VoIP Networks

Network Requirements

For the phone to successfully operate as an endpoint in your network, your network must meet the following requirements:

- VoIP Network
  - VoIP is configured on your Cisco routers and gateways.
  - Cisco Unified Communications Manager is installed in your network and is configured to handle call processing.
- IP network that supports DHCP or manual assignment of IP address, gateway, and subnet mask

Note The phone displays the date and time from Cisco Unified Communications Manager. If the user turns off **Automatic date and time** in the Settings application, the time may become out of sync with the server time.

Network Protocols

The Cisco Wireless IP Phone 8821 and 8821-EX supports several industry-standard and Cisco network protocols required for voice communication. The following table provides an overview of the network protocols that the phones support.

Table 1: Supported Network Protocols

Network protocol	Purpose	Usage notes
Bluetooth	Bluetooth is a wireless personal area network (WPAN) protocol that specifies how devices communicate over short distances.	The phones support Bluetooth 4.0.
Bootstrap Protocol (BootP)	BootP enables a network device, such as the Cisco IP Phone, to discover certain startup information, such as the IP address.	None
Cisco Audio Session Tunnel (CAST)	The CAST protocol allows Cisco IP Phones and associated applications to discover and communicate with the remote IP Phones without requiring changes to the traditional signaling components, such as Cisco Unified Communications Manager (CM) and gateways.	The phones use CAST as an interface between CUVA and Cisco Unified Communications Manager using the Cisco IP Phone as a SIP proxy.
Cisco Discovery Protocol (CDP)	CDP is a device-discovery protocol that runs on all Cisco-manufactured equipment. Using CDP, a device can advertise its existence to other devices and receive information about other devices in the network.	The phones use CDP to communicate information such as auxiliary VLAN ID, per port power management details, and Quality of Service (QoS) configuration information with the Cisco Catalyst switch.
Cisco Peer-to-Peer Distribution Protocol (CPPDP)	CPPDP is a Cisco proprietary protocol used to form a peer-to-peer hierarchy of devices. This hierarchy is used to distribute firmware files from peer devices to their neighboring devices.	CPPDP is used by the Peer Firmware Sharing feature.
Dynamic Host Configuration Protocol (DHCP)	DHCP dynamically allocates and assigns an IP address to network devices. DHCP enables you to connect an IP phone into the network and the phone to become operational without the need to manually assign an IP address or to configure additional network parameters.	DHCP is enabled by default. If disabled, you must manually configure the IP address, subnet mask, gateway, and a TFTP server on each phone locally. We recommend that you use DHCP custom option 150. With this method, you configure the TFTP server IP address as the option value. For more information, see the documentation for your particular Cisco Unified Communications Manager release. **Note** If you cannot use option 150, you may try using DHCP option 66.
Hypertext Transfer Protocol (HTTP)	HTTP is the standard way of transferring information and moving documents across the Internet and the web.	The phones use HTTP for XML services and for troubleshooting purposes.
Hypertext Transfer Protocol Secure (HTTPS)	Hypertext Transfer Protocol Secure (HTTPS) is a combination of the Hypertext Transfer Protocol with the SSL/TLS protocol to provide encryption and secure identification of servers.	Web applications with both HTTP and HTTPS support have two URLs configured. Phones that support HTTPS choose the HTTPS URL.

Network protocol	Purpose	Usage notes
IEEE 802.1X	The IEEE 802.1X standard defines a client-server-based access control and authentication protocol that restricts unauthorized clients from connecting to a LAN through publicly accessible ports. Until the client is authenticated, 802.1X access control allows only Extensible Authentication Protocol over LAN (EAPOL) traffic through the port to which the client is connected. After authentication is successful, normal traffic can pass through the port.	The phones implement the IEEE 802.1X standard by providing support for the following authentication methods: • EAP-FAST • EAP-TLS • PEAP-GTC • PEAP-MSCHAPV2
IEEE 802.11n/802.11ac	The IEEE 802.11 standard specifies how devices communication over a wireless local area network (WLAN).	802.11n operates in the 2.4 GHz and 5 GHz band. 802.11ac operates in the 5 GHz band.
Internet Protocol (IP)	IP is a messaging protocol that addresses and sends packets across the network.	To communicate using IP, network devices must have an assigned IP address, subnet, and gateway. IP addresses, subnets, and gateway identifications are automatically assigned if you are using the phone with Dynamic Host Configuration Protocol (DHCP). If you are not using DHCP, you must manually assign these properties to each phone locally. The phones do not support IPv6.
Real-Time Transport Protocol (RTP)	RTP is a standard protocol for transporting real-time data, such as interactive voice, over data networks.	The phones use the RTP protocol to send and receive real-time voice traffic from other phones and gateways.
Real-Time Control Protocol (RTCP)	RTCP works in conjunction with RTP to provide QoS data (such as jitter, latency, and round-trip delay) on RTP streams.	RTCP is enabled by default.
Session Description Protocol (SDP)	SDP is the portion of the SIP protocol that determines which parameters are available during a connection between two endpoints. Conferences are established by using only the SDP capabilities that all endpoints in the conference support.	SDP capabilities, such as codec types, DTMF detection, and comfort noise, are normally configured on a global basis by Cisco Unified Communications Manager or Media Gateway in operation. Some SIP endpoints may allow configuration of these parameters on the endpoint itself.
Session Initiation Protocol (SIP)	SIP is the Internet Engineering Task Force (IETF) standard for multimedia conferencing over IP. SIP is an ASCII-based application-layer control protocol (defined in RFC 3261) that can be used to establish, maintain, and terminate calls between two or more endpoints.	Like other VoIP protocols, SIP addresses the functions of signaling and session management within a packet telephony network. Signaling allows transportation of call information across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

Network protocol	Purpose	Usage notes
Transmission Control Protocol (TCP)	TCP is a connection-oriented transport protocol.	The phones use TCP to connect to Cisco Unified Communications Manager and to access XML services.
Transport Layer Security (TLS)	TLS is a standard protocol for securing and authenticating communications.	Upon security implementation, the phones use the TLS protocol when securely registering with Cisco Unified Communications Manager.
Trivial File Transfer Protocol (TFTP)	TFTP allows you to transfer files over the network. On the Cisco IP Phone, TFTP enables you to obtain a configuration file specific to the phone type.	TFTP requires a TFTP server in your network that the DHCP server can automatically identify. If you want a phone to use a TFTP server other than the one that the DHCP server specifies, you must manually assign the IP address of the TFTP server by using the Network Configuration menu on the phone. For more information, see the documentation for your particular Cisco Unified Communications Manager release.
User Datagram Protocol (UDP)	UDP is a connectionless messaging protocol for delivery of data packets.	UDP is used by the phones for signaling.

Related Topics

Manually Set Up the Phone Network from the Settings Menu
Cisco Unified Communications Manager Interaction, on page 14
802.11 Standards for WLAN Communications, on page 8
Startup Sequence

Cisco Wireless IP Phone 882x Deployment Guide

The *Cisco Wireless IP Phone 882x Deployment Guide* contains useful information about the wireless phone in the Wi-Fi environment. You can find the deployment guide at this location:

https://www.cisco.com/c/en/us/support/collaboration-endpoints/unified-ip-phone-8800-series/products-implementation-design-guides-list.html

Wireless LAN

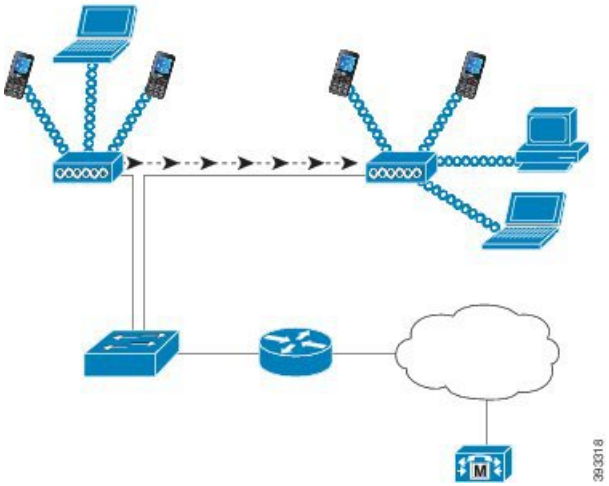
Note For detailed Cisco Wireless IP Phone 8821 and 8821-EX deployment and configuration instructions, see the *Cisco Wireless IP Phone 8821 Series Deployment Guide*.

Devices with wireless capability can provide voice communication within the corporate WLAN. The device depends on and interacts with wireless access points (AP) and key Cisco IP Telephony components, including Cisco Unified Communications Manager Administration, to provide wireless voice communication.

The wireless phones exhibit Wi-Fi capabilities that can use 802.11a, 802.11b, 802.11g, and 802.11n Wi-Fi.

The following figure shows a typical WLAN topology that enables the wireless transmission of voice for wireless IP telephony.

Figure 1: Typical WLAN Topology



When a phone powers on, it searches for and associates with an AP if the device wireless access is set to On. If remembered networks are not within range, you can select a broadcasted network or manually add a network.

The AP uses the connection to the wired network to transmit data and voice packets to and from the switches and routers. Voice signaling is transmitted to the call control server for call processing and routing.

APs are critical components in a WLAN because they provide the wireless links or hot spots to the network. In some WLANs, each AP has a wired connection to an Ethernet switch, such as a Cisco Catalyst 3750, that is configured on a LAN. The switch provides access to gateways and the call control server to support wireless IP telephony.

Some networks contain wired components that support wireless components. The wired components can comprise switches, routers, and bridges with special modules to enable wireless capability.

For more information about Cisco Unified Wireless Networks, see https://www.cisco.com/c/en/us/products/wireless/index.html.

Wi-Fi Network Components

The phone must interact with several network components in the WLAN to successfully place and receive calls.

AP Channel and Domain Relationships

Access points (APs) transmit and receive RF signals over channels within the 2.4GHz or 5GHz frequency band. To provide a stable wireless environment and reduce channel interference, you must specify nonoverlapping channels for each AP.

For more information about AP channel and domain relationships, see the “Designing the Wireless LAN for Voice” section in the *Cisco Wireless IP Phone 8821 Series Deployment Guide*.

AP Interactions

Wireless phones use the same APs as wireless data devices. However, voice traffic over a WLAN requires different equipment configurations and layouts than a WLAN that is used exclusively for data traffic. Data transmission can tolerate a higher level of RF noise, packet loss, and channel contention than voice transmission. Packet loss during voice transmission can cause choppy or broken audio and can make the call inaudible. Packet errors can also cause blocky or frozen video.

Wireless phones users are mobile and often roam across a campus or between floors in a building while connected to a call. In contrast, data users remain in one place or occasionally move to another location. The ability to roam while maintaining a call is one of the advantages of wireless voice, so RF coverage needs to include stairwells, elevators, quiet corners outside conference rooms, and passageways.

To ensure good voice quality and optimal RF signal coverage, you must perform a site survey. The site survey determines settings that are suitable to wireless voice and assists in the design and layout of the WLAN; for example AP placement, power levels, and channel assignments.

After deploying and using wireless voice, you should continue to perform postinstallation site surveys. When you add a group of new users, install more equipment, or stack large amounts of inventory, you are changing the wireless environment. A postinstallation survey verifies that the AP coverage is still adequate for optimal voice communications.

Note Packet loss occurs during roaming; however, the security mode and the presence of fast roaming determines how many packets are lost during transmission. Cisco recommends implementation of Cisco Centralized Key Management (CCKM) to enable fast roaming.

For more information about Voice QoS in a wireless network, see the *Cisco Wireless IP Phone 8821 Series Deployment Guide*.

Access Point Association

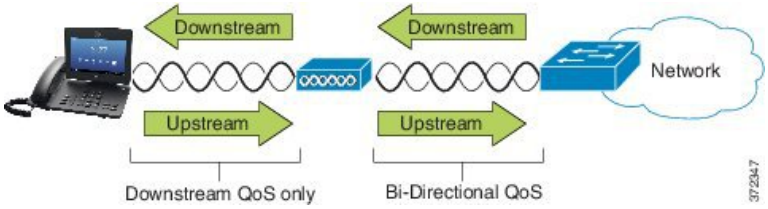
At startup, the phone scans for APs with SSIDs and encryption types that it recognizes. The phone builds and maintains a list of eligible APs and selects the best AP, based on the current configuration.

QoS in a Wireless Network

Voice and video traffic on the wireless LAN, like data traffic, is susceptible to delay, jitter, and packet loss. These issues do not impact the data end user, but can seriously impact a voice or video call. To ensure that voice and video traffic receives timely and reliable treatment with low delay and low jitter, you must use Quality of Service (QoS).

By separating the devices into a voice VLAN and marking voice packets with higher QoS, you can ensure that voice traffic gets priority treatment over data traffic, which results in lower packet delay and fewer lost packets.

Unlike wired networks with dedicated bandwidths, wireless LANs consider traffic direction when implementing QoS. Traffic is classified as upstream or downstream relative to the AP as shown in the following figure.



The Enhanced Distributed Coordination Function (EDCF) type of QoS has up to eight queues for downstream (toward the 802.11b/g clients) QoS. You can allocate the queues based on these options:

- QoS or Differentiated Services Code Point (DSCP) settings for the packets
- Layer 2 or Layer 3 access lists
- VLANs for specific traffic
- Dynamic registration of devices

Although up to eight queues on the AP can be set up, you should use only three queues for voice, video, and signaling traffic to ensure the best possible QoS. Place voice in the Voice queue (UP6), video in the Video queue (UP5), signaling (SIP) traffic in the Video queue (UP4), and place data traffic in a best-effort queue (UP0). Although 802.11b/g EDCF does not guarantee that voice traffic is protected from data traffic, you should get the best statistical results by using this queuing model.

The queues are:

- Best Effort (BE) - 0, 3
- Background (BK) - 1, 2
- Video (VI) - 4, 5
- Voice (VO) - 6, 7

Note The device marks the SIP signaling packets with a DSCP value of 24 (CS3) and RTP packets with DSCP value of 46 (EF).

Note Call Control (SIP) is sent as UP4 (VI). Video is sent as UP5 (VI) when Admission Control Mandatory (ACM) is disabled for video (Traffic Specification [TSpec] disabled). Voice is sent as UP6 (VO) when ACM is disabled for voice (TSpec disabled).

The following table provides a QoS profile on the AP that gives priority to voice, video, and call control (SIP) traffic.

Table 2: QoS Profile and Interface Settings

Traffic Type	DSCP	802.1p	WMM UP	Port Range
Voice	EF (46)	5	6	UDP 16384-32767
Interactive Video	AF41 (34)	4	5	UDP 16384-32767

Traffic Type	DSCP	802.1p	WMM UP	Port Range
Call Control	CS3 (24)	3	4	TCP 5060-5061

To improve reliability of voice transmissions in a nondeterministic environment, the device supports the IEEE 802.11e industry standard and is Wi-Fi Multimedia (WMM) capable. WMM enables differentiated services for voice, video, best effort data and other traffic. For these differentiated services to provide sufficient QoS for voice packets, only a certain amount of voice bandwidth can be serviced or admitted on a channel at one time. If the network can handle “N” voice calls with reserved bandwidth, when the amount of voice traffic is increased beyond this limit (to N+1 calls), the quality of all calls suffers.

To help address issues with call quality, an initial Call Admission Control (CAC) scheme is required. With SIP CAC enabled on the WLAN, QoS is maintained in a network overload scenario by limiting the number of active voice calls so as not to exceed the configured limits on the AP. During times of network congestion, the system maintains a small bandwidth reserve so wireless device clients can roam into a neighboring AP, even when the AP is at “full capacity.” After the voice bandwidth limit is reached, the next call is load-balanced to a neighboring AP so as not to affect the quality of the existing calls on the channel.

The phones use TCP for SIP communications, and call control system registrations can potentially be lost if an AP is at full capacity. Frames to or from a client that has not been "authorized" through the CAC can be dropped, leading to call control system deregistration. Therefore, we recommend that you disable SIP CAC.

Set up Flexible DSCP

Procedure

Step 1 In Cisco Unified Communications Manager Administration, go to **System** > **Service Parameters**.

Step 2 In Clusterwide Parameters (System - Location and Region), set Use Video BandwidthPool for Immersive Video Calls to **False**.

Step 3 In Clusterwide Parameters (Call Admission Control), set Video Call QoS Marking Policy to **Promote to Immersive**.

Step 4 Save your changes.

802.11 Standards for WLAN Communications

Wireless LANs must follow the Institute of Electrical and Electronics Engineers (IEEE) 802.11 standards that define the protocols that govern all Ethernet-based wireless traffic. The wireless phones support the following standards:

- 802.11a: Uses the 5 GHz band that provides more channels and improved data rates by using OFDM technology. Dynamic Frequency Selection (DFS) and Transmit Power Control (TPC) support this standard.
- 802.11b: Specifies the radio frequency (RF) of 2.4 Ghz for both transmission and receipt of data at lower data rates (1, 2, 5.5, 11 Mbps).
- 802.11d: Enables access points to advertise their currently supported radio channels and transmit power levels. The 802.11d-enabled client then uses that information to determine the channels and powers to use. The phone requires World mode (802.11d) to determine which channels are legally allowed for any

given country. For supported channels, see the table that follows. Ensure that 802.11d is properly configured on the Cisco IOS Access Points or Cisco Unified Wireless LAN Controller.

- 802.11e: Defines a set of Quality of Service (QoS) enhancements for wireless LAN applications.
- 802.11g: Uses the same unlicensed 2.4 Ghz band as 802.11b, but extends the data rates to provide greater performance by using Orthogonal Frequency Division Multiplexing (OFDM) technology. OFDM is a physical-layer encoding technology for transmission of signals through use of RF.
- 802.11h: Supports 5 GHz spectrum and transmit power management. Provides DFS and TPC to the 802.11a Media Access Control (MAC).
- 802.11i: Specifies security mechanisms for wireless networks.
- 802.11n: Uses the radio frequency of 2.4 GHz or 5 GHz for both transmission and receipt of data with speeds up to 150 Mbps, and enhances data transfer through the use of multiple input, multiple output (MIMO) technology, channel bonding, and payload optimization.

Note The wireless phones have a single antenna and use the Single Input Single Output (SISO) system, which supports MCS 0 to MCS 7 data rates only (72 Mbps with 20 MHz channels and 150 Mbps 40MHz channels). Optionally, you can enable MCS 8 to MCS 15 if 802.11n clients are using MIMO technology that can take advantage of those higher data rates.

- 802.11r: Specifies requirements for fast secure roaming.
- 802.11ac: Uses the radio frequency of 5 GHz for both transmission and receipt of data with speeds up to 433 Mbps.

Table 3: Supported Channels

Band Range	Available Channels	Channel Set	Channel Width
2.412 - 2.472 GHz	13	1 - 13	20 MHz
5.180 - 5.240 GHz	4	36, 40, 44, 48	20, 40, 80 MHz
5.260 - 5.320 GHz	4	52, 56, 60, 64	20, 40, 80 MHz
5. 500 - 5.700 GHz	11	100 - 140	20, 40, 80 MHz
5.745 - 5.825 GHz	5	149, 153, 157, 161, 165	20, 40, 80 MHz

Note Channels 120, 124, 128 are not supported in the Americas, Europe, or Japan, but may be in other regions around the world.

For information about supported data rates, Tx power and Rx sensitivity for WLANs, see the *Cisco Wireless IP Phone 8821 Series Deployment Guide*.

World Mode (802.11d)

The wireless phones use 802.11d to determine the channels and transmit power levels to use. The phone inherits its client configuration from the associated AP. Enable World mode (802.11d) on the AP to use the phone in World mode.

Note Enablement of World mode (802.11d) may not be necessary if the frequency is 2.4 GHz and the current access point is transmitting on a channel from 1 to 11.

Because all countries support these frequencies, you can attempt to scan these channels regardless of World mode (802.11d) support.

For more information on enabling World mode and 2.4 GHz support, see the *Cisco Wireless IP Phone 8821 Series Deployment Guide*.

Enable World mode (802.11d) for the corresponding country where the access point is located. World mode is enabled automatically for the Cisco Unified Wireless LAN Controller.

Radio Frequency Ranges

WLAN communications use the following radio frequency (RF) ranges:

- 2.4 GHz—Many devices that use 2.4 GHz can potentially interfere with the 802.11b/g connection. Interference can produce a Denial of Service (DoS) scenario, which may prevent successful 802.11 transmissions.
- 5 GHz—This range divides into several sections called Unlicensed National Information Infrastructure (UNII) bands, each of which has four channels. The channels are spaced at 20 MHz to provide nonoverlapping channels and more channels than 2.4 GHz provides.

Security for Communications in WLANs

Because all WLAN devices that are within range can receive all other WLAN traffic, security of voice communications is critical in WLANs. To ensure that intruders do not manipulate or intercept voice traffic, the Cisco SAFE Security Architecture supports wireless phones and Cisco Aironet APs. For more information about security in networks, see https://www.cisco.com/c/en/us/solutions/enterprise/design-zone-security/index.html.

Authentication Methods

The Cisco Wireless IP Telephony solution provides wireless network security that prevents unauthorized sign-ins and compromised communications through use of the following authentication methods that wireless phones support:

- WLAN Authentication
  - WPA (802.1x authentication + TKIP or AES encryption)
  - WPA2 (802.1x authentication + AES or TKIP encryption)

- WPA-PSK (Pre-Shared key + TKIP encryption)
- WPA2-PSK (Pre-Shared key + AES encryption)
- EAP-FAST (Extensible Authentication Protocol – Flexible Authentication via Secure Tunneling)
- EAP-TLS (Extensible Authentication Protocol – Transport Layer Security)
- PEAP (Protected Extensible Authentication Protocol) MS-CHAPv2 and GTC
- CCKM (Cisco Centralized Key Management)
- Open (None)

- WLAN Encryption
  - AES (Advanced Encryption Scheme)
  - TKIP / MIC (Temporal Key Integrity Protocol / Message Integrity Check)
  - WEP (Wired Equivalent Protocol) 40/64 and 104/128 bit

Note Dynamic WEP with 802.1x authentication and Shared Key authentication are not supported.

For more information about authentication methods, see the "Wireless Security" section in the *Cisco Wireless IP Phone 8821 Series Deployment Guide*.

Authenticated Key Management

The following authentication schemes use the RADIUS server to manage authentication keys:

- WPA/WPA2: Uses RADIUS server information to generate unique keys for authentication. Because these keys are generated at the centralized RADIUS server, WPA/WPA2 provides more security than WPA pre-shared keys that are stored on the AP and device.
- Cisco Centralized Key Management (CCKM): Uses RADIUS server and a wireless domain server (WDS) information to manage and authenticate keys. The WDS creates a cache of security credentials for CCKM-enabled client devices for fast and secure reauthentication.

With WPA/WPA2 and CCKM, encryption keys are not entered on the device, but are automatically derived between the AP and device. But the EAP username and password that are used for authentication must be entered on each device.

Encryption Methods

To ensure that voice traffic is secure, the wireless phones support WEP, TKIP, and Advanced Encryption Standards (AES) for encryption. When these mechanisms are used for encryption, voice Real-Time Transport Protocol (RTP) packets are encrypted between the AP and the device.

WEP

When WEP is used in the wireless network, authentication happens at the AP through open or shared-key authentication. The WEP key that is set up on the phone must match the WEP key that is configured at

the AP for successful connections. The phones support WEP keys that use 40-bit encryption or a 128-bit encryption and remain static on the device and AP.

TKIP

WPA and CCKM use TKIP encryption, which has several improvements over WEP. TKIP provides per-packet key ciphering and longer initialization vectors (IVs) that strengthen encryption. In addition, a message integrity check (MIC) ensures that encrypted packets are not altered. TKIP removes the predictability of WEP that helps intruders decipher the WEP key.

AES

An encryption method used for WPA2 authentication. This national standard for encryption uses a symmetrical algorithm that has the same key for encryption and decryption.

For more information about encryption methods, see the "Wireless Security" section in the *Cisco Wireless IP Phone 8821 Series Deployment Guide*.

AP Authentication and Encryption Options

Authentication and encryption schemes are set up within the wireless LAN. VLANs are configured in the network and on the APs and specify different combinations of authentication and encryption. An SSID associates with a VLAN and the particular authentication and encryption scheme. In order for wireless phones to authenticate successfully, you must configure the same SSIDs with their authentication and encryption schemes on the APs and on the phone.

Note

- When you use WPA pre-shared key or WPA2 pre-shared key, the pre-shared key must be statically set on the phone. These keys must match the keys that are on the AP.
- The wireless phones do not support auto EAP negotiation; to use EAP-FAST mode, you must specify it.

The following table provides a list of authentication and encryption schemes that are configured on the Cisco Aironet APs that the phones support. The table shows the network configuration option for the device that corresponds to the AP configuration.

Table 4: Authentication and Encryption Schemes

Cisco WLAN Configuration			**Phone Configuration**
Authentication	**Key management**	**Common encryption**	**Authentication**
Open	None	None	None
Static WEP	None	WEP	WEP
EAP-FAST	WPA or WPA2 with optional CCKM	TKIP or AES	802.1x EAP > EAP-FAST
PEAP-MSCHAPv2	WPA or WPA2 with optional CCKM	TKIP or AES	802.1x EAP > PEAP > MSCHAPV2

Cisco WLAN Configuration			Phone Configuration
PEAP-GTC	WPA or WPA2 with optional CCKM	TKIP or AES	802.1x EAP > PEAP > GTC
EAP-TLS	WPA or WPA2 with optional CCKM	TKIP or AES	802.1x EAP > TLS
WPA/WPA2-PSK	WPA-PSK or WPA2-PSK	TKIP or AES	WPA/WPA2 PSK

For more information, see the *Cisco Wireless IP Phone 8821 Series Deployment Guide*.

Certificates

The phones support the following certificates.

- X.509 digital certificate for EAP-TLS or to enable PEAP + Server Validation for WLAN authentication
- Simple Certificate Enrollment Protocol (SCEP) for certificate enrollment and auto-renewal
- 1024, 2048, 4096 bit keys
- SHA-1 and SHA-256 signature types
- DER and Base-64 (PEM) encoding types
- User Installed Certificate in PKCS #12 format (.p12 or .pfx extension), which also contains the private key
- Server (Root CA) Certificate with .crt or .cer extension

You install certificates on the phones in one of these ways:

- Use the Administration web page. For more information, see Cisco IP Phone Administration Page.
- Use an SCEP server to manage and install the certificates. For more information, see SCEP Setup

If your users set up their phones themselves and their phones need certificates, you need to give them the type of certificate when you give them the other configuration settings. If you don't use SCEP for certificate installation, then you need to install the certificates yourself.

WLANs and Roaming

The wireless phones support Cisco Centralized Key Management (CCKM), a centralized key management protocol that provides a cache of session credentials on the wireless domain server (WDS).

For details about CCKM, see the *Cisco Fast Secure Roaming Application Note* at:

http://www.cisco.com/en/US/products/hw/wireless/ps4570/prod_technical_reference09186a00801c5223.html

The phones also support 802.11r. For more information, see the *Cisco Wireless IP Phone 8821 Series Deployment Guide*.

Cisco Unified Communications Manager Interaction

Cisco Unified Communications Manager is an open, industry-standard call processing system. Cisco Unified Communications Manager software sets up and tears down calls between phones, integrating traditional PBX functionality with the corporate IP network. Cisco Unified Communications Manager manages the components of the telephony system, such as the phones, the access gateways, and the resources necessary for features such as call conferencing and route planning. Cisco Unified Communications Manager also provides:

- Firmware for phones
- Certificate Trust List (CTL) and Identity Trust List (ITL) files using the TFTP and HTTP services
- Phone registration
- Call preservation, so that a media session continues if signaling is lost between the primary Communications Manager and a phone

For information about configuring Cisco Unified Communications Manager to work with the phones described in this chapter, see the documentation for your particular Cisco Unified Communications Manager release.

Note If the phone model that you want to configure does not appear in the Phone Type drop-down list in Cisco Unified Communications Manager Administration, install the latest device package for your version of Cisco Unified Communications Manager from Cisco.com.

Voice Messaging System Interaction

Cisco Unified Communications Manager lets you integrate with different voice messaging systems, including the Cisco Unity Connection voice messaging system. Because you can integrate with various systems, you must provide users with information about how to use your specific system.

To enable the ability for a user to transfer to voicemail, set up a *xxxxx dialing pattern and configure it as Call Forward All to Voicemail. For more information, see the Cisco Unified Communications Manager documentation.

Provide the following information to each user:

- How to access the voice messaging system account.
- Initial password for accessing the voice messaging system.

  Configure a default voice messaging system password for all users.
- How the phone indicates that voice messages are waiting.

  Use Cisco Unified Communications Manager to set up a message waiting indicator (MWI) method.