
<Report_to_Server ua="na">
Periodically
</Report_to_Server>
<!available options: On Request | On Local Change| Periodically-->
<periodic_upload_to_server ua="na">
3600
</periodic_upload_to_server><User_Configurable_Resync ua="na">
Yes
</User_Configurable_Resync_</td></tr>
</table>

Field	Description
Upload Delay On Local Change:	Defines the delay (in seconds) that the phone waits after a change is made, and then reports the configuration. This field is used only when **Report to Server** is set to **On Local Change**. Default: 60 Minimum: 10 Maximum: 900 Example XML configuration: **<Upload_Delay_On_Local_Change ua="na">** 60 **</Upload_Delay_On_Local_Change>**

Firmware Upgrade

Parameter	Description
Upgrade Enable	Allows firmware update operations independent of resync actions. Default: Yes

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.11-1-1MPP-221.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).
	
	You can also include the credentials that are used to access the server. Then, the upgrade rule is:
	
	`[--uid $userID --pwd $password]protocol://server[:port]/profile_pathname`
	
	For example,
	
	`[--uid TEST --pwd TestAbC123]tftp://192.168.1.5/image/sip88xx.11-1-1MPP-221.loads`
	
	If the user ID or the password contains special characters (/ [& } (*) # , etc.), you need to quote them in the upgrade rule. There are two options for quoting special characters:
	
	• Put the user ID or the password that contains special characters into double quotation marks (" "). This option doesn't work for some of the special characters, such as " " [ ].
	
	For example,
	
	`[--uid TEST --pwd "Test#\AbC123"]tftp://192.168.1.5/image/sip88xx.11-1-1MPP-221.loads`
	
	• Use the octal encoding of the special characters.
	
	For example, escape the pond (#) with “\043” and the backslash with “\057” for the password “Test#\AbC123” in the following rule:
	
	`[--uid TEST --pwd Test\043\057AbC123]tftp://192.168.1.5/image/sip88xx.11-1-1MPP-221.loads`
	
	Default: Blank

Parameter	Description
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
Peer Firmware Sharing	Enables or disables the Peer Firmware Sharing feature. Select **Yes** or **No** to enable or to disable the feature. Default: Yes
Peer Firmware Sharing Log Server	Indicates the IP address and the port to which the UDP message is sent. For example: 10.98.76.123:514 where, 10.98.76.123 is the IP address and 514 is the port number.

For more information about the Provisioning page, see the *Cisco IP Phone 7800 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	Specifies the path to the PRT upload script. You can enter the path in the format:
	`https://proxy.example.com/prt_upload.php`
	or
	`http://proxy.example.com/prt_upload.php`
	If **PRT Max Timer** and **PRT Upload Rule** fields are empty, problem reports are not generated.

PRT Upload Method	Determines the method used to upload PRT logs to the remote server. Options are: HTTP POST and PUT. Default: POST
PRT Max Timer	Determines at what interval (minutes) the phone starts generating problem report automatically. The interval range that you can set is 15 minutes to 1440 minutes. Default: Empty If **PRT Max Timer** and **PRT Upload Rule** fields are empty, problem reports are not generated. a
PRT Name	Defines a name for the generated PRT file. Enter the name in the format: **`prt-string1-$MACRO`**

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.
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.
Off Hook Warning Tone	Played when the phone receiver has been off hook after a period of time.
Ring Back Tone	Played during an outbound call when the far end is ringing.

Parameter	Description
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.
Mute Tone	Played when the **Mute** button is pressed to mute the phone.
Unmute Tone	Played when the **Mute** button is pressed to unmute the phone.
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).
Cadence 3	Cadence script for distinctive ring 3. Defaults to 60(.8/.4,.8/4).

Parameter	Description
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.
Secure One Call Act Code	Makes a secure call. Default: *18.
Secure One Call Deact Code	Disables secure call feature. Default: *19.
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.

Parameter	Description
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.
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.

Parameter	Description
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.
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

Parameter	Description
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.
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).

Parameter	Description
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	Use this field to specify the language options for the phone display, and the dictionary and font files required for each language. See Set Up Dictionaries and Fonts. Default: Blank
Language Selection	Use this field to specify the default language. The value must match one of the languages supported by the dictionary server. See Specify a Language for the Phone Display. You can configure the language through the XML Configuration file. For example: `<Language_Selection ua="na"> Spanish </Language_Selection>` The language name can have up to 512 characters.
Locale	Use this drop-down list box to see the supported languages. See Supported Languages for the Phone Display.

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

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

Each line key has a set of settings.

Parameter	Description

Extension	Specifies the n extension to be assigned to Line Key n.
	Default: n
	XML configuration examples:
	To set the line key 1 to extension 1:
	`<Extension_1_ ua="na">1</Extension_1_>`
	To disable the extension function for line key 2:
	`<Extension_2_ ua="na">Disabled</Extension_2_>`

Parameter	Description
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 any of the following features or functions to unused line keys on the phone: • Busy Lamp Field • Call Pickup • Speed Dial

Miscellaneous Line Key Settings

Parameter	Description
Line ID Mapping	Specifies the shared call appearance line ID mapping. If Vertical First is set, the second call makes the next available line ID LED flash. If Horizontal first is set, the second call will make the same LED flash on which the first call is received. Also, the behavior is same for both outgoing and incoming calls. **Note** 7811 Cisco IP Phone does not support Line ID Mapping. Default: Horizontal First
SCA Barge-In Enable	Enables the SCA Barge-In. Default: No
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 or disable three-way conference service. Default: Yes

Parameter	Description
Attn Transfer Serv	Enable or disable attended-call-transfer service. Default: Yes
Blind Transfer Serv	Enable or disable blind-call-transfer service. Default: Yes
DND Serv	Enable or disable do not disturb service. Default: Yes
Block ANC Serv	Enable or disable block-anonymous-call service. Default: Yes
Block CID Serv	Enable or disable blocking outbound Caller-ID service. Default: Yes
Secure Call Serv	Enable or disable secured call services. Default: Yes
Cfwd All Serv	Enable or disable call-forward-all service. Default: Yes
Cfwd Busy Serv	Enable or disable call-forward-on-busy service. Default: Yes
Cfwd No Ans Serv	Enable or disable call-forward-no-answer service. Default: Yes
Paging Serv	Enable or disable paging service on the phone. Default: Yes
Call Park Serv	Enable or disable call park services on the phone. Default: Yes
Call Pick Up Serv	Enable or disable call pick up services on the phone. Default: Yes
ACD Login Serv	Enable or disable ACD login services on the phone. Default: Yes
Group Call Pick Up Serv	Enable or disable group call pick up services on the phone. Default: Yes

Parameter	Description
Service Annc Serv	Enable or disable the vertical service announcement services on the phone. Default: No
Call Recording Serv	Enable or disable the call recording services on the phone. Default: No
Reverse Phone Lookup Serv	Enable or disable reverse name lookup for the phone. When enabled, the phone can searches the personal address book and call history, server directory, and either the configured LDAP or XML directory. Default: Yes

Ringtone

Parameter	Description
Ring1 to Ring12	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
Session Timer(m)	Specifies the duration of the phone session.
Countdown Timer(s)	Specifies the duration for which it waits before it logs out. Default: 10

Parameter	Description
Preferred Password Input Mode	Options to specify the password input method of extension mobility PIN. Options are: Alpha-numeric and Numeric. Default: Alphanumeric

XSI Phone Service

Parameter	Description
XSI Host Server	Enter the name of the server; for example, xsi.iop1.broadworks.net. **Note** XSI Host Server uses http protocol by default. To enable XSI over HTTPS, you can specify `https://` in the server. Default: Blank
XSI Authentication Type	Determines the XSI authentication type. Select **Login Credentials** to authenticate access with XSI id and password. Select **SIP Credentials** to authenticate access with the register user ID and password of the SIP account registered on the phone. Default: Login Credentials
Login User ID	BroadSoft User ID of the phone user; for example, johndoe@xdp.broadsoft.com. Enter SIP Auth ID when you select **Login Credentials** or **SIP Credentials** for XSI authentication type. When you choose SIP Auth ID as **SIP Credentials**, you must enter Login User ID. Without Login User ID, the BroadSoft directory will not appear under the phone Directory list. Default: Blank
Login Password	Alphanumeric password associated with the User ID. Enter login password, when you select **Login Credentials** for XSI authentication type. Default: Blank
SIP Auth ID	The registered user ID of the SIP account registered on the phone. Enter SIP Auth ID when you select **SIP Credentials** for XSI authentication type.

Parameter	Description
SIP Password	The password of the SIP account registered on the phone. Enter SIP password when you select **SIP Credentials** for XSI authentication type.
Directory Enable	Enables BroadSoft directory for the phone user. Select **Yes** to enable the directory and select **No** to disable it. Default: No
Directory Name	Name of the directory. Displays on the phone as a directory choice. Default: Blank
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
CallLog Enable	Enables to log XSI calls. Select **Yes** to log XSI calls and select **No** to disable it. Default: No
CallLog Associated Line	Allows you to select a phone line for which you want to display the recent call logs. You can select line numbers ranges from 1 to 10.
Display Recents From	Allows you to set which type of recent call logs the phone will display. Choose **Server** to display BroadSoft XSI recent call logs and select **Phone** to display local recent call logs. **Note** The **Display recents from** is added to the **Recents** screen of the phone only when you set **CallLog Enable** to **Yes** and **Display Recents From** type to **Server**.

Broadsoft XMPP

Parameter	Description
XMPP Enable	Set to Yes to enable the BroadSoft XMPP directory for the phone user. Default: No
Server	Enter the name of the XMPP server; for example, xsi.iop1.broadworks.net. Default: Blank
Port	Server port for the directory. Default: Blank
User ID	BroadSoft User ID of the phone user; for example, johndoe@xdp.broadsoft.com. Default: Blank
Password	Alphanumeric password associated with the User ID. Default: Blank
Login Invisible	When enabled, the user's presence information is not published when the user signs in. Default: No
Retry Intvl	Interval, in seconds, to allow a reconnect without a log in after the client disconnects from the server. After this interval, the client needs to reauthenticate. Default: 30

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 Application Service Name	Name of the XML application. Displays on the user's phone as a web application choice.
XML Application Service URL	URL where the XML application is located.
XML User Name	XML service username for authentication purposes Default: Blank

Parameter	Description
XML Password	XML service password for authentication purposes Default: Blank
CISCO XML EXE Enable	Enables or disables Cisco XML EXE authentication. Default: No
CISCO XML EXE Auth Mode	Specifies the authentication mode for Cisco XML EXE. The available options are: • Trusted—No authentication is performed (local user password is set or not). • Local Credential—Authentication is based on digest authentication using the local user password, if the local user password is set. If not set, then no authentication is performed. • Remote Credential—Authentication is based on digest authentication using the remote username/password as set in the XML application on the web page (to access an XML application server). Default: Trusted

Multiple Paging Group Parameters

Feature	Description
Group Paging Script	Enter a string to configure group paging and priority paging (out of band paging) that doesn’t require the phone registration.

LDAP

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

Parameter	Description
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 cn=”David Lee”,dc=cv2bu,dc=com username@domain is the client DN format for a Windows server For example, DavidLee@cv2bu.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

Parameter	Description
Last Name Filter	Use this field to specify how the phone must perform searches based on the last name or surname (sn), when users search for contacts. Examples: **`sn:(sn=$VALUE*)`** instructs the phone to find all last names that begin with the entered search string. **`sn:(sn=*$VALUE*)`** instructs the phone to find all last names in which the entered search string appears anywhere in the last name. This method is more inclusive and retrieves more search results. This method is consistent with the search method in other directories such as the Broadsoft directories and the user's personal address book on the phone. Default: Blank
First Name Filter	Use this field to specify how the phone must perform searches based on the first name or common name (cn), when users search for contacts. Examples: **`cn:(cn=$VALUE*)`** instructs the phone to find all first names that begin with the entered search string. **`cn:(cn=*$VALUE*)`** instructs the phone to find all first names in which the entered search string appears anywhere in the first name. This method is more inclusive and retrieves more search results. This method is consistent with the search method in other directories such as the Broadsoft directories and the user's personal address book on the phone. 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.
Missed Call Key List	Softkeys that display when there is a missed call.
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. To silence an incoming call, you can add **Ignore** softkey.
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.

Extension

General

Parameter	Description
Line Enable	To enable this line for service, select yes. Otherwise, select No. Default: Yes Example XML configuration: To disable service on the line associated with extension 2: `<Line_Enable_2_ ua="na">No</Line_Enable_2_>`

Video Configuration

Parameter	Description
H264 BP0 Enable	Enables the H264 Base Profile 0 codec when you select **Yes** and disables it when you select **No**. Default: Yes
H264 HP Enable	Enables the H264 High Profile codec when you select **Yes** and disables it when you select **No**. Default: Yes
Encryption Method	Selects the encryption method to be used during a secured call. Options are **AES 128** and **AES 256 GCM**. Default: **AES 128**

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

Parameter	Description
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
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. Default: 0x68.
RTP ToS/DiffServ Value	Value for the ToS field of voice data packets. Sets the priority for voice packets in data traffic. Default: 0xb8.

SIP Settings

Parameter	Description
SIP Transport	Select the transport protocol for SIP messages: • UDP • TCP • TLS • AUTO **AUTO** allows the phone to select the appropriate protocol automatically, based on the NAPTR records on the DNS server. See Configure the SIP Transport for more details. Default: UDP
SIP Port	The phone's port number for SIP message listening and transmission. Specify the port number here only when you are using UDP as the SIP transport protocol. If you are using TCP, the system uses a random port within the range specified in **SIP TCP Port Min** and **SIP TCP Port Max** on the **Voice** > **SIP** tab. If you need to specify a port of SIP proxy server, you can specify it using the **Proxy** field (Proxy and Registration, on page 82) or the **XSI Host Server** field (XSI Line Service, on page 85). Default: 5060
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.

Parameter	Description
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
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. Default: 0

Parameter	Description
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.
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 Recording Protocol	Determines the type of recording protocol that the phone uses. Options are: • SIPINFO • SIPREC Default: SIPREC

Parameter	Description
Privacy Header	Sets user privacy in the SIP message in the trusted network. The privacy header options are: • Disabled (default) • none—The user requests that a privacy service applies no privacy functions to this SIP message. • header—The user needs a privacy service to obscure headers which cannot be purged of identifying information. • session—The user requests that a privacy service provide anonymity for the sessions. • user—The user requests a privacy level only by intermediaries. • id—The user requests that the system substitute an id that doesn't reveal the IP address or host name. Default: Disabled
P-Early-Media Support	Controls whether the P-Early-Media header is included in the SIP message for an outgoing call. To include the P-Early-Media header, select **Yes**. Otherwise, select **No**. 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
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.

Parameter	Description
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 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.
Auto Ans Page On Active Call	Determines the behavior of the phone when a page call arrives.
Feature Key Sync	Enable the synchronization of settings between the line and the server if necessary. Feature Key Sync must be enabled for lines that are configured for the following functions or users: • Call forward all • DND
Call Park Monitor Enable	BroadSoft server-only feature. If call park is enabled on the server or on any of the programmable line keys, you need to enable this field for call park notification to work. Default: No

Parameter	Description
Enable Broadsoft Hoteling	When this parameter is set to yes, the phone sends out subscription messages (without body) to the server. Default: No
Hoteling Subscription Expires	An expiration value that is added in the subscription message. Default value is 3600.
Secure Call Option	The secure call feature works only when the **SIP Transport** on the **Ext (n)** tab is set to **TLS**. Enables secured calls on an extension. Options are: • Optional: The phone maintains the current behavior for secure calls. • Required: The phone rejects nonsecure calls from other phones. Default: Optional

ACD Settings

Parameter	Description
Broadsoft ACD	Enables the phone for Automatic Call Distribuion (ACD). Select **Yes** to enable or **No** to disable. Default: No
Call Information Enable	Enables the phone to display details of a call center call. Select **Yes** to enable or **No** to disable. Default: No
Disposition Code Enable	Enables the user to add a disposition code. Select **Yes** to enable or **No** to disable. Default: No
Trace Enable	Enables the user to trace the last incoming call. Select **Yes** to enable or **No** to disable. Default: No
Emergency Escalation Enable	Enables the user to escalate a call to a supervisor in case of emergency. Select **Yes** to enable or **No** to disable. Default: No
Queue Status Notification Enable	Displays the call center status and the agent status. Select **Yes** to enable or **No** to disable. Default: No

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. If you don't specify a port, the default port 5060 is used for UDP, and the default port 5061 is used for TLS. When you need to refer to this proxy in another setting, for example, the speed dial line key configuration, use the `$PROXY` macro variable.
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

Parameter	Description
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
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

Parameter	Description
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
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. When you need to refer to this user ID in another setting, for example, the short name for a line key, use the `$USER` macro variable.
Password	Password for this line. Default: Blank (no password required)
Auth ID	Authentication ID for SIP authentication. Default: Blank

Parameter	Description
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.

XSI Line Service

Parameter	Description
XSI Host Server	Enter the name of the server; for example, . **`xsi.iop1.broadworks.net`** **Note** XSI Host Server uses http protocol by default. To enable XSI over HTTPS, you can specify `https://` in the server. For example: **`https://xsi.iop1.broadworks.net`** You can also specify a port for the server. For example: **`https://xsi.iop1.broadworks.net:5061`** If you don't specify a port. The default port for the specified protocol is used. Default: Blank
XSI Authentication Type	Determines the XSI authentication type. Select **Login Credentials** to authenticate access with Login User ID and Login Password. Select **SIP Credentials** to authenticate access with the register Auth ID and Password of the SIP account registered on the phone. Default: Login Credentials

Parameter	Description
Login User ID	BroadSoft User ID of the phone user; for example, johndoe@xdp.broadsoft.com. For any XSI Authentication Type, you must enter **Login User ID**. Without **Login User ID**, the BroadWorks Anywhere feature does not work. Default: Blank
Login Password	Alphanumeric password associated with the Login User ID. Enter Login Password, when you select **Login Credentials** for XSI authentication type. Default: Blank
Anywhere Enable	Enables BroadWorks Anywhere feature on an extension. If you choose **Yes**, Anywhere is enabled on this line, and the user can use the phone menu to add multiple locations to this specific line. Default: Yes
Block CID Enable	Enables XSI Caller ID blocking on a line. Choose **Yes** to enable the synchronization of blocking caller id status with the server using XSI interface. Choose **No** to use the phone's local blocking caller id settings.
CFWD Enable	Enables or disables call forwarding status sync on a line via XSI service. Choose **Yes** to enable the phone to synchronize the call forwarding status with the server using the XSI service. Choose **No** to disable this feature. **Note** • When **Feature Key Sync** is set to **Yes**, FKS takes precedent over XSI synchronization. • If XSI host server and credentials are not entered and the **CFWD Enable** field is set to **Yes**, the phone user can't forward calls on the phone.

Parameter	Description
DND Enable	Enables or disables DND status sync on a line via XSI service. Choose **Yes** to enable the phone to synchronize DND status with the server using the XSI service. Choose **No** to disable this feature. **Note** • When **Feature Key Sync** is set to **Yes**, FKS takes precedent over XSI synchronization. • If XSI host server and credentials are not entered and the **DND Enable** field is set to **Yes**, the phone user can't turn on DND mode on the phone.

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

Parameter	Description
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
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.

Parameter	Description
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).

E911 Geolocation Configuration

E911 Geolocation Configuration

Parameter	Description
Company UUID	The Universally Unique Identifier (UUID) assigned to the customer by the emergency call services provider. Maximum identifier length is 128 characters. Defaults to blank.
Primary Request URL	Encrypted HTTPS phone location request. The request uses the phone IP addresses, MAC address, Network Access Identifier (NAI), and chassis ID and port ID assigned by the network switch manufacturer. The request also includes the location server name and the customer identifier. The server used by the emergency call services provider responds with an Emergency Response Location (ERL) that has a location Uniform Resource Identifier (URI) tied to the user phone IP address. Defaults to blank.
Secondary Request URL	Encrypted HTTPS request sent to the emergency call services provider's backup server to obtain the user's phone location. Defaults to blank.

See Emergency Call Support Terminology for terms that describe emergency call support for phones.

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
Hold Reminder Ringtone	Specifies the volume of the timer ringtone.

Call Forward

Parameter	Description
Cfwd Setting	Select **Yes** to enable call forwarding.

Parameter	Description
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

Parameter	Description
Speed Dial Name (2 to 9)	Name assigned to a specific speed dial number. Default: Blank
Speed Dial Number 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.
Handset LED Alert	Enables or disables LED alert on the handset. Options are: Voicemail and Voicemail, Missed Call. Default: Voicemail

Parameter	Description
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. 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 Volume

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

Parameter	Description
Electronic Hookswitch Control	Enables or disables the Electronic HookSwitch (EHS) feature. After EHS is enabled, the AUX port does not output phone logs.

Audio Compliance

Parameter	Description
Compliant Standard	Specifies the compliance standard for the phone audio. The available options are: • **ETSI**: A set of standards for speech and multimedia transmission for the narrowband and wideband terminals from European Telecommunications Standards Institute (ETSI). • **TIA**: A set of standards from US Telecommunications Industry Association (TIA). The standards are for narrowband and wideband audio transmission via wired telephones. Default: TIA

Screen

Parameter	Description
Screen Saver Enable	Enables a screen saver on the phone. When the phone is idle for a specified time, it enters screen saver mode. Default: No
Screen Saver Type	Types of screen saver. Options you can choose: • **Clock**: Displays a digital clock on a plain background. • **Download Picture**: Displays a picture pushed from the phone webpage. • **Lock** : Enables locking of the screensaver.
Screen Saver Wait	Amount of idle time before screen saver displays. enter the number of seconds of idle time to elapse before the screen saver starts. Default: 300

Parameter	Description
Screen Saver Refresh Period	Number of seconds before the screen saver should refresh (if, for example, you chose a rotation of pictures).
Back Light Timer	Number of seconds for which the back light timer will be on.
LCD Contrast	Value for desired contrast.
Logo Type	Type of logo displayed on the phone screen. Options you can choose: • Default • Download Picutre • Text Logo

Text Logo	Text logo to display when the phone boots up. A service provider, for example, can enter logo text as follows:
	• Up to 2 lines of text
	• Each line must be fewer than 32 characters
	• Insert a new line character (\n) between lines
	• Insert escape code %0a
	For example,
	`Super\n%0aTelecom`
	displays:
	`Super`
	`Telecom`
	Use the + character to add spaces for formatting. For example, you can add multiple + characters before and after the text to center it.

Picture Download URL	URL locating the (.png) file to display on the phone screen background. For more information, see the Phone Information and Display Settings.

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
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
Server Type	Specifies the server type with which the phone is connected. Options available: • BroadSoft • SPA9000 • Asterisk • RFC3265_4235 • Sylantro

Parameter	Description
BLF List URI	The Uniform Resource Identifier (URI) of the Busy Lamp Field (BLF) list that you have set up for a user of the phone, on the BroadSoft server. This field is only applicable if the phone is registered to a BroadSoft server. The BLF list is the list of users whose lines the phone is allowed to monitor. See Phone Configuration for Monitoring Other Phones for details. The BLF List URI must be specified in the format **<URI name>@<server>**. The BLF List URI specified must be the same as the value configured for the **List URI: sip** parameter on the BroadSoft server. Default: Blank Example XML configuration: <BLF_List_URI ua="na">MonitoredUsersList@sipurash22.com</BLF_List_URI>
Use Line Keys For BLF List	Controls whether the phone uses its line keys to monitor the BLF list, when monitoring of the BLF list is active. This setting only has significance when **BLF List** is set to **Show**. Default: No Example XML configuration: <Use_Line_Keys_For_BLF_List ua="na">Yes</Use_Line_Keys_For_BLF_List>

Parameter	Description
Customizable PLK Options	Features that users are allowed to configure on line keys. To allow a feature, add the corresponding option as shown below. Separate options with the semi-colon (;). • Speed dial: sd • Busy Lamp Field (BLF) key to monitor a user: blf • Call pickup from a monitored line: cp **Note** This option is only effective when the blf option is added. Default: sd; **Note** Adding the sd option automatically allows users to configure speed dial to a monitored line (speed dial with BLF) when the blf option is added. Example, to allow all features: sd;blf;cp Example XML configuration: <Customizable_PLK_options ua="na">sd;</Customizable_PLK_options>
BLF List	Activates or deactivates monitoring of the BLF list. When set to **Show**, the phone assigns available line keys in sequence, to monitor the BLF list entries. The labels of the BLF list keys show the names of the monitored users and the status of the monitored lines. This setting only has significance when **BLF List URI** is configured. Example XML configuration: <BLF_List ua="rw">Show</BLF_List>
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

Parameter	Description
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
BXfer To Remote Party Number Enable	When set to **Yes**, the phone performs a blind transfer to a remote number. When set to no, blind transfer to remote number is disabled.
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.
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. Default: Disabled

Parameter	Description
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.
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 Inform 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.

Parameter	Description
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.
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**.