Cisco Unified Communications Platforms for Small Business

Feature Description Guide

June 2011

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Introduction

This document lists and describes all of the Cisco® Unified Communications platforms for Small Business system features and capabilities.

Features and capabilities are listed by:

- Gateways and terminations
- Installation and configuration
- Localization
- Call control features
- Voicemail
- Automated attendant
- Hunt group
- Music on hold
- Phone
- Video
- Conferencing
- Fax
- Session Initiation Protocol (SIP) trunking
- Networking
- Wireless
- Security
- Firewall
- Teleworkers
- Multi-site
- Management and reporting
- Serviceability
- Basic automatic call distributor (ACD)
- Applications
- Third-party integration

For more information on the Cisco Unified Communications 300 Series for Small Business, go to www.cisco.com/go/uc300. For more information on the Cisco Unified Communications 500 Series for Small Business, go to www.cisco.com/go/uc500.

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Gateways and Terminations			
Foreign Exchange Office (FXO) ports	Support for FXO analog ports. FXO designates a telephone signaling interface that receives POTS, or "plain old telephone service."	4	4
Foreign Exchange Station (FXS) (analog) ports	Support for 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.	1	4
Session Initiation Protocol (SIP) trunk	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.	4	1
T1	ISDN line provides 23 simultaneous calls.		X
E1	ISDN line provides 30 simultaneous calls.		X
Basic Rate Interface (BRI)	Support for BRI. This protocol is used primarily in Europe. On the Cisco UC 320, BRI is supported using the Mediatrix Gateway.	X	X
Primary Rate Interface (PRI)	PRI is a high-capacity digital trunk used to carry voice and data between the service provider and customer. The Cisco UC 500 Series supports only voice PRI; shared data and voice circuits are not supported.		X
Installation and Configuration			
Automatic notification of firmware upgrade	Administrator is alerted automatically when a firmware upgrade is available for download. On the UC 540 and UC 560, Cisco Certification Agent (CCA) periodically checks for available CCA upgrades.	X	X
Automatic phone configuration (plug and dial)	Plug and play configuration for phones. Phones are automatically detected during the installation.	X	X
Automatically upgrade phone firmware	Phone firmware is automatically upgraded with the voice platform firmware upgrade.	X	X
Auto-registration (directory number and dial tone)	Automatic registration of phones with directory number and dial tone.	X	
Built-in dial plan (for instance, local, national, international)	Built-in dial plan for local, national, and international calling.	X	X
Bulk provisioning	Users and phones can be provisioned in bulk by uploading a spreadsheet or comma-separated value (CSV) file.		X
Easy replacement of phones	On phone malfunction, a return materials authorization (RMA) alert is sent to the manufacturer automatically.		
First-time setup wizard	A setup program guides users through a step-by-step installation and configuration process.	X	X
Plug and play compatibility with select Cisco Switches	Plug and play compatibility with select Cisco switches. For example, the Cisco ESW 500 Series switches are pre-configured with VLANs for seamless integration with the UC 320, UC 540, and UC 560.	X	X
Plug and play	Devices are discovered by the voice platform as they are connected to the system.	X	X
Remote access for management (methods)	A remote access method for configuring the voice platform for ongoing maintenance and management.	https	VPN
Localization			
Dual-language support for auto attendant	Administrator can set two languages for the auto attendant. The auto attendant provides incoming callers with the option to choose a language of their preference for the voice prompts.		X
Localized admin user interface (UI)	Administration utility is translated into the local language where product is sold	X	X
Localized documentation	Documentation is translated for the specific country where the product is sold.	X	X

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Localized phone UI	The phone UI is translated for the specific country where the product is sold.	X	X
Localized PSTN dial plan	The PSTN dial plan is customized for the specific country where the product is sold.	X	X
Support for multiple languages	Install multiple concurrent languages on the same voice platform. The IP phone display and voicemail prompts can be different on a per-user basis.	X	X
Call Control Features			
Feature access code (FAC)	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 soft key.	X	X
Account code entry	Enter account codes during call setup or when connected to an active call using the account soft key (a non-forced option). Account codes are inserted into call detail records (CDRs) on the voice platform for later interpretation by billing software.	X	Third-party application
Adaptive jitter buffer	Adaptive jitter buffer reorders User Datagram Protocol/Routing Transport Protocol (UDP/RTP) packets and adjusts the buffer size to maintain good quality and low latency.		X
Blocked numbers	Tables used to control a list of numbers or dial patterns that are not allowed to be dialed.		X
Blocking caller ID	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.	X	X
Built in router capability	Voice platform provides basic routing capability.	Optional	X
Busy timeout	Busy timeout sets the length of time after which calls that are transferred to busy destinations are disconnected.		X
Call admission control	Ability to cap inter-site calls to maintain quality of service (QoS).		Command-line interface (CLI)
Call blast	Calls simultaneously ring multiple phones.	X	X
Call forwarding–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.	X	X
Call forwarding–all calls	All incoming calls are diverted based on destination configured by the administrator or entered by a user.	X	X
Call forwarding–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.	X	X
Call forwarding–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.	X	X
Call hold	By default, the system allows you to place a call on hold using the hold soft key on any active call.	X	X
Differentiated services code point (DSCP)	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.	X	X
Echo cancellation (G.165/G.168)	This feature removes echoes from the voice stream for clear call quality.	X	X

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Extension mobility	Extension mobility offers the benefit of phone mobility. A user login service allows you to temporarily access a physical phone other than your own phone and use your personal settings, such as directory number, speed-dial lists, and services, as if the phone is your own desk phone. You can make and receive calls on that phone using the same personal directory number as on your own desk phone.		X
Feature control	The administrator can control which features a phone user has access to by hiding or exposing specific soft keys. Soft key		X
Frame loss concealment	This feature provides an error-correction algorithm to mask the effect of packet loss in VoIP.		X
G.711	G.711 is a high bit rate (64 kbps) standard codec. It is the industry standard codec for voice.		X
G.722	G.722 is a Wideband codec for high-definition voice.		
G.726	G.726 is a speech codec standard covering the transmission of voice at rates of 16, 24, 32, and 40 kbps.		
G.729	G.729 is a low-bandwidth codec used for carrying voice over low-bandwidth networks.		X
Key system	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 so you can put a call on hold on one phone and resume the call on another phone without invoking call transfer.	X	X
Manual backup and restore	Back up voicemail	X	X
Media encryption (SRTP)	Media encryption (SRTP) encrypts the voice media for calls made between IP phones registered to the same Cisco Unified Communications system without the need for a dedicated VPN tunnel.		X
Network Time Protocol (NTP) reference	The Cisco Unified Communications system needs the current time and date to show on IP phones, and time stamp for voice messages. NTP reference service allows the application to receive the current time and date from an external server, which allows all devices in a network to stay synchronized.	X	X
Online help	Online help is a detailed, transparent help function embedded in Cisco configuration tools. It provides an extensive glossary and a powerful search engine that helps you quickly and easily find the information you need to apply specific settings. With these online help features, you can often troubleshoot and resolve problems without calling technical support.	X	X
Outgoing call restrictions	The class-of-restrictions (COR) feature allows you to deny 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.		X
Park	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.	X	X

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Private branch exchange (PBX) system	When setting up a Cisco Unified Communications system, you need to decide if call handling should be similar to that of a PBX, a keyswitch, or a hybrid of both. The Cisco Unified Communications system 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.	X	X
Privacy	With shared lines set up with barge or cBarge, you can block users from barging into an active (non-hold) call unless privacy is off.		X
PSTN failover	Automatically route calls over the PSTN network if the IP network is down.	X	X
Restore to factory defaults	This operation restores the system to its factory default settings. It resets all the configuration parameters, as well as any database maintained by the system.	X	X
Same call-control agent for analog phones and IP phones	Configure analog voice ports (FXS) by the call-control agent using the same protocol that is used to communicate with IP phones. You can configure the analog phones with most of the features available for IP phones, including voicemail.	X	X
Support for electronic business card (vCard) information from remote subscribers	Allow electronic business card (vCard) information from remote subscribers to update their directory entries.		X
T1 and E1 digital trunk interface	PBX support for direct digital trunk interfaces, such as ISDN PRI and T1 or E1 CAS circuits.		X
Time zones	Support for time zones allows the correct time and date information to be displayed on IP phones and played back on the voicemail.	X	X
Toll restrictions	This feature allows you to prevent specified phones from making long distance (toll) calls.		X
Trace (debugging)	Real-time troubleshooting tool for on-demand debugging.	X	X
Transcoder resources	This feature provides translation of RTP streams from one codec format into another.		X
Translation rules and profiles	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 (Digital Nunber Identification Service [DNIS]) • Calling numbers (asynchronous network interface [ANI]) • Redirected called numbers The Cisco Unified Communications 500 Series 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.		X
Trunk groups	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.		X
Trunk-to-trunk connections	Connect outside callers to another PSTN trunk when a phone is set to forward (or signal-to-noise ration [SNR]) or transferred by a user.		X
Voice activity detection (VAD)	When the system detects no audio between endpoints, it does not send packets. This saves bandwidth and offers a better QoS experience.		X

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Voice messaging	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.	X	X
Voicemail			
Minutes stored per message (maximum)	Refers to the storage capacity for each voicemail message.	10 minutes	Variable (default 775 seconds)
Message waiting indicator (MWI)	MWI is a signal that a voicemail is waiting. This is usually displayed in the form of a visual flashing light on the phone.	X	X
New voicemail subscriber feature	When first calling into the system, new voicemail users are given a mini-tutorial on use, recording their voice name, recording a greeting, and setting a PIN.	X	X
Pre-recorded generic voice mail prompts	A set of pre-recorded voicemail greetings are available.	X	X
Remote access to voicemail	Access the voicemail system remotely by dialing into the main number.	X	X
Speech to text	Voice transcription service to convert voice recording into text.		
Support for caller ID information for voicemail messages	Play back the caller ID number of the caller who recorded the message when you listen to new voicemail messages.	X	X
Transfer to voicemail	Transfer to voicemail allows you or the automated 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.	X	X
Undelete voicemail messages	During a voicemail session you can restore a deleted voicemail message for playback.		X
Virtual mailboxes	Group voice mail boxes are available.	X	X
Visual voicemail	The visual voicemail application is a Cisco Unified Communications Widget that delivers rich messaging experience on Cisco Unified IP Phones. You can view, listen, and respond to Cisco Unity® and Cisco Unity Connection messages right from the Cisco Unified IP Phone display without having to dial into your corporate voicemail box. On the UC 540 and UC 560 this is called the Voice View Express.		X
Voicemail message storage capacity	Voice mail storage capacity.	12 hours	32 hours On the UC 560, it can be extended to 142 hours.
Voice commands	Use voice commands to interact with the phone system instead of using buttons on your phone. For example, respond to auto-attendant voice prompts to reach an extension.		
Voice dialing (Speech Connect)	Speech Connect is a speech-enabled automated attendant (SEAA) for the enterprise that is available in installations that include a Cisco Unity or Cisco Unity Connection system. Internal or external callers speak the name of an employee into the phone and are connected to that employee.		
Voice mailbox PIN-less login	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.		X
Voicemail to email and unified messaging (wave file)	With the voicemail-to-email feature, you can receive your voicemail messages as a .wav email attachment sent to your email account.	X	X (IMAP)
Auto Attendant			
Auto-attendant support for business and holiday schedules	The automated attendant offers holiday and business schedules to enable time-of-the-day routing of incoming calls. On holidays, the auto attendant plays the closed greeting.	X	X

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Auto-attendant answer delay	Configure the number of rings before the auto attendant intercepts the call.		
Auto-attendant levels	The Multi-level Auto Attendant mode allows you to set up a hierarchical menu structure that provides callers with better self-service access to the department or person they are trying to reach.	Two nested levels	Three nested levels
Auto-attendants schedule (day, time)	You can configure separate prompts and actions for business hours and night hours.	X	X
Automated attendant	A call processing system for answering telephone calls and for helping to direct a caller to a requested party without the assistance of an operator. It is navigated using touch-tone entries from your phone. The automated attendant can have different options based on time of day and holiday schedules. It can also redirect to another menu layer in the automated attendant.	X	X
Fixed holidays	The automated-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 automated attendant plays a greeting that indicates the business is closed.	X	X
Night service	You can configure a different set of phones to ring when night service is enabled.	X	X
Number of automated-attendant ports	Number of simultaneous calls you can make to the automated-attendant.	Eight	Six for UC 540 12 of UC 560
Number of automated attendants	The number of individual automated attendants.	Two	Three
Pre-recorded generic automated attendant	A pre-recorded generic automated attendant bundled with the voice platform.	X	X
Hunt Group			
Blast hunt group	Incoming calls simultaneously ring multiple phones or multiple destinations. Any of the phones in the group can answer the incoming call.	X	X
Dynamic hunt group membership	An authorized agent can dynamically join and leave hunt groups. For example, you can choose to participate in a hunt group and receive incoming calls that are bound for that hunt group.		
Hunt group agent status control	Manually activate a toggle to temporarily enter a not-ready state in which hunt-group calls bypass the agent's phone.		X
Hunt group announcements	Ability for a supervisor to make an announcement to a hunt group. On the UC 540 and UC 560 this is available only with basic ACD (limit of 10 ACD hunt groups per system).		X
Hunt group automatic-agent status to ready	Put an agent's phone in a not-ready state after a specified number of hunt-group calls are unanswered by the agent's phone. On the UC 540 and 560, this is available only with basic ACD.		X
Hunt group call queuing	Queue calls in the hunt group for next available agent.		X
Hunt groups	Hunt groups allow you to direct incoming calls from a specific number (pilot number) to a defined group of extension numbers.	X	X
Hunt group login	Users can log in and log out of the hunt group. When you log in the extension becomes available to accept calls coming into the hunt group.		X
Longest-idle ringing	New calls to this hunt-group type ring the directory number that has been idle for the longest duration.	X	X
Night service group	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 the week and hours per day.	X	X
Out-of-service group	This feature puts a specified hunt group into out-of-service or night service mode.		X

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Overflow group	Calls that ring the hunt group members without being answered can be redirected to an overflow group. In the UC 540 and 560 you can use the Call forward no answer (CFNA) feature to address this functionality.	X	X
Queue threshold alert	When the number of calls queued against a hunt group exceed a threshold, the system can be configured to alert at a selected extension.		
Circular ringing	Phones that participate in a hunt group ring in a circular order picking up from the last phone that did not ring	X	X
Sequential ringing	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.	X	X
Music on Hold			
WAV, MP3, music-on-hold (MOH) source	Outside callers placed on hold will hear recorded music. The source of the music is a .wav file saved to local flash memory.		X
Default MOH source	Outside callers placed on hold will hear pre-recorded music set by the manufacturer.	X	X
Network-based MOH source	Outside callers placed on hold will hear recorded music. The music is streamed from the network.		
MOH multicast	With multicast music on hold, the music stream is broadcasted to all nodes, but only callers placed on hold hear the music. This helps save bandwidth on the network.		X
External MOH through an audio player	You can connect an MP3 player or an iPod using a 3.5 inch jack and stream it as the MOH source for the voice platform.	X	X
Phone			
Direct station select (DSS)	DSS allows a multi-button 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.	X	X
Abbreviated dialing speed dial	Dial a phone number by entering an assigned index code 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.	X	X
Administration using the telephone	Access the greeting management system (GMS) for recording alternate greetings and prompts through the phone (requires administrator privileges). For example, administrators can change automated-attendant greetings remotely for bad weather days.	X	X
Analog phone support	You can add analog phones for fax machines or other devices by connecting them to the built-in analog station (FXS) ports on the voice platform. On the UC 320, users can connect additional analog phones using a Cisco SPA 8800 IP Telephony Gateway with 4 FXS and 4 FXO Ports. On the UC 540 and 560, users can add modular virtual interface cards (VIC) and virtual WAN interface cards (VWICs) to connect additional analog phones.	X	X
Attendant Console	Attendant Console is an application that supports the traditional role of a manual attendant-console hardware device. Associated with an IP phone, the application allows the attendant to quickly accept and dispatch calls to enterprise users. An integrated directory service provides traditional busy-lamp-field (BLF) and DSS functions for any line in the system. On the UC 540 and 560, this is supported using Smart Call Connector Operator Console.		X
Audio paging	Announce an audio message to phones that are designated to receive paging. When a caller dials the paging number, each idle 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.	X	X

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Auto-answer with headset and speaker	Configure lines on specific phones to automatically connect to incoming calls when the headset key is activated. The phone cannot be busy with an active call, and the headset key must be engaged to automatically answer calls. Incoming calls are automatically answered one by one on the phone as long as the headset light remains lit. This feature is available in Cisco Unified Communications Manager Express 4.0 and later versions.		X
Automatic line selection	Picking up the handset answers the first ringing line or, if no line is ringing, selects the first idle line.	X	X
Barge (cBarge)	The barge feature enables phone users who share a directory number to join an active call on the shared line by pressing a soft key on supported phone models.		X
Barge privacy	This feature allows you to prevent anyone from barging into your calls (on supported phone models).		X
Busy lamp field (BLF)	This feature provides a visible phone line status indicating whether or not the line is in use. 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.	X	X
Call history	You can view the list of missed calls, placed calls, and received calls from the phone user interface.	X	X
Call logs	You can view records of your placed, received and missed calls on your phone display.	X	X
Call park	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.	X	X
Call transfer	Call transfer allows you to transfer the current active call on your phone to a different destination.	X	X
Call transfer–blind	Call transfers can be blind, where the transferring extension connects you to the target destination before the target phone rings.	X	X
Call transfer–consultative	On a consultative call transfer, the transferring extension either connects you to a ringing phone or speaks with the target destination before connecting you to the target.	X	X
Call transfer recall	This feature returns a transferred call to the phone that initiated the transfer if the target destination is busy or does not answer.		
Call waiting	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 will hear an audible call-waiting tone such as beep or ring, and will also visually see the calling-party information on your phone screen.	X	X
Called-name display	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.	X	X
Caller ID blocking	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.	X	X
Call waiting beep	The audible indication for call waiting on the system.	X	X
CFNA timer	Customize the time-out period associated with no answer on an incoming call.	X	X
Configuration of essential network parameters using the phone	Network parameters that are required for the phone to operate in an IP network can be configured from the phone user interface.	X	X
Customizable logos and screensavers	Display a JPEG image as a logo or screensaver on the phone’s high-resolution display.	X	X

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Directed call pickup	Any local phone user can pick up a ringing call on another phone by pressing a soft key and then dialing the extension. You do not need to belong to a pick-up group to use this method. The soft key that you press, either GPickUp or PickUp, depends on your configuration.	X	X
Distinctive ring	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 system or is routed through the local FXS port.		X
Do not disturb (DND)	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. You can answer the call if desired.	X	X
Door phone	A door phone is a system that enables you to receive calls from your front door or some other door on the outside of an office building. When someone arrives at an outside door, they press a button that alerts one or multiple inside stations and then the people inside the building can talk to them by pressing a button on an inside station. On the UC 540 and 560, this feature is supported by an external third-party application.		X
External paging	Announce an audio message to the speaker system that is designated to receive paging. When a caller dials the paging number, the announcement is heard on the speaker system. On the UC 540 and 560, this feature is supported by an external third-party application.	X	X
Feature ring	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.		X
Flash soft key and FXO hookflash	The Flash soft key 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 soft key provides this function for IP phones.		X
Group call pickup	Answer a ringing phone in a different pick-up group by pressing the GPickUp soft key and then dialing the pickup group number.	X	X
Group voicemail notification	Notification to indicate message waiting in a group mailbox. This can be mapped to any phone button. In the UC 540 and 560 shared lines have group voicemail notification.	X	X
Handset paging	Audio-paging provides a one-to-many voice announcement 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.	X	X
Hold	When a call is in progress, you can use the hold soft key to place the call on hold. The person on the other end typically hears music while on hold. Pressing resume reconnects you to the caller.	X	X
Hold notification	The on-hold indicator is an optional feature that generates a ring burst on idle IP phones that have placed a call on hold. An option is available to generate call waiting beeps for occupied phones that have placed calls on hold. This feature is disabled by default.	X	X
Incoming caller ID mapping	With caller ID mapping you can specify how to map a caller ID number for display on screen and recorded into call logs.		X

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Intercom lines	An intercom line is a dedicated two-way audio path between two phones. 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 sounds when the call is auto-answered 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.	X	X
International languages and tones	Support for international languages for text displays and the country-specific tones and cadences required for connection to the local telephone network.	X	X
IP phone password setting	You can change your phone password from your user options webpage. Phone passwords are used for computer telephony integration/Telephony Application Programming Interface (CTI/TAPI) integrations, extension mobility, toll bar override, and user page login.	X	X
Key system mode (tie line to trunk)	In this mode the Cisco Unified Communications system will emulate a squared key system (for example, direct access to trunk, line status, intercom, and more).	X	X
Line status indication	Line status indication displays whether the monitored line is in use, idle, ringing, or in a do-not-disturb state.	X	X
Local directory	The local directory lists all internal users with extension number by name. You can also add additional entries to the directory.	X	X
Multiple lines per phone	Multi-button IP phones offer the option to support more than one directory number per phone. The Cisco Unified Communications 500 Series also supports up to 10 directory numbers on a per-button basis using overlays.	X	X
Multiple ring tones	You can define multiple tones for shared point adapter (SPA) phones.	X	X
Number of phones–base license	The number of phones supported by the base license.	24	Eight for UC 540; 16 for UC 560
Maximum number of phones with extended license	The maximum number of phones that can be supported on the platform.	24	32 for UC 540; 104 for UC 560
Off-premises extension	Supports remote teleworker phones and off-premises phones. The phones can be IP-based using VPN or analog using telco-provided wiring.		X
On-hold notification	Also known as hold reminder, this feature sends a reminder ring burst to an IP phone that has placed a call on hold after a defined number of seconds.		X
On-hook dialing	Dial a number (internal or external) without going off hook on the IP phone.	X	X
On-hook transfer	The call transfer feature supports the on-hook (hang-up) 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.		X
Overhead paging	Paging over a loud speaker system.	X	X

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Paging	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 other 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).	X	X
Personal address book (PAB)	PAB is a directory for personal contacts. It is stored locally on the Cisco SPA525G 5-line IP Phone with Color Display.	X	X
Personal directory	Store contact information on the personal address book stored in the Cisco Call Manager LDAP directory. You can synchronize your Outlook and Outlook Express Contacts with the personal directory.		X
Personal speed dial	With this feature, you can configure a maximum of 24 personal speed-dial numbers per phone. The numbers can be accessed through the directory, personal speed dial listing. Each phone can store up to 99 fast dials. You can configure each phone using the options webpage or through the phone using services, MyPhoneApps, speed dial.	X	X
Phone button customization	Customize the button layout on Cisco SPA phones.	X	X
Phone display	Phone display features caller name and system messages in the lower part of the display window on display-capable IP phones.	X	X
Phone header bar display	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.	X	X
Phone labels	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. This allows phone users with multiple lines to select a line by label instead of by phone number, thus eliminating the need to consult in-house phone directories.	X	X
Phone lock (Cisco Unified Wireless IP Phone 7921G and 7925G phone models)	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.		X
Photo album	With the Cisco Unified Communications System, it is possible to display photographs or images on the high-resolution display of selected phones. This feature can be used to display advertising on a lobby phone, or to display photographs from the latest staff outing.	X	X
PIN setting	This feature sets a PIN to be used by a phone user to access voicemail.	X	X

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Placed call list	You can view records of your placed calls. While viewing call logs, you can use soft keys 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, end call) for handling the first call before placing the second call.	X	X
Pre-dial	You can enter a phone number before getting a dial tone and complete the call by going off hook (lifting the handset or pressing the speakerphone button).	X	X
Private-line automated ring-down (PLAR)	Administrators can set up an analog or IP phone to automatically ring a predetermined number, internal or external, when the phones goes off hook. For example, a phone placed inside an elevator would automatically ring the receptionist when the phone goes off hook.		
Programmable buttons	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.	X	X
Redial	You can automatically dial the last call that you placed from your IP phone.	X	X
Resume	This soft key 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 connection is established between the users.	X	X
Ringer settings	This setting selects the ring tone on an IP phone. You can set ringers on a per-directory number basis.	X	X
Ringing-line preference	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.	X	X
Ring tone setting	You can select the ring tone on your IP phone.	X	X
Shared lines	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.	X	X
Silent ringing	You can suppress audible rings and the call waiting beep for incoming calls on a Cisco IP phone.	X	
Single number reach (SNR)	SNR allows you to have incoming calls to a specific number simultaneously ring an office, home. or mobile phone.		X
Soft keys	Soft keys are keys that appear on the bottom of the IP phone liquid crystal display (LCD). They allow you to access various features such as call forward, call transfer, conferencing, and call park. The soft keys available for use change dynamically according to whether the phone is in connected, ringing, idle, or seized (handset is lifted) mode.	X	X
Speakerphone mode	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.	X	X
Speed dialing	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.	X	X
Station volume controls	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.	X	X

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Support for Cisco Unified IP Phones 6900 Series	Support for Cisco Unified IP Phones 6900 Series. (Support for Cisco SPA6945 is not available on the UC 540 or UC 560.)		X
Support for Cisco Unified IP Phones 7900 Series	Support for Cisco Unified IP Phones 7900 Series.		X
Support for Cisco SPA300 Series IP Phones	Support for Cisco SPA 300 Series IP Phones.	X	X
Support for Cisco SPA 500 Series IP Phones	Support for Cisco SPA 500 Series IP Phones.	X	X
System message display	Specify a custom text or display message to appear in the lower part of the display window on your IP phone.		X
Time of day routing	Configure day or night schedule for routing calls.	X	X
Touchscreen	On IP Phones with touch-sensitive phone screens, you can press the phone screen to choose menu items, soft keys, and feature tabs.		X
Transcoding	Ability to convert the voice stream from one codec to another. This is useful for multi-site deployments.		X
Trunks mapped to phone buttons	You can configure IP phones running Skinny Client Control Protocol (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.	X	X
Types of phones supported	Support for IP, digital, or analog phones.	IP and analog	IP and analog
User downloadable ringtones	You can download ringtones for your IP phone.	X	X
Video streaming	Stream a live video feed from a Cisco PVC2300 Business Internet Video or Cisco WVC2300 Wireless-G Business Internet Video Cameras to the display of a Cisco SPA525G2 5-line IP phone.		X
Video support	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.		X
View conference list	You can see all the users currently participating in a conference call by using the view participant IP phone soft key option. (This feature is not supported with SIP-based phones or when using third-party conferencing).		X
Viewing angle settings	With this feature of the Cisco Unified IP Phone 7940, 7960, and 7970 models, you can tilt your phone to change the viewing angle.		X
VoiceView Express	Smart Business Productivity Application for Cisco IP phones with a larger display allows you to interact with the voicemail system. You can listen to new messages, forward, send, save, or delete messages, set greetings, change your password, and manage groups and notification options–all without listening to the voicemail system prompts.		X
Volume settings	You can set the ringing, handset, and headset volume on your IP phone.	X	X
Whisper	Ability to intercom a party when they are on another line. This feature requires an Octal line.		X

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Whisper intercom	Whisper intercom, a feature for supervisors and administrators, allows administrators to place a one-way intercom call to an in-use supervisor's phone to notify the supervisor of another call or other important information. The outside party on the supervisor's phone does not hear the administrator's voice, and the supervisor cannot reply without placing the caller on hold. This feature requires an Octal line.		X
Wireless IP phones (digital enhanced cordless communication [DECT])	Support for DECT cordless telephones.		X
Wireless IP hones (WiFi)	Support for WiFi-enabled phones. The UC 320 supports only the Cisco SPA525G phones. The UC 540 and 560 supports\ the Cisco SPA525G2 and the 7900 Series wireless phones.	X	X
Video			
Video surveillance	You can connect the Cisco PVC2300 Business Internet Video Camera to the Cisco Unified Communication System to access workplace video.		X
Point-to-point video telephony	With this capability, you can interact and collaborate point to point using video conferencing over the network.		X
Conferencing			
Unplanned conferencing	Create a conference with the "Confrn IP phone" soft key or access code. Conference calls can include other IP phones, analog phones, or external calls through SIP or PSTN trunks. (Note: Unplanned conferencing is often referred to as "ad hoc" conferencing)	X	X
Conference gain control	Gain control on conference calls keeps the volume for callers at the same volume for ease of use.		X
Conferencing	Conferencing allows you to connect three or more parties in a telephone conversation.	X	X
Hardware-based conferencing	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 unplanned conferencing (with a maximum of eight parties) uses digital signal processors (DSPs) to allow more parties than software-based unplanned conferencing, which allows three 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.		X
Meet-me conference	A meet-me conference is a voice conference bridge initiated by one IP phone user, where other users join by dialing an internal or external number. This is useful when conferencing more than two users.		X
Fax			
Automatic fax detection on automated attendant	The automated-attendant default system prompts can detect whether an incoming call is a voice call or fax.		X
Fax	The fax feature offers fax-machine support using analog (FXS) lines.	X	X
Fax–G.711	G.711 fax passthrough makes it possible to send IP faxes through G.711-compliant networks.	X	X
Fax relay	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 comprehensive transport of the facsimile data.	X	X

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Fax to email support	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.		X
T.37 fax server	The T.37 fax server converts incoming fax calls to an email message with a TIFF attachment. The fax email message and attachment are stored in the voice platform and can be retrieved using an IMAP email client such as Microsoft Outlook. The server can also convert an email message with a TIFF attachment into a traditional fax format that can be delivered to a standard fax machine or the PSTN.		X
T.38 fax support	Support for T.38 for fax codec when using SIP gateways.		
SIP Trunking			
Alternate route selection–backup	Ability to use an alternate backup service provider if the default is unavailable. On the UC 540 and 560 you can set alternate PSTN or SIP as backup routes.	X	X
Alternate route selection–time of day	Ability to select alternate SIP service based on time of day.		
Direct inward dialing (DID)	Support for DID on SIP trunks will allow the assignment of unique telephone numbers to individuals within the organization to efficiently route calls directly to their destination without having to first go through a receptionist or auto attendant.	X	X
Maximum number of simultaneous calls per trunk	Maximum number of simultaneous sessions per SIP trunk.	8	unlimited (only bandwidth constrained)
Network Address Translation (NAT) support–NAT keepalive	NAT Traversal (NAT-T) method.	X	
NAT support–static mapping	NAT-T method	X	
NAT support–serial tunnel (STUN)	NAT-T method	X	
NAT support– traversal using relay NAT (TURN)	NAT-T method		
Networking			
Back up to PSTN on power failure	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.	X	X
Dynamic Host Configuration Protocol (DHCP) Option 66 for Trivial File Transfer Protocol (TFTP) server address	This feature enables ready-to-use capability on the devices.	X	X
Differentiated Services (DiffServ) and type of service (ToS) packet marking	DiffServ is a new model in which traffic is treated by intermediate systems with relative priorities based on the ToS field. DiffServ increases the number of definable priority levels by reallocating bits of an IP packet for priority marking.	X	X
Domain Name System Server (DNS)SRV) for proxy redundancy	SIP trunk redundancy	X	
Multiple A records for proxy redundancy	SIP trunk redundancy	X	
Network Time Protocol	You can use this protocol to get the local time from the network.	X	X
Point-to-Point Protocol over Asynchronous Transfer Mode (PPPoA)	PPPoA		

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Point-to-Point Protocol over Ethernet (PPPoE) client	PPPoE	X	X
Real-time control protocol (RTCP)	RTCP used to calculate Microsoft Office SharePoint Server (MOSS) score for QoS.		X
Support for Cisco Catalyst switches	Support for Cisco Catalyst switches.		
Support for Cisco Small Business Switches	Support for Cisco Small Business ESW 500 Series Switches.	X	X
VLAN ID and priority tagging (802.1Q/p)	This feature enables the platform to separate voice and data networks, and allows the data to be sent through the same cable.	X	X
DHCP options- configuration and relay	When a Cisco Unified IP phone is connected to the Cisco Unified Communications System, it automatically queries for a DHCP 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 TFTP server through DHCP option 150. Then the phone registers with the Cisco Unified Communications System and attempts to get configuration and phone firmware files from the TFTP server.	X	X
Internet Group Management Protocol (IGMP) proxy	IGMP proxy enables the system to issue IGMP host messages on behalf of hosts that the system discovered through standard IGMP interfaces. The system acts as a proxy for its hosts.		X
Multilink Point-to-Point Protocol (MLPPP)	MLPPP is a communications protocol that enables a personal computer to use two Point-to-Point Protocol (PPP) communications ports as if they were a single port of greater bandwidth.		X
NAT support one-to-one mapping	NAT is a one-to-one static port mapping for TCP and User Datagram Protocol (UDP) ports.	X	X
NAT Support 1-many mapping	Network Address Translation for 1-to-many static port mapping for TCP and User Datagram Protocol (UDP) ports.		X
Password Authentication Protocol (PAP) and Challenge Handshake Authentication Protocol (CHAP)	Support for PAP and CHAP.		X
PPP	PPP is a data link protocol commonly used in establishing a direct connection between two networking nodes. It can provide connection authentication, transmission encryption privacy, and compression. Most Internet service providers (ISPs) use PPP for customer dial-up access to the Internet.		X
QoS and Bandwidth shaping	This feature allows the voice platform to smooth the flow of data traffic in the network to remove spikes and troughs in bandwidth utilization. This ensures that the network does not impact voice quality.	X	X
Routing Information Protocol (RIP) version 2	RIP is a dynamic routing protocol used in LANs and WANs.		X
Static routing	Routing between networks is set up manually by entering routes directly into the routing table of a router. Static routing has the advantage of being predictable and simple to set up. It is easy to manage in small networks but does not scale well.		X
VPN Client	VPN client allows the user to make a secure connection to the enterprise network.		SSL, IP security (IPsec)
Wireless			
Wireless–802.11	802.11 is a set of standards for implementing wireless local area network.	X	X
Wireless–WiFi Protected Access 2 (WPA2)	WPA2 is a security standard that provides stronger data protection and network access control.	X	X

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Security			
Device authentication	Phones or other devices need to register with the voice platform.	X	X
Encrypting stored PINs	Voicemail PIN codes are stored in encrypted form for security reasons.		X
Password and PIN security protection	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.		X
Password-protected factory reset	Factory reset requires administration password authentication.		X
Password-protected voicemail	Access to voicemail requires user password authentication.	X	X
Password-protected web UI access	Access to the web UI requires administration password authentication.	X	X
Phone lock	You can secure access to your Cisco Unified wireless IP phone 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. In the UC 540 and UC 560, this can be enabled with Extension Mobility.		X
Secure remote provisioning	Access to configuration UI using https.	X	X
Secure Socket Layer (SSL) phone client	The SSL phone client on the SPA 525G IP phone provides secure connectivity to the UC voice platform or the SR 500 Series Secure Router over the Internet.		X
SIP authentication support	Registration and authentication with the SIP trunk service provider.	X	X
SIP over Transport Layer Security (TLS)	SIP over TLS enables a SIP session to be encrypted on a hop-by-hop basis between the source and destination, providing better security than basic Message Digest Algorithm 5 (MD5) authentication.		X
sRTP	Secure Real-Time Transfer protocol.		
VPN	Support for VPN for secure communication over the network. This enables a remote user to virtually be part of the local network.		X
VPN devices (integrated and external)	VPN functionality is embedded onto the platform or requires additional components.		Integrated
Access control list (ACL)	ACL allows you to open or close ports on your network to maintain security. In UC 540 and 560, ACLs are adjusted automatically based on configuration made with CCA.	Basic	X
Teleworker			
Number of teleworkers	Total number of teleworkers (remote yet virtually local workers) supported.		10 teleworkers for UC 540 20 teleworkers for UC 560
Teleworker VPN	VPN support for teleworkers		X
Multi-Site			
Multi-site deployment	The voice platform can be deployed at multiple sites, each connected the others through a WAN. This type of deployment allows for toll-free calling between the sites.		X
Number of sites	Total number of sites supported for multi-site deployment.		Five
Management and Reporting			
Asynchronous notification of update and upgrade availability	Firmware update notifications are sent to the voice platform for upgrade.	X	
Billing records	Collect accounting data for each call event created on the voice platform. You can use this information for activities such as generating billing records and network analysis. A third-party application may be necessary.	X	X

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Congestion indication on phones	Phones will display packet drop, jitter, and latency when there is congestion on the network.		
IP camera portal	Automatically discover Cisco PVC2300 Business Internet Video Camera and Cisco WVC 2300 Internet Video Camera, and view video from the camera's webpage.		X
Measure Mean Opinion Scores (MOS) for QoS monitoring on Phones	Reporting on the measure of voice quality on the phones.		X
Remote Firmware Upgrades	Upgrade firmware by remotely connecting to the Configuration utility. On the UC 540 and UC 560, you can remotely upgrade the firmware for all devices on the network.	X	X
Remote monitoring	Monitor the status of the voice platform remotely.		X
Remote provisioning	Add users and phones remotely using the configuration utility.	X	X
Report generation and event logging	Generate reports on critical system events.		X
Report of performance indicators	Generate reports on performance parameters.		X
Simple Network Management Protocol (SNMP)	Monitor the status of the voice platform remotely via SNMP		X
Syslog and debug records	Syslog and debug records for troubleshooting.	X	X
TR 069/111/104	This is a service provider management protocol for remotely provisioning the voice platform.		
Web UI	Browser-based UI for configuration and maintenance.	X	
Serviceability			
Backup and restore	Back up configuration, voicemail, and automated attendant prompts to USB, PC, or another storage device.	X	X (Local)
Call detail records (CDR) offloading	Offload CDRs onto another server.		X
Health summary	Administration UI provides a summary of CPU performance, trunks, usage statistics, and more.	X	X
Single button log collection	Use a single button to trigger log collection.		X
SNMP alerts and traps	An SNMP trap allows for a device on the network to contact the network management when there is a significant event. This is done using unsolicited SNMP messages.		X
Upgrade rollback	Roll back to a prior release of firmware following an upgrade.		X
UI for monitoring	Includes a UI for monitoring system performance statistics.		
Basic-Automatic Call Distribution (B-ACD)			
ACD call monitor	The ACD call monitor feature allows agents to monitor the number of calls in the queue. Agent phones have flashing message-waiting indication (MWI) LEDs and text displays that indicate the number of calls in the queue.		X
ACD login	Users can log in and log out of their ACD group.		X
ACD time-based logoff	Ability to log off users at a preset time based on users' working hours. On the UC 540 and 560, you can set a number of missed calls after which the agent will be automatically logged off.		X
Acquire call	Ability to acquire a call in the queue that is not bound for a user's extension.		

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Basic ACD	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.		X
Applications			
Location (extension to emergency location identification number [ELIN] mapping)	Ability for 911 dispatcher to be able to ring back to the extension where the 911 call originated.		X
Cisco Unified Video Advantage	Cisco Unified Video Advantage (formerly Cisco VT Advantage) adds video telephony features to Cisco Unified IP phones (Cisco Unified IP Phone 7900 models and the Cisco IP Communicator softphone). Cisco Unified Video Advantage offers a Cisco Video Advantage camera (or third-party USB camera) with Windows software. The camera is connected to the Windows PC, and the PC is connected to the Cisco Unified IP Phone 7900 Series, 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.		X
Click to dial application	Click to dial allows you to easily place calls from your PC using any supported phone device. Example applications include Webdialer, SCC, and instant messaging clients (MOC, CUPC, WebBx, and UCC).		X
Directory integration	Integration of the voice platform with the corporate directory to enable user lookups (sometimes called the "white pages" service) from IP phones or other voice and video endpoints. This allows users to dial contacts quickly after looking up their numbers in the directory.		X
Instant messaging	Instant messaging offers a real-time text-based communication between two or more people for enhanced productivity. You can send instant messages to your colleagues at work and to external contacts on compatible IM servers. This feature requires the Cisco Smart CallConnector Advanced Client.		X
Integrated messaging	Access and manage voice messages using email clients such as Microsoft Outlook and Outlook Express. Integrated messaging brings voice, fax, and email together at the email client.	X	X
Mobile voice access and direct inward system access	This feature allows a user to make a call from a mobile or home phone to an outside number and make it appear as if the call is originating from the user's office phone. The benefits of this feature are: limiting toll charges to the enterprise, and masking the caller ID of the mobile phone.		
PC notification–incoming call	A pop-up notification window provides information on incoming calls, with caller names from the directory and with buttons to answer a call or route it to voicemail. This feature requires the Cisco Smart CallConnector Advanced Client application.		X
PC operator console	Windows-based operator console application designed specifically for the calling, messaging, and contact management requirements of small business attendant positions.		X
Phone conversation recording	Record phone conversation for monitoring customer satisfaction or for legal reasons. This feature requires a third-party application. In the UC 540 and 560, this is supported using the live record feature.		X
Recording to email	Ability to record a phone conversation and send to email. In the UC 540 and 560, this is supported using the live record feature and IMAP support.		X
Timecard view	Timecard view is an application that automatically tracks an employee's working hours and allows supervisors to see the employee's real-time status on their IP phone.		X

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Feature Name	Feature Description	Unified Communications (UC) 320W 1.0	Unified Communications (UC) 540 and 560 with Cisco Configuration Agent (CCA) 3.0
Single number reach (SNR)	The SNR feature allows users to answer incoming calls to their extension on either their desktop IP phone or at a remote destination, such as a mobile phone. Users can pick up active calls on the desktop phone or the remote phone without losing the connection. This enables callers to dial a single number to reach the phone user. Calls that are not answered can be forwarded to voice mail.		X
SoftPhone	A PC-based phone client application that is supported on the system. It typically acts as a remote teleworker phone for users at remote sites.		X
Speech recognition for dialing	You can use speech recognition with a corporate directory (or personal address book) to dial a telephone number by saying the name of the user.		
System directory	Includes a contact list for employees within a company or system that can be accessed from the desk phone.	X	X
Third-party integration			
CRM (Microsoft and Sales Force) integration	This feature provides the ability to invoke dialing from SalesForce.com or another CRM suite.		X
Computer telephony integration (CTI) and Telephony Application Programming Interface (TAPI_ for Third-party integration	Available software development kit for integration with third-party applications.		X
Mobile clients (iPhone, Blackberry, Android, and others)	Available client application for mobile devices like the iPhone or Android phone.		
Phone services (XML and Java on phone)	Available weather and news services running on the phone using XML integration.	X	X
Presence (BLF)	This feature provides the ability to detect presence of a user with the BLF functionality.	X	X (Basic)
Rich presence (away, calendar, policy)	This feature provides the ability to view busy, away, and in a meeting status. The UC 540 and 560 require the SCC advanced server and client.		X

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Printed in USA C07-670182-00 06/11

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