Cisco SPA IP Phone Field Reference

This appendix describes the fields within the following sections of the phone web user interface:

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Info Tab

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

System Information

Parameter	Description
Connection Type	Indicates the type of internet connection for the phone: • DHCP • Static IP • PPPoE (only applicable to Cisco SPA525G or Cisco SPA525G2)
Current IP	Displays the current IP address assigned to the IP phone.

Parameter	Description
Host Name	Displays the current host name assigned to the SPA9000 (defaults to SipuraSPA).
Domain	Displays the network domain name of the SPA9000.
Current Netmask	Displays the network mask assigned to the SPA9000.
Current Gateway	Displays the default router assigned to the SPA9000.
Primary DNS	Displays the primary DNS server assigned to the SPA9000.
Secondary DNS	Displays the secondary DNS server assigned to the SPA9000.

Cisco SPA525G or Cisco SPA525G2-Specific Parameters:

Parameter	Description
NTP Enable	Shows if Network Time Protocol is enabled.
Primary NTP Server	IP Address of the primary NTP server.
Secondary NTP Server	IP Address of the secondary NTP server.
TFTP Server	Address of the TFTP server for provisioning.
Bluetooth Enabled	Shows if Bluetooth is enabled.
Bluetooth Firmware Version	Displays the Bluetooth firmware version.
Bluetooth Connected	Shows if a Bluetooth device is connected to the phone.
Bluetooth MAC	Shows the hardware address of the Bluetooth device.
Connected Device ID	Shows the name of the connected Bluetooth device.
Wireless Enabled	Shows if Wireless-G is enabled on the phone.
Wireless Connected	Shows if the phone is connected to the wireless network.
Wireless MAC	Shows the hardware address of the Wireless-G controller.
SSID	Shows the SSID, or name of the wireless router to which the phone is connected.
Standard Channel	Shows the wireless channel being used in the wireless connection.
Security Mode	Shows if wireless security is configured on the phone (yes or no).

Reboot History

The IP phone stores the reasons for the last five reboots or refreshes. When the phone is reset to factory defaults, this information is deleted.

The reboot history is displayed in reverse chronological order, with the reasons for the latest reboot displayed in the **Reboot Reason 1** field.

Each Reboot Reason field displays the reason for the reboot and a time stamp indicating when the reboot took place as in the following examples:

```
Reboot Reason 1: Provisioning(06/22/2011 13:29:33)
Reboot Reason 2: Upgrade(06/22/2011 13:01:43)
Reboot Reason 3: Provisioning(06/22/2011 10:40:12)
```

The following is a list of the supported reboot/refresh reasons:

Reason	Description
Upgrade	An upgrade operation caused a reboot (regardless whether the upgrade completed or failed).
Provisioning	Changes made to parameter values by using the phone LCD or Web GUI, or a resync caused a reboot.
SIP Triggered	A SIP request caused a reboot.
RC	A remote customization caused a reboot.
User Triggered	The user manually triggered a warm reboot.
Software Req	A remote server triggered a warm reboot.
System *n*	System events (for example, running out of resources) triggered a warm reboot.
IP Changed	The phone IP address was changed triggering a warm reboot.

You can view the reboot history from the phone Web GUI, the phone LCD screen, and the phone SPA Status Dump file (http://*phoneIP*/status.xml or http://*phoneIP*/admin/status.xml).

Viewing the Reboot History on the Phone LCD Screen

A new menu, **Reboot History**, was added under the **Setup menu**. On the **Reboot History** Page, the 5 reboot entries are displayed in reverse chronological order, just like the Web GUI.

Viewing the Reboot History in the SPA Status Dump File

The reboot history is stored in the SPA Status Dump file (http://<*phone_IP_address*>/admin/status.xml). In this file, tags **Reboot_Reason_1** to **Reboot_Reason_5** store the reboot history, as shown in this example:

```
<Reboot_History><Reboot_String/>
<Reboot_Reason_1>Provisioning(06/13/2011 14:03:43)</Reboot_Reason_1>
<Reboot_Reason_2>Provisioning(06/13/2011 13:58:15)</Reboot_Reason_2>
<Reboot_Reason_3>Provisioning(06/13/2011 12:08:58)</Reboot_Reason_3>
<Reboot_Reason_4>Provisioning(05/26/2011 15:26:49)</Reboot_Reason_4>
<Reboot_Reason_5>System 4(05/24/2011 10:20:06)</Reboot_Reason_5>
<Reboot_History/>
```

The Web GUI and the LCD screen get the reboot history from these tags.

Network Configuration (SPCP)

Parameter	Description
TFTP Server	Address of the TFTP server for provisioning.
Call Manager	IP address of the Unified Communications server.
Directories URL	Populated by the Unified Communications Server; points to the directory application server.
Services URL	Populated by the Unified Communications Server; points to the Cisco XML application server.

Parameter	Description
Authentication URL	Populated by the Unified Communications Server; points to the authentication server.
DHCP Address Released	Populated by the Unified Communications Server; indicates if the DHCP address has been released.

VPN Status (Cisco SPA525G or Cisco SPA525G2 Only)

Parameter	Description
VPN Connected	Indicates if the phone is connected to a VPN.
Client Address	IP address given to the phone from the VPN server.
Client Netmask	Netmask given to the phone from the VPN server.
Bytes Sent	Size of data sent from the phone.
Bytes Recv	Size of data received by the phone.

Product Information

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

Phone Status

Parameter	Description
Current Time	Current date and time of the system; for example, 10/3/2003 16:43:00.
Elapsed Time	Total time elapsed since the last reboot of the system; for example, 25 days and 18:12:36.
Broadcast Pkts Sent	Total number of broadcast packets sent.
Broadcast Bytes Sent	Total number of broadcast packets received.
Broadcast Pkts Recv	Total number of broadcast bytes sent.
Broadcast Bytes Recv	Total number of broadcast bytes received and processed.
Broadcast Pkts Dropped	Total number of broadcast packets received but not processed. Most codecs can handle up to 5% random packet drops as long as the packets are random and not in groups of two or more. Concurrent packet drops result in voice quality issues.
Broadcast Bytes Dropped	Total number of broadcast bytes received but not processed.
RTP Packets Sent	Total number of RTP packets sent (including redundant packets).
RTP Bytes Sent	Total number of RTP packets received (including redundant packets).
RTP Packets Recv	Total number of RTP bytes sent.
RTP Bytes Recv	Total number of RTP bytes received.
SIP Messages Sent	Total number of SIP messages sent (including retransmissions).
SIP Bytes Sent	Total number of SIP messages received (including retransmissions).
SIP Messages Recv	Total number of bytes of SIP messages sent (including retransmissions).
SIP Bytes Recv	Total number of bytes of SIP messages received (including retransmissions).
External IP	External IP address used for NAT mapping.
Operational VLAN ID	ID of the VLAN currently in use if applicable. **Note** Not applicable to Cisco WIP310.
SW Port (Cisco SPA300 Series and Cisco SPA500 Series)	Displays the type of Ethernet connection from the IP phone to the switch.
PC Port (Cisco SPA303 and Cisco SPA500 Series)	Indicates whether the link from the IP phone to a device plugged into the PC port on the phone is up or down.

Ext Status

The following parameters show for each extension on the phone.

Parameter	Description
Registration State	Shows “Registered” if the phone is registered, “Not Registered” if the phone is not registered to the ITSP.
Last Registration At	Last date and time the line was registered.
Next Registration In	Number of seconds before the next registration renewal.
Message Waiting	Indicates whether the phone user has a new voice mail waiting: Yes or No. This is updated when voice mail notification is received.
Mapped SIP Port	Port number of the SIP port mapped by NAT.
Hoteling State	Indicates the status of the phone with the guest account. This parameter is of 6 types: • **Disabled:** When the parameter, <Enable Broadsoft Hoteling> is set to **no**. • **Unsubscribed:** When the parameter, <Enable Broadsoft Hoteling> is set to **yes**, and the Hoteling State is not subscribed to server. • **Associating xxx:** When the phone is trying for guest login, where xxx stands for the guest User ID. • **Associated xxx:** When the guest login is successful on the phone.. • **Disassociating:** When the phone is trying to disassociate with the guest account. • **Disassociated:** When the phone has disassociated with the guest account.

Line/Call Status

The following parameters show for each line and call on the phone.

Parameter	Description
Call State	Status of the call.
Tone	Type of tone used by the call.
Encoder	Codec used for encoding.
Decoder	Codec used for decoding.
Type	Direction of the call.
Remote Hold	Indicates whether the far end has placed the call on hold.
Callback	Indicates whether the call was triggered by a call back request.
Peer Name	Name of the internal phone.

Parameter	Description
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.
Mapped RTP Port	The port mapped for Real Time Protocol traffic for the call.
Media Loopback	If the call is a loopback call, displays the loopback mode (source or mirror) and type (media or packet). If the call is not loopback, the field appears blank.
Decode Latency	Number of milliseconds for decoder latency.
Jitter	Number of milliseconds for receiver jitter.
Round Trip Delay	Number of milliseconds for delay in the RTP-to-RTP interface round trip.
End System Delay	Number of milliseconds for delay in the internal round trip within the reporting endpoint.
Packets Lost	Number of packets lost.
Packet Error	Number of invalid packets received.
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).
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 describing 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/MOS Listening	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/MOS Conversational	The estimated mean opinion score for conversational quality (MOS-CQ) is defined as including the effects of delay and other effects that would affect conversational quality. Defined in RFC-3611—RTP Control Protocol Extended Reports (RTCP XR).

Downloaded Ring Tone

Parameter	Description
Status	Indicates whether the phone is downloading a ring tone (and from where) or if it is idle.
Ring Tone 1	Information about the user downloaded ring tone 1: name, size, and time-stamp of the tone.
Ring Tone 2	Information about the user downloaded ring tone 2: name, size, and time-stamp of the tone.

Custom CA Status

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

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

Custom CA certificates are configured in the Provisioning tab. For more information about custom CA certificates, see the *Cisco Small Business IP Telephony Devices Provisioning Guide*.

System Tab

System Configuration

Parameter	Description
Restricted Access Domains (SIP)	This feature is used when implementing software customization.
Enable Web Server	Enable/disable web server of the IP phone. Defaults to yes.

Parameter	Description
Web Server Port	Port number of the phone web user interface. Defaults to 80.
SPA525-http-write (Cisco SPA525G or Cisco SPA525G2 SPCP only)	Allow Cisco Configuration Assistant (CCA) or other application to write XML file parameters directly to the phone using HTTP. Choose **yes** to allow this feature, or no to disable this feature.
Enable Web Admin Access	Lets you enable or disable local access to the phone web user interface. Select yes or no from the drop-down menu. Defaults to yes.
Admin Passwd	Password for the administrator. Defaults to no password.
User Password	Password for the user. Defaults to no password.
User Web Password	Eliminates the user requirement for daily password entry on the LCD GUI to use the basic functions. Basic functions include CFwd, Directory, Redial, and Corporate Directory. The options are No and Yes: • If the User Password field is not empty, using this field restricts user access through the web interface and LCD GUI. • If the User Password field is empty, no password is required to access the LCD GUI. However, the user must enter a password to access the web interface. This is the User Web Password parameter that was set earlier. Default value is empty. **Note** This field is supported in Firmware Release 7.6.2 and later.
SPA525-protocol (Cisco SPA525G or Cisco SPA525G2 only)	Allows you to choose the type of protocol for the phone: • SIP—Session Initiation Protocol. Choose if the phone is used with a SIP call control system, such as the Cisco SPA9000 or a SIP call control system from another provider such as BroadSoft or Asterisk. • SPCP—Smart Phone Control Protocol. Choose if the phone is used with a Cisco Unified Communications Series server, such as the Cisco Unified Communications 500 Series for Small Business.
SPA525-auto-detect-sccp (Cisco SPA525G or Cisco SPA525G2 only)	Choose if the phone should automatically detect the type of protocol used on the network to which it is connected. If set to yes, the phone automatically discovers if it is connected to a call control system using SPCP.
SPA525-readonly (Cisco SPA525G or Cisco SPA525G2 only)	If set to **yes**, the Signaling Protocol and Auto Detect SCCP Settings on the phone are read only. If set to **no**, the above settings on the phone can be changed by the end user.

Parameter	Description
Phone-UI-user-mode	Allows you to restrict the menus and options that phone users see when they use the phone interface. Choose **yes** to enable this parameter and restrict access. The default is no. Specific parameters are then designated as “na” or “ro” using provisioning files. Parameters designated as “na” will not appear on the phone interface. Parameters designated as “ro” will not be editable by the user. **Note** For the SPA525 phones, both the parameters, na and ro, work in a similar fashion.
WiFi User Access Mode (Cisco SPA525G or Cisco SPA525G2 only)	Setting the parameter to **yes** allows a user to view and configure Wi-Fi settings, even if the user cannot change other phone parameters because the **Phone-UI-user-mode** parameter is set to yes. If set to **no**, users cannot view and configure WiFi settings if the **Phone-UI-user-mode** parameter is set to yes.

Power Settings

Parameter	Description
PoE Power Required	Specifies the PoE power setting: Normal (default) or Maximum. When one or more attendant consoles are attached to the phone, use **Maximum** to advertise to a PoE switch that the phone will consume up to 12 watts. When no attendant consoles are attached, use **Minimum** to advertise a required power budget of 6.5 watts. This parameter is not applicable to Cisco SPA301 or SPA303 as they do not have Cisco Attendant Console support.

Internet Connection Type

Parameter	Description
Connection Type	Choose the type of internet connection: • DHCP • Static IP • PPPoE (not applicable to Cisco WIP310)
Use Backup IP (Cisco SPA 30X and Cisco SPA5XX only)	When this field is set to **yes**, the IP phone does not renew its IP address when the DHCP server is not reachable during the phone's startup sequence, but uses the last-known IP address. If set to **no**, the phone keeps retrying the DHCP server to obtain a new IP address. This field is set to **yes** by default.

Static IP Settings

Parameter	Description
Static IP	If static IP was chosen as the type of internet connection, displays the static IP address assigned to the phone.
Netmask	If static IP was chosen as the type
Gateway	Default router IP address. Blank if DHCP assigned.
LAN MTU (Cisco SPA525G or Cisco SPA525G2 SIP mode only)	LAN Maximum Transmission Unit size. Default value: 1500.
Ethernet MTU (Cisco SPA525G or Cisco SPA525G2 SPCP mode only)	Ethernet Maximum Transmission Unit size. Default value: 1500.
Duplex Mode (Cisco SPA525G or Cisco SPA525G2 SPCP mode only)	Duplex Mode—Choose one of the following to configure the speed/duplex for the phone Ethernet ports: • Auto • 10MBps/Duplex • 10MBps/Half • 100Mbps/Duplex • 100MBps/Half

PPPoE Settings (Cisco SPA525G or Cisco SPA525G2 Only)

Parameter	Description
PPPoE Login Name	Specifies the account name assigned by the ISP for connecting on a Point-to-Point Protocol over Ethernet (PPPoE) link.
PPPoE Login Password	Specifies the password assigned by the ISP for connecting on a Point-to-Point Protocol over Ethernet (PPPoE) link.
PPPoE Service Name	Specifies the service name assigned by the ISP for connecting on a Point-to-Point Protocol over Ethernet (PPPoE) link.

Optional Network Configuration

Parameter	Description
Host Name	The host name of the SPA9000.
Domain	The network domain of the SPA9000.
Primary DNS	DNS server used by SPA9000 in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the primary DNS server. Defaults to 0.0.0.0.
Secondary DNS	DNS server used by SPA9000 in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the secondary DNS server. Defaults to 0.0.0.0.
DNS Server Order	Specifies the method for selecting the DNS server. The options are Manual, Manual/DHCP, and DHCP/Manual.
DNS Query Mode	Do parallel or sequential DNS Query. With parallel DNS query mode, the SPA9000 sends the same request to all the DNS servers at the same time when doing a DNS lookup, the first incoming reply is accepted by the SPA9000. Defaults to parallel.
DNS Query TTL Ignore (Not applicable to Cisco SPA525G or Cisco SPA525G2.)	If enabled, the phone continues to use the previous cached DNS result if the DNS server does not respond when the phone tries to renew its DNS query. When disabled, the phone uses the previous TTL value and clears the DNS query result cache.
Syslog Server	Specify the syslog server name and port. This feature specifies the server for logging IP phone system information and critical events. If both Debug Server and Syslog Server are specified, Syslog messages are also logged to the Debug Server.
Debug Server	The debug server name and port. This feature specifies the server for logging IP phone debug information. The level of detailed output depends on the debug level parameter setting.

Parameter	Description
Debug Level	The debug level from 0-3. The higher the level, the more debug information is generated. Zero (0) means no debug information is generated. To log SIP messages, you must set the Debug Level to at least 2. Defaults to 0.
Layer 2 Logging	Used for IP phone network layer debugging purposes. Do not use except when advised to do so by Cisco technical support, as this may impact system performance. Set to **no** by default.
Primary NTP Server	IP address or name of primary NTP server.
Secondary NTP Server	IP address or name of secondary NTP server.
Enable Bonjour (Cisco SPA525G or Cisco SPA525G2 only)	Enable Bonjour networking that is used by Office Manager and Cisco Configuration Assistant to discover the Cisco IP phones. Choose **yes** to enable or **no** to disable.
Enable SSLv3	Choose **Yes** to enable SSLv3. Choose **no** to disable. Defaults to NO.

VLAN Settings

Not applicable to the Cisco WIP310.

Parameter	Description
Enable VLAN	Choose **Yes** to enable VLAN. Choose **no** to disable.
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.
Enable PC Port VLAN Tagging	Enables VLAN and priority tagging on the phone data port (802.1p/q). This feature facilities tagging of the VLAN ID (802.1Q) and priority bits (802.1p) of the traffic coming from the PC port of the IP phone. Defaults to No. Choose **Yes** to enable the tagging algorithm.
PC Port VLAN Highest Priority	0-7 (default 0). The priority applied to all frames, tagged and untagged. The phone modifies the frame priority only if the incoming frame priority is higher than this value.
PC Port VLAN ID	0-4095 (default 0). Value of the VLAN ID. The phone tags all the untagged frames coming from the PC (it will not tag frames with an existing tag).
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.

Parameter	Description
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.

Wi-Fi Settings (Cisco SPA525G or Cisco SPA525G2 Only)

Parameter	Description
SPA525-wifi-on	Set to yes to enable Wireless-G service.

Bluetooth Settings (Cisco SPA525G or Cisco SPA525G2 Only)

Parameter	Description
Enable BT	Set to yes to enable support for Bluetooth devices.

VPN Settings (Cisco SPA525G or Cisco SPA525G2 Only)

Parameter	Description
VPN Server	The IP address of the VPN server to which the phone connects.
VPN User Name	Username configured on the VPN server for the phone.
VPN Password	Password associated with the username configured on the VPN for the phone.
VPN Tunnel Group	(Optional) The tunnel group, if required by the VPN server.
Connect on Bootup	If the phone should attempt to connect to the VPN each time it is powered on. Choose **yes** to have the phone try to automatically connect, or no to keep the default behavior.

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 Tab

This section describes the fields for the SIP tab.

SIP Parameters

Parameter	Description
Max Forward	SIP Max Forward value, which can range from 1 to 255. Defaults to 70.
Max Redirection	Number of times an invite can be redirected to avoid an infinite loop. Defaults to 5.
Max Auth	Maximum number of times (from 0 to 255) a request may be challenged. Defaults to 2.
SIP User Agent Name	Used in outbound REGISTER requests. Defaults to $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. Defaults to $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. Defaults to blank.
SIP Accept Language	Accept-Language header used. To access, click the SIP tab, and fill in the SIP Accept Language field. There is no default (this indicates SPA9000 does not include this header). If empty, the header is not included.

Parameter	Description
DTMF Relay MIME Type	MIME Type used in a SIP INFO message to signal a DTMF event. This field must match that of the Service Provider. Defaults to application/dtmf-relay.
Hook Flash MIME Type	MIME Type used in a SIP INFO message to signal a hook flash event. The default is application/hook-flash.
Remove Last Reg	Lets you remove the last registration before registering a new one if the value is different. Select yes or no from the drop-down menu. Defaults to no.
Use Compact Header	Lets you use compact SIP headers in outbound SIP messages. Select yes or no from the drop-down menu. If set to yes, the phone uses compact SIP headers in outbound SIP messages. If set to no, the phone uses normal SIP headers. If inbound SIP requests contain compact headers, the phone reuses the same compact headers when generating the response regardless the settings of the <Use Compact Header> parameter. If inbound SIP requests contain normal headers, the phone substitutes those headers with compact headers (if defined by RFC-261) if <Use Compact Header> parameter is set to yes. Default: no
Escape Display Name	Lets you 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. Any occurrences of or \ in the string is escaped with \ and \\ inside the pair of double quotes. Otherwise, select no. Defaults to yes.
Escape Special Character	Normally, the IP phone sends a “%23” (escape) as part of the message when the special character # is included in the SIP INVITE message. This can cause problem s with some telephony servers that need to receive the # character. When this parameter is set to **yes**, the phone sends a “%23” (escape) as part of the message when the # is included. When set to **no**, the # is sent directly and the escape (%23) is not used. Defaults to yes.
SIP-B Enable	Enables SIP for Business (supports Sylantro call flows) call features.
Talk Package	Enables support for the BroadSoft Talk Package, which enables a user to answer or resume a call by clicking a button in an external application.
Hold Package	Enables support for the BroadSoft Hold Package, which enables a user to place a call on hold by clicking a button in an external application.
Conference Package	Enables support for the BroadSoft Conference Package, which enables a user to start a conference by clicking a button in an external application.

Parameter	Description
Notify Conference	If enabled, the unit will send out a NOTIFY with event=conference when starting a conference.
RFC 2543 Call Hold	If set to yes, unit will include 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. Defaults to yes.
Random REG CID On Reboot	If set to yes, the Cisco IP 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. Defaults to no.
Mark All AVT packets	If set to yes, all audio video transport (AVT) tone packets (encoded for redundancy) have the marker bit set. If set to no, only the first packet has the marker bit set for each DTMF event. Defaults to yes.
SIP TCP Port Min	Specifies the lowest TCP port number that can be used for SIP sessions. Defaults to 5060.
SIP TCP Port Max	Specifies the highest TCP port number that can be used for SIP sessions. Defaults to 5080.
CTI Enable	The CTI interface allows a third-party application to control and monitor the state of a phone that has registered with the Cisco SPA9000. With this interface, an application can control a phone to initiate an outgoing call or answer an incoming call with a mouse click from a PC.
Caller ID Header	Provides the option to take the caller ID from PAID-RPID-FROM, P-ASSERTEDIDENTITY, REMOTE-PARTY-ID, or FROM header.
SRTP Method	Selects the method to use for SRTP. Two choices are available: • x-sipura—legacy SRPT method • s-descriptor—new method compliant with RFC-3711 and RFC-4568 The default value is "x-sipura.” **Note** Not applicable to Cisco WIP310.
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 value is "no,” where the call leg is not held. **Note** Not applicable to Cisco WIP310.
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.

Parameter	Description
Keep Referee When REFER Failed	Set this parameter to yes to configure the phone to immediately handle NOTIFY sipfrag messages. You can also configure this parameter in the configuration file: `<Keep_Referee_When_REFER_Failed ua="na">Yes </Keep_Referee_When_REFER_Failed>`

Display Diversion Info	Set to **yes** to cause the IP phone to parse and display the Diversion header in the incoming INVITE messages. The phone displays the information from the first Diversion header; if there are multiple Diversion headers, the others are ignored. The Diversion header follows the definition from RFC 5806. For example, for this diversion header in the INVITE messages:
	`Diversion:`
	`<sip:WeSellFlowers@p4.isp.com>;reason=time-of-day`
	The screen should display:
	`From xxx`
	`12345678`
	`Via: WeSellFlowers.`
	The reason field in the header is ignored. If the header field is not properly formed, Diversion header (all) will be ignored. If the privacy field is not set or is set to *off*, only the URI is displayed due to screen size limitation. If the privacy field is set to *full*, “anonymous” is displayed. If the privacy field is set to *name*, only the display name is displayed.

Display Anonymous From Header	Set to **yes** to show the caller ID from the SIP INVITE message “From” header, even if the call is an anonymous call. When the parameter is set to **no**, the phone displays "Anonymous Caller" as the caller ID.
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, the following happens: – Disables the display name in “Directory” and “Call History” in the “To” header during an outgoing call. – Ignores the CLID in the “UPDATE” message. – Redial list is populated based on 18x or 200 OK PAID header with or without Display Name. **Note** This field is supported in Firmware Release 7.6.2 and later.

SIP Timer Values (sec)

Parameter	Description
SIP T1	RFC 3261 T1 value (RTT estimate) that can range from 0 to 64 seconds. Defaults to .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. Defaults to 4 seconds.
SIP T4	RFC 3261 T4 value (maximum duration a message remains in the network) that can range from 0 to 64 seconds. Defaults to 5 seconds.
SIP Timer B	INVITE time-out value that can range from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer F	Non-INVITE time-out value that can range from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer H	INVITE final response, time-out value that can range from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer D	ACK hang-around time that can range from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer J	Non-INVITE response hang-around time that can range from 0 to 64 seconds. Defaults to 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 19999999999999999999999999999999. Defaults to 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 19999999999999999999999999999999. Defaults to 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. Defaults to 1 second.
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. Defaults to 7200.
Reg Retry Intvl (See note below)	Interval to wait before the SPA9000 retries registration after failing during the last registration. Defaults to 30.

Parameter	Description
Reg Retry Long Intvl (See note below)	When registration fails with a SIP response code that does not match<Retry Reg RSC>, the SPA9000 waits for the specified length of time before retrying. If this interval is 0, the SPA9000 stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. Defaults to 1200.
Reg Retry Random Delay	Random delay range (in seconds) to add to <Register Retry Intvl> when retrying REGISTER after a failure. This feature was added in Release 5.1. Defaults to blank, which disables this feature.
Reg Retry Long Random Delay	Random delay range (in seconds) to add to <Register Retry Long Intvl> when retrying REGISTER after a failure. This feature was added in Release 5.1. Defaults to blank, which disables this feature.
Reg Retry Intvl Cap	The maximum value to cap the exponential back-off retry delay (which starts at <Register Retry Intvl> and doubles on every REGISTER retry after a failure). In other words, the retry interval is always at <Register Retry Intvl> seconds after a failure. If this feature is enabled, <Reg Retry Random Delay> is added on top of the exponential back-off adjusted delay value. This feature was added in Release 5.1. Defaults to blank, which disables the exponential back-off feature.
Sub Min Expires	This value sets the lower limit of the REGISTER expires value returned from the Proxy server.
Sub Max Expires	This value sets the upper limit of the REGISTER min-expires value returned from the Proxy server in the Min-Expires header. Defaults to 7200.
Sub Retry Intvl	This value (in seconds) determines the retry interval when the last Subscribe request fails. Defaults to 10.

Note Cisco IP phones can use a RETRY-AFTER value when 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 retrying 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* parameter.

Response Status Code Handling

Parameter	Description
SIT1 RSC	SIP response status code for the appropriate Special Information Tone (SIT). For example, if you set the SIT1 RSC to 404, when the user makes a call and a failure code of 404 is returned, the SIT1 tone is played. Reorder or Busy Tone is played by default for all unsuccessful response status code for SIT 1 RSC through SIT 4 RSC. Defaults to blank.
SIT2 RSC	SIP response status code to INVITE on which to play the SIT2 Tone. Defaults to blank.
SIT3 RSC	SIP response status code to INVITE on which to play the SIT3 Tone. Defaults to blank.
SIT4 RSC	SIP response status code to INVITE on which to play the SIT4 Tone. Defaults to blank.
Try Backup RSC	This parameter may be set to invoke failover upon receiving specified response codes. Defaults to blank.
Retry Reg RSC	Interval to wait before the SPA9000 retries registration after failing during the last registration. Defaults to blank.

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 16402. Defaults to 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 16402. Defaults to 16482.
RTP Packet Size	Packet size in seconds, which can range from 0.01 to 0.16. Valid values must be a multiple of 0.01 seconds. Defaults to 0.030.

Parameter	Description
Max RTP ICMP Err	Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the SPA9000 terminates the call. If value is set to 0, the SPA9000 ignores the limit on ICMP errors. Defaults to 0.
RTCP Tx Interval	Interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. During an active connection, the SPA9000 can be programmed to send out compound RTCP packet on the connection. Each compound RTP packet except the last one contains a SR (Sender Report) and a SDES (Source Description). The last RTCP packet contains an additional BYE packet. Each SR except the last one contains exactly 1 RR (Receiver Report); the last SR carries no RR. The SDES contains CNAME, NAME, and TOOL identifiers. The CNAME is set to <User ID>@<Proxy>, NAME is set to <Display Name> (or Anonymous if user blocks caller ID), and TOOL is set to the Vendor/Hardware-platform-software-version (such as Cisco/SPA9000-1.0.31(b)). The NTP timestamp used in the SR is a snapshot of the SPA9000 local time, not the time reported by an NTP server. If the SPA9000 receives a RR from the peer, it attempts to compute the round trip delay and show it as the <Call Round Trip Delay> value (ms) in the Info section of SPA9000 web page. Defaults to 0.
No UDP Checksum	Select yes if you want the SPA9000 to calculate the UDP header checksum for SIP messages. Otherwise, select no. Defaults to no.
Symmetric RTP	Enable symmetric RTP operation. If enabled, sends RTP packets to the source address and port of the last received valid inbound RTP packet. If disabled (or before the first RTP packet arrives) sends RTP to the destination as indicated in the inbound SDP. Defaults to no.
Stats In BYE	Determines whether the IP phone includes the P-RTP-Stat header or response to a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the drop-down menu. The format of the P-RTP-Stat header is: P-RTP-State: PS=<packets sent>,OS=<octets sent>,PR=<packets received>,OR=<octets received>,PL=<packets lost>,JI=<jitter in ms>,LA=<delay in ms>,DU=<call duration in s>,EN=<encoder>,DE=<decoder>. Defaults to no.

SDP Payload Types

The configured dynamic payloads are used for outbound calls only where the Cisco SPA9000 presents the SDP offer. For inbound calls with a SDP offer, the Cisco SPA9000 follows the caller dynamic payload type assignments.

The Cisco SPA9000 uses the configured codec names in its outbound SDP. The SPA9000 ignores the codec names in incoming SDP for standard payload types (0-95). For dynamic payload types, the Cisco SPA9000 identifies the codec by the configured codec names. Comparison is case-insensitive.

Parameter	Description
AVT Dynamic Payload	AVT dynamic payload type. Ranges from 96-127. Defaults to 101.
INFOREQ Dynamic Payload	INFOREQ dynamic payload type. Defaults to blank.
G726r16 Dynamic Payload	G.726-16 dynamic payload type. Ranges from 96-127. Defaults to 98. **Note** Not applicable to Cisco SPA525G or Cisco SPA525G2 or Cisco WIP310.
G726r24 Dynamic Payload	G.726-24 dynamic payload type. Ranges from 96-127. Defaults to 97. **Note** Not applicable to Cisco SPA525G or Cisco SPA525G2 or Cisco WIP310.
G726r32 Dynamic Payload	G726r32 dynamic payload type. The default is 2.
G726r40 Dynamic Payload	G.726-40 dynamic payload type. Ranges from 96-127. Defaults to 96. **Note** Not applicable to Cisco SPA525G or Cisco SPA525G2 or Cisco WIP310.
G729b Dynamic Payload	G729b Dynamic Payload type. Defaults to 99.
EncapRTP Dynamic Payload	EncapRTP Dynamic Payload type. Defaults to 112.
RTP-Start-Loopback Dynamic	RTP-Start-Loopback Dynamic Payload. Defaults to 113.
RTP-Start-Loopback Codec	RTP-Start-Loopback Codec. Select one of following: G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a, or G723. Cisco SPA525G or Cisco SPA525G2: G711u, G711a, G726-32, G729a, G722. Defaults to G711u.
AVT Codec Name	AVT codec name used in SDP. Defaults to telephone-event.

Parameter	Description
G711u Codec Name	G.711u codec name used in SDP. Defaults to PCMU.
G711a Codec Name	G.711a codec name used in SDP. Defaults to PCMA.
G726r16 Codec Name	G.726-16 codec name used in SDP. Defaults to G726-16. **Note** Not applicable to Cisco SPA525G or Cisco SPA525G2 or Cisco WIP310.
G726r24 Codec Name	G.726-24 codec name used in SDP. Defaults to G726-24. **Note** Not applicable to Cisco WIP310, Cisco SPA525G or Cisco SPA525G2.
G726r32 Codec Name	G.726-32 codec name used in SDP. Defaults to G726-32.
G726r40 Codec Name	G.726-40 codec name used in SDP. Defaults to G726-40. **Note** Not applicable to Cisco SPA525G or Cisco SPA525G2 or Cisco WIP310.
G729a Codec Name	G.729a codec name used in SDP. Defaults to G729a.
G729b Codec Name	G.729b codec name used in SDP. Defaults to G729ab.
G722 Codec Name	G.722 codec name used in SDP. Defaults to G722. **Note** Not supported on the Cisco WIP310.
EncapRTP Codec Name	EncapRTP codec name used in SDP. Defaults to encaprtp.

NAT Support Parameters

Parameter	Description
Handle VIA received	If you select yes, the phone processes the received parameter in the VIA header (this is inserted by the server in a response to any of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu. Defaults to no.
Handle VIA rport	If you select yes, the SPA9000 processes the rport parameter in the VIA header (this is inserted by the server in a response to any of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu. Defaults to no.
Insert VIA received	Inserts the received parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Select yes or no from the drop-down menu. Defaults to no.
Insert VIA rport	Inserts the rport parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Select yes or no from the drop-down menu. Defaults to no.
Substitute VIA Addr	Lets you use NAT-mapped IP:port values in the VIA header. Select yes or no from the drop-down menu. Defaults to no.
Send Resp To Src Port	Sends responses to the request source port instead of the VIA sent-by port. Select yes or no from the drop-down menu. Defaults to no.
STUN Enable	Enables the use of STUN to discover NAT mapping. Select yes or no from the drop-down menu. Defaults to no.
STUN Test Enable	If the STUN Enable feature is enabled and a valid STUN server is available, the SPA9000 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 SPA9000 detects symmetric NAT or a symmetric firewall, NAT mapping is disabled. Defaults to 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.

Parameter	Description
EXT IP	External IP address to substitute for the actual IP address of the SPA9000 in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed. If this parameter is specified, the SPA9000 assumes this IP address when generating SIP messages and SDP (if NAT Mapping is enabled for that line). However, the results of STUN and VIA received parameter processing, if available, supersede this statically configured value. Defaults to blank.
EXT RTP Port Min	External port mapping number of the RTP Port Min. 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. Defaults to blank.
NAT Keep Alive Intvl	Interval between NAT-mapping keep alive messages. Defaults to 15.
Redirect Keep Alive	If enabled, the IP phone redirects the keepalive message when SIP_301_MOVED_PERMANENTLY is received as the registration response.

Linksys Key System Parameters

Parameter	Description
Linksys Key System	Enable or disable the Linksys Key System on the IP phone. Defaults to yes.
Multicast Address	The multicast address is used by the Cisco SPA9000 to communicate with the Cisco SPA IP phones. Defaults to 224.168.168.168:6061.
Key System Auto Discovery	Enables or disables auto discovery of the call control server (for example, the Cisco SPA9000). Disable this feature for teleworkers or other scenarios where multicast does not work.
Key System IP Address	IP address of the call control server IP. Enter the IP address for teleworkers or other scenarios where multicast does not work.
Force LAN Codec	The choices are: none, G.711u, or G.711a. Defaults to none.
Auto Ans GrPage On Active Call	Used with the Cisco UC320. Allows you to enable or disable “auto answer” of a group page when there is an active call on the phone. When set to **yes** (the default), a group page is automatically answered by the phone even if the user is on an active call. When set to **no**, the page is not automatically answered, and the user can decide to answer the page or stay on the active call.

Provisioning Tab

For information about the Provisioning page, see the *Cisco Small Business IP Telephony Devices Provisioning Guide*.

Regional Tab

This section describes the fields for the Regional tab.

Call Progress Tone Description

Parameter	Description
Dial Tone	Prompts the user to enter a phone number. Defaults to 350@-19,440@-19;10(*/0/1+2).
Bluetooth Dial Tone (Cisco SPA525G or Cisco SPA525G2 only)	Indicates a bluetooth headset is paired and the user can make a call. Defaults to 350@-19,440@-19;1(0/*/0);10(*/0/1+2).
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. Defaults to 420@-16;10(*/0/1).
Prompt Tone	Prompts the user to enter a call forwarding phone number. Defaults to 520@-19,620@-19;10(*/0/1+2).
Busy Tone	Played when a 486 RSC is received for an outbound call. Defaults to 480@-19,620@-19;10(.5/.5/1+2).
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. Defaults to 480@-19,620@-19;10(.25/.25/1+2).
Off Hook Warning Tone	Played when the caller has not properly placed the handset on the cradle. Off Hook Warning Tone is played when Reorder Tone times out. Defaults to 480@10,620@0;10(.125/.125/1+2).
Ring Back Tone	Played during an outbound call when the far end is ringing. Defaults to 440@-19,480@-19;*(2/4/1+2).
Call Waiting Tone	Played when a call is waiting. Defaults to 440@-10;30(.3/9.7/1)
Confirm Tone	Brief tone to notify the user that the last input value has been accepted. Defaults to 600@-16; 1(.25/.25/1).

Parameter	Description
SIT1 Tone	Alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on theIP phone screen. Defaults to 985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0).
SIT2 Tone	Alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the IP phone screen. Defaults to 914@-16,1371@-16,1777@-16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0).
SIT3 Tone	Alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the IP phone screen. Defaults to 914@-16,1371@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT4 Tone	This is an alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the IP phone screen. Defaults to 985@-16,1371@-16,1777@-16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0).
MWI Dial Tone	Played instead of the Dial Tone when there are unheard messages in the caller's mailbox. Defaults to 350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2).
Cfwd Dial Tone	Played when all calls are forwarded. Defaults to 350@-19,440@-19;2(.2/.2/1+2);10(*/0/1+2).
Holding Tone	Informs the local caller that the far end has placed the call on hold. Defaults to 600@-19*(.1/.1/1,.1/.1/1,.1/9.5/1).
Conference Tone	Played to all parties when a three-way conference call is in progress. Defaults to 350@-19;20(.1/.1/1,.1/9.7/1).
Secure Call Indication Tone	Played when a call has been successfully switched to secure 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. Defaults to 397@-19,507@-19;15(0/2/0,.2/.1/1,.1/2.1/2).
Page Tone	Specifies the tone transmitted when the paging feature is enabled. Defaults to 600@-16;.3(.05/0.05/1).
Alert Tone	Played when an alert occurs. Defaults to 600@-19;.2(.05/0.05/1).

Parameter	Description
System Beep	Audible notification tone played when a system error occurs. Defaults to 600@-16;.1(.05/0.05/1). **Note** Applies to Cisco SPA525G or Cisco SPA525G2 or Cisco WIP310 only.
Call Pickup Tone	This field provides the ability to configure an audio indication for call pickup. The call pickup tone is the tone to play when the BLF button is configured and there is an incoming alert on the button. The default value for this parameter is 440@-10;30(.3/9.7/1), which is the same as the call waiting tone. This feature appears as follows in the phone configuration file: `<Call_Pickup_Tone ua="na">440@-10;30(.3/9.7/1)</Call_Pickup_Tone>`

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).
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. Defaults to 5.
Call Back Expires	Expiration time in seconds of a call back activation. Range: 0–65535 seconds. Defaults to 1800.
Call Back Retry Intvl	Call back retry interval in seconds. Range: 0–255 seconds. Defaults to 30.
Call Back Delay	Delay after receiving the first SIP 18x response before declaring the remote end is ringing. If a busy response is received during this time, the SPA9000 still considers the call as failed and keeps on retrying. Defaults to 0.5.
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. Defaults to 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. Defaults to 3.

Vertical Service Activation Codes

The following Vertical Service Activation Codes are automatically appended to the dial plan.

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 *98.

Parameter	Description
Call Back Act Code	Starts a callback when the last outbound call is not busy. Defaults to *66.
Call Back Deact Code	Cancels a callback. Defaults to *86.
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.
Block CID Deact Code	Removes caller ID blocking on all outbound calls. Defaults to *68.
Block CID Per Call Act Code	Blocks caller ID on the next outbound 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.

Parameter	Description
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 the next outbound call secure. (It is redundant if all outbound calls are secure by default.) Defaults to *18.
Secure One Call Deact Code	Makes the next outbound call not secure. (It is redundant if all outbound calls are not secure by default.) Defaults to *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.
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.
Media Loopback Code	The star code used for media loopback. Defaults to *03.

Parameter	Description
Referral Services Codes	These codes tell the SPA9000 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, etc. 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 SPA9000 to perform a blind transfer to a target number that is prepended by the service *code. For example, after the user dials *98, the SPA9000 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 SPA9000 sends a blind REFER to the holding party with the Refer-To target equals to *98<target_number>. This feature allows the Cisco SPA9000 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 Cisco SPA9000. You can empty the corresponding *code that you do not want to Cisco SPA9000 to process.
Feature Dial Services Codes	These codes tell the Cisco SPA9000 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 Cisco SPA9000 to call the target number prepended by the *code. For example, after user dials *72, the Cisco SPA9000 plays a prompt tone awaiting the user to enter a valid target number. When a complete number is entered, the Cisco SPA9000 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).

Parameter	Description
Feature Dial Services Codes (continued	The *codes should not conflict with any of the other vertical service codes internally processed by the Cisco SPA9000. You can empty the corresponding *code that you do not want to Cisco SPA9000 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‘l*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 Cisco SPA9000 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 SPA9000 send INVITE *73@..... as usual when user dials *73.

Vertical Service Announcement Codes

- Service Annc (Announcement) Base Number: Defaults to blank.
- Service Annc (Announcement) Extension Codes: Defaults to blank.

Outbound Call Codec Selection Codes

These codes automatically appended to the dial plan. You do not need to include them in the dial plan.

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

Parameter	Description
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. **Note** Not supported on the Cisco WIP310.
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. **Note** Not supported on the Cisco WIP310.
Prefer L16 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *01016.
Force L16 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *02016.
Prefer G723 Code	Makes this codec the preferred codec for the associated call. Defaults to *01723. **Note** Not applicable to Cisco WIP310, Cisco SPA525G or Cisco SPA525G2.
Force G723 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *02723. **Note** Not applicable to Cisco WIP310, Cisco SPA525G or Cisco SPA525G2.
Prefer G726r16 Code	Makes this codec the preferred codec for the associated call. Defaults to *0172616. **Note** Not applicable to Cisco WIP310, Cisco SPA525G or Cisco SPA525G2.
Force G726r16 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *0272616. **Note** Not applicable to Cisco WIP310, Cisco SPA525G or Cisco SPA525G2.

Parameter	Description
Prefer G726r24 Code	Makes this codec the preferred codec for the associated call. Defaults to *0172624. **Note** Not applicable to Cisco WIP310, Cisco SPA525G or Cisco SPA525G2.
Force G726r24 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *0272624. **Note** Not applicable to Cisco WIP310, Cisco SPA525G or Cisco SPA525G2.
Prefer G726r32 Code	Makes this codec the preferred codec for the associated call. Defaults to *0172632.
Force G726r32 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *0272632.
Prefer G726r40 Code	Makes this codec the preferred codec for the associated call. Defaults to *0172640. **Note** Not applicable to Cisco WIP310, Cisco SPA525G or Cisco SPA525G2.
Force G726r40 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *0272640. **Note** Not applicable to Cisco WIP310, Cisco SPA525G or Cisco SPA525G2.
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.

Time (Cisco SPA525G or Cisco SPA525G2 Only)

Parameter	Description
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. Defaults to GMT-08:00.
Time Offset	This specifies the offset from GMT to use for the local system time.
Ignore DHCP Time Offset	See **Ignore DHCP Time Offset** in Miscellaneous.

Parameter	Description
Daylight Saving Time Rule	See **Daylight Saving Time Rule** in Miscellaneous.
Daylight Saving Time Enable	Select **yes** to enable Daylight Saving Time.

Language (Cisco SPA525G or Cisco SPA525G2 Only)

Parameter	Description
Dictionary Server Script.	See **Dictionary Server Script** in Miscellaneous.
Language Selection	See **Language Selection** in Miscellaneous.

Miscellaneous

Parameter	Description
Set Local Date (mm/dd)	Sets the local date (mm represents the month and dd represents the day). The year is optional and uses two or four digits. **Note** Not applicable to the Cisco SPA525G or Cisco SPA525G2.
Set Local Time (HH/mm)	Sets the local time (hh represents hours and mm represents minutes). Seconds are optional. **Note** Not applicable to the Cisco SPA525G or Cisco SPA525G2.
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. Defaults to GMT-08:00. **Note** Found in the **Time** section for the Cisco SPA525G or Cisco SPA525G2.
Time Offset (HH/mm)	This specifies the offset from GMT to use for the local system time. **Note** Found in the **Time** section for the Cisco SPA525G or Cisco SPA525G2.
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. The default value is **yes**. **Note** Found in the **Time** section for the Cisco SPA525G or Cisco SPA525G2.

Parameter	Description
Daylight Saving Time Rule	Enter the rule for calculating daylight saving time; it should include the start, end, and save values. This rule is comprised of three fields. Each field is separated by ; (a semicolon) as shown below. Optional values inside [ ] (the brackets) are assumed to be 0 if they are not specified. Midnight is represented by 0:0:0 of the given date. This is the format of the rule: Start = <start-time>; end=<end-time>; save = <save-time>. The <start-time> and <end-time> values specify the start and end dates and times of daylight saving time. Each value is in this format: <month> /<day> / <weekday>[/HH:[mm[:ss]]] The <save-time> value is the number of hours, minutes, and/or seconds to add to the current time during daylight saving time. The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition. The <save-time> value is in this format: [/[+|-]HH:[mm[:ss]]] The <month> value equals any value in the range 1-12 (January-December). The <day> value equals [+|-] any value in the range 1-31. If <day> is 1, it means the <weekday> on or before the end of the month (in other words the last occurrence of < weekday> in that month).
Daylight Saving Time Rule (continued)	The <weekday> value equals any value in the range 1-7 (Monday-Sunday). It can also equal 0. If the <weekday> value is 0, this means that the date to start or end daylight saving is exactly the date given. In that case, the <day> value must not be negative. If the <weekday> value is not 0 and the <day> value is positive, then daylight saving starts or ends on the <weekday> value on or after the date given. If the <weekday> value is not 0 and the <day> value is negative, then daylight saving starts or ends on the <weekday> value on or before the date given. Where: HH stands for hours (0-23). mm stands for minutes (0-59). ss stands for seconds (0-59). The default Daylight Saving Time Rule is start=4/1/7;end=10/-1/7;save=1. **Note** Found in the **Time** section for the Cisco SPA525G or Cisco SPA525G2.
Daylight Saving Time Enable	Select yes to enable Daylight Saving Time. **Note** Found in the **Time** section for the Cisco SPA525G or Cisco SPA525G2.
DTMF Playback Level	Local DTMF playback level in dBm, up to one decimal place. Defaults to -16.
DTMF Playback Length	Local DTMF playback duration in milliseconds. Defaults to .1.

Parameter	Description
Inband DTMF Boost	Controls the amount of amplification applied DTMF signals. Choices are OdB, 3dB, 6dB, 9dB, 12dB, 15dB, or 18dB. Defaults to 12dB.

Dictionary Server Script/SCCP Dictionary Server Script (Cisco SPA525G or Cisco SPA525G2 SCCP only)	Defines the location of the dictionary server, the languages available and the associated dictionary. The syntax is as follows:
	`<Dictionary_Server_Script ua="na">`
	`</Dictionary_Server_Script>`
	Defaults to blank and the maximum number of characters is 512. The detailed format is as follows:
	`serv={server ip port and root path};`
	`d0=<language0>;x0=<dictionary0 filename>;`
	`d1=<language1>;x1=<dictionary1 filename>;`
	`d2=<language2>;x2=<dictionary2 filename>;`
	`d3=<language3>;x3=<dictionary3 filename>;`
	`d4=<language4>;x4=<dictionary4 filename>;`
	`d5=<language5>;x5=<dictionary5 filename>;`
	`d6=<language6>;x6=<dictionary6 filename>;`
	`d7=<language3>;x7=<dictionary7 filename>;`
	`d8=<language8>;x8=<dictionary8 filename>;`
	`d9=<language5>;x9=<dictionary9 filename>;`

	The following is an example value:
	`<Dictionary_Server_Script ua="na">`
	`serv=tftp://192.168.1.119/;d0=English;x0=enS_v101.xml;d1=S`
	`panish;x1=esS_v101.xml </Dictionary_Server_Script>`
	Note Not applicable to the Cisco WIP310.

Language Selection/SCCP Language Selection (Cisco SPA525G or Cisco SPA525G2 SCCP Only)	Specifies the default language. The value needs to match one of the languages supported by the dictionary server. The script (dx value) is as follows:
	`<Language_Selection ua="na">`
	`</Language_Selection>`
	Defaults to blank and the maximum number of characters is 512. The following is an example:
	`<Language_Selection ua="na"> Spanish`
	`</Language_Selection>`
	Note Not applicable to the Cisco WIP310.

Default Character Encoding (Cisco SPA303, Cisco SPA500 Series)	The default is ISO-8859-1 for backward compatibility with Cisco SPA900 series phones. If set to UTF-8, line keys and other labels entered by using the phone web user interface containing UTF-8 characters are displayed correctly on the phone. (SIP only)
Locale	Choose the locale that should be set in the HTTP Accept-Language header.

Phone Tab

This section describes the fields for the Phone tab.

General

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>Station Display Name</td><td>Name to identify the IP phone; appears on the IP phone screen on models that have a display. You can use spaces in this field and the name does not have to be unique. If both the Station Display Name and Station Name fields are populated, the Station Display Name field takes precedence and is displayed on the phone.</td></tr>
<tr><td>Station Name</td><td>Name to identify this IP phone; appears on the IP phone screen on models that have a display. No spaces are allowed and the name must be unique.</td></tr>
<tr><td>Voice Mail Number</td><td>Phone number or URL to check voice mail. The service provider often hosts a voice mail service. The advantages of hosted voice mail include:
Advanced features such as voice mail to email conversion.
Calls can go to voice mail when the broadband connection is down.</td></tr>
<tr><td>Text Logo</td><td>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, <code>Super\n%0aTelecom</code> displays:<pre>Super
Telecom</pre>Use the + character to add spaces for formatting. For example, you can add multiple + characters before and after the text to center it.
Note Not applicable to the Cisco WIP310, Cisco SPA301, or Cisco SPA501G. On the Cisco SPA525G or Cisco SPA525G2, this setting is located on the User tab.</td></tr>
<tr><td>BMP Picture Download URL</td><td>URL locating the bitmap (.BMP) file to display on the IP phone screen background.
For more information, see Configuring Phone Information and Display Settings.
Note Not applicable to the Cisco WIP310, Cisco SPA301, or Cisco SPA501G. On the Cisco SPA525G or Cisco SPA525G2, this setting is located on the User tab.</td></tr>
</table>

Parameter	Description
Select Logo	Select from Default, BMP Picture, Text Logo, or None. Defaults to Default. **Note** Not applicable to the Cisco WIP310, Cisco SPA301, or Cisco SPA501G. On the Cisco SPA525G or Cisco SPA525G2, this setting is located on the **User** tab.
Select Background Picture	Select from Default, BMP Picture, or None. Defaults to Default. **Note** Not applicable to the Cisco WIP310, Cisco SPA301, or Cisco SPA501G. On the Cisco SPA525G or Cisco SPA525G2, this setting is located on the **User** tab.
Softkey Labels Font	Choose the font width for the softkey labels to display on your phone. See Customizing Phone Softkeys. **Note** Not applicable to the Cisco WIP310, Cisco SPA301, or Cisco SPA501G.
Screen Saver Enable	Enables a screen saver on the IP phone screen. When the phone is idle for a specified time, it enters screen saver mode. (Users can set up screen savers directly using phone Setup button.) When the screen saver is active, the screen shows "Press any key to unlock your phone." User input is accepted if the screen saver is active (for example, if a user begins to dial a number while the screen saver is active, the key press is accepted). If the phone LCD is displaying the Home screen and the screen saver becomes active, and if there are no active calls, the following events generated by user input are passed to the Home screen: • Numeric keys • Line keys • Speaker key • Headset key • Mail Box key • Handset off hook All other key events are not passed. **Note** Not applicable to the Cisco WIP310, Cisco SPA301, or Cisco SPA501G. On the Cisco SPA525G or Cisco SPA525G2, this setting is located on the **User** tab.
Screen Saver Wait	Amount of idle time before screen saver displays. **Note** Not applicable to the Cisco WIP310, Cisco SPA301, or Cisco SPA501G. On the Cisco SPA525G or Cisco SPA525G2, this setting is located on the **User** tab.

Parameter	Description
Screen Saver Icon	In screen saver mode, the IP phone screen can display: • A background picture. • The station time in the middle of the screen. • A moving padlock icon. When the phone is locked, the status line displays a scrolling message “Press any key to unlock your phone.” • A moving phone icon. • The station date and time in the middle of the screen. **Note** Not applicable to the Cisco WIP310, Cisco SPA301, or Cisco SPA501G. On the Cisco SPA525G or Cisco SPA525G2, this setting is located on the **User** tab.
JPEG Logo Download URL (Cisco SPA525G or Cisco SPA525G2)	URL from which to download a .jpg file for the phone logo display.
JPEG Wallpaper Download URL (Cisco SPA525G or Cisco SPA525G2)	URL from which to download a .jpg file for the phone wallpaper.
Enable SMS	Enables sending and receiving of SMS text messages on the phone. **Note** Cisco WIP310 only.
Show DTMF Digits when connected	The phone can be configured to show the digits entered by a local user while a call is connected or is proceeding. The display options are "No"/"Yes"/"Masked". 'No'— No digits are displayed. 'Yes'— Upto 15 most recently dialed digits are displayed. ‘Masked—Behaves like 'Yes,' but the digits are masked with '*****'s. **Note** Display location of DTMF digits is slightly different on Cisco SPA5xx and Cisco SPA525 phones. **SPA5xx**: Dialed digits are displayed on the second line of the call screen. If the call is diverted while a new digit is entered, the line switches from "Via: XXXX" to digits for 10 seconds. If no digits are entered during the last 10 seconds, the line will switch back to "Via:XXXX." **SPA525**: Dialed digits are displayed on the fourth line. For a conference call, two call appearances present on the GUI, and the digits are displayed only on the highlighted call.

Line Key

When used in the configuration profile, parameters in this section must be appended with *n*, where *n* represents line 1, 2, 3, 4, 5 or 6. For more information on these parameters, see Configuring Lines.

Note Does not apply to the Cisco WIP310.

Parameter	Description
Extension	Extension number of the line key.
Short Name	A short label shown on the IP phone screen for Line Key 1 through Line Key 6.
Share Call Appearance	**Yes** indicates that Line Key 1/2/3/4/5/6 is a shared call appearance. Otherwise this call appearance is not shared (it is private). Defaults to no.
Extended Function	Use to assign Busy Lamp Field, Call Pickup, and Speed Dial Functions to Idle Lines on the IP phone. Syntax is: `fnc=type;sub=stationname@$PROXY;ext=extension#@$PROXY` where: • fnc: function • blf: busy lamp field • cp: call pickup • sub: station name (not needed for speed dial) • ext or usr: extension or user (the usr and ext keywords are interchangeable) **Note** Not applicable to the Cisco WIP310, Cisco SPA301, or Cisco SPA501G.

Miscellaneous Line Key Settings

Does not apply to the Cisco WIP310.

Parameter	Description
Line ID Mapping	Specifies the call appearance line ID mapping when the call appearance per line is set to two. Applied to both shared line and private lines. Effective for incoming calls only. Choose **Vertical First** or **Horizontal First**. Each LED can hold multiple calls and the first call on an LED makes it light up. **Horizontal first** means the second incoming call makes the same line LED flash. **Vertical first** means the second incoming call lights up the next line LED with the same account. For example, if Extension 101 is assigned to two LEDs, and Vertical First is selected, the second incoming call on Extension 101 lights up the second LED. The third call makes the first LED flash, and the fourth call makes the second LED flash. If Horizontal First is selected, the second incoming call on Extension 101 makes the first LED flash. The third call lights up the second LED, and the fourth call makes the second LED flash.
SCA Barge-In Enable	Enables the SCA Barge-In. Defaults to no.
SCA Sticky Auto Line Seize	When enabled, taking the phone off-hook will not automatically pick up an incoming call on a shared line.
SCA Unseize Delay	Allows shared line unseize delay to be Configurable. Default is 0.
Line Navigation (Does not apply to the Cisco SPA525G or Cisco SPA525G2.)	This parameter controls the way that the IP phone navigates between call appearances when the user presses the up or down navigation button. The default value is “per line.” When **Per Call** is configured, and the user presses the up and down navigation button, the phone displays each call for a line before moving to subsequent lines. For example, if there are two calls on Line 1, pressing the up or down navigation button scrolls between the two calls first, before moving to display any calls that are on Line 2. We recommend you enable programmable softkeys when choosing this option. When **Per Line** is configured, and the user presses the up and down navigation button, the phone displays one call appearance for a line, before moving to the next line. For example, if there are two calls on Line 1, pressing the up or down navigation button moves directly to display any calls on Line 2 (rather than the second call on Line 1).

Parameter	Description
Call Appearance Per Line	This parameter allows you to choose the number of calls per line button. You can choose a value from 2 (the default) to 10. When you increase the number of calls per line to a value greater than 2, the phone automatically sets the **Line ID Mapping** option to **Horizontal First**. Changing the **Line ID Mapping** value has no effect if the value of the Call Appearance Per Line setting is greater than 2. **Note** The maximum number of calls per an SPA525G or 525G2 phone is 10. When the maximum numbers of calls per phone is reached, the phone does not allow you to make a new call and rejects incoming calls. This parameter is not supported on the Cisco SPA501G and Cisco SPA301 phones. Also, this parameter is only supported when the phones are operating in the SIP mode. In version 7.4.8a, only private call appearances are supported. In versions 7.5.1a and higher private and shared call appearances are supported.
Softkey Navigation Style	If **scrollable** is chosen, users must use the navigation button on the IP phone to scroll to see the additional softkeys. If **more** is chosen, a “More” softkey is displayed, which, when pressed, displays the additional softkeys. **Note** The Cisco SPA525G and Cisco SPA525G2 always display the “More” softkey when in an idle state.

Line Key LED Pattern

Does not apply to the Cisco WIP310.

Parameter	Description
Idle LED	The call appearance is not is in use and is available to make a new call. Leaving this entry blank indicates the default value of c=g.
Remote Undefined LED	The shared call state is undefined (the phone is still waiting for the state information from the application server). Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Seized LED	This phone has seized the call appearance to prepare for a new outbound call. Leaving this entry blank indicates the default value of c=r.
Remote Seized LED	The shared call appearance is seized by another phone. Not application if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Progressing LED	This phone is attempting on this call appearance an outgoing call that is in proceeding (i.e. the called number is ringing). Leaving this entry blank indicates the default value of c=r.

Parameter	Description
Remote Progressing LED	Another phone is attempting on this shared call appearance an outbound call that is progressing. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Ringing LED	The call appearance is ringing. Leaving this entry blank indicates the default value of c=r;p=f.
Remote Ringing LED	The shared call appearance is in ringing on another phone. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Active LED	The call appearance is engaged in an active call. Leaving this entry blank indicates the default value of c=r.
Remote Active LED	Another station is engaged in an active call on this shared call appearance. Not applicable is this call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Held LED	The call appearance is held by this phone. Leaving this entry blank indicates the default value of c=r;p=s.
Remote Held LED	Another phone has placed this call appearance on hold. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r,p=s.
Register Failed LED	The corresponding extension has failed to register with the proxy server. Leaving this entry blank indicates the default value of c=a.
Disabled LED	The Call Appearance is disabled (not available for any incoming or outgoing call). Leaving this entry blank indicates the default value of c=o.
Registering LED	The corresponding extension is trying to register with the proxy server. Leaving this entry blank indicates the default value of c=r;p=s.
Call Back Active LED	Call Back operation is currently active on and this call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=s.
Trunk In-Use LED	A shared trunk is in use.
Trunk No Service LED	A shared trunk is not in service.
Trunk Reserved LED	A shared trunk has been reserved.

Supplementary Services

Enable or disable the corresponding supplementary services on the phone. A value of **yes** indicates enabled; **no** indicates disabled.

Parameter	Description
Conference Serv	Enable/disable Three way conference service. Defaults to yes.
Attn Transfer Serv	Enable/disable attended-call-transfer service. Defaults to yes.
Blind Transfer Serv	Enable/disable blind-call-transfer service. Defaults to yes.
DND Serv	Enable/disable do-not-disturb service. Defaults to yes.
Block ANC Serv	Enable/disable block-anonymous-call service. Defaults to yes.
Call Back Serv	Enable/disable call-back (aka. repeating dialing) service. Defaults to yes.
Block CID Serv	Enable/disable blocking outbound Caller-ID service. Defaults to yes.
Secure Call Serv	Enable/disable secure-call service. Defaults to yes.
Cfwd All Serv	Enable/disable call-forward-all service. Defaults to yes.
Cfwd Busy Serv	Enable/disable call-forward-on-busy service. Defaults to yes.
Cfwd On No Ans Serv	Enable/disable call-forward-on-no-answer service. Defaults to yes.
Paging Serv	Enable/disable the paging service. Defaults to yes.
Call Park Serv	Enable/disable the call park service. Defaults to yes.
Call Pick Up Serv	Enable/disable the call pickup service. Defaults to yes.
ACD Login Serv	Enable/disable the ACD Login Service, used for call centers. Typically enabled with the <SIP-B> parameter. Defaults to no.
Group Call Pick Up Serv	Enable/disable the group call pickup service. Defaults to yes.

Parameter	Description
Group Call Pick Up Serv	Enable/disable the group call pickup service. Defaults to yes.
ACD Ext	The extension used for handling ACD calls. Select from 1, 2, 3, 4, 5, or 6. Defaults to 1.
Service Annc Serv	Enable/disable sending announcement requests to a customer-supplied announcement server. Defaults to no.
Web Serv (Cisco SPA525G or Cisco SPA525G2 only)	Enable/disable the web server. Defaults to yes.
SMS Serv (Cisco SPA525G or Cisco SPA525G2 only)	Enable/disable the SMS text messaging server.

Ring Tone (Cisco SPA300 Series and Cisco SPA500 Series)

Each entry defines a ring tone to be used on the phone, with an ID between 1 and 12. The ID can be used in a DirEntry to indicate which ring tone to use when the corresponding caller calls.

Parameter	Description
Ring1	Ring tone script for ring 1. Defaults to n=Classic-1;w=3;c=1.
Ring2	Ring tone script for ring 2. Defaults to n=Classic-2;w=3;c=2.
Ring3	Ring tone script for ring 3. Defaults to n=Classic-3;w=3;c=3.
Ring4	Ring tone script for ring 4. Defaults to n=Classic-4;w=3;c=4.
Ring5	Ring tone script for ring 5. Defaults to n=Simple-1;w=2;c=1.
Ring6	Ring tone script for ring 6. Defaults to n=Simple-2;w=2;c=2.
Ring7	Ring tone script for ring 7. Defaults to n=Simple-3;w=2;c=3.
Ring8	Ring tone script for ring 8. Defaults to n=Simple-4;w=2;c=4.
Ring9	Ring tone script for ring 9. Defaults to n=Simple-5;w=2;c=5.
Ring10	Ring tone script for ring 10. Defaults to n=Office;w=4;c=1.
Ring11	(Cisco SPA300 Series and Cisco SPA500 Series) Ring tone script for ring 11. Defaults to n=Pulse;w=5;c=1. (Cisco SPA525G or Cisco SPA525G2) Ring tone script for ring 11. Defaults to n=Pulse;w=file://Pulse1.raw;c=1.

Parameter	Description
Ring12	(Cisco SPA300 Series and Cisco SPA500 Series) Ring tone script for ring 12. Defaults to n=Du-dut;w=6;c=1. (Cisco SPA525G or Cisco SPA525G2) Ring tone script for ring 11. Defaults to n=Du-dut;w=file://Ring7.raw;c=1.
Silent Ring Duration	(Cisco SPA 300 Series and Cisco SPA500 Series) 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.

In addition to these two ring tones, four user-configurable ring tones were added. See the "Configuring Ring Tones" section on page 3-18 for more information.

Label	Value of the w Parameter
Warble	(Cisco SPA300 Series and Cisco SPA500 Series) w=7 ((Cisco SPA525G or Cisco SPA525G2)) w=file://Warble.raw
Low	(Cisco SPA300 Series and Cisco SPA500 Series) w=8 ((Cisco SPA525G or Cisco SPA525G2)) w=file://Low.raw
Floor	(Cisco SPA300 Series and Cisco SPA500 Series) w=9 ((Cisco SPA525G or Cisco SPA525G2)) w=file://Floor.raw
Reverb	(Cisco SPA300 Series and Cisco SPA500 Series) w=10 ((Cisco SPA525G or Cisco SPA525G2)) w=file://Reverb.raw

Ring Tone (Cisco WIP310)

Parameter	Description
Keypad Tone	Select **yes** to enable the keypad tone to be played when a key on the keypad is pressed. Select no to silence the keypad.
Keypad Tone Volume	Corresponds to the volume of the keypad tone. Default is 5.

Audio Input Gain (dB)

Note Does not apply to the Cisco WIP310.

Parameter	Description
Handset Input Gain	The amount of amplification to apply to the audio input signal for the handset. Defaults to zero.
Headset Input Gain	The amount of amplification to apply to the audio input signal for the headset. Defaults to zero. **Note** Not applicable to the Cisco SPA301 or the Cisco SPA501G.
Speakerphone Input Gain	The amount of amplification to apply to the audio input signal for the speakerphone. Defaults to zero. **Note** Not applicable to the Cisco SPA301 or the Cisco SPA501G.
Bluetooth Input Gain (Cisco SPA525G or Cisco SPA525G2 only)	The amount of amplification to apply to the audio input signal for the Bluetooth device. Defaults to zero.
Handset Additional Input Gain	Applies additional input gain to the handset. The maximum value is 9. **Note** Does not apply to the Cisco SPA525G or Cisco SPA525G2.
Headset Additional Input Gain	Applies additional input gain to the headset. The maximum value is 9. **Note** Does not apply to the Cisco SPA525G or Cisco SPA525G2, or the Cisco SPA501G.
Speakerphone Additional Input Gain	Applies additional input gain to the speakerphone. The maximum value is 9. **Note** Does not apply to the Cisco SPA301 or Cisco SPA525G or Cisco SPA525G2 or Cisco SPA501G.
Bluetooth Additional Input Gain (Cisco SPA525G or Cisco SPA525G2 only)	Applies additional input gain to the Bluetooth device. Defaults to zero. The maximum value is 9.

Multiple Paging Group Parameters

You can configure a phone as part of a paging group by using a Group Paging Script. Users can then direct pages to specific groups of phones. A phone can be part of no more than two paging groups, and user can page a maximum of five paging groups.

Syntax:

```
[pggrp=ip-address:port;[name=xxx;]num=xxx;
[listen={yes|no}]];
```

Where:

IP address: Multicast IP address of the phone that listens for and receives pages.

port: Port on which to page; you must use different ports for each paging group.

name (optional): The name of the paging group. In this name, do not use the pggrp string because it is reserved. Using it causes the script not to work, as in these examples:

```
pggrp=224.168.168.168:3141;name=ITGPgGrp;
num=800;listen=yes;
pggrp=224.168.168.168:3141;name=PgGrp;num=800;listen=yes;
```

num: The number users dial to access the paging group; must be unique to the group.

listen: If the phone should listen on the page group. Only the first two groups with listen set to yes will listen to group pages. If the field is not defined, the default value is no, so you must set this field to listen to the group pages.

BroadSoft Settings

The Cisco SPA300 Series and Cisco SPA500 Series supports the BroadSoft directory feature and synchronization of Do Not Disturb and Call Forward.

Parameter	Description
Directory Enable	Set to **yes** to enable BroadSoft directory for the phone user. Defaults to no. **Note** Not applicable to the Cisco SPA301 or Cisco SPA501G.
XSI Host Server	Enter the name of the server; for example, xsp.xdp.broadsoft.com. **Note** Not applicable to the Cisco SPA301 or Cisco SPA501G.
Directory Name	Name of the directory. Displays on the user phone as a directory choice. **Note** Not applicable to the Cisco SPA301 or Cisco SPA501G.

Parameter	Description
Directory Type	Select the type of BroadSoft directory: Enterprise (default): 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. **Note** Not applicable to the Cisco SPA301 or Cisco SPA501G.
Directory UserID	BroadSoft User ID of the phone user; for example, johndoe@xdp.broadsoft.com. **Note** Not applicable to the Cisco SPA301 or Cisco SPA501G.
Directory Password	Alphanumeric password associated with the User ID. **Note** Not applicable to the Cisco SPA301 or Cisco SPA501G.
Call Feature Sync Ext	Allows the phone to synchronize with the call server so that if Do Not Disturb or Call Forwarding settings are changed on the phone, changes are also made on the server; if changes are made on the server, they are propagated to the phone. This feature is disabled by default. Choose the extension (1 through 5) that is registered to the BroadSoft server. **Note** This parameter was removed in software release 7.4.9, and replaced by the Feature Key Sync parameter in the Extension tab.

LDAP Corporate Directory Search

If using Active Directory with authentication set to MD5, you must first configure the following:

- Click the **System** tab. In the **Optional Network Configuration** section, under **Primary DNS**, enter the IP address of the DNS server.
- In the **Optional Network Configuration** section, under **Domain**, enter the Lightweight Directory Access Protocol (LDAP) domain.

Note Does not apply to the Cisco WIP310, Cisco SPA301, or Cisco SPA501G.

Parameter	Description
LDAP Dir Enable	Choose **yes** to enable LDAP.
LDAP Corp Dir Name	Enter a free-form text name, such as “Corporate Directory.”

Parameter	Description
LDAP Server	Enter a fully qualified domain name or IP address of LDAP server, in the following format: `nnn.nnn.nnn.nnn`
LDAP 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.

LDAP Client DN	Enter the distinguished name domain components [dc] ; for example:
	`dc=cv2bu,dc=com`
	If 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`

LDAP Username	Enter the username for a credentialed user on the LDAP server.
LDAP Password	Enter the password for the LDAP username.
LDAP 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`
LDAP Last Name Filter	This defines the search for surnames [sn], known as last name in some parts of the world. For example, sn:(sn=*$VALUE*). This search allows the provided text to appear anywhere in a name, beginning, middle, or end.
LDAP First Name Filter	This defines the search for the common name [cn]. For example, cn:(cn=*$VALUE*). This search allows the provided text to appear anywhere in a name, beginning, middle, or end.
LDAP Search Item 3	Additional customized search item. Can be blank if not needed.
LDAP Item 3 Filter	Customized filter for the searched item. Can be blank if not needed.
LDAP Search Item 4	Additional customized search item. Can be blank if not needed.
LDAP Item 4 Filter	Customized filter for the searched item. Can be blank if not needed.

Parameter	Description
LDAP Display Attrs	Format of LDAP results display 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, then 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 will be ignored. • p—phone number When p is assigned to a type attribute, example t=p, then the retrieved number is dialable by the phone.
LDAP 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.

XML Service

The Cisco SPA300 Series and Cisco SPA500 Series IP phones support XML services, such as an XML Directory Service or other XML applications. (Not applicable to the Cisco SPA301 or the Cisco SPA501G.)

Parameter	Description
XML Directory Service Name	Name of the XML Directory. Displays on the user's phone as a directory choice.
XML Directory Service URL	URL where the XML Directory is located.

Parameter	Description
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.
XML Password	XML service password for authentication purposes.
Cisco XML EXE Auth Mode	Select the authentication mode for *http://ip/CGI/Execute*. There are 3 modes: **Trusted:** This is the default value. **Local Credential:** Use the password to authenticate. **Remote Credential:** Use XML User Name and XML Password to authenticate.

Extension Mobility

For more information, see Configuring Extension Mobility.

Note Does not apply to the Cisco WIP310.

Parameter	Description
Extension Mobility Enable	Enable or disable extension mobility. Defaults to no (disabled).
EM User Domain	The user domain for extension mobility. Defaults to blank.

Programmable Softkeys

The Cisco SPA300 Series and Cisco SPA500 Series IP phones (models with display screens) have four softkeys on the screen that, when pressed, perform certain actions.

You can customize the softkeys displayed on the phone, and create your own softkeys for speed dials or XML scripts. Customized softkey information is entered in the PSK1 through PSK16 fields. (See Customizing Phone Softkeys for more information.)

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 a call has been missed.
Off Hook Key List	Softkeys that display when the receiver is lifted, or the headphone or speakerphone buttons are pressed.

Parameter	Description
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. (Cisco SPA525G or Cisco SPA525G2 only)
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 disconnecting. (Cisco SPA525G or Cisco SPA525G2 only)
Hold Key List	Softkeys that display when one or more calls are on hold. (Cisco SPA525G or Cisco SPA525G2only)
Ringing Key List	Softkeys that display when a call is incoming.
Shared Active Key List	Softkeys that display when a call is active on a shared line. (Cisco SPA525G or Cisco SPA525G2 only)
Shared Held Key List	Softkeys that display when a call is on hold on a shared line. (Cisco SPA525G or Cisco SPA525G2 only)
PSK1 through PSK16	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. For more information on the types of softkeys that can be created and the string syntax to enter, see the "Programmable Softkeys" section on page 3-15.

Call Audio Recording (Cisco SPA525G and SPA525G2)

Parameter	Description
Record Enable	The Cisco SPA525G2 supports recording a call to a USB flash drive. Select yes to enable this feature. The user can record and play back the call using the phone interface. The following limitations apply: • Supported in SIP mode only. • Recording is limited to a single active call. Multiple calls cannot be recorded at the same time. If the user performs a blind transfer, call recording stops immediately. If the user performs an attended transfer, call recording stops after the local call exits from the bridged call. Call recording stops when a call is parked. If an incoming call is answered while on the first call, the first call continues to be recorded after the second call is connected. The calls must be joined by conference to record both calls. If a call is put on hold and the user makes another call, the first call continues to be recorded unless the calls are joined by conference. Call recording can only be performed on conferenced calls when the original recorded call is still active, or when the recording is started after both calls are joined by conference. If call recording is stopped on a call and started on another call, a new, separate call recording file is created. The remote call number and call name are part of the filename of the record file. If the filename contains any of these characters (\ / : * ? " < >), that character is replaced by '-' in the call recording filename.
Record Beep Reminder	When enabled, the phone plays a beep when call recording is started. This feature is enabled by default.

Ext Tab

The Ext tabs vary by phone and depend on the number of extensions the phone model supports. In a configuration profile, the Line parameters must be appended with the appropriate numeral to indicate the line to which the setting applies. For example:

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

General

Line Enable: To enable this line for service, select **yes**. Otherwise, select **no**.

Defaults to yes.

Share Line Appearance

Parameter	Description
Share Ext	Indicates whether this extension is to be shared with other Cisco SPA IP phones or private. If the extension is not shared, then a call appearance assigned to this extension is not shared, regardless the setting of `<Share Call Appearance>` for that call appearance. If the extension is shared, then whether or not a call appearance assigned to this extension is shared follows the setting of `<Share Call Appearance>` for that call appearance. The choices are shared or private. Defaults to shared.
Shared User ID	The user identified assigned to the shared line appearance.
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. Defaults to 60 seconds.
Restrict MWI	When enabled, the message waiting indicator lights only for messages on private lines.
Monitor User ID (Cisco SPA300 Series, Cisco SPA500 Series)	This field is for future use.

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**. Defaults to no.
NAT Keep Alive Enable	To send the configured NAT keep alive message periodically, select **yes**. Otherwise, select **no**. Defaults to 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. Defaults to $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. Defaults to $PROXY.

Network Settings

Parameter	Description
SIP TOS/DiffServ Value	TOS/DiffServ field value in UDP IP packets carrying a SIP message. Defaults to 0x68.
SIP CoS Value	CoS value for SIP messages. Defaults to 3.
RTP TOS/DiffServ Value	ToS/DiffServ field value in UDP IP packets carrying RTP data. Defaults to 0xb8.
RTP CoS Value	CoS value for RTP data. Defaults to 6.
Network Jitter Level	Determines how jitter buffer size is adjusted by the Cisco SPA9000. Jitter buffer size is adjusted dynamically. The minimum jitter buffer size is 30 milliseconds or (10 milliseconds + current RTP frame size), whichever is larger, for all jitter level settings. However, the starting jitter buffer size value is larger for higher jitter levels. This setting controls the rate at which the jitter buffer size is adjusted to reach the minimum. Select the appropriate setting: low, medium, high, very high, or extremely high. Defaults to high.
Jitter Buffer Adjustment	Controls how the jitter buffer should be adjusted. Select the appropriate setting: up and down, up only, down only, or disable. Defaults to up and down.

SIP Settings

Parameter	Description
SIP Transport	Select from UDP, TCP, or TLS. Defaults to UDP.
SIP Port	Port number of the SIP message listening and transmission port. Defaults to 5060.
SIP 100REL Enable	To enable the support of 100REL SIP extension for reliable transmission of provisional responses (18x) and use of PRACK requests, select **yes**. Otherwise, select **no**. Defaults to no.
EXT SIP Port	The external SIP port number.

Parameter	Description
Auth Resync-Reboot	If this feature is enabled, the Cisco SPA9000 authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select **yes**. Otherwise, select **no**. Defaults to 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	To use the Remote-Party-ID header instead of the From header, select **yes**. Otherwise, select **no**. Defaults to yes.
Referor Bye Delay	Controls when the SPA9000 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 Referor Bye Delay, enter the appropriate period of time in seconds. Defaults to 4.
Refer-To Target Contact	To contact the refer-to target, select **yes**. Otherwise, select **no**. Defaults to no.
Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. Defaults to 0.

Parameter	Description
SIP Debug Option	SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Choices are as follows: none—No logging. 1-line—Logs the start-line only for all messages. 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. 1-line excl. OPT|NTFY|REG—Logs the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses. full—Logs all SIP messages in full text. full excl. OPT—Logs all SIP messages in full text except OPTIONS requests/responses. full excl. NTFY—Logs all SIP messages in full text except NOTIFY requests/responses. full excl. REG—Logs all SIP messages in full text except REGISTER requests/responses. full excl. OPT|NTFY|REG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. Defaults to none.
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. Defaults to 0.
Sticky 183	If this feature is 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**. Defaults to no.
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy.
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). **Note** Not applicable to the Cisco WIP310.

Parameter	Description
Use Anonymous With RPID	This parameter applies only if <SIP Remote-Party-ID> is set to **yes**; otherwise, it is ignored. If the parameter is set to **yes**, the FROM header's display-name and user-id fields are set to anonymous when the caller blocks his caller-id. If the parameter is set to **no**, the FROM header's display-name and user-id are not masked. The Remote-Party-ID header indicates privacy=full when the caller wishes to block his caller-id. Defaults to yes. **Note** Not applicable to the Cisco WIP310.
Set G729annexb	Configure G.729 Annex B settings. **Note** Not applicable to Cisco SPA525G or Cisco SPA525G2.

Parameter	Description
Voice Quality Report Address	For configuration of a SIP event package, SIP PUBLISH, that enables the collection and reporting of metrics that measure the quality for VoIP sessions. Voice call quality information derived from RTCP-XR and call information from SIP is conveyed from a User Agent in a session to the third party in SIP PUBLISH method. To configure, first configure RTCP-XR (see RTP Parameters). Configure the RTCP Tx Interval. In the Voice Quality Report Address field, enter the name of the collector that collects the statistics from the SIP PUBLISH events. For example, enter **collector@*fully-qualified-domain-name*** (collector@reports.cisco.com) or ***collector@IP-address*** (collector@192.168.5.1). After RTCP-XR feature is enabled, the call status information is updated on **Voice** > **Info** during an active call. Additionally, RTCP-XR packets containing a voice metrics block report are sent with the interval specified in the RTCP Tx Interval. When the call session is ended, a SIP PUBLISH with voice metrics information is sent to the collector endpoint. This parameter supports a full SIP URI. Examples of valid addresses are: • collector@domain.com • 123.collect@123.123.123.123:5555 • 5678@domain.com:5656 For example to configure for extension 1, edit the phone configuration file as follows: <Voice_Quality_Report_Address_1_ ua="na">collector@domain.com </Voice_Quality_Report_Address_1_> <Voice_Quality_Report_Address_1_ ua="na">123.collect@123.123.123.123:5555 </Voice_Quality_Report_Address_1_> <Voice_Quality_Report_Address_1_ ua="na">5678@domain.com:5656 </Voice_Quality_Report_Address_1_>
User Equal Phone	When a tel URL is converted to a SIP URL and the telephone number is represented by the user portion of the URI, the SIP URL includes the optional :user=phone parameter (RFC3261). For example: `To: sip:+12325551234@example.com;user=phone` To enable this optional parameter, choose **yes**. The default value is **no**.

Call Feature Settings

Parameter	Description
Blind Attn-Xfer Enable	Enables the IP 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 IP 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**. Defaults to no.
MOH Server	User ID or URL of the auto-answering streaming audio server. When only a user ID is specified, the current or outbound proxy is contacted. Music-on-hold is disabled if the MOH Server is not specified. Defaults to imusic when used with a Cisco SPA9000 IP PBX.
Message Waiting	Indicates whether the Message Waiting Indicator on the phone is lit. This parameter is toggled by a message from the SIP proxy to indicate if a message is waiting. You can manually modify it to clear or set the flag in the Ext 1-6 tab. Setting this value to Yes can activate stutter tone and VMWI signal. This parameter is stored in long-term memory and survives after reboot or power cycle. Defaults to No.
Auth Page	Specifies whether to authenticate the invite before auto answering a page. Defaults to No.
Default Ring	Type of ring heard. This corresponds to the Ring Tone on the Phone tab. Choose from No Ring, 1 through 10, User 1, or User 2. Defaults to 1.
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. Defaults to blank.
Conference Bridge URL	This is the URL used to join into a conference call, generally in the form of the word **conference** or **user@IPaddress:port**. Defaults to blank.
Auth Page Password	Identifies the password used when the Auth Page parameter is set to **yes**. This parameter accepts alphanumeric characters. Defaults to blank.
Mailbox ID	Identifies the voice mailbox number/ID for the phone. Defaults to blank.
Voice Mail Server	Identifies the SpecVM server for the phone, generally the IP address and port number of the VM server.

Parameter	Description
Voice Mail Subscribe Interval	The expiration time, in seconds, of a subscription to a voice mail server.
State Agent	Reserved feature.
CFWD Notify Serv	Specifies whether to enable a SIP-B feature regarding the sending of a Notify to the phone when a call is forwarded elsewhere. Defaults to No.
CFWD Notifier	Typically, this field is configured with the SIP proxy information.
User ID with Domain (Cisco SPA300 Series and SPA 50X IP Phones)	When this field is set to **yes**, the IP phone will show the Caller ID followed by domain name, and the domain name is also shown in the received calls list. This parameter is used when calls are made between different branches of the same phone system. For example, if user John@domain1.com receives a call from Mary@domain2.com, by default the phone only shows the call as being from Mary, so John is not able to pick up or call back Mary from the received call list. With this parameter set to **yes**, the phone logs the call as being from Mary@domain2.com, and John can dial Mary from the received call list.
BroadSoft ACD	This parameter enables support for basic BroadSoft Automatic Call Distribution (ACD). The supported values for this option are **Yes** and **No** (default). If you set **Broadsoft ACD** to **Yes**, the phone sends a Subscribe message according to the BroadSoft specification. If you set **Broadsoft ACD** to **No**, the phone may still send out a Subscribe message because another feature is using ACD, but the phone ignores any Notify message from the BroadSoft server related to ACD. Limitations: • Cisco SPA301 or Cisco SPA501G—ACD is not supported. The ACD Login and Status keys are not visible. • Cisco SPA509—Lines 9–12 cannot be used as ACD Agents since the Lines cannot be selected for Login/Logout and Agent status. You can also configure BroadSoft ACD support by adding the following line to your configuration file to configure this feature on line 1: **<Broadsoft_ACD_1_ ua="na">Yes</Broadsoft_ACD_1_>**

Parameter	Description
Auto Ans Page on Active Call	In conjunction with the global Auto Answer Page parameter (found in the Phone tab), this parameter determines the behavior of the phone when a page call arrives. When set to **no**, page calls are not auto answered (even if the phone is idle), regardless of the value of the global Auto Answer Page setting in the Phone tab. When the global Auto Answer Page parameter is set to **yes**, whether page calls are auto answered or not during an active call depends on the *per line* Auto Answer Page setting (this parameter). Both Auto Answer Page and Auto Answer Page on Active Call are enabled by default.
Feature Key Sync	Allows the phone to synchronize with the call server. If **yes** is selected, if Do Not Disturb or Call Forwarding settings are changed on the phone, changes are also made on the server; if changes are made on the server, they are propagated to the phone. This feature is disabled by default.
Call Park Monitor Enable	When this parameter is set to **yes**, the phone monitors the call park status against that extension. When the extension is private, the phone sends out the call park SUBSCRIBE message to the server to request call park notification. When the extension is shared, the SCA SUBSCRIBE includes an ACCEPT header with the *x-broadworks-callpark-info* tag. By default, a BLF button always sends out a SUBSCRIBE message, and the server notifies the BLF button if there is a call parked against the monitor extension. The phone presents this information to the user without any client-side configuration. When this parameter is set to **no**, the primary extension or shared extension will not monitor the call park status.
Enable Broadsoft Hoteling	Default is no. When this parameter is set to yes, the phone sends out subscription message (without body) to the server; the phone requests “GuestIn” on the registered extension line with input “User ID” and “Password.” The phone then sends out subscription message (with body) to the server with the 'guestAddress' attribute containing the "User ID" and "Password" for authentication. When this parameter is set to no, the phone sends out unsubscribe message (with empty body) to the server to request Host-Guest Association termination.
Hoteling Subscription Expires	An expiration value is put in the subscription message. Default value is 3600. **Note** If this parameter is set to 0, the default 3600 is used instead.

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.
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop.
Alternate Proxy Alternate Outbound Proxy	These fields are used with the Verizon dual registration/fast fallback feature. This feature provides fast fallback 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 fallback 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**.

Parameter	Description

Use Outbound Proxy	Enables an outbound proxy (for example, 172.20.2.1:5060—port is optional) or a domain name such as sip.server.com as long as this name is a fully-qualified domain name. If set to no, the Outbound Proxy and Use OB Proxy in Dialog fields are ignored.
	Defaults to no.
	Optionally, the proxy can be configured (Cisco SPA500 Series only) for Survivable Remote Site Telephony (SRST) support. The proxy is configured with an extension that includes a statically-configured DNS SRV record or DNS A record. Configuring the proxy allows for failover and fallback functionality with a secondary proxy server. For example:
	For SRV Record:
	`sip.server.com:SRV=node1.sip.server.com:5060:p=1:w=50|`
	`node2.sip.server.com:5060:p=2:w=50`
	Set Use DNS SRV to **no** and DNS SRV Auto Prefix to **no**.
	For A Record:
	`sip.server.com:A=172.20.2.1,172.20.2.2`
	Set Use DNS SRV to **no** and DNS SRV Auto Prefix to **no**.

Use OB Proxy In Dialog	Whether to force SIP requests to be sent to the outbound proxy within a dialog. Ignored if <Use Outbound Proxy> is no or <Outbound Proxy> is empty. Defaults to yes.
Register	Enable periodic registration with the <Proxy>. This parameter is ignored if <Proxy> is not specified. Defaults to yes.
Make Call Without Reg	Allow making outbound calls without successful (dynamic) registration by the unit. If no, the dial tone will not play unless registration is successful. Defaults to no.
Register Expires	Allow answering inbound calls without successful (dynamic) registration by the unit. If proxy responded to REGISTER with a smaller Expires value, the phone will renew registration based on this smaller value instead of the configured value. If registration failed with an Expires too brief error response, the phone will retry with the value given in the Min-Expires header in the error response. Defaults to 60.
Ans Call Without Reg	If enabled, the user does not have to be registered with the proxy to answer calls. Defaults to no.
Use DNS SRV	Whether to use DNS SRV lookup for Proxy and Outbound Proxy. Defaults to no.

Parameter	Description
DNS SRV Auto Prefix	If enabled, the phone will automatically prepend the Proxy or Outbound Proxy name with _sip._udp when performing a DNS SRV lookup on that name. Defaults to no.
Proxy Fallback Intvl	This parameter sets the delay (sec) after which the phone will retry from the highest priority proxy (or outbound proxy) servers after it has failed over to a lower priority server. This parameter is useful only if the primary and backup proxy server list is provided to the phone via DNS SRV record lookup on the server name. (Using multiple DNS A record per server name does not allow the notion of priority and so all hosts will be considered at the same priority and the phone will not attempt to fall back after a fail over). Defaults to 3600
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, the phone uses normal, then inspects the port number based on the first listed proxy port. Defaults to Normal.
Dual Registration	Set to yes to enable the Verizon dual registration/fast fallback feature. You must also configure the alternate proxy/alternate outbound proxy fields in the Proxy and Registration section.
Auto Register when Failover	This parameter controls the failover behavior when there is an error. Defaults to no. There are 2 Failover options: • Unregister with current proxy, register to backup proxy and then re-send to backup proxy. This option can be used under the following conditions: • The parameter Auto Register when Failover is set to yes. • The failed message should be an INVITE message. • The return code should match the parameter Try Backup RSC, otherwise the message times out. **Note** When the parameter Auto Register when Failover is set to no, if the return code matches with Try Backup RSC or the message times out, the phone re-sends this message directly to backup proxy. • Re-send directly to backup proxy.

Subscriber Information

Parameter	Description
Display Name	Display name for caller ID.
User ID	Extension number for this line.
Password	Password for this line. Defaults to blank.
Use Auth ID	To use the authentication ID and password for SIP authentication, select yes. Otherwise, select no to use the user ID and password. Defaults to no.
Auth ID	Authentication ID for SIP authentication. Defaults to blank.
Reversed Auth Realm	The Reversed Authentication Realm field. The default value is empty; the proxy address is used as the authentication realm. To use a different authentication realm, enter the IP address to use in the Reversed Authentication Realm field.
Mini Certificate	Base64 encoded of Mini-Certificate concatenated with the 1024-bit public key of the certificate authority (CA) signing the mini-certificate of all subscribers in the group. Defaults to blank.
SRTP Private Key	Base64 encoded of the 512-bit private key per subscriber for establishment of a secure call. Defaults to blank.
Resident Online Number	For Skype use. Enter the Skype Number associated with the registration user ID.
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 is a SIP or SIPS URI that contains no "@" character, then the actual URI used in the SIP signaling should be automatically formed by appending this parameter with an "@" character followed by the IP address of the device.

Audio Configuration

A codec resource is considered as allocated if it has been included in the SDP codec list of an active call, even though it eventually might not be the one chosen for the connection. So, if the G.729a codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the call actually uses G.729a. If the G.729a resource is already allocated and since only one G.729a resource is allowed per device, no other low-bit-rate codec might be allocated for subsequent calls; the only choices are G711a and G711u. On the other hand, two G.723.1/G.726 resources are available per device.

Therefore, it is important to disable the use of G.729a in order to guarantee the support of two simultaneous uses of the G.723/G.726 codecs.

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, G722, G726-16, G726-24, G726-32, G726-40, G729a, or G723. Cisco SPA525G or Cisco SPA525G2: G711u, G711a, G726-32, G729a, and G722. G.723 (not available on Cisco SPA300 Series or Cisco SPA500 Series). G722 not available on Cisco WIP310. Defaults to G711u.
Use Pref Codec Only	To use only the preferred codec for all calls, select yes. (The call fails if the far end does not support this codec.) Otherwise, select no. Defaults to no.
Second Preferred Codec	The second preferred codec when the preferred codec cannot be used. If *Use Pref Codec Only* is enabled (set to **yes**), this parameter is not used. Defaults to Unspecified.
Third Preferred Codec	The third preferred codec when the preferred codec and second preferred codec cannot be used. If *Use Pref Codec Only* is enabled (set to **yes**), this parameter is not used. Defaults to Unspecified.
G711u Enable	Enables use of the G711u codec. Defaults to yes.
G711a Enable	Enables use of the G711a codec. Defaults to yes.
G729a Enable	To enable the use of the G.729a codec at 8 kbps, select yes. Otherwise, select no. Defaults to yes.
G722 Enable	Enables use of the G.722 codec. Defaults to yes. **Note** Not applicable to the Cisco WIP310.
G723 Enable	To enable the use of the G.723a codec at 6.3 kbps, select **yes**. Otherwise, select **no**. Defaults to yes. **Note** G.723 is not supported on the Cisco SPA300 Series, Cisco SPA500 Series, or Cisco WIP310.

Parameter	Description
G726-16 Enable	To enable the use of the G.726 codec at 16 kbps, select **yes**. Otherwise, select **no**. Defaults to yes. **Note** Not supported on the Cisco SPA525G or Cisco SPA525G2.
G726-24 Enable	To enable the use of the G.726 codec at 24 kbps, select **yes**. Otherwise, select **no**. Defaults to yes. **Note** Not supported on the Cisco SPA525G or Cisco SPA525G2 or Cisco WIP310.
L16 Enable	To enable the use of the L16 codec, select **yes**. Otherwise, select **no**. Defaults to yes. **Note** Cisco SPA525G or Cisco SPA525G2 only.
G726-32 Enable	To enable the use of the G.726 codec at 32 kbps, select **yes**. Otherwise, select **no**. Defaults to yes.
G726-40 Enable	To enable the use of the G.726 codec at 40 kbps, select **yes**. Otherwise, select **no**. Defaults to yes. **Note** Not applicable to the Cisco SPA525G or Cisco SPA525G2.
Release Unused Codec	Allows the release of codecs not used after codec negotiation on the first call so that other codecs can be used for the second line. To use this feature, select yes. Defaults to yes.
DTMF Process AVT	Select **yes** to process RTP DTMF events. Otherwise, select **no**. If this parameter is set to no, the AVT payload type is not included in outbound SDP. Defaults to yes.
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select **yes**. Otherwise, select no. Defaults to **no**.
DTMF Tx Method	Select the method to transmit DTMF signals to the far end: InBand, AVT, INFO, Auto, InBand+INFO, or AVT+INFO. InBand sends DTMF using the audio path. AVT sends DTMF as AVT events. INFO uses the SIP INFO method. Auto uses InBand or AVT based on the outcome of codec negotiation. Defaults to Auto.

Parameter	Description
DTMF Tx Volume for AVT Packet	Allows you to manually configure the AVT Tx volume. The value of this parameter is inserted into the volume field of the payload in the AVT packet. Values are based on the AVT specification as described in RFC 2833, *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*. According to RFC 2833, the volume field is represented by 6 bits, and describes the power level of the tone, expressed in dBm0 after dropping the sign. Valid range for this parameter is 0 to 63. If the provisioned value is negative, it will be negated first. Thereafter, if the value is beyond the high limit of 63, it will be clipped to 63. The default value is 0, and is the recommended setting. However, some gateways do not accept this volume setting. If the gateway does not accept the value of 0, the DTMF tone is not relayed to the remote end. As a workaround for the phone to interoperate with those gateways, you can change the value to a value greater than 0.
Use Remote Pref Codec	If set to **yes**, the phone communicates using the remote phone preferred codec. If set to **no**, the Cisco IP phone communicates using its own preferred codec (as indicated in the **Preferred Codec** field and in the SDP by order of preferences). The default vale is no.
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.

A codec resource is considered allocated if it has been included in the SDP codec list of an active call, even though it eventually might not be chosen for the connection. If the G.729a codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the call actually uses G.729a. If the G729a resource is already allocated (and since only one G.729a resource is allowed per phone), no other low-bit-rate codec can be allocated for subsequent calls. The only choices are G711a and G711u.

Since two G.723.1/G.726 resources are available per IP phone, you should disable the use of G.729a to guarantee support for two simultaneous G.723/G.726 codecs.

Dial Plan Script

The default dial plan script for each line is as follows:

```
(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxx|xxxxxxxxxxxx.).
```

Parameter	Description
Dial Plan	Dial plan script for this line. The default is `(<9:>xx.)` `(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)` The dial plan syntax is expanded in the Cisco SPA IP phones to allow the designation of three parameters to be used with a specific gateway: • uid— authentication user-id • pwd—authentication password • nat— if this parameter is present, use NAT mapping Each parameter is separated by 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. (Not applicable to Cisco WIP310.)
Enable IP Dialing	Enable or disable IP dialing. Defaults to no.

User Tab

This section describes the fields for the User tab.

Call Forward

Parameter	Description
Cfwd Setting	The overall control to turn on all forward functions. Defaults to yes.
Cfwd All Dest	Enter the extensions to forward calls to.
Cfwd Busy Dest	Enter the extensions to forward calls to when the line is busy. Defaults to voice mail.
Cfwd No Ans Dest	Enter the extension to forward calls to when the call is not answered. Defaults to voice mail.
Cfwd No Ans Delay	Enter the time delay in seconds to wait before forwarding a call that is not answered. Defaults to 20 seconds.

See Vertical Service Activation Codes for more information on call forwarding parameters.

Speed Dial

You can configure speed dials on the Cisco SPA300 Series and Cisco SPA500 Series IP phones. Speed dial configuration is on a separate tab on the Cisco SPA525G or Cisco SPA525G2. It is not configurable from the phone web user interface on the Cisco WIP310. Speed dial configuration for the Cisco WIP310 is done on the IP phone screen.

Speed Dial 2 through 9: Target phone number (or URL) assigned to speed dial 2, 3, 4, 5, 6, 7, 8, or 9.

Defaults to blank.

See the respective phone user guides for the phone for more information.

Supplementary Services

The IP phone provides native support of a large set of enhanced or supplementary services. All of these services are optional. A supplementary service should be disabled if the user has not subscribed to it or the service provider intends to support similar service by using other means.

A value of **yes** indicates the service is enabled; **no** indicates it is disabled.

Parameter	Description
CW Setting	Enables or disables call waiting.
Block CID Setting	Enables or disables the Block Caller ID feature (caller ID is not transmitted).
Block ANC Setting	Enables or disables the blocking of anonymous calls.
DND Setting	Enables or disables Do Not Disturb.
Secure Call Setting	Enables or disables Secure Call.
Dial Assistance	Enables or disables Dial Assistance.
Auto Answer Page	Enables or disables automatic answering of paged calls.
Speakerphone DTMF Masking	Masks the DTMF playback when using the speakerphone.

Parameter	Description
Preferred Audio Device	Designates the headset or the speaker as the preferred audio device. When the user answers a call by pressing the line button, the audio is directed to the chosen destination. Options for SPA30X, SPA50X, and SPA51X phones: • Speaker–Default value. The audio will direct to the speaker. • Headset–The audio will direct to the headset. Options for SPA525G or SPA525G2 phones: • Speaker–The audio will direct to the speaker. • Headset–The audio will direct to the headset. If Bluetooth is connected, direct to Bluetooth. • Bluetooth–The audio will direct to Bluetooth. If the Bluetooth headset is not connected, direct to headset. • None–Default value. The audio will direct to the last used audio device. If no last used audio device, the speaker is used. If offhook, direct to the handset
Send Audio to Speaker	If enabled, sends the audio to the speaker when the handset is lifted. The handset operates normally, but the speaker is muted.
Time Format	Choose the time format for the phone (12 or 24 hour).
Date Format	Choose the date format for the phone (month/day or day/month).
Miss Call Shortcut	Enables or disables a shortcut that appears as a softkey on the phone screen when a call is missed. The “miss” softkey, when pressed, returns the missed call.
Miss Call Banner	Enables or disables the notification on the phone screen that tells the user that a call has been missed.
Accept Media Loopback Request	Chooses the following behavior of the phone when responding to calls from an IP media loopback service (normally for testing purposes). The options are: • Never—Do not accept media loopback calls. • Automatic—Automatically accept media loopback calls. • Manual—Manually accept media loopback calls.
Media Loopback Mode	For media loopback calls, choose the loopback mode. One end is source, the other end is mirror.
Media Loopback Type	For media loopback calls, choose the loopback type (media or packet).
Text Message	Enables or disables receipt and display of text messages on Cisco SPA303 and Cisco SPA5XX phones by using SIP (RFC-3428). When this feature is enabled, the IP phone screen displays messages up to 255 characters in length. The message appears on the IP phone screen along with the date and time.
Text Message From 3rd Party	To enable receipt of text messages from a third party directly without proxy involvement, choose **yes** to enable.

Parameter	Description
Alert Tone Off	Turn on or turn off the alert tone upon incoming paging or instant messages.
Log Missed Calls for Ext<number>	Enables or disables missed call logging per extension. For example, if you have set up a line to monitor another user line, you can disable missed call logging for the monitored line.
Shared Line DND Cfwd Enable	For shared lines, allows the phone to synchronize with the call server. If **yes** is selected, if Do Not Disturb or Call Forwarding settings are changed on the phone, changes are also made on the server; if changes are made on the server, they are propagated to the phone. This feature is disabled by default.
DND Cfwd Active Warning	(Applicable to SPA508 and SPA509 only). If "Yes", DND and CFWD message is shown as in other phone models.

Camera Settings (Cisco SPA525G or Cisco SPA525G2)

The Cisco SPA525G or Cisco SPA525G2 works with the Cisco WVC2300 Wireless-G Business Internet Video Camera and the Cisco PVC2300 Business Internet Video Camera to provide simple video monitoring from your IP phone. See Entering Camera Information Into the Cisco SPA525G or Cisco SPA525G2 Configuration Utility.

Web Information Service Settings (Cisco SPA525G or Cisco SPA525G2)

These parameters apply only to the Cisco SPA525G or Cisco SPA525G2. For configuration information, see Configuring RSS Newsfeeds (Cisco SPA525G or Cisco SPA525G2).

Audio (SPA5XX)/Audio Volume (SPA525G/525G2)

Note Does not apply to the Cisco WIP310.

Parameter	Description
Ringer Volume	Sets the default volume for the ringer.
Speaker Volume	Sets the default volume for the full-duplex speakerphone.
Handset Volume	Sets the default volume for the handset.
Headset Volume	Sets the default volume for the headset.

Handset Version	Change the handset version manually. **Auto—**Phone automatically sets the handset version based on the hardware version and model. (Default) **Original**—Handset set to Version 2 and below. **V3**—Handset set to Version 3.
Deep Bass	Configures the audio bass settings for the phone handset. For SPA50X and SPA51X phones, the settings are: • Default—Default equalization (EQ) taking advantage of the wideband handset. • HiDef—Alternative EQ that accentuates the high frequencies. • Standard—Emulates the sound of a traditional, narrowband handset. For the SPA525G/525G2 phone, the settings are: • Yes—Default equalization (EQ) taking advantage of the wideband handset. • No—Alternative EQ that accentuates the high frequencies.
Bluetooth Volume	Volume of the Bluetooth device. **Note** Applies to the Cisco SPA525G or Cisco SPA525G2 only.
Electronic HookSwitch Control (EHC) - (SPA525G/SPA525G2) EHS Type (SPA51X)	Some headsets enable you to answer and end phone calls by using controls located on the headset. Set this parameter to **yes** if the user is using a headset with a control function. This feature has been tested with Plantronics base/headsets (that require adapter APC45): • Savi W7xx series (Savi W710, W720, W730, W740, and W745) • CS5xx series (CS 540, CS 510, and CS520) The default setting is no.
Speakerphone Enable	Enables or disables the speakerphone. If the parameter is set to **yes** (the default setting), the speakerphone is enabled. If the parameter is set to **no**, the speakerphone is disabled, and pressing the Speakerphone button on the phone sends the audio to the phone handset instead of the speaker.
Mute Enable	Allows you to enable or disable the Mute button. The default value is Yes. If the parameter is set to **No**, the user cannot mute the audio. **Note** This field is supported in Firmware Release 7.6.2 and later.

Screen (Cisco SPA525G or Cisco SPA525G2)

Parameter	Description
Screen Saver Enable	Enables a screen saver on the IP phone screen. When the phone is idle for a specified time, it enters screen saver mode. (Users can set up screen savers directly using phone Setup button.) Any button press or on/off hook event triggers the phone to return to its normal mode. (The screen shows "Press any key to unlock your phone.") If a user password is set, the user must enter it to exit screen saver mode.
Screen Saver Type	Choose the type of screen saver: • **Black Background**—Displays a black screen. • **Gray Background**—Displays a gray screen. • **Black/Gray Rotation**—The screen incrementally cycles from black to gray. • **Picture Rotation**—The screen rotates through available pictures on the phone. • **Digital Frame**—Shows the background picture.
Screen Saver Trigger Time	Number of seconds that the phone remains idle before the screen saver turns on.
Screen Saver Refresh Time	Number of seconds before the screen saver should refresh (if, for example, you chose a rotation of pictures).

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" will display:
	`Super`
	`Telecom`
	For more information, see the "Configuring Phone Information and Display Settings" section on page 3-2.

BMP Picture Download URL	URL locating the bitmap (.BMP) or .jpg file to display on the IP phone screen background. For more information, see the "Configuring Phone Information and Display Settings" section on page 3-2.
Logo Type	Select from Default, Download BMP Picture, or Text Logo. Defaults to Default. For more information, see the "Configuring Phone Information and Display Settings" section on page 3-2.

Parameter	Description
Background Picture Type	Select from Default, Download BMP Picture, or None. Defaults to Default. For more information, see the "Configuring Phone Information and Display Settings" section on page 3-2.
LCD Contrast	Enter a number value from 1 to 30. The higher the number, the greater the contrast on the IP phone screen.
Back Light Enable	Select yes to enable the IP phone screen back light.
Back Light Timer (sec)	Enter the number of seconds before the back light should turn off.

Attendant Console Tab (Cisco SPA500 and Cisco SPA500DS)

General

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. Defaults to 1800.
Subscribe Retry Interval	Specifies the length of time to wait to try again if subscription fails.
Unit 1 Enable	Enables or disables the first Cisco Attendant Console unit (each IP phone can have up to two Cisco Attendant Consoles attached).
Subscribe Delay	Length of delay before attempting to subscribe. Defaults to 1.
Unit 2 Enable	Enables or disables the second Cisco Attendant Console unit (each IP phone can have up to two Cisco Attendant Consoles attached).
Server Type	Selects the type of server used (Cisco SPA9000, BroadSoft, or Asterisk).
Test Mode Enable	Enables or disables test mode. When test mode is enabled, the LEDs are turned on when keys are pressed, going from off to green to red, and back to off. In test mode, when all the buttons on the Cisco Attendant Console are returned to off, all the keys become orange. The IP phone must be rebooted after the test is completed.
Attendant Console Call Pickup Code	The star code used for picking up a ringing call. Defaults to *98.
Attendant Console Call Park Code Attendant Console Call unPark Code	Not supported on Cisco SPA501, SPA502, and SPA301 phones. You can configure a dedicated call park button to park a call, unpark a call and monitor the parking lot status. For more information on configuring call park buttons, see the "Unused line keys can be enabled to allow call park (for the MetaSwitch soft switch) on the Cisco SPA300 and SPA500 series phones. Users can press this line button to park a call or retrieve a parked call." section on page 2-10. The Call Park code field contains the star code used for parking a call. Defaults to *68. The unPark code field contains the star code used for retrieving a parked call. Defaults to *88. You can change these values to the values used by the call server (for example, the BroadSoft server).
BLF List URI	Automatically configures Busy Lamp Field (BLF) subscriptions for all users on a monitored list. **Note** This feature generates the BLF configuration to the attendant console keys, not the regular line keys.

Parameter	Description
Use Line Keys For BLF List	Set the parameter to **yes**, to use the the line keys first for Broadsoft BLF list feature. Default is **no.** **Note** Previously, the phone used only side-car keys for Broadsoft BLF list feature
BLF Label Display Mode	This parameter displays both name and extension for BLF keys. **Note** Can be configured via phone UI also. The values are "Name/Ext/Both." Default is "Name". If set to "Both:" **Line keys**—Display alternates between name and extension in 5 seconds intervals. **Sidecar keys**—Both name and extension will be shown. The phone tries to render a single line using the selected font. If there is less space, it tries successively with smaller size fonts. If it fails even with 8pt, it renders name and extension in two lines.
Call Pickup Audio Notification	By default, this parameter is set to **no**. If you set it to **yes**, the phone plays the Call Pickup tone when there are incoming calls to any of the lines that the user is monitoring with the Call Pickup function.
Attendant Console LCD Contrast (SPA500DS)	The contrast between the text, lines, and background on the attendant console display. Enter a number value from 1 to 30. The higher the number, the greater the contrast on the display.
Attendant Console Font Size (SPA500DS)	Font size of the text on the attendant console display. Choose 10 or 12 point font.
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.
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.
Unit 1/2 Key 1-32 (SPA500S) Unit 1/2 Key 1-30 (SPA500DS)	Enter a strings that define the extension and other parameters associated with each lighted button on the first Cisco Attendant Console. Keywords and values are case-sensitive.

Attendant Key LED Patterns

The configuration for the attendant key LED patterns is the same as the line key LED script described in Chapter 9, “Configuring LED Patterns.”

Parameter	Description
Application LED	Extended applications (for example, speed dial, xml, mp3, weather, news) are ready for use.
Serv Subscribe Failed LED	BLF service subscription failed.
Serv Subscribing LED	BLF service subscription is on proceeding.
SNRM Day Mode LED	(Used with Cisco UC320W.) The System Night Ringing Mode (SNRM) is turned off. The system is in day mode.
SNRM Night Mode LED	(Used with Cisco UC320W.) The SNRM is turned on. The system is in night mode.
Parking Lot Idle LED	These parameters are used to define the LED pattern when the extension is in the parked state. By default, when there is a call parked on the monitored line (parking lot busy), the phone line LED is in a very slow red blinking state (150ms on, 150ms off). When there is no call parked on the monitored line (parking lot idle), the phone line LED will be in off (or line idle) state. If the button is the primary line, shared line or BLF line, the button LED will reflect the current button status and not reflect the Parking Lot Idle status if the parking lot is idle. If the button is a call park button, the button will reflect the Parking Lot Idle or Parking Lot Busy statuses. To define a pattern other than the default (slow red blinking), enter a pattern in these fields. See Chapter 9, “Configuring LED Patterns.”
Parking Lot Busy LED	
BLF Idle LED	Busy lamp field service: the monitored phone or line is in the idle state.
BLF Ringing LED	Busy lamp field service: the monitored phone or line is in the ringing state.
BLF Busy LED	Busy lamp field service: the monitored phone or line is in the busy state.
BLF Held LED	Busy lamp field service: the monitored phone or line is in the held state.

Attendant Console Status

This page provides two tabs to display the status of up to two Cisco Attendant Consoles that are supported by a single IP phone:

- Unit 1—Displays information about the first Cisco Attendant Console.
- Unit 2—Displays information about the second Cisco Attendant Console.

Each tab provides the read-only fields described in the following table:

Parameter	Description
Unit Enable	Indicates that the Cisco Attendant Console is enabled or disabled.
Subscribe Expires	When the current subscription expires. After the subscription expires, the Cisco Attendant Console automatically requests a new subscription.
HW Version	Version of the hardware.
Unit Online	Indicates that the Cisco Attendant Console is powered on and connected.
Subscribe Retry Interval	Length of time the Cisco Attendant Console waits to try again if subscription fails.
SW Version	Version of the software.
Key	Key number on the attendant console.
Name	Name assigned to each key.
Type	Function enabled for each key.
Line	Extension assigned to each key.
Station	Displays the subscribe URI configured for each key.
Subscribed	Subscription status of the unit/key. The value can be Yes, Fail, or N/A. N/A indicates that the feature/function (fnc) of that line does not require a subscription (such as a speed dial).

TR-069 Tab

This tab is available on the Cisco SPA51X and Cisco SPA525G/525G2 phones. For more information on TR-096, see the "Using TR-069" section on page 6-12.

Field	Description
Enable TR-069	From the drop-down menu, select **yes** to enable TR-069, or **no** to disable TR-069.
ACS URL	Enter the URL of the ACS using 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 using SSL or TLS.
ACS Username	Enter the username that authenticates the CPE to the ACS by using the CPE WAN Management Protocol. This username is used only for HTTP-based authentication of the CPE.
ACS Password	Enter the password grants access to the ACS for this user. This password is used only for HTTP-based authentication of the CPE.
ACS URL In Use	Displays the ACS URL.
Connection Request URL	Displays the ACS making the connection request to the CPE.

Field	Description
Connection Request Username	Enter the username that authenticates the ACS making the connection request to the CPE.
Connection Request Password	Enter the password used to authenticate the ACS making a connection request to the CPE.
Periodic Inform Interval	The duration in seconds of the interval between CPE attempts to connect to the ACS when Periodic Inform Enable is set to **yes**.
Periodic Inform Enable	From the drop-down menu, select **yes** to enable CPE connection requests. Enter **no** to disable connection requests.
TR-069 Traceability	From the drop-down menu, select **yes** to enable TR-069 transaction traceability. Enter **no** to disable traceability.
CWMP V1.2 Support	From the drop-down menu, select **yes** to enable CPE WAN Management Protocol (CWMP) support. Enter **no** to disable CWMP support such that the device does not send any Inform messages to the ACS or accept any connection requests from the ACS.
TR-069 VoiceObject Init	From the drop-down menu, select **yes** to initialize all voice objects to factory default values. Enter **no** to retain the current values.
TR-069 DHCP Option Init	From the drop-down menu, select **yes** to initialize the DHCP settings from the ACS. Enter **no** to leave the settings unchanged.
TR-069 IGD Support	From the drop-down menu, select **yes** to enable TR-069 on the Internet Gateway Device (IGD). Enter **no** to disable traceability. (This is used for debugging purposes.)
TR-069 Fallback Support	From the drop-down menu, select **yes** to enable TR-069 fallback support. Enter **no** to disable fallback support. If the SPA phone first attempt to discover the ACS by using DHCP, it attempts to use DNS to resolve ACS IP address.
TR-069 DHCP Inform Timer	Enter the interval in seconds that the phone should poll the DHCP server.
BACKUP ACS URL	Enter the backup URL of the ACS using 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 using SSL or TLS.
BACKUP ACS User	Enter the backup username that authenticates the CPE to the ACS by using the CPE WAN Management Protocol. This username is used only for HTTP-based authentication of the CPE.
BACKUP ACS Password	Enter the backup password grants access to the ACS for the backup user. This password is used only for HTTP-based authentication of the CPE.

Cisco SPA525G or Cisco SPA525G2-Specific Tabs

The tabs described in this section appear on the Cisco SPA525G or Cisco SPA525G2.

Wi-Fi

Enable or disable the Wireless-G service on the phone from this tab.

Parameter	Description
Wireless Enable	Click **on** to enable the wireless controller.
Wi-Fi Device	Choose the method of wireless setup: • Wi-Fi Profile—Create a wireless profile by manually entering the information. • Wi-Fi Protected Setup—If your router has a WPS button, you can use Wi-Fi Protected Setup to add a new wireless network profile.
Wireless Status	Information about the wireless network.
Wi-Fi Profile	Contains up to 3 wireless profiles for the phone. Includes a wireless profile for the Cisco Unified Communications Server.

Bluetooth

For more information on configuring Bluetooth, see Configuring Bluetooth (Cisco SPA525G or Cisco SPA525G2 only).

Parameter	Description
Bluetooth Device	Click **on** to enable Bluetooth.
Bluetooth Status (Cisco SPA525G2 only)	Name and status of any connected Bluetooth devices.
Bluetooth Mode (Cisco SPA525G2 only)	Shows the method of Bluetooth connection chosen: • **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 (see Configuring Bluetooth (Cisco SPA525G or Cisco SPA525G2 only). The Cisco SPA525G2 connects to only one Bluetooth device at a time.

Parameter	Description
Bluetooth Profiles (Cisco SPA525G2 only)	This table shows the MAC (hardware) address, device name, and other information for the Bluetooth device that is associated with a Cisco SPA525G or Cisco SPA525G2. If multiple Bluetooth devices are in range of a Cisco SPA525G or Cisco SPA525G2, the phone attempts to pair with the devices in order the shown in the list. Highlight an entry and click the arrow keys to move devices up and down the list, changing the priority. You can choose **yes** or **no** to indicate if the phone should connect to a Bluetooth device automatically. You can also remove devices from the list.
Bluetooth Device List (Cisco SPA525G2 only)	Click **Scan for Bluetooth Devices** to locate Bluetooth devices in the area. Found devices are shown with the type of device, MAC address, and device name.

Personal Address Book

Address book for the phone. For more information, see the respective Cisco Small Business IP Phone User Guide.

Call History

Displays the call history for the phone. To change the information displayed, select the type of call history from the drop-down list:

- All Calls
- Received Calls
- Placed Calls
- Missed Calls

Speed Dials

See Speed Dial.

Firmware Upgrade

Used to upgrade the firmware for the Cisco SPA525G or Cisco SPA525G2. See Updating Firmware.