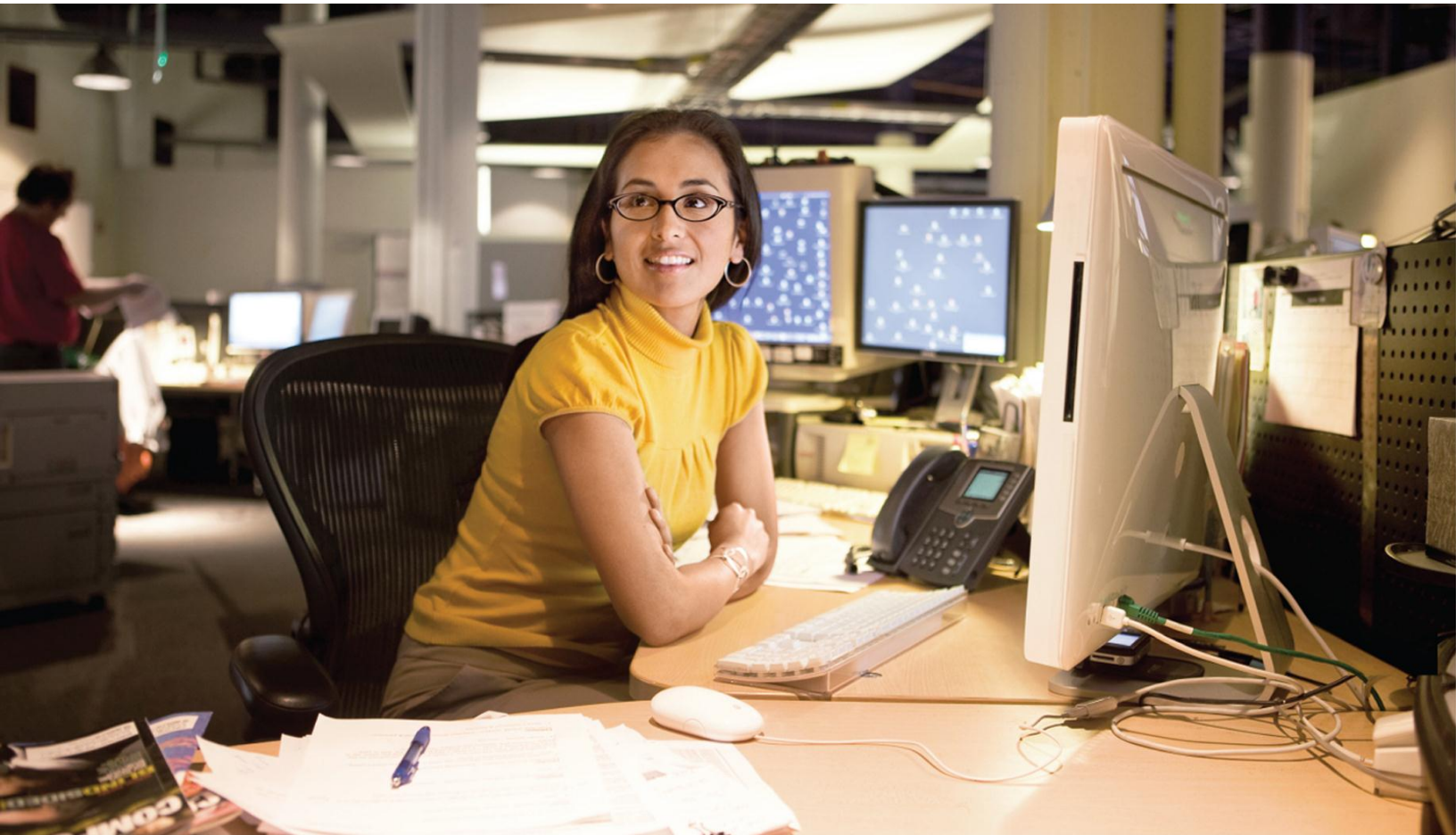
Cisco Smart Business Communications System

Feature Reference Guide

June 2012



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Introduction

Cisco® Configuration Assistant simplifies the task of configuring, deploying, and administering the Cisco Smart Business Communications System (SBCS) – a complete system of advanced voice, data, security, and wireless networking products designed specifically for small and medium-sized businesses. Cisco Configuration Assistant improves network security and performance and substantially reduces deployment and configuration time. This PC-based application features a simple graphical user interface (GUI) and provides everything required to quickly set up a small office network.

Cisco Configuration Assistant provides:

- Configuration, deployment, and ongoing network management support for the Cisco Smart Business Communications System
- Multiple network views
- Simplified network reporting
- Drag-and-drop software updates
- Troubleshooting

This document lists Cisco SBCS system features that can be configured using Cisco Configuration Assistant versions 3.1 and 3.2. Feature configuration information is categorized by network and voice, switching, wireless, and security. A comprehensive, A-to-Z guide to SBCS system features is also included. For device- and platform-specific features, see the data sheets and platform reference guides available on Cisco.com (www.cisco.com/go/sbcs)

Download Cisco Configuration Assistant at no charge at www.cisco.com/go/configassist by selecting the "Download Software" link.

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Section One: Feature Comparisons by Release

Cisco Configuration Assistant Network and Voice Feature Support

Table 1 lists network and voice features that are supported by Cisco Configuration Assistant versions 3.1 and 3.2.

Table 1. Cisco Configuration Assistant Network and Voice Feature Support

Cisco Configuration Assistant Network and Voice Feature Support				
Category	**Feature**	**V3.1**	**v3.2.1**	**Description**
Basic Network Configuration				
	WAN IP address	X	X	
	Dynamic Host Configuration Protocol (DHCP)	X	X	
	Disable DHCP server on UC500	X	X	Cisco Configuration Assistant 3.1 allows users to disable the DHCP server on UC500 series routers.
	Time zone	X	X	
	Data and voice VLANs	X	X	
	Routing	X	X	Only static routing supported
Voice Deployment Scenarios				
	Key System mode	X	X	Supported for Foreign Exchange Office (FXO) and T1/E1 channel associated signaling (CAS)
	Private Branch Exchange (PBX) mode	X	X	Support for direct inward dialing (DID) and direct outward dialing (DOD)
	Remote teleworker	X	X	
	Multisite SBCS deployment	X	X	Up to 5-site multisite deployment
	Phone VPN client	X	X	
	Video streaming to IP phone	X	X	
Cisco Unified Communications (UC) 500 Series Platforms Supported				
	UC520	X	X	
	UC540	X	X	
	UC560	X	X	
Basic Managed Device Support				
	System time	X	X	
	Users and passwords	X	X	
	IP address	X	X	
	Device access	X	X	
	Access device configuration utility from Cisco Configuration Assistant Topology View	X	X	
	Simple Network Management Protocol (SNMP)	X	X	Varies by device
	Save configuration	X	X	
	Backup and restore configuration	X	X	
	Firmware upgrade	X	X	
	Inventory	X	X	

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Cisco Configuration Assistant Network and Voice Feature Support				
Category	**Feature**	**V3.1**	**v3.2.1**	**Description**
	System messages (logging)	X	X	
	Hostname	X	X	
	Reset to factory defaults	X	X	
Dial Plan				
	Inbound call handling	X	X	Call handling for FXO, Basic Rate Interface (BRI)/T1/E1 incoming call handling mechanism
	Outbound call handling	X	X	Ability to specify multiple emergency numbers, customized call blocking capability
	Public Switched Telephone Network (PSTN) number mapping (DID)	X	X	Ability to perform 1-to-1 and 1-to-many DID to internal extension mapping.
	Call forward busy	X	X	
	Call forward no answer	X	X	
	Inbound caller ID support	X	X	Supported on FXO, BRI, Primary Rate Interface (PRI), Session Initiation Protocol (SIP)
	Outbound caller ID support	X	X	Supported on BRI, PRI, and SIP. If DID range is noncontiguous, outbound caller ID is the auto attendant (AA) number.
				Supported for configuring caller ID per PSTN trunk and overriding trunk-level caller ID for individual extensions.
	Caller ID blocking	X	X	
	Call blocking	X	X	Added ability to customize up to 5 call block numbers or ranges
	Customizable dialplan	X	X	
	Trunk failover	X	X	Support for SIP trunk to PSTN or PSTN to SIP trunk failover
	Multiple access codes	X	X	Support for multiple access codes for any string of digits in the dial plan
	Digit collection timeout	X	X	
	Dial plan test diagnostic	X	X	
PSTN Trunks				
	Central office (CO) trunk (FXO)	X	X	Call handling per FXO port can be customized to redirect to AA/operator/hunt group/shared line; Cisco Configuration Assistant 2.2(5) adds support for overlay on CO trunk line. Cisco Configuration Assistant 3.0 adds support for detailed FXO trunk settings.
	Analog DID	X	X	Supported only on the expansion voice interface card (VIC) slot
	BRI	X	X	
	T1 (PRI) including fractional	X	X	
	E1 (PRI) including fractional	X	X	
	T1 (CAS) including fractional	X	X	Ear and mouth (E&M) (Wink Start and Immediate Start signaling types supported). For Foreign Exchange Station (FXS)/FXO, loop start and ground start signaling types supported.
	E1 (CAS) including fractional	X	X	E&M (Wink Start and Immediate Start signaling types supported). For FXS/FXO, loop start and ground start signaling types supported.
	Analog station ports (integrated FXS)	X	X	
	ISDN BRI/PRI parameter customization	X	X	Configurable bearer-cap and static TEI settings
	International call progress tones	X	X	Configurable call progress tone for international locales

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Cisco Configuration Assistant Network and Voice Feature Support				
Category	**Feature**	**V3.1**	**v3.2.1**	**Description**
	SIP trunks	X	X	Support for AT&T, Cbeyond, Nuvox, Paetec, XO, Broadview and British Telecom, FiberNet, and generic SIP provider. Cisco Configuration Assistant 3.0 added support for **CableVision, Qwest, and Portugal Telecom**. Cisco Configuration Assistant 3.1 adds support for CableVision and Megapath. Cisco Configuration Assistant 3.2 adds support for Integra, Keyyo, and OpenIP.
	Generic SIP trunk template		X	Cisco Configuration Assistant 3.2 adds support for generic SIP trunk template including timer and retry management and toll fraud protection for SIP Trunks
	Unknown plan type for International calls for ISDN carrier		X	
Voice System Features				
	Call pickup	X	X	
	Call park	X	X	Cisco Configuration Assistant 3.0 adds support for call park extension timeout and recall settings.
	Call hold	X	X	Cisco Configuration Assistant 3.0 adds support for call hold audible alert settings.
	Automatic redial	X	X	Cisco Configuration Assistant 3.1 allows the user to automatically redial with the outgoing prefix.
	Hunt groups	X	X	Includes sequential, peer, simultaneous (call blast), and longest idle hunt group types
	Paging groups	X	X	Cisco Configuration Assistant 2.2(5) adds support for combined paging groups with one level of nesting
	Intercom (with or without mute)	X	X	Including multiple intercom support and ability to configure intercom on nonprimary buttons
	Dialable intercom (with or without mute, overlay option)	X	X	Dialable intercom on primary button is not supported.
	Whisper intercom	X	X	Cisco Configuration Assistant 2.2(5) adds whisper intercom support on all buttons except the primary button.
	Single Number Reach (SNR)	X	X	Cisco Configuration Assistant 3.0 adds support for SNR timeout settings.
	Conferencing	X	X	Cisco Configuration Assistant 3.0 adds support for **enabling and disabling conference join and leave tones**
	Ad hoc conferencing	X	X	For UC540 system, max number of conferences is 24 (up to 8 concurrent users per conference). For UC560 system, this limit is 56 (up to 8 concurrent users per conference)
	Meet-me conferencing	X	X	For UC540 system, max number of conferences is 6 (up to 32 concurrent users per conference). For UC560 system, this limit is 12 (up to 32 concurrent users per conference). Cisco Configuration Assistant 3.1 allows external callers to initiate a meet-me conference.
	cBarge and privacy	X	X	Cisco Configuration Assistant 2.2(5) adds support for configuring cBarge with or without Privacy button (requires shared octo-line extensions on phones)
	Basic automatic call distribution	X	X	
	Night service	X	X	Support for night service schedule, night service bell, night service call forward, and night service toggle code
	Phone services URL provisioning	X	X	
	Fax support	X	X	
	T.37 fax to email	X	X	Support for enabling voice mailboxes to receive incoming faxes, voice and fax detection, fax printing, and integration with voicemail notifications or IMAP to allow users to receiving email notifications with faxes attached in TIFF format. Administrators can use the default system prompts for voice and fax detection or record a custom prompt.

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Cisco Configuration Assistant Network and Voice Feature Support				
Category	**Feature**	**V3.1**	**v3.2.1**	**Description**
	T.38 fax relay	X	X	Supported for Cisco Certified SIP trunk providers. Generic SIP trunk providers supported in Cisco Configuration Assistant 2.4.
	Arc Express 2.1 compatibility	X	X	Arc Express client must be configured separately.
	FXO hook flash	X	X	
	Advanced FXO port settings	X	X	Configure advanced FXO port settings, including supervisory disconnect, audio, and timer settings.
	Busy lamp field (BLF)	X	X	
	Line monitoring	X	X	
	Trunk monitors	X	X	
Users/Phones/Extensions				
	Multiple extensions per phone	X	X	
	Floating/phantom extensions	X	X	Create extensions that are not associated with any phone. Within Cisco Configuration Assistant, these are called floating extensions. Cisco Configuration Assistant 3.1 adds support for mapping a single digit to a floating extension.
	Extension mobility	X	X	Support normal and shared line only.
	Email and voicemail notifications	X	X	
	Shared line	X	X	
	2 calls per line	X	X	
	Phone display header	X	X	Supports editing the description displayed in the phone header
	Overlay DN	X	X	Supports configuring an overlay on the CO Line button. Cisco Configuration Assistant 3.0 added support for configuring overlay with intercom.
	Octo-lines	X	X	Supports octo-lines on shared extensions.
	Busy line monitoring	X	X	
	Custom button label	X	X	
	PSTN line appearance	X	X	
	Personal speed dials	X	X	
	System speed dials	X	X	
	Soft key templates	X	X	
	Soft key customization	X	X	Cisco Configuration Assistant 3.1 allows custom URL and softkey definition per template, where a template can be applied to any IP phone on the UC500, on a per-phone basis.
	Phone background customization	X	X	
	Phone ringtone customization	X	X	
	Call Ring Features		X	Cisco Configuration Assistant 3.2 adds support for No call-waiting beep, Call-waiting ring, Distinctive Ring, Missed calls, HuntStop Channel, Silent Ring, Feature Ring, Beep with no ring
	Editable IP phone URLs		X	
	Cisco SPA300 Series phones	X	X	
	Cisco SPA500 Series phones	X	X	

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Cisco Configuration Assistant Network and Voice Feature Support				
Category	**Feature**	**V3.1**	**v3.2.1**	**Description**
	Cisco SPA525G/G2 wireless IP phone	X	X	
	Cisco 7900 phones	X	X	Supports Cisco 7902, 05, 06, 10, 11, 12 and 31 Cisco 7920, 21, 25 Wireless phones Cisco 7940,41,41G-GE,42,45 Cisco 7960,61, 61GE, 62,65 Cisoc 7970,71,71GE,75, 85 models only
	Cisco SPA 500s, Cisco 7915 and 7916 sidecar	X	X	
	Cisco 7935, 7936, 7937 conference station	X	X	
	Cisco Unified IP phones 6900 Series	X	X	Supports Cisco Unified IP Phone 6901, 6911, 6921, 6941, and 6961 models only
	Cisco IP Communicator	X	X	
	Limited support for Cisco 8961, 9951, and 9971		X	Limited feature support is available for Cisco Unified IP Phone 8961, 9951, and 9971
	FXS support with SPA8800 and SPA8000		X	Support for additional FXS ports for analog phones using Cisco SPA8800 (no FXO) and SPA8000 gateways
	Limited support for Cisco Jabber™ for iPhone		X	Limited feature support is available for Cisco Jabber for iPhone.
Voicemail				
	Personal mailbox	X	X	
	General delivery mailbox	X	X	
	Floating/phantom mailbox	X	X	Assign a voice mailbox to an extension that is not associated with a physical phone (floating extension).
	Bilingual voice messaging	X	X	Cisco Configuration Assistant 3.1 allows the caller to the mailbox to hear voicemail greetings in one of two languages installed on the Cisco Unity® Express.
	Unified messaging (Internet Message Access Protocol [IMAP])	X	X	Automatically configured by Cisco Configuration Assistant. IMAP client must be configured separately.
	VoiceView Express	X	X	
	Voicemail notification	X	X	Cisco Configuration Assistant 3.0 added support for voicemail notifications sent to phone or email. Voicemail notifications can be used in conjunction with T.37 fax-to-mail to enable users to receive incoming faxes as attachments.
	Greeting	X	X	
	Call recording	X	X	Support for Live Record softkey. Cisco Configuration Assistant 3.0 adds support for Live Record beep tone settings.
	Voice messaging zero-out options	X	X	Cisco Configuration Assistant 3.1 allows customization of how the call flow should proceed in response to keys pressed at the mailbox greeting for leaving a voicemail.
	Direct transfer to voicemail	X	X	
	Play caller ID	X	X	
Auto Attendant (AA)				
	Multiple AAs	X	X	Prior releases of Cisco Configuration Assistant allow configuration of up three AAs on a single Cisco Unified Communications 500 Series. Cisco Configuration Assistant 3.1 allows configuration of up five AAs on a single Cisco Unified Communications 500 Series.
	Multilevel AA	X	X	Cisco Configuration Assistant allows configuration of up to three levels of AA menus.

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Cisco Configuration Assistant Network and Voice Feature Support				
Category	**Feature**	**V3.1**	**v3.2.1**	**Description**
	Bilingual AA		X	Cisco Configuration Assistant 3.1 allows for configuration of the language attribute of each AA so that the AA will play the standard greetings and associated prompts in the selected language.
	No-option transfer	X	X	In prior releases, the AA_SBCS_v03 AA script becomes the default AA script, which supports configuration of an optional drop-through number. In Cisco Configuration Assistant 3.1, the AA_SBCS_v04 AA script allows the user to specify a Maximum Menu Prompt Attempts setting. This parameter dictates the number of times the AA replays its greeting before terminating, or transferring an idling call (a call without any key input) to a no-option-transfer-to number.
	Separate prompts and actions for open/closed hours	X	X	
	Dial by first name	X	X	In Cisco Configuration Assistant 3.1, the AA_SBCS_v04 AA script supports a variation of the dial-by-name feature that allows the caller to enter the first name before the last name.
	Alternate greeting	X	X	In Cisco Configuration Assistant 3.1, the AA_SBCS_v04 AA script allows the user to designate the newly recorded prompt as the alternate greeting.
	Custom script management	X	X	Custom scripts can be uploaded. For custom scripts, only the AA extension and PSTN main number can be configured using Cisco Configuration Assistant.
	Prompt management	X	X	AA prompt management allows assignment of prompt recording and management privileges to users and configuration of prompt management extension.
	Bilingual prompt management	X	X	Cisco Configuration Assistant 3.1 allows the user to configure the language attribute of each AA so that the AA will play the standard greetings and associated prompts in the selected language.
	Holiday/business hours	X	X	Configure up to 4 separate business schedules; up to 26 holidays.
	Night service schedule	X	X	AA closed hour prompts and actions are presented during night service hours.
Cisco Unified Communications 500 Series Administration				
	Backup/restore of configuration/data	X	X	Cisco Configuration Assistant can back up and restore both Cisco Unified Communications 500 Series configuration and user data, such as voicemails and passwords.
	Reset to factory default	X	X	
	Disk cleanup	X	X	
	Cisco IOS® Software upgrade	X	X	Drag and drop or through the Cisco Unified Communications 500 Series software pack upgrade.
	Cisco Unified Communications Manager Express/Cisco Unity Express localization	X	X	Cisco Unified Communications 500 Series locale packs and language files posted on free software download page. Administrators can switch between active and alternate phone languages and voicemail languages.
	Phone load management	X	X	Phone loads should be in the phone load folder in Cisco Configuration Assistant. Drag-and-drop upgrades are supported for Cisco SPA 500 Series and SPA 300 Series IP Phones.
	Flash file management	X	X	Cisco Configuration Assistant 3.0 added support for uploading and downloading files to the Cisco Unified Communications 500 Series flash from the File Management window.
	Music on hold (MOH)	X	X	
	Cisco Unity Express upgrade	X	X	Supported through Cisco Unified Communications 500 Series software pack upgrade.
	Cisco Unified Communications 500 Series license upgrade	X	X	Starting with Cisco Configuration Assistant 3.1, UC540 and UC560 platforms will come equipped with 24 user licenses, and the scale of the UC560 platform will increase to a maximum number of 138 phones.
				Cisco UC540 and UC560 license upgrade through a product authorization key (PAK).

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Cisco Configuration Assistant Network and Voice Feature Support				
Category	**Feature**	**V3.1**	**v3.2.1**	**Description**
	Reboot phone from topology view	X	X	
	Display phone detailed status GUI from topology view	X	X	
	Telephony setup wizard	X	X	Requires factory default Cisco Unified Communications 500 Series
	VPN phone setup wizard	X	X	Step-by-step wizard for enabling and configuring SSL VPN client settings for Cisco SPA525G and SPA525G2 phones (day 0 and day 2).
	Cisco WebEx® Phone Connect	X	X	
	Timecard View	X	X	
	System Dashboard	X	X	
	Network Topology View	X	X	
	Network diagnostics	X	X	Ping, trace, DHCP bindings, and system status.
	Voice diagnostics and debugging	X	X	Collect per phone debugs for SIP, H.323, and Skinny Client Control Protocol (SCCP) troubleshooting, perform T1/E1 loopback circuit diagnostics. Supports PCM capture, SIP trunk registration diagnostics, and Cisco Unity Express connectivity diagnostics.
	Security diagnostics	X	X	Collect VPN and firewall/NAT debug logs.
	Expert mode telephony monitoring and reporting	X	X	View detailed reports for phones and extensions, hunt groups, call blast groups, TFTP server files, dial peers, translation profiles, SIP trunk status, phone templates, voicemail status, DSP status, and software pack status.
	Expert mode monitoring and troubleshooting	X	X	Enter Cisco IOS Software and Cisco Unity Express exec mode commands and view output.
				View show command output for selected Cisco IOS Software and Cisco Unity Express commands.
	CLI postview	X	X	Enable display of Cisco IOS Software commands sent to Cisco Unified Communications 500 Series when telephony configuration is applied.
	Online Help	X	X	
Remote Teleworker Deployment				
	Soft phone (Cisco IP Communicator)	X	X	
	Cisco 871W Integrated Services Router	X	X	EzVPN client on 871W, EzVPN server on Cisco Unified Communications 500 Series
	Cisco SR500 Series Secure Routers	X	X	EzVPN client on 871W, EzVPN server on Cisco Unified Communications 500 Series
	Codec configuration	X	X	
	Firewall/NAT Traversal (MTP)	X	X	Configured automatically for Cisco IP Communicator phones in Cisco Configuration Assistant
	Transcoding	X	X	
	VPN Phone Setup Wizard	X	X	Step-by-step wizard for enabling and configuring SSL VPN client settings for Cisco SPA525G and SPA525G2 phones (day 0 and day 2)
Multisite SBCS Deployment				
	H.323 and SIP voice over IP (VoIP) parameters	X	X	
	Call admission control (CAC)	X	X	
	Dynamic DNS (DDNS)	X	X	
	Quality of service (QoS) traffic shaping	X	X	

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Cisco Configuration Assistant Network and Voice Feature Support				
Category	**Feature**	**V3.1**	**v3.2.1**	**Description**
	Site-to-Site VPN and Site-to-Site dialing		X	

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Cisco Configuration Assistant Switching Feature Support

Table 2 lists the switching features that are supported by Cisco Configuration Assistant 3.1 and Cisco Configuration Assistant 3.2.

Table 2. Cisco Configuration Assistant Switch Feature Support

Cisco Configuration Assistant Switch Feature Support				
Category	**Feature**	**V3.1**	**v3.2**	**Description**
Layer 2 Switching				
	Cisco 300 Series Managed Switches	X	X	Cisco Configuration Assistant 3.1 adds support for all Cisco 300 Series Managed Switches.
	Cisco 200 Series Smart Switches	X	X	Cisco Configuration Assistant 3.1 adds support for Cisco 200 Series Smart Switches (excluding SG200-08 and SG200-08P models).
	Cisco Small Business ESW 500 Series Switches	X	X	Access the ESW 500 Series Switch Configuration Utility from the Cisco Configuration Assistant Topology View to configure additional features.
	Cisco Catalyst® Express 500 Series Switches	X	X	
	Spanning Tree Protocol	X	X	Catalyst Express 500 Series only
				Fast convergence using 802.1w, enabled by default
				Portfast: supported on “Desktop,” “Phone + Desktop,” “Printer,” and “Server” Smartports roles
	MAC addresses	X	X	Catalyst Express 500 Series only Read-only display of MAC address table from switch
	Virtual LANs (VLANs)	X	X	Up to 32 VLANs (1000 range) and support for 802.1Q trunking using Cisco Smartports
	Number and type of queues	X	X	4 queues per port
				Shaped Round Robin (SRR) queuing
				Support for differentiated services code point (DSCP) and class of service (CoS) using Cisco Smartports
	Port grouping	X	X	Catalyst Express 500 Series only
				Support for Cisco Ether Channel and IEEE 802.3ad
				Up to 6 groups
				Up to 8 ports per group
Smartports (Preset Cisco Recommended Network Enhancements, QoS, and Security)				
	Desktop	X	X	Optimized for desktop connectivity
				Configurable VLAN setting
				Port security enabled to limit unauthorized access to the network
	IP phone plus desktop	X	X	Optimized QoS for IP phone plus desktop configurations
				Voice traffic is placed on “Cisco-Voice” VLAN
				Configurable data VLAN
				QoS level assures VoIP traffic takes precedence
				Port security enabled to limit unauthorized access to the network
	Router	X	X	Configured for optimal connection to a router or firewall for WAN connectivity
	Switch	X	X	Configured as an uplink port to a backbone switch for fast convergence
				Enables 802.1Q trunking
	Access point	X	X	Configured for optimal connection to a wireless access point
				Configurable VLAN
	Server	X	X	Catalyst Express 500 Series only

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Cisco Configuration Assistant Switch Feature Support				
Category	**Feature**	**V3.1**	**v3.2**	**Description**
				Can be classified as trusted, critical, business, or standard server:
				• Trusted: For use with Cisco Communications Manager Express; same QoS setting as voice (VoIP traffic is prioritized)
				• Critical: For critical servers with QoS set higher than default
				• Business: Default setting; QoS higher than desktop Internet traffic
				• Standard: For servers set to same level as regular desktop Internet traffic; configurable VLAN port security enabled to limit unauthorized access to the network
	Printer	X	X	Catalyst Express 500 Series only
				QoS settings for “Printer” are the same as “Desktop,” “Access Point,” and “Standard Server.”
				Configurable VLAN
				Port security enabled to limit unauthorized access to the network.
	Guest	X	X	Catalyst Express 500 Series only
				Guests are allowed access to the Internet, but not to the company network.
				All guest ports are placed on the “Cisco-Guest” VLAN.
				Port security enabled to limit unauthorized access to the network.
	Other	X	X	“Other” Smartports role allows for flexible connectivity of nonspecified devices and diagnostic on ESW 500 Series.
				Configurable VLAN
				No security
				No QoS policy
	Diagnostic	X	X	Customers can connect diagnostics devices to monitor traffic on other switches (configurable using Cisco Configuration Assistant only). On ESW500 Series switches, this functionality is provided by setting the Smartports role to Other and using port mirroring.
Switch Security				
	SSL	X	X	SSL support: Encrypts all HTTP traffic, allowing secure access to the browser-based management GUI in the switch Can be configured through Cisco Configuration Assistant; for Catalyst Express 500 Series only
	Security policy slider	X	X	Catalyst Express 500 Series only
				Three security levels: Low, medium, and high (configurable using Cisco Configuration Assistant only):
				• Low: For business environments where there is limited guest access; limited number of devices are allowed per port (broadcast storm control and port security enabled).
				• Medium: For business environments where security is important; only authorized devices (by MAC address) are allowed on the company network.
				• High: For business environments where security is critical; only authorized devices (by MAC address) and authenticated users (using IEEE 802.1x) are allowed on the company network.
	Multicast	X	X	Available through Cisco Configuration Assistant for Catalyst Express 500 Series only
				High-bandwidth video traffic is optimized so that it does not affect other applications on the network
				Internet Group Management Protocol (IGMP) (v1, v2, and v3) snooping: IGMP snooping constrains multicast traffic at Layer 2 by configuring Layer 2 LAN ports dynamically to forward multicast traffic only to those ports that want to receive it.

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Cisco Configuration Assistant Wireless Feature Support

Table 3 lists the wireless features that are supported by Cisco Configuration Assistant 3.1 and Cisco Configuration Assistant 3.2.

Table 3. Cisco Configuration Assistant Wireless Feature Support

Cisco Configuration Assistant Wireless Feature Support				
Category	**Feature**	**v3.1**	**v3.2**	**Description**
Basic Device Configuration				
	Host name	X	X	Cisco 521 Wireless Express Access Point, Cisco AP541N Wireless Access Pont
	System time	X	X	Cisco 521, AP541N
	Daylight saving time	X	X	Cisco 521, AP541N
	Time zone	X	X	Cisco 521, AP541N
	NTP	X	X	Cisco 521, AP541N
	HTTP authentication	X	X	Cisco 521, AP541N
	Enable password	X	X	Cisco 521, AP541N
	Local user name and password	X	X	Cisco 521, AP541N
	Telnet and console passwords	X	X	Cisco 521 only
	WLAN	X	X	For Cisco 521, VLAN is created as part of WLANs. Maximum three WLANs supported; AP541N
	Data, voice, and guest VLANs usability	X	X	Cisco 526 Wireless Express Mobility Controller, AP541N
	Save configuration	X	X	
WLAN Deployment Scenarios				
	Wireless Voice Setup Wizard	X	X	Simple setup wizard for configuring wireless LAN security, QoS for Cisco Unified Communications 500W, 521, or AP541N and SPA525G wireless IP phones
	Data WLAN usability	X	X	Cisco 526 controller only with version 4.2.x.x, AP541N
	Voice WLAN usability	X	X	Cisco 526 controller only with version 4.2.x.x, AP541N
	Guest VLAN usability	X	X	Cisco 526 controller only with version 4.2.x.x, AP541N
	Secure authentication	X	X	Wired Equivalent Privacy (WEP), Extensible Authentication protocol (EAP), Wi-Fi Protected Access (WPA), WPA Preshared Key (WPA-PSK), WPA2, WPA2-PSK, MAC, MAC+EAP, AP541
	Web authentication	X	X	Cisco 526 controller only with version 4.2.x.x
	Fast roaming (Cisco Centralized Key Management)	X	X	Cisco 526 controller only with version 4.2.x.x
	Voice CAC	X	X	Cisco 526 controller only with version 4.2.x.x
	Support for 10 autonomous access points	X	X	Cisco 526 controller, AP541N
WLAN Users				
	Guest user	X	X	Cisco 526 controller only with version 4.2.x.x
	Nonguest user	X	X	Cisco 526 controller only with version 4.2.x.x
	Web login page	X	X	Cisco 526 controller only with version 4.2.x.x; both internal and customized web login page
Convert to LAP				
	Convert authoritative access point (AAP) to lightweight access point (LAP)	X	X	Cisco 521 only

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Cisco Configuration Assistant Wireless Feature Support				
Category	**Feature**	**v3.1**	**v3.2**	**Description**
Reports				
	Inventory	X	X	Cisco 521LAPs are displayed under their respective Cisco 526 controller, AP541N
	Wireless radios	X	X	Cisco 526 controller, AP541N
	Wireless clients	X	X	Cisco 526 controller, AP541N
Views				
	Topology	X	X	
	Front panel view	X	X	Cisco 526 controller only
Monitor				
	Events	X	X	Includes acknowledgments
	System messages	X	X	
Maintenance				
	Software upgrade	X	X	
	Configuration archive	X	X	Includes RMA setup
	Restart (that is, reboot)	X	X	
	Reset to factory default	X	X	
	Upload troubleshooting log	X	X	
Out-of-Box Setup				
	Wireless Setup Wizard	X	X	Cisco 521, AP541N, SPA525G, and SPA525G2 IP phones operating in Wireless-G mode

Cisco Configuration Assistant Security Feature Support

Table 4 lists the security features that are supported by Cisco Configuration Assistant 3.1 and Cisco Configuration Assistant 3.2.

Table 4. Cisco Configuration Assistant Security Feature Support

Cisco Configuration Assistant Security Feature Support				
Category	**Feature**	**v3.1**	**v3.2**	**Description**
Firewall				
	Application firewall	X	X	Provides high, medium, and low security levels for firewall policy settings to enable accelerated and easy deployment:
				• Low: For business environments that do not need to track peer to peer (P2P) and IM applications on the network or check for protocol conformance
				• Medium: For business environments where security is important and there is a need to track the use of IM and P2P applications and check for HTTP and email protocol conformance
				• High: For business environments where security is critical, and there is a need for protocol anomaly detection services to drop nonconformant HTTP and email traffic and prevent use of P2P and IM applications
	Zone-based firewall	X	X	Advanced firewall supported by default on Cisco SR500 Series Secure Router.
	URL filtering	X	X	Supported on Cisco SR500 Series Secure Router only.
	Intrusion prevention system (IPS)	X	X	Supported on Cisco SR500 Series Secure Router only.
	ACL Management		X	

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VPN				
	Cisco Easy VPN Remote	X	X	Scalable, easy-to-manage, secure remote access for teleworkers for Cisco SR500 Series
	Cisco Easy VPN Server	X	X	Offers wizard-based configuration of remote-access VPN server configuration for Cisco Unified Communications 500 Series
	Secure Sockets Layer (SSL) VPN	X	X	
	Editable SSL VPN port		X	
	Ability to accept DHCP-assigned WAN IP address for SSL VPN		X	
	Split tunneling	X	X	
	Disable split tunneling	X	X	Uses Dynamic Virtual Tunnel Interface (DVTI) to allow WAN/Internet access only from VPN hub site
Security Features				
	Security Setup Wizard	X	X	Cisco SA500 software version 1.1.42 and earlier are not supported by Cisco Configuration Assistant. All other SA500 features are configured through the SA500 Configuration Utility, which is accessible from the Cisco Configuration Assistant Topology view.
	SSL- and SSHv2-based secure remote access	X	X	Provides for secure management between PC and Cisco Unified Communications 500 Series.
	Network Address Translation (NAT)	X	X	1-to-1 static port mapping for TCP and User Datagram Protocol (UDP) ports. VoIP pass-through enabled by default on Cisco SR500 Series Secure Router. Cisco Configuration Assistant 3.0 adds support for 1-to-many static NAT mappings.
	Remove NAT and firewall	X	X	Remove NAT and firewall from Cisco Unified Communications 500 Series and Cisco SR500 Series Secure Router for deployments in network with existing firewall
	DMZ	X	X	A DMZ network enables Internet users to access a company's public servers, including web and FTP servers, while maintaining security for the company's private LAN.
	Security audit	X	X	Assesses vulnerability of existing Cisco Unified Communications 500 Series and Cisco SR500 Series Secure Router.
				Provides quick compliance with best-practices (Cisco Technical Assistance Center [TAC], ICSA recommendations) security policies for Cisco Unified Communications 500 Series and Cisco SR500 Series Secure Router.
	Security diagnostics	X	X	Collect firewall/NAT and VPN debug logs.
	Monitoring	X	X	EzVPN client and server, site-to-site VPN, SSL VPN, firewall, NAT, and VPN status reports.
Video Monitoring/IP Cameras				
	Cisco PVC2300 and WVC2300 Business Internet Video Camera support	X	X	
	Video monitor setup wizard	X	X	Step-by-step wizard for enabling viewing of video from Cisco PVC2300 or WVC2300 Business Internet Video Cameras on SPA525G and SPA525G2 IP phones
	Bonjour discovery	X	X	Ability to discover Cisco and third-party devices running the Bonjour protocol. Devices that can run Bonjour include video cameras and network printers. Cisco Configuration Assistant 2.2 can launch the web GUI of Bonjour discovered devices that are not natively supported in Cisco Configuration Assistant.

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Section Two: SBCS Feature Descriptions

Table 5 lists and describes all of the available Cisco Smart Business Communications System features. Not all of these features require configuration.

Table 5. SBCS Feature Descriptions

Feature	Category	Description
Abbreviated dialing speed dial	Users, phones, and extensions	This feature allows you to quickly dial a phone number by entering an assigned index code (1-99) on the phone keypad. Abbreviated dialing can be useful if your phone model does not provide speed-dial buttons or if you want to configure more speed-dial numbers than the number of speed-dial buttons on your phone.
		You can assign index codes from the User Options webpage or on the phone using Services, My Phone Apps, or Speed Dial.
		The associated softkey is AbbrDial.
Access to a greeting management system from the telephone user interface (TUI)	Voicemail	Subscribers with administrative privileges can access a greeting management system (GMS) for recording alternate greetings and prompts through their phone. This feature is also referred to as Administration via Telephone (AvT) or prompt management. Administrators can change Auto Attendant greetings remotely for snow days, for example.
Account code entry	Users, phones, and extensions	Account code entry allows you to enter account codes during call setup or when connected to an active call using the Acct softkey (a nonforced option). Account codes are inserted into call detail records (CDRs) on the Cisco Unified Communications 500 Series for Small Business for later interpretation by billing software.
		The account code also appears in the “account-code” attribute of the CDR. To enter an account code during call setup or when in a connected state, press the Acct softkey, enter the account code using the phone keypad, and then press the # key to notify the Cisco Unified Communications 500 that you have entered the last digit of the code. The Cisco Unified Communications 500 processes the account code digits upon receipt of the #.
Ad hoc conferencing	Voice-system features	You can have multiparty impromptu conferencing. Conference calls can include other IP phones, analog phones, or external calls through SIP or PSTN trunks. With an analog phone you can set up only three-party conferences.
Agent availability and hunt groups	Voice-system features	Three options increase the flexibility of hunt groups:
		• Dynamic hunt-group membership allows an authorized agent to join and leave hunt groups.
		• Agent status control allows an agent to manually activate a toggle to temporarily enter a not-ready state, in which hunt-group calls bypass the agent’s phone.
		• Automatic agent status not-ready automatically puts an agent’s phone in a not-ready state after a specified number of hunt-group calls are unanswered by the agent’s phone.
Alternate automated-attendant greetings and prompts	Voicemail	You can record alternate automated-attendant greetings and prompts that you can upload or download as needed. These alternate greetings and prompts are in addition to the default greetings and prompts that ship with the Cisco Unified Communications 500.
Analog phone support	Voice-system features	You can add analog phones for fax machines or other devices that you can connect to built-in analog station (FXS) ports on the Cisco Unified Communications 500. In user mode, these analog FXS ports require a user license and can use IP phone features such as voicemail and conferencing. In common area or fax mode, these ports do not consume a user license and are restricted to making and receiving calls.
		You can connect additional analog phones using an analog telephone adapter (ATA).
ATA (Cisco Analog Telephone Adapters)	Users, phones, and extensions	The Cisco ATA analog telephone adaptors are handset-to-Ethernet adaptors that allow regular analog telephones to operate on IP-based telephony networks.
Audio paging	Voice-system features	Audio paging provides a one-to-many voice message to phones that are designated to receive paging. When a caller dials the paging number, each idle IP phone that is configured with the paging number automatically answers using its speakerphone. Only the voice from the phone originating the page is heard. The phones receiving the audio page can only listen.

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Feature	Category	Description
Automatic agent status not-ready; electronic phone (ephone) hunt groups	Voice-system features	This feature automatically puts an agent's phone in a not-ready state after a specified number of hunt-group calls are unanswered by the agent. Typically used with basic automatic call distributors (BACDs), this feature is also called auto logout.
Automatic line selection	Users, phones, and extensions	Picking up the handset answers the first ringing line or, if no line is ringing, selects the first idle line.
Auto Attendant	Voicemail	The Auto Attendant provided with Cisco Unified Communications 500 provides simplified self-service for callers by allowing them to quickly reach a person, hear an audio directory, or hear recorded guidance without the assistance of an operator. It is navigated using touch-tone entries from your phone. The Auto Attendant can have different options based on time of day and holiday schedules. It can also redirect to another menu layer in the Auto Attendant up to 5 layers deep.
Auto Attendant script	Voice-system features	The Auto Attendant is included in the Cisco Unified Communications 500 as aa_sbcs_v04.aef script. It supports dial-by-extension, dial-by-name, alternate and business-hours greetings, bilingual Auto Attendant, bilingual prompt management, and no-option transfer.
Basic Automated Call Distribution (Basic ACD)	Voice-system features	Basic ACD provides automatic answering and distribution of incoming calls through interactive menus and hunt groups. A Basic ACD application consists of one call queue service and up to 10 Basic ACD services. For each Basic ACD service, users configure a pilot number for the service, hunt group parameters, prompts, destination for unanswered calls, timeout, number of retries, and other settings. The Basic ACD call flow implemented in Configuration Assistant is limited to drop- through mode, in which the Auto Attendant serves as the top-level entry point and control is transferred to Basic ACD for second-level menu actions.
Barge, Conference Barge (cBarge)	Voice-system features	The barge feature allows phone users who share a directory number to join an active call on the shared line by pressing a softkey. When the initiator barges into a call, a conference is created between the barge initiator, the target party, and the other party connected in the call. Parties see the call information on their phones and, if the conference join tone is configured, they hear a tone.
		If a phone that is using the shared line has privacy enabled, call information does not appear on the other phones that share the line and the call cannot be barged. Connected parties hear the barge tone (single beep) after the conference is set up. When a party leaves the conference, a barge leave tone is played to the remaining parties.
Bilingual voice messaging	Voice-system features	Bilingual voice messaging allows the caller to the mailbox to hear the voicemail greetings in one of two languages installed on the Cisco Unity Express.
Billing records	Voice-system features	The accounting process collects accounting data for each call leg created on the Cisco Unified Communications 500. You can use this information for activities such as generating billing records and network analysis. The feature captures accounting data in the form of CDRs containing attributes defined by Cisco. The feature can send CDRs to a RADIUS server, syslog server, or to a file in comma-separated value (CSV) format for storing to flash memory or an FTP server.
Blast hunt group	Voice-system features	With this feature enabled, incoming calls simultaneously ring multiple phones or multiple destinations, including PSTN, SIP trunk, and multisite numbers. This feature is also called parallel hunt groups. In Cisco Unified Communications 500 Software Pack 7.0.2 and later versions, this feature is supported across all IP phones.
Busy lamp field (BLF)	Users, phones, and extensions	This feature provides a visible line status indicating whether or not the line is in use. A monitor-line lamp is off or unlit only when its line is in the idle call state. The idle state occurs before a call is made and after a call is completed. For all other call states, the monitor line lamp is lit. A receptionist who monitors the line can see that it is in use and can decide not to send additional calls to that extension, assuming that other transfer and forwarding options are available. BLF for phones is also available; refer to the BLF notification feature that follows.
BLF notification ("watch mode")	Users, phones, and extensions	A line button that is configured for watch mode on one phone provides BLF notification for all lines on another phone (watched phone) for which the watched directory number is the primary line. Watch mode allows a phone user, such as a receptionist, to visually monitor the in-use status of an individual phone. The line and line button on the watching phone are available in watch mode for visual status only. Calls cannot be made or received using a line button that has been set in watch mode. Incoming calls on a line button that is in watch mode do not ring and do not display caller ID or call-waiting caller ID.
Blocking caller ID	Voice-system features	You can selectively choose to block your name or number on outbound calls. Caller ID blocking on outbound calls does not apply to PSTN calls through analog FXO ports. Caller ID features on analog FXO-connected subscriber lines are under the control of the PSTN service provider, who may require you to use the provider's caller ID blocking service.
Busy timeout	Voice-system features	Busy timeout sets the length of time after which calls that are transferred to busy destinations are disconnected.

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Feature	Category	Description
Call forwarding	Voice-system features	Call forwarding allows you to divert incoming calls to an extension to a specific destination under different conditions: • All calls: All incoming calls are diverted based on destination configured by the administrator or entered by a user. • No answer: All incoming calls are diverted when the extension does not answer the call for a specific timeout to a destination such as voicemail configured by the administrator. • Busy: All incoming calls are diverted when the extension is busy and call waiting is not active to a destination such as voicemail configured by the administrator. • Night service: All incoming calls are automatically diverted during designated hours, defined by night service schedule, to a destination such as an automated attendant configured by the administrator. An extension can have all four types of call forwarding defined at the same time, with a different forwarding destination defined for each type of call forwarding.
Call history	Voice-system features	This feature helps collect call-detail information for reporting activities such as generating billing records and voice network analysis. The system captures accounting data in the form of CDRs that can be stored as a text file (CSV format) for storing to flash memory or exporting to an FTP server for third-party application integration.
Call hold	Voice-system features	The system by default allows you to place a call on hold by using the Hold softkey on any active call. For calls placed on hold, the system can enable the optional on-hold indicator, which generates a ring burst on idle phones or call-waiting beeps for phones on another call.
Call hunt	Voice-system features	Call hunt allows you to use the same extensions on multiples phones or lines on the same phone to provide coverage for a single called number. It uses preference to control the order in which the extensions are matched and a huntstop option to determine when the call should be forwarded to the no answer destination.
Call park	Voice-system features	Call park allows you to place a call on hold at a special extension called “park slot” such that you can retrieve the parked call from any other phone in the system. The system supports basic call park; directed call park, where you can transfer the call to a predefined park-slot extension; and dedicated call park, which reserves specific call-park slots for a specific phone.
Call pickup	Voice-system features	Call pickup allows you to answer a call that is ringing on another phone. The system supports three types of call pickup: • Directed call pickup, where you can pick up a ringing call on another phone • Local group pickup, where you can pick up a ringing call on another phone by pressing the GPickupsoftkey and then the asterisk (*) if both phones are in the same pickup group • Group pickup, where you can answer a ringing phone in a different pickup group by pressing the GPickUpsoftkey and then dialing the pickup group number
Call transfer	Voice-system features	Call transfer allows you to transfer the current active call on your phone to a different destination. Call transfers can be blind, where the transferring extension connects you to the target destination before the target phone rings, or consultative, where the transferring extension either connects you to a ringing phone or speaks with the target destination before connecting you to the target.
Call waiting	Users, phones, and extensions	Call waiting allows you to be alerted when you receive an incoming call while you are on another call. While you are on an active call, you get an audible call-waiting tone such as beep or ring and also visually see the calling-party information on your phone screen.
Call waiting for overlaid ephone directory-number extensions	Users, phones, and extensions	For extensions in an overlay set (refer to “Ephone-dn, overlaid” later in this table), by default call waiting is disabled. If enabled, call waiting causes idle phones to ring and phones with active calls to generate an audible call-waiting notification such as a beep or ring along with visual call-waiting notification by displaying caller ID on the phone screen.
Call-waiting beep	Users, phones, and extensions	By default, the audible indication for call waiting on the system is to use beeps. The administrator can turn this indication off if required.
Called-name display	Users, phones, and extensions	When phone users answer calls for several different departments, it is often helpful for them to see a display of the name of the called extension, rather than the number. The called-name display feature can display either the name associated with an extension in a local directory or the name associated with an overlay extension.
Caller ID blocking	Users, phones, and extensions	You can block the display of caller ID information for outgoing calls from an extension on a per-call basis, allowing you to maintain your privacy when necessary. The system administrator defines a code for caller ID blocking, which you then dial before making any call on which you do not want your caller ID sent. Alternatively, the system can also block caller ID on specific dial patterns or trunks if required. Outbound caller ID blocking does depend on the PSTN trunks being used - you cannot use it on analog lines because the control of caller ID in this case is with the service provider.
Call detail records	Voice-system features	Refer to “Call history”
Channel huntstop	Users, phones, and extensions	The system by default allows two calls per line on each phone. If this feature is enabled, a second incoming call is sent to a line to the call-forward busy target if an active call is already on that line. (Refer also to “Call hunt.”)

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Feature	Category	Description
Cisco IP Communicator	Users, phones, and extensions	Cisco IP Communicator is the Windows softphone client that is supported on the system. It typically acts as a remote teleworker phone for users at remote sites (teleworkers).
Conference gain control	Voice-system features	Gain control on conference calls keeps the volume for callers at the same volume for ease of use.
Conferencing	Voice-system features	Conferencing allows you to connect three or more parties in a telephone conversation. Conferences can be hardware- or software-based, depending on the number of parties.
Consulting call transfers (SIP call control only)	Voice-system features	Voicemail permits attended and semiattended call-transfer modes in addition to blind transfers. You cannot configure this feature through the GUI.
Dedicated FXO trunk lines	Voice-system features	You can configure IP phones running SCCP to have buttons for dedicated PSTN FXO trunk lines, also known as FXO lines. FXO lines are ideal for companies whose employees require private PSTN numbers. FXO lines can use PSTN service provider voicemail: when the line button is pressed, the line is seized, allowing you to hear the stutter dial tone provided by the PSTN to indicate that voice messages are available.
		Because FXO lines behave as private lines, you do not have to dial a prefix, such as 9 or 8, to reach an outside line. To reach phone users within the company, FXO-line users must dial numbers that use the company's PSTN number. For calls to non-PSTN destinations, such as local IP phones, a second directory number must be provisioned.
		Calls placed to or received on an FXO line have restricted Cisco Unified Communications 500 services and cannot be transferred by the application. However, phone users are able to access hookflash-controlled PSTN services using the Flash softkey.
DHCP setup	Voice-system features	When a Cisco Unified IP Phone is connected to the Cisco Unified Communications 500 system, it automatically queries for a Dynamic Host Configuration Protocol server (onboard or external to the application). The DHCP server responds by assigning an IP address to the Cisco Unified IP Phone and providing the IP address of the Trivial File Transfer Protocol (TFTP) server through DHCP option 150. Then the phone registers with the Cisco Unified Communications 500 and attempts to get configuration and phone firmware files from the TFTP server.
Differentiated Services Code Point	Voice-system features	DSCP packet marking specifies the class of service for each packet. Cisco Unified IP Phones get their DSCP information from the configuration file that is downloaded to the device.
Directed call pickup	Voice-system features	Any local phone user can pick up a ringing call on another phone by pressing a softkey and then dialing the extension. You do not need to belong to a pickup group to use this method. The softkey that you press, either GPickUp or PickUp, depends on your configuration.
Directories	Voice-system features	Local, called-name display, and directory search use directory services.
Called-name display	Users, phones, and extensions	The called-name display feature can display either of the following types of names: • Name for a directory number in a local directory • Name associated with an overlay directory number Calls to the first directory number in a set of overlay numbers display a caller ID. Calls to the remaining directory numbers in the overlay set display the name associated with the directory number.
Display phone header bar	Users, phones, and extensions	You can customize the content of an IP phone header bar, which is the top line of the IP phone display. The IP phone header bar, or top line, of a Cisco Unified IP Phone normally replicates the text that appears next to the first line button. The header bar can contain a user-definable message instead of the extension number. For example, the header bar can be used to display a name or the full E.164 number of the phone. If no description is specified, the header bar replicates the extension number that appears next to the first button on the phone.
Phone system-message display	Users, phones, and extensions	The system-message display feature allows you to specify a custom text or display messages to appear in the lower part of the display window on display-capable IP phones. If you do not set a custom text or display message, a default message is displayed.
Distinctive ring	Users, phones, and extensions	Distinctive ring is used to identify internal and external incoming calls. An internal call is defined as a call originating from any Cisco Unified IP Phone that is registered in the Cisco Unified Communications 500 or is routed through the local FXS port.
Do not disturb (DND)	Users, phones, and extensions	The DND feature prevents incoming calls from audibly ringing a phone. When DND is enabled, the phone flashes an alert to visually indicate an incoming call instead of ringing, and you can answer the call if desired.
DSP	Voice-system features	A digital-signal-processor (DSP) chip provides analog FXS and FXO and digital BRI/PRI to IP connectivity, in addition to conferencing features for audio calls.
Direct station select (DSS)	Users, phones, and extensions	DSS allows a multibutton phone user to transfer calls to an idle monitored line by pressing the Transfer key and the appropriate monitored line button. A monitored line is one that appears on two phones; one phone can use the line to make and receive calls and the other phone simply monitors whether the line is in use. Consultative transfers can occur during DSS transfers (transferring calls to idle monitored lines).

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Feature	Category	Description
Dual-tone multifrequency (DTMF) relay (SIP call control only)	Voice-system features	DTMF relay handles incoming and outgoing DTMF signals for SIP calls.
DTMF relay for H.323 networks	Voice-system features	For IP phones on H.323 networks, DTMF is relayed using the H.245 alphanumeric method, which is defined by the ITU H.245 standard. This method separates DTMF digits from the voice stream and sends them as ASCII characters in H.245 user input indication messages through the H.245 signaling channel instead of the Real-Time Transport Protocol (RTP) channel.
DTMF relay for SIP trunks	Voice-system features	To use remote voicemail or interactive-voice-response (IVR) applications on SIP networks from Cisco Unified Communications 500 phones, the DTMF digits used by these phones must be converted to the RFC 2833 in-band DTMF relay mechanism used by SIP phones. The SIP DTMF relay method is needed in the following situations: • When SIP is used to connect a Cisco Unified Communications 500 system to a remote SIP-based IVR or voicemail applications • When SIP is used to connect a Cisco Unified Communications 500 system to a remote SIP-PSTN voice gateway that goes through the PSTN to a voicemail or IVR application; SIP phones natively support in-band DTMF relay as specified in RFC 2833
Encrypting stored personal identification numbers (PINs)	Voice-system features	Voicemail PIN codes are stored in encrypted form for security reasons.
Ephone	Voice-system features	Ephone is a term for Cisco Unified Communications 500 configuration for phones using SCCP or the single-channel-per-carrier (SCPC) protocol.
Extension mobility	Users, phones, and extensions	Extension mobility provides the benefit of phone mobility for end users. It offers a user login service that allows phone users to temporarily access a physical phone other than their own phone and use their personal settings, such as directory number, speed-dial lists, and services, as if the phone were their own desk phone. The phone user can make and receive calls on that phone using the same personal directory number as on their own desk phone.
Hunt groups (ephone-hunt)	Voice-system features	Hunt groups allow you to direct incoming calls to a specific number (pilot number) to a defined group of extension numbers. There are four different types of hunt groups: sequential, peer, longest-idle, and parallel. Each type uses a different strategy to determine the first number that rings for successive calls to the pilot number, as described later in the table.
ephone-dn overview	Voice-system features	You can associate buttons on a phone with a line. Lines are of various types; some allow only a single call, whereas others allow two or more calls. You can associate a phone with multiple line buttons with multiple lines, allowing one phone to have multiple numbers. You can also share a given line across multiple phones, allowing one number to ring multiple phones.
ephone-dn, overlay	Voice-system features	Overlaid lines allow you to increase the capacity of a line to up to 25 simultaneous calls. This feature is useful in a helpdesk- or call center-like setup where calls to a single number are answered by multiple people at different phones.
Feature access code (FAC)	Voice-system features	FACs are special patterns of characters that are dialed from a telephone keypad to invoke particular features. For example, you might press **1, and then press 2345 to forward all incoming calls to extension 2345. FAC is typically used only on an analog phone that does not have softkey.
Fax	Users, phones, and extensions	The fax feature allows the voicemail system to include fax-machine support using analog (FXS) lines. Fax traffic is handled using SCCP and pass-through by default, or it can be set up for T.37 or T.38 with support for G2 and G3 speeds.
Fax relay	Voice-system features	The fax-relay feature allows two fax machines to exchange faxes over an IP network. It recognizes that a call is a fax and not a voice call, and by doing so it provides a more robust transport of the facsimile data.
Feature ring	Users, phones, and extensions	When a phone has more than one line associated with it, you can configure one of the lines with a feature ring. The feature ring allows you to easily recognize that an incoming call is ringing a specific line.
Fixed holidays	Voice-system features	The Auto Attendant greetings are controlled by a calendar. You can identify certain days as fixed holidays because they fall on the same date every year. On holidays the Auto Attendant plays a greeting that is appropriate for a day when the business is closed.
Fixed line and feature button set	Users, phones, and extensions	The Cisco Unified IP Phone 7931G has a large number of buttons. You can select from two fixed button-layout formats to assign functions to certain line buttons on the phone to support key-system phone behavior.
Flash softkey and FXO hookflash	Users, phones, and extensions	The Flash softkey provides hookflash functions for calls made on analog trunks. Certain PSTN services, such as three-way calling and call waiting, require hookflash intervention from a phone user. The Flash softkey provides this function for IP phones.

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Feature	Category	Description
Forwarding	Users, phones, and extensions	Call forwarding diverts calls to an alternative specified number under one or more of the following conditions: • Always • Upon no answer • Upon busy • When night service is active
FXS (analog) ports	Users, phones, and extensions	Call processing supports FXS analog phone ports. FXS ports behave like a regular phone line from the phone company and allow you to connect a regular analog phone or fax machine.
Group call pickup	Voice-system features	With group call pickup, you can answer a ringing phone in any pickup group by pressing the GPickUpsoftkey and then dialing the pickup group number. If only one pickup group is defined in the Cisco Unified Communications 500 system, you can pick up the call simply by pressing the GPickUpsoftkey. You do not need to belong to a pickup group to use this method.
Hairpin call routing	Voice-system features	The Cisco Unified Communications 500 supports hairpin call routing. When a call that originally terminated on a FXO port from the PSTN is transferred or forwarded by a phone back out to the PSTN (for example, an outside caller makes a call to a Cisco Unified Communications 500 phone and the call is forwarded to the callee's cell phone), the application reoriginates the call and routes it back out to the PSTN through another FXO port. This looping back from and to the PSTN is called hairpinning. Hairpin routing of transferred and forwarded calls also causes the generation of separate billing records for each call leg, so that the transferred or forwarded call leg is typically billed to the user who initiates the transfer or forward.
Hardware-based conferencing	Voice-system features	Conferencing allows you to connect three or more parties in a telephone conversation. Conferences can be hardware- or software-based, depending on the number of parties. Hardware-based Ad Hoc conferencing (maximum of 8 parties) uses DSPs to allow more parties than software-based Ad Hoc conferencing, which allows 3 parties only. Meet-me hardware-based conferences (maximum of 32 parties) are created by parties calling a designated conference number. If you configure software-based conferencing, you cannot have meet-me conferences.
Header-bar display	Users, phones, and extensions	The header-bar display is the text on the top line of the IP phone display.
Historical reports	Voice-system features	Historical reports refers to the ability of the Cisco Unified Communications 500 to provide reports about call activities and application activities on the system.
Hold	Users, phones, and extensions	When a call is in progress, you can use the Hold softkey to place the call on hold. The person on the other end typically hears music while on hold. Pressing Resume reconnects to the caller.
Hunt groups	Voice-system features	Hunt groups allow you to direct incoming calls to a specific number (pilot number) to a defined group of extension numbers.
Password and PIN security protection	Voice-system features	This feature provides both temporary and permanent lockout for passwords and PINs to help prevent security breaches; it includes set minimum lengths and expiry times for passwords and PINs.
Integrated messaging	Voice-system features	Cisco Unified Communications 500 voicemail subscribers can access and manage their voice and fax messages using an Internet Message Access Protocol (IMAP)-enabled email client such as Microsoft Outlook and Outlook Express. Integrated messaging brings voice, fax, and email together at the email client.
Intercom lines	Users, phones, and extensions	An intercom line is a dedicated two-way audio path between two phones. The Cisco Unified Communications 500 supports intercom functions for one-way and press-to-answer voice connections using a dedicated pair of intercom directory numbers on two phones that speed-dial each other.
		When you press an intercom speed-dial button, a call is speed-dialed to the directory number that is the other half of the dedicated pair. The called phone automatically answers the call in speakerphone mode with mute activated, providing a one-way voice path from the initiator to the recipient. A beep is sounded when the call is autoanswered to alert the recipient to the incoming call. To respond to the intercom call and open a two-way voice path, the recipient deactivates the mute function by pressing the Mute button.
Interdigit timeout	Voice-system features	Interdigit timeout is the number of seconds the system waits between dialed digits. The default is 10 seconds.
International languages and tones	Users, phones, and extensions	International languages and tones refer to the language used for text displays and the country-specific tones and cadences required for connection to the local telephone network.
Key system	Voice-system features	In a key system, most phones have nearly identical configurations, in which each phone can answer any incoming PSTN call on any line without the aid of a receptionist, an automated-attendant service, or (expensive) direct-inward-dialing (DID) lines. Also, the lines act as shared lines - you can put a call on hold on one phone and resume the call on another phone without invoking call transfer.
Keyswitch telephone system	Voice-system features	This Cisco Unified Communications 500 configuration option sets the system up to act as a key system; refer also to "Key system" and "Dedicated FXO trunk lines."
Languages and tones	Voice-system features	Refer to "International languages and tones."

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Feature	Category	Description
Leaving multiple voice messages in the same session	Voice-system features	You can leave multiple voice messages for the same or different subscribers without having to hang up and call back to the system each time.
License upgrades	Voice-system features	An enforced system determines how many phones can register to the system using the Cisco Licensing System. Customers wanting additional phones can purchase a software license upgrade, receive a PAK ID, register this PAK ID on Cisco.com, and receive a license file by email (or within Cisco Configuration Assistant); the file is applied to the system, stored in ROM, and used to allow additional phones to connect or register.
Live record	Applications	The live-record feature enables IP phone users in a Cisco Unified Communications 500 system to record a phone conversation with the Cisco Unified Communications 500 voicemail system. Callers will hear a brief pause and optional tone at the start of call recording. By option (required in some states or countries), an audible notification, by either announcement or periodic beep, alerts participants that the conversation is being recorded.
Live reply	Voice-system features	Live reply enables voicemail subscribers to make a phone call to the sender of a voice message while listening to the message by pressing 4-4.
Local directory	Voice-system features	The local directory lists all internal users with extension number by name plus the option to add 250 external additional names (including hunt groups and call blast groups) and phone numbers that a user can search then autodial using the IP phone display.
Local group pickup	Voice-system features	You can pick up another phone ringing in the same group without dialing the group number.
Locale installer	Voice-system features	You can easily add new IP phone country locales (text on display phones plus tones heard).
Longest-idle ephone hunt groups	Voice-system features	New calls to this hunt-group type ring the directory number that has been idle for the longest duration. An option is for the phone instead of the directory number to be used for timing. This feature is used with BACDs.
Manual backup and restore	Voice-system features	Cisco Unified Communications 500 configuration includes backing up voicemail using Cisco Configuration Assistant by choosing the backup option. Scheduled backups are not currently supported.
Maximum calls per button	Users, phones, and extensions	You can configure each button on an IP phone to handle one to eight calls at one time. Only one call is active; other states include call waiting, hold, and hold and place new call. The Cisco Unified IP Phone 7940, 7960, and 7970 models support one to eight calls per button; the Cisco Unified IP Phone 7931 supports only one call; and the Cisco Unified Communications 500 Series phones support one or two calls per button.
Meet-me conference	Voice-system features	A meet-me conference is a voice conference bridge initiated by one IP phone user, where other users (internal or external by DID or Auto Attendant) join. No scheduling or password option exists for meet-me conferencing. Up to 32 phones can join a single meet-me session on Cisco Unified Communications 500.
Message properties (envelope) customization	Voice-system features	This systemwide option lets you choose the information about each voicemail message that is played (envelope information) before the message is played. Options include sender, time and date, receive date (if more than 30 minutes from send time), message number, priority, and type.
Monitor mode	Users, phones, and extensions	You can assign a button on your IP phone to show the idle or busy status of a shared directory number. Also known as busy lamp field (BLF), in this case this feature is only for a directory number, not a phone. You should use watch mode to show the status of the phone.
Monitor-line speed dial	Users, phones, and extensions	You can assign a button on your IP phone to show the idle or busy status of another directory number on the system; this feature is also known as BLF with an additional option to have the Idle button act as a speed-dial button to call that phone. For example, an administrator will use the monitor-line speed dial feature to watch the supervisor's phone, and can press the button when idle to easily call the supervisor.
MTP	Voice-system features	Media termination point (MTP) is an IP phone feature that forces all RTP voice streams to transverse though the Cisco Unified Communications 500 unit (like a proxy) for proper call handling with NAT and firewalls.
Multiple lines per phone	Users, phones, and extensions	Multibutton IP phones offer the option to support more than one directory number per phone. Cisco Unified Communications 500 also supports up to 10 directory numbers on a per-button basis using overlays.
Multi-site	Voice-system features	Multi-site is the common name for the ability to connect multiple UC or SR 500s across the internet using VPN connection(s). The VPN connection(s) encrypt data, voice and video traffic for secure communications.
		You can connect up to 5 sites using a full-mesh topology. That means each site can have up to 4 direct VPN connections to other sites.
Music on hold	Voice-system features	Outside callers placed on hold will hear recorded music. The source can be a WAV file saved to flash memory or a live source. Only one source for all phones is supported.
New voicemail subscriber feature	Voice-system features	New voicemail users when first calling in to the system are given a minitutorial on use, recording their voice name, recording a greeting, and setting a PIN.

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Feature	Category	Description
Network Time Protocol (NTP) reference	Voice-system features	Cisco Unified Communications 500 needs the current time and date to show on IP phones and time stamp for voice messages. NTP reference service allows the application to get the current time and date from an external server, which allows all devices in a network to be kept synchronized.
Night service	Voice-system features	You can configure a different set of phones to ring when night service is enabled. You can activate night service by dialing a code, using speed dial, or automatically setting it for certain days of week and hours per day.
Octo-lines	Users, phones, and extensions	The octo-line feature supports up to eight active calls (octo) per directory number assigned to a button. This feature is supported on Cisco Unified IP Phone 794x, 796x, and 797x models.
Off-premises extension	Users, phones, and extensions	The Cisco Unified Communications 500 system supports remote teleworker phones and off-premises phones. The phones can be IP-based using VPN or analog using telco-provided wiring.
On-hook dialing	Users, phones, and extensions	An IP phone user can dial a number (internal or external) without going off hook.
On-hook transfer	Users, phones, and extensions	The call transfer feature supports the on-hook (hangup) action as a possible last step to complete a call transfer. With the on-hook transfer implementation, user B can hang up after dialing the number of user C, and the transfer completes.
Online help	Voice-system features	Online help is a detailed, transparent help function embedded in Cisco Configuration Assistant that provides an extensive glossary and powerful search engine that helps you quickly and easily find the information you need to apply specific settings. With these online help features, you often can troubleshoot and resolve problems without having to call for technical support.
Outgoing call restrictions	Voice-system features	The class-of-restrictions (COR) feature allows you to deny certain call attempts based on the incoming and outgoing class of restrictions provisioned on the dial peers. This function provides flexibility in network design, allows you to block calls (for example, to 900 numbers), and applies different restrictions to call attempts from different originators. COR is used to specify which incoming dial peer can use which outgoing dial peer to make a call. You can provision each dial peer with an incoming and an outgoing COR list. The incoming COR list indicates the capability of the dial peer to initiate certain classes of calls. The outgoing COR list indicates the capability required for an incoming dial peer to deliver a call through this outgoing dial peer. If the capabilities of the incoming dial peer are not the same or are a superset of the capabilities required by the outgoing dial peer, the call cannot be completed using this outgoing dial peer.
Paging	Voice-system features	You can define a paging number to relay audio pages to a group of designated phones. When a caller dials the paging number, each idle IP phone that is configured with the paging number automatically answers using its speakerphone mode. Displays on the phones that answer the page show the caller ID that has been set using the name command under the paging ephone-dn. When the caller finishes speaking the message and hangs up, the phones are returned to their idle states. Audio paging provides a one-way voice path to the phones that have been designated to receive paging. It does not have a press-to-answer option like the intercom feature. A paging group is created using a dummy ephone-dn, known as the paging ephone-dn, which can be associated with any number of local IP phones. The paging ephone-dn can be dialed from anywhere, including on-net. After you have created two or more simple paging groups, you can unite them into combined paging groups. By creating combined paging groups, you provide phone users with the flexibility to page a small local paging group (for example, paging four phones in a store's jewelry department) or to page a combined set of several paging groups (for example, paging a group that consists of both the jewelry and the accessories departments).
Park	Voice-system features	The call park feature allows you to place a call on hold at a special extension so you can retrieve it from any other phone in the system. You can park the call at the extension, known as the call-park slot, by pressing the Park softkey. Cisco Unified Communications 500 chooses the next available call-park slot and displays that number on the phone. A user on another phone can then retrieve the call by dialing the extension number of the call-park slot. You can define either a single extension number or a range of extension numbers to use as call-park slots. Each call-park slot can hold one call at a time, so the number of calls that you can park is equal to the number of slots you create. If the secondary number is used to group calls together, calls are retrieved in the order in which they were parked; the call that has been parked the longest is the first call retrieved from the call-park slot. A caller who is parked in a park slot hears the MOH audio stream if the call uses the G.711 codec or if the call uses G.729 with transcoding; otherwise, the caller hears a tone on hold. Users who attempt to park a call at a busy slot hear a busy tone.
Parallel hunt groups (call blast)	Voice-system features	In a parallel hunt group, calls simultaneously ring multiple phones. Using parallel hunt groups is also referred to as application-level forking because it enables the forking of a call to multiple destinations. In versions earlier than Cisco Unified Communications 500 Version 4.3, only SIP phones support parallel hunt groups. In Cisco Unified Communications 500 Version 4.3 and later versions, SCCP phones also support voice hunt groups. You can enable functions similar to parallel hunt groups on SCCP phones by using the ephone-dn overlay feature for shared lines. The number of ringing calls that a parallel hunt group can support depends on whether call waiting is enabled on the SIP phones.

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Feature	Category	Description
IP phone password setting	Users, phones, and extensions	You can change your phone password from your User Options webpages. Phone passwords are used for computer telephony integration/Telephony Application Programming Interface (CTI/TAPI) integrations, extension mobility, toll-bar override, and user page login.
PBX system	Voice-system features	When setting up a Cisco Unified Communications 500 system, you need to decide if call handling should be similar to that of a PBX, similar to that of a keyswitch, or a hybrid of both. The Cisco Unified Communications 500 Series provides significant flexibility in this area, but you must have a clear understanding of the model that you choose. The simplest model is the PBX model, in which most of the IP phones in your system have a single, unique extension number. Incoming PSTN calls are routed to a receptionist at an attendant console or to an automated attendant. Phone users can be in separate offices or geographically separated and therefore often use the telephone to contact each other.
Peer ephone hunt groups	Voice-system features	This feature defines a hunt group in which the first extension to ring is the number to the right (in the list) of the extension that was the last one to ring when the hunt group was last called. Ringing proceeds in a circular manner, left to right, for the number of hops specified when the ephone hunt group is defined.
Personal address book (PAB)	Users, phones, and extensions	PAB is a directory for personal contacts. It is stored locally on the Cisco SPA525G 5-line IP Phone with Color Display.
Personal speed dial	Users, phones, and extensions	With this feature, you can configure a maximum of 24 personal speed-dial numbers per phone; the numbers are accessed through the Directory, Personal Speed Dial listing. Each phone can have up to 99 fast dials. You can configure each phone using the Options webpage or through the phone using Services, MyPhoneApps, Speed dial.
Phantom extension	Users, phones, and extensions	This feature allows you to create an extension without associating it with any phone.
Phantom mailbox	Voicemail	This feature lets you create a mailbox without associating it with any phone.
Phone display	Users, phones, and extensions	Phone display features comprise the following functions: • Caller-name display: When phone agents answer calls for several different departments or people, it is often helpful for them to see a display of the name, rather than the number, of the called party. The caller-name display feature can display either of the following types of name: name for a directory number in a local directory or name associated with an overlay directory number. • Header bar: Refer to “Phone header bar display.” • System message display: The system message display feature allows you to specify a custom text or display message to appear in the lower part of the display window on display-capable IP phones. If you do not set a custom text or display message, the default message “UC5X0” is displayed. When you specify a text message, the number of characters that can be displayed is not fixed because IP phones typically use a proportional (as opposed to fixed-width) font. There is room for approximately 30 alphanumeric characters.
Phone header bar display	Users, phones, and extensions	You can customize the content of an IP phone header bar, which is the top line of the IP phone display. The header bar can contain a user-definable message instead of the extension number. If no description is specified, the header bar replicates the extension number that appears next to the first button on the phone.
Phone labels	Users, phones, and extensions	Phone labels are configurable text strings that can be displayed instead of extension numbers next to line buttons on a Cisco Unified IP Phone. By default, the number that is associated to a directory number and assigned to a phone is displayed next to the applicable button. The label feature allows you to enter a meaningful text string for each directory number so that a phone user with multiple lines can select a line by label instead of by phone number, thus eliminating the need to consult in-house phone directories.
Phone lock (Cisco Unified Wireless IP Phone 7921G and 7925G Wi-Fi phone models)	Users, phones, and extensions	You can secure access to your Cisco Unified Wireless IP Phone 7920 by enabling the phone-lock feature. After powering on the phone, you must enter a password before the phone can authenticate with the wireless network. This feature is not available with desktop phones.
Phone services	Users, phones, and extensions	With this feature, you can use Cisco Unified IP Phones to deploy customized client services with which users can interact through the keypad and display. Services deploy using HTTP from standard web servers. You can access these features using the Services and Directories buttons or menu options (availability varies by phone model). When you press the Services button (or choose the Services menu item), a menu of configured services appears. You can then choose a service from the list, and the phone displays the service. The following list gives typical services that are supplied to the phones when connected to the Cisco Unified Communications 500 system: • Visual voicemail (VoiceView Express) • System speed dial • Personal speed dial • Call history • Local directory

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Feature	Category	Description
Phone softkeys	Users, phones, and extensions	With this feature, you can customize the display and order of softkeys that appear during various call states on individual IP phones. Softkeys that are appropriate in each call state are displayed by default. Using phone templates, you can delete softkeys that would normally appear or change the order in which the softkeys appear. For example, you might want to display the CFwdAll and Confrnsoftkeys on a manager's phone and remove these softkeys from a receptionist's phone. You change the softkey order by defining a phone template and applying the template to one or more phones. You can create up to 10 phone templates for SCCP phones.
Pickup groups	Voice-system features	With this feature, you can answer a ringing phone in any pickup group by pressing the GPickUpsoftkey and then dialing the pickup group number. If only one pickup group is defined in the Cisco Unified Communications 500 system, you can pick up the call simply by pressing the GPickUpsoftkey. You do not need to belong to a pickup group to use this method. If both phones are in the same pickup group, you can pick up a ringing call on another phone by pressing a softkey and then the asterisk (*). The softkey that you press, either GPickUp or PickUp, depends on your configuration.
PIN setting	Users, phones, and extensions	This feature sets a PIN to be used by a phone user to access voicemail.
Placed call list	Voice-system features	You can view records of your placed calls. While viewing call logs, you can use softkeys to display details for a call record, erase call records, and dial from call records. If you are on another call when dialing, your phone might prompt you with options (Hold, Transfer, Conference, EndCall) for handling the first call before placing the second call.
Power failure transfer	Voice-system features	With no power, the first FXO trunk will be connected to the first FXS port, allowing for calls to be answered or placed until power is restored.
Predial	Users, phones, and extensions	You can enter a phone number before getting a dial tone and complete the call by going off hook (lifting the handset, pressing the speakerphone button, and so on).
Primary Rate Interface (PRI)	Voice-system features	The PRI is a high-capacity digital trunk for carrying voice and data between the service provider and customer; it is based on the ISDN telecommunications standard. In the United States PRI is based on a T1 line and offers 23 channels; in Europe it is based on an E1 line and offers 29 channels.
		All data and voice channels operate at 64 kbps. The primary difference between T1/E1 channel associated signaling (CAS) and PRI is that PRI reserves one channel for signaling and providing features not available with standard T1/E1 circuits such as DID and feature-rich caller ID. The Cisco Unified Communications 500 system supports only voice PRI; shared data and voice circuits are not supported.
Private-line automated ringdown (PLAR)	Users, phones, and extensions	Private-line automated ringdown allows the administrator to set up an analog or IP phone on going off hook to automatically ring a predetermined number. An example is that a phone placed inside an elevator on going off hook would automatically ring the receptionist or other defined number, internal or external.
Programmable buttons	Users, phones, and extensions	Administrators can choose button features for Cisco IP Phones, including a directory number, a shared line for trunks, speed dial, BLF showing the status of another phone with speed dial to that phone, or blank.
PSTN failover	Voice-system features	PSTN failover automatically routes calls over the PSTN network if the IP network is down.
Redial	Users, phones, and extensions	You can automatically dial the last placed call from your IP phone.
Remote teleworker phones	Voice-system features	IP phones used by remote teleworkers are called remote teleworker phones. A remote teleworker is a user who is not physically present at the main office and uses secure IP network to connect remotely.
Remove conference participants	Voice-system features	This feature allows removal of the last user who joined a conference call.
Repeat last number dialed	Users, phones, and extensions	You can automatically dial the last placed call from your IP phone; this feature is also known as redial. You can also use directory, placed calls to redial any of the last 30 calls placed.
Resetting phones	Users, phones, and extensions	You can reset an IP phone remotely from the call control agent (IP PBX). The call control agent sends out control messages to the IP phone to reset itself and start the bootup process.
Resetting TAPI session	Users, phones, and extensions	TAPI sessions are used for controlling IP phones. A Windows desktop application Telephony Service Provider (TSP) is installed and configured with an application like Microsoft Outlook to control the user's IP Phone. The TSP using the Windows telephony API communicates to the UC 500 call-control agent to control the operation of the IP phone. Resetting a TAPI session flushes out the current session parameters and forces the TSP to reestablish communication with the call-control agent. This is sometimes necessary due to TSP or application issues.
Restriction tables	Voice-system features	These tables are used to control a list of numbers (or dial patterns) that are not allowed to be dialed.
Restore to factory defaults	Voice-system features	This operation restores the system to its factory default setting. It resets all the configuration parameters as well as any database maintained by the system.
Resume	Users, phones, and extensions	This softkey feature on IP phones allows you to reclaim a call on hold. While on hold, the other end hears MOH. When resumed, a two-way audio is established between the users.

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Feature	Category	Description
Ringer settings	Users, phones, and extensions	This setting selects the ring tone on an IP phone. You can set ringers on a per-directory number basis.
Ringing-line preference	Users, phones, and extensions	This feature enables you to pick up the handset and get connected to the line that is ringing on your IP phone. This feature is used when multiple lines are configured on an IP phone.
Ringing timeout	Users, phones, and extensions	Ringing timeout is associated with no answer on an incoming call. By default, Cisco Configuration Assistant configures this value to 30 seconds on IP phones; for voice ports it is 180 seconds.
Ring-tone setting	Users, phones, and extensions	You can select the ring tone on your IP phone. You can set ringers on a per directory number basis.
Rollover buttons for overlaid ephone-dns	Voice-system features	Phones with overlaid ephone-dns can use the button command with the x keyword to dedicate one or more additional buttons to receive overflow calls. If an overlay button is busy, an incoming call to any of the other ephone-dns in the overlay set rings on the first available overflow button on each phone that is configured to receive the overflow.
SCCP-controlled analog (FXS) ports	Users, phones, and extensions	You can configure analog voice ports (FXS) to be controlled by the call-control agent using the same protocol that is used to communicate with IP phones. The FXS ports appear as IP phones to the Cisco Unified Communications 500, and you can configure the phones with most of the features available for IP phones, including voicemail.
Schedules for holidays and business hours	Voice-system features	The Auto Attendant on Cisco Unified Communications 500 offers holiday and business schedules to enable time-of-the-day routing of incoming calls.
Secondary dial tone	Voice-system features	A secondary dial tone is available for Cisco Unified IP Phones connected to Cisco Unified Communications 500 in the PBX mode. The secondary dial tone is generated when you dial a predefined PSTN access prefix and terminates when you dial additional digits.
Secure Socket Layer (SSL) Phone Client	Voice-system features	The SSL Phone Client on the SPA525G IP phone provides secure connectivity to a UC500 or SR500 series over the internet. The SSL Phone Client at the remote site does not need a teleworker router, making it an ideal solution for simple scenarios that require voice-only connectivity.
Sequential ephone hunt groups	Voice-system features	The sequencing method of hunting always starts with the first member of the hunt group and hunts through all the members in the sequential order.
Session Initiation Protocol (SIP)	Voice-system features	SIP is a signaling protocol widely used for controlling multimedia communication sessions such as voice and video calls over IP. Cisco Unified Communications 500 supports Cisco SIP endpoint devices as well as SIP trunks to SIP providers.
SIP IP Phone dial plans	Voice-system features	SIP dial plans enable call routing using the SIP protocol. The dial plans route calls toward a SIP server, which could be either another Cisco Unified Communications 500 at a remote site or a SIP server hosted by a SIP provider. A dial plan is a set of dial patterns that SIP phones use to determine when digit collection is complete after you go off-hook and dial a destination number. Dial plans allow SIP phones to perform local digit collection and recognize dial patterns as your input is collected. After a pattern is recognized, the SIP phone sends an INVITE message to the Cisco Unified Communications 500 to initiate the call to the number matching your input. All of the digits you entered are presented as a block to the Cisco Unified Communications 500 for processing. Because digit collection is done by the phone, dial plans reduce signaling messages overhead compared to Keypad Markup Language (KPML) digit collection. SIP dial plans eliminate the need for you to press the Dial softkey or # key, or to wait for the interdigit timeout to trigger an outgoing INVITE. You can configure a SIP dial plan and associate the dial plan with a SIP phone. The dial plan is downloaded to the phone in the configuration file.
SIP IP phones	Users, phones, and extensions	IP phones can communicate using several protocols, including H.323, SIP, and Media Gateway Control Protocol (MGCP). SIP IP phones use the SIP protocol to communicate with a SIP server.
SIP trunk	Voice-system features	SIP trunks provide an alternative to the traditional PSTN (digital and analog) connectivity options. SIP trunks are provisioned through SIP service providers that provide PSTN connectivity.
SIP supplementary services, disabling	Voice-system features	The SIP call transfer and call forwarding supplementary services feature introduces the ability of SIP gateways to initiate blind, or attended, call transfers by passing the call control back to the originating devices. Disabling these supplementary services forces the SIP gateway to handle these call conditions locally.
Shared lines	Users, phones, and extensions	A shared directory number allows the same number to appear on two different IP phones. A call made to a shared directory number rings all the IP phones that have a button assigned to the shared number. A call made from the shared directory number ties up the shared-dn-buttons on the rest of the IP phones. If a call on the shared directory number is put on hold, any of the IP phones can resume the on-hold call.
Shared-line overlay ephone-dns	Users, phones, and extensions	The overlay feature allows a single IP phone button to be associated with multiple directory numbers. A call to any of the associated directory numbers rings the IP phone on the overlaid button. Shared directory numbers can also be part of the overlay configuration. Primary extensions cannot be part of the shared overlay ephone-dns.

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Feature	Category	Description
Single number reach (SNR)	Voice-system features	SNR allows you to have incoming calls to a single number simultaneously ring an IP phone and a remote destination such as a home or mobile phone. You can answer incoming calls to your SNR number on your IP phone or at your remote destination and pick up in-progress calls on your desktop phone or the remote destination without losing the connection. If you do not answer the call within 5 seconds, the Cisco Unified Communications 500 system rings the remote number while continuing to ring your IP phone extension. If you answer the call on your IP phone, you can send the call to the remote phone by pressing the Mobility softkey. If you answer the call on your remote phone, you can pull back the call to the IP phone by pressing the Resume softkey. You can also change the SNR remote destination using the IP Phone menu.
Softkeys	Users, phones, and extensions	Softkeys are keys that appear on the bottom of the IP phone LCD. They allow you to access various features such as call forward, call transfer, conferencing, and call park. The softkeys available for use change dynamically according to whether the phone is in connected, ringing, idle, or seized (handset is lifted). Additional softkeys are also automatically enabled when advanced features, such as SNR and live record, are enabled.
Software-based conferencing	Voice-system features	Software conferencing is the type of voice conferencing that is supported by default on the Cisco Unified Communications 500. It does not require any hardware resources; the audio mixing is done within the application software. Software conferencing allows a maximum of three parties in a conference, with maximum of eight simultaneous conferences.
Speakerphone mode	Users, phones, and extensions	Speakerphone mode allows you to talk and listen hands-free (without using a handset or headset). It is typically activated by pressing the Speakerphone button on an IP phone. Conference phones such as the Cisco Unified IP Phone 7936 and 7937G use speakerphone for all calls. Speakerphones on lower-end Cisco Unified IP Phone 7906G and 7911G model phones allow you to hear audio only.
Speed dialing	Users, phones, and extensions	Speed dial allows you to quickly dial a number from a list. The different types of speed dial include local speed-dial menu, personal speed-dial menu, and speed-dial buttons. The local speed-dial menu is configured by the administrator and is shared between all IP phones on the system. You can configure personal speed dials and speed-dial buttons on your individual IP phones.
SRTP	Voice-system features	Media encryption (SRTP) encrypts the voice media for calls made between IP phones registered to the same Cisco Unified Communications 500, without the need for a dedicated VPN tunnel.
Station volume controls	Users, phones, and extensions	You can adjust the volume for your incoming call ringer and phone speaker. You can also adjust the outgoing volume of the phone microphone on a hands-free call.
Support for caller ID information in incoming messages	Users, phones, and extensions	The Cisco Unified Communications 500 can play back the caller ID number of the caller who recorded the message, when you listen to new voicemail messages.
Support for multiple languages	Voice-system features	You can install multiple concurrent languages on a single Cisco Unified Communications 500. The IP phone display and voicemail prompts can be different on a per-user basis.
Support for vCard information from remote subscribers	Voice-system features	This feature permits vCard information from remote subscribers to update their directory entries.
System message display	Voice-system features	This feature allows you to specify a custom text or display message to appear in the lower part of the display window on your IP phones. If you do not set a custom text or display message, the default message “UC5X0” is displayed.
Teleworker remote phones	Voice-system features	IP phones or instances of Cisco IP Communicator can be connected to a Cisco Unified Communications 500 over a VPN connection to support teleworkers who are at remote locations from the application. Cisco IP Communicator Softphones can connect to the Cisco Unified Communications 500 over a VPN tunnel established using Cisco VPN client installed on a PC. Hardware IP phones connect to the Cisco Unified Communications 500 over a VPN tunnel established using a Cisco Secure Router 500 Series teleworker router The product data sheet is at http://www.cisco.com/cisco/web/solutions/small_business/products/routers_switches/500_series_secure_routers/index.html. The Cisco Unified Communications 500 supports a maximum of 10 teleworker sites, with a maximum of 5 teleworker phones at each site. The VPN client on Cisco SPA525G IP phones can be configured through the Cisco Configuration Assistant Phone VPN Setup Wizard and deployed at remote sites.
T1/E1 digital trunk interface	Voice-system features	This feature provides PBX support for direct digital trunk interfaces, such as ISDN PRI and T1 or E1 CAS circuits.
T1 CAS (channel associated signaling)	Voice-system features	CAS is also referred to as robbed-bit signaling. In this type of signaling, the least significant bit of information in a T1 signal is “robbed” from the channels that carry voice and is used to transmit framing and clocking information. This protocol is sometimes called “in-band” signaling. CAS is a method of signaling each traffic channel rather than having a dedicated signaling channel (such as ISDN). In other words, the signaling for a particular traffic circuit is permanently associated with that circuit. The most common forms of CAS signaling are loopstart and groundstart. In addition to receiving and placing calls, CAS signaling also processes the receipt of Dialed Number Identification Service (DNIS) and automatic-number-identification (ANI) information.

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Feature	Category	Description
Time zones	Voice-system features	Support for time zones on the Cisco Unified Communications 500 allows the correct time and date information to be displayed on IP phones and played back on the voicemail. Correct time-zone information is also essential for integration with applications that use time information, such as Cisco WebEx PhoneConnect.
Toll-fraud prevention	Voice-system features	The Cisco Unified Communications 500 implements several security features to prevent unauthorized phones from both internal and external users when using the Cisco Configuration Assistant tool for configuration.
Toll restrictions	Voice-system features	This feature allows you to prevent specified phones from making long-distance (toll) calls.
Touchscreen	Users, phones, and extensions	You can press the phone screen to choose menu items, softkeys, and feature tabs (on Cisco Unified IP Phones with touch-sensitive phone screens).
Trace (debugging)	Voice-system features	Real-time troubleshooting tools are available through on-demand debugs for the Cisco Unified Communications 500 and within Cisco Configuration Assistant 2.1 and later.
Transcoder resources	Voice-system features	This feature provides translation of RTP streams from one codec format into another.
Transfer	Users, phones, and extensions	When you are connected to another party, call transfer allows you to shift the connection of the other party to a different number. The transferred call (transferee) and transfer destination (transfer-to) can be a local extension or PSTN number. On the Cisco Unified Communications 500, IP phones, analog phones, and Auto Attendant can initiate call transfers.
Transfer to voicemail	Voice-system features	Transfer to voicemail allows you or the Auto Attendant feature to transfer incoming calls directly to a voice mailbox. Unlike normal call transfers, callers who are transferred to a voice mailbox can leave a message immediately, instead of waiting until the phone rings and call forwards to voicemail. IP phone users can transfer calls to voicemail by pressing the TrnsfVMsoftkey, or by entering the transfer to voicemail prefix.
Translation rules and profiles	Voice-system features	Translation rules manipulate dialed numbers to conform to internal or external numbering schemes. Voice translation profiles allow you to group translation rules together and apply them to the following types of numbers: • Called numbers (DNIS) • Calling numbers (ANI) • Redirected called numbers The Cisco Unified Communications 500 uses translation rules and profiles for many purposes, including translating incoming numbers from ISDN or SIP trunks to internal extension numbers and stripping off the access code for outbound dialing to the PSTN.
Trunk groups	Voice-system features	Trunk groups are an administrator-controlled feature that allows administrators to easily configure outbound and inbound calls to have common call-handing properties. For example, all FXO and analog trunks would be in one trunk group and SIP trunks would be in another.
Trunk-to-trunk connections	Voice-system features	Outside callers can be connected to another PSTN trunk when a phone is set to forward (or SNR) or transferred by a user.
Undelete voice messages	Voice-system features	During a voicemail session you can listen once again to a deleted voicemail message.
Vibration alert	Users, phones, and extensions	This feature is supported on Cisco Unified IP Phone 7921 and 7925 models for incoming calls; it is also referred to as vibration mode.
Video Streaming	Voice-system features	This feature allows a live video feed from a Cisco 2300 Series Business Internet Video Camera to be streamed to the display of a Cisco SPA525G IP Phone.
Video support	Voice-system features	Video telephony is as easy as making a voice call. When a Cisco Unified Video Advantage camera and a Cisco Unified IP Phone 7900 Series phone or video endpoint is in use by two users when connected over the LAN or WAN, then a video telephony session will automatically start between the two devices. With no special dialing or button to push, it is very easy to set up and use.
Voice hunt group	Voice-system features	You can set up a voice hunt group to ring all phones at one time; the voice hunt group is also known as a blast or parallel hunt group
VoiceView Express	Applications	This Cisco Unified Communications 500 Smart Productivity Application for Cisco IP Phones with a larger display allows you to interact with the voicemail system as an XML application; it is included without charge. You can listen to new messages play though the phone, forward, send a message, save, delete, set greetings, change password, and manage groups and notification options - all without listening to the voicemail system prompts.
View conference list	Voice-system features	With this View Participant IP Phone softkey option when participating on a conference call (ad hoc or meet-me), you can see all the users currently participating in the conference. (This feature is not supported with SIP-based phones or when using third-party conferencing).
Viewing angle settings	Users, phones, and extensions	With this feature of the Cisco Unified IP Phone 7940, 7960, and 7970 models, you can tilt your phone to change the viewing angle.

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Feature	Category	Description
Voice activity detection (VAD)	Voice-system features	This call-processing feature sends no packets when no audio is heard between endpoints, saving bandwidth and offering you a better quality-of-service (QoS) experience. Note that with Cisco solutions side tones are provided, so even with VAD enabled (by default) you still feel as if connected.
Voice messaging	Voice-system features	Whether you are an external or internal caller, you can record an audio message to be played back to the receiver at a later time. In most cases when calling a phone that is not answered or busy, you have the option to leave a voice message. When you leave a voice message, notification of the new message is made available to the person you called by a light on the phone, email notification, or by the system calling that person and playing a new-message prompt. Automated voice-messaging systems allow business users to receive and listen to messages without the need for an operator or message desk employees, saving costs.
Voice messaging, bilingual	Voice-system features	Refer to “Bilingual Voice Messaging.”
Voice messaging zero-out options	Voice-system features	Allows the user to customize how the call flow should proceed in response to keys pressed at the mailbox greeting for leaving a voicemail.
Voice mailbox PINless login	Voice-system features	This feature allows you to call in to your voicemail internally without entering a PIN or code. You should enable this feature only when voicemail security is not required.
Voice translation rules and profiles	Voice-system features	Refer to “Translation rules and profiles.”
Voicemail notification	Voicemail	You can enable notifying subscribers of new messages in their voice mailboxes. These notifications can be sent to phone or email.
Volume settings	Users, phones, and extensions	You can set the ringing, handset, and headset volume on your IP phone.
Cisco Unified Video Advantage	Users, phones, and extensions	Cisco Unified Video Advantage (formerly Cisco VT Advantage) adds video telephony features to Cisco Unified IP Phones (Cisco Unified IP Phone 7900 models plus Cisco IP Communicator softphone). Cisco Unified Video Advantage offers a Cisco Video Advantage camera (or third-party USB camera) plus Windows software. The camera is connected to the Windows PC, and the PC is connected to the Cisco Unified IP Phone 7900, which is then connected to the network. When you make a call from one video-enabled endpoint (within the site, or site to site over the WAN), video is displayed automatically.
Watch button	Users, phones, and extensions	You can configure a button on your IP phone or sidecar to show the status of another IP phone on the network; this button is also known as the busy-lamp-field (BLF) button. Watch status shows a phone idle or in use, no matter which directory number on the phone is in use. Refer also to “Busy lamp field.”
Cisco WebEx PhoneConnect	Applications	With this free application with Cisco Unified Communications 500, an XML application shows Cisco WebEx meetings you are either hosting or being invited to. In addition, a notification on the phone reminds you of upcoming Cisco WebEx meetings; a softkey call button autodials you in to the meeting.

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