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Physical and Operating Environment Specifications

The following table shows the physical and operating environment specifications for the Cisco IP Phone 7800 Series with Multiplatform Firmware.

Table 1: Physical and Operating Specifications

Specification	Value or Range
Operating temperature	32° to 104°F (0° to 40°C)
Operating relative humidity	10% to 90% (noncondensing)
Storage temperature	14° to 140°F (–10° to 60°C)
Height	8.14 in. (207 mm)
Width	• Cisco IP Phone 7811— 7.67 in. (195 mm) • Cisco IP Phone 7821 — 8.11 in. (206 mm) • Cisco IP Phone 7841 — 8.11 in. (206 mm) • Cisco IP Phone 7861— 10.42 in. (264.91 mm)

Specification	Value or Range
Depth	1.1 in. (28 mm)
Weight	• Cisco IP Phone 7811— 0.84 kg • Cisco IP Phone 7821 — 0.867 kg • Cisco IP Phone 7841 — 0.868 kg • Cisco IP Phone 7861— 1.053 kg
Power	• 100-240 VAC, 50-60 Hz, 0.5 A—When using the AC adapter • 48 VDC, 0.2 A—When using the in-line power over the network cable
Cables	Cisco IP Phone 7811, 7821, 7841, and 7861: • Category 3/5/5e/6 for 10-Mbps cables with 4 pairs • Category 5/5e/6 for 100-Mbps cables with 4 pairs Cisco IP Phone 7841: Category 5/5e/6 for 1000-Mbps cables with 4 pairs **Note** Cables have 4 pairs of wires for a total of 8 conductors.
Distance Requirements	As supported by the Ethernet Specification, it is assumed that the maximum cable length between each Cisco IP Phone and the switch is 100meters (330feet).

For more information, see the *Cisco IP Phone 7800 Series Data Sheet*: https://www.cisco.com/c/en/us/products/collaboration-endpoints/ip-phone-7800-series-multiplatform-firmware/datasheet-listing.html

Cable Specifications

- RJ-9 jack (4-conductor) for handset and headset connection.

> **Note** The Cisco IP Phone 7811 does not contain a headset jack.

- RJ-45 jack for the LAN 10/100BaseT connection (on Cisco IP Phones 7811, 7821, and 7861) and the LAN 1000BaseT connection (on the Cisco IP Phone 7841).
- RJ-45 jack for a second 10/100BaseT compliant connection (on Cisco IP Phones 7811, 7821, and 7861) and the LAN 1000BaseT connection (on the Cisco IP Phone 7841).
- 48-volt power connector.

Network and Computer Port Pinouts

Although both the network and computer (access) ports are used for network connectivity, they serve different purposes and have different port pinouts:

- The network port is the 10/100 SW port; the Cisco IP Phone 7841 has a 10/100/1000 SW network port.
- The computer (access) port is the 10/100 PC port; the Cisco IP Phone 7841 has a 10/100/1000 PC computer port.

Network Port Connector

The following table describes the network port connector pinouts.

Table 2: Network Port Connector Pinouts

Pin Number	Function
1	BI_DA+
2	BI_DA-
3	BI_DB+
4	BI_DC+
5	BI_DC-
6	BI_DB-
7	BI_DD+
8	BI_DD-
Note BI stands for bidirectional, while DA, DB, DC, and DD stand for Data A, Data B, Data C, and Data D respectively.	

Computer Port Connector

The following table describes the computer port connector pinouts.

Table 3: Computer (Access) Port Connector Pinouts

Pin Number	Function
1	BI_DB+
2	BI_DB-
3	BI_DA+

Pin Number	Function
4	BI_DD+
5	BI_DD-
6	BI_DA-
7	BI_DC+
8	BI_DC-
Note BI stands for bidirectional, while DA, DB, DC, and DD stand for Data A, Data B, Data C, and Data D respectively.	

Phone Power Requirements

The Cisco IP Phone can be powered with external power or with Power over Ethernet (PoE). A separate power supply provides external power. The switch can provide PoE through the phone Ethernet cable.

Note When you install a phone that is powered with external power, connect the power supply to the phone and to a power outlet before you connect the Ethernet cable to the phone. When you remove a phone that is powered with external power, disconnect the Ethernet cable from the phone before you disconnect the power supply.

Table 4: Guidelines for Cisco IP Phone Power

Power Type	Guidelines
External power: Provided through the CP-PWR-CUBE-3= external power supply	The Cisco IP Phone uses the CP-PWR-CUBE-3 power supply.
External power—Provided through the Cisco IP Phone Power Injector	The Cisco IP Phone Power Injector may be used with most Cisco IP Phones. The phone datasheet identifies if the phone can use the power injector. Functioning as a midspan device, the injector delivers inline power to the attached phone. The Cisco IP Phone Power Injector connects between a switch port and the IP Phone, and supports a maximum cable length of 100m between the unpowered switch and the IP phone.
PoE power—Provided by a switch through the Ethernet cable attached to the phone.	To ensure uninterruptible operation of the phone, make sure that the switch has a backup power supply. Make sure that the CatOS or IOS version that runs on your switch supports your intended phone deployment. See the documentation for your switch for operating system version information.

The documents in the following table provide more information on the following topics:

- Cisco switches that work with Cisco IP Phones
- Cisco IOS releases that support bidirectional power negotiation
- Other requirements and restrictions about power

Document topics	URL
PoE Solutions	http://www.cisco.com/c/en/us/solutions/enterprise-networks/power-over-ethernet-solutions/index.html
Cisco Catalyst Switches	http://www.cisco.com/c/en/us/products/switches/index.html
Integrated Service Routers	http://www.cisco.com/c/en/us/products/routers/index.html
Cisco IOS Software	http://www.cisco.com/c/en/us/products/ios-nx-os-software/index.html

Power Outage

Your access to emergency service through the phone requires that the phone receive power. If a power interruption occurs, service or emergency calling service dialing does not function until power is restored. If a power failure or disruption occurs, you may need to reset or reconfigure the equipment before you can use service or emergency calling service dialing.

Power Reduction

You can reduce the amount of energy that the Cisco IP Phone consumes by using Power Save mode.

Power Save

In Power Save mode, the backlight on the screen is not lit when the phone is not in use. The phone remains in Power Save mode until the user lifts the handset or presses any button. Set up each phone to enable or disable Power Save settings.

Note The Cisco IP Phone 7811 does not support Power Save because the phone screen does not have a backlight.

Power Negotiation Over LLDP

The phone and the switch negotiate the power that the phone consumes. Cisco IP Phone operates at multiple power settings, which lowers power consumption when less power is available.

After a phone reboots, the switch locks to one protocol (CDP or LLDP) for power negotiation. The switch locks to the first protocol (containing a power Threshold Limit Value [TLV]) that the phone transmits. If the system administrator disables that protocol on the phone, the phone cannot power up any accessories because the switch does not respond to power requests in the other protocol.

Cisco recommends that Power Negotiation always be enabled (default) when connecting to a switch that supports power negotiation.

If Power Negotiation is disabled, the switch may disconnect power to the phone. If the switch does not support power negotiation, disable the Power Negotiation feature before you power up accessories over PoE. When the Power Negotiation feature is disabled, the phone can power the accessories up to the maximum that the IEEE 802.3af-2003 standard allows.

Note

- When CDP and Power Negotiation are disabled, the phone can power the accessories up to 15.4W.

Network Protocols

Cisco IP Phones support several industry-standard and Cisco network protocols that are required for voice communication. The following table provides an overview of the network protocols that the phones support.

Table 5: Supported Network Protocols on the Cisco IPPhone

Network Protocol	Purpose	Usage Notes
Bootstrap Protocol (BootP)	BootP enables a network device, such as the Cisco IP Phone, to discover certain startup information, such as its IP address.	—
Cisco Discovery Protocol (CDP)	CDP is a device-discovery protocol that runs on all Cisco-manufactured equipment. A device can use CDP to advertise its existence to other devices and receive information about other devices in the network.	The Cisco IPPhone uses 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.
Domain Name Server (DNS)	DNS translates domain names to IP addresses.	Cisco IP Phones have a DNS client to translate domain names into IP addresses.
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 have the phone 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, and gateway on each phone locally. We recommend that you use the DHCP custom option 160, 159.

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Network Protocol	Purpose	Usage Notes
Hypertext Transfer Protocol (HTTP)	HTTP is the standard protocol for transfer of information and movement of documents across the Internet and the web.	CiscoIP Phones use HTTP for XML services, provisioning, upgrade 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. Cisco IP Phones that support HTTPS choose the HTTPS URL. A lock icon is displayed to the user if the connection to the service is via HTTPS.
Internet Protocol (IP)	IP is a messaging protocol that addresses and sends packets across the network.	To communicate with IP, network devices must have an assigned IP address, subnet, and gateway. IP addresses, subnets, and gateways identifications are automatically assigned if you are using the Cisco IPPhone with Dynamic Host Configuration Protocol (DHCP). If you are not using DHCP, you must manually assign these properties to each phone locally.
Link Layer Discovery Protocol (LLDP)	LLDP is a standardized network discovery protocol (similar to CDP) that is supported on some Cisco and third-party devices.	The Cisco IP Phone supports LLDP on the PC port.
Link Layer Discovery Protocol-Media Endpoint Devices (LLDP-MED)	LLDP-MED is an extension of the LLDP standard developed for voice products.	The Cisco IPPhone supports LLDP-MED on the SW port to communicate information such as: • Voice VLAN configuration • Device discovery • Power management • Inventory management For more information about LLDP-MED support, see the *LLDP-MED and Cisco Discovery Protocol* white paper at this URL: [illegible]

Network Protocol	Purpose	Usage Notes
Network Transport Protocol (NTP)	NTP is a networking protocol for clock synchronization between computer systems over packet-switched, variable-latency data networks.	Cisco IP Phones have an NTP client integrated into the software.
Real-Time Transport Protocol (RTP)	RTP is a standard protocol for transporting real-time data, such as interactive voice and video, over data networks.	Cisco IP 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 disabled 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 a Third-Party Call Control System or a 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 is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.
Secure Real-Time Transfer protocol (SRTP)	SRTP is an extension of the Real-Time Protocol (RTP) Audio/Video Profile and ensures the integrity of RTP and Real-Time Control Protocol (RTCP) packets providing authentication, integrity, and encryption of media packets between two endpoints.	Cisco IP Phones use SRTP for media encryption.
Transmission Control Protocol (TCP)	TCP is a connection-oriented transport protocol.	—

Network Protocol	Purpose	Usage Notes
Transport Layer Security (TLS)	TLS is a standard protocol for securing and authenticating communications.	When security is implemented, Cisco IP Phones use the TLS protocol when securely registering with the third-party call control system.
Trivial File Transfer Protocol (TFTP)	TFTP allows you to transfer files over the network. On the Cisco IPPhone, TFTP enables you to obtain a configuration file specific to the phone type.	TFTP requires a TFTP server in your network, which can be automatically identified from the DHCP server.
User Datagram Protocol (UDP)	UDP is a connectionless messaging protocol for delivery of data packets.	UDP is used only for RTP streams. SIP uses UDP, TCP, and TLS.

Related Topics

Verify the Network Setup
Verify Phone Startup

VLAN Interaction

The Cisco IP Phone contains an internal Ethernet switch, enabling forwarding of packets to the phone, and to the computer (access) port and the network port on the back of the phone.

If a computer is connected to the computer (access) port, the computer and the phone share the same physical link to the switch and share the same port on the switch. This shared physical link has the following implications for the VLAN configuration on the network:

- The current VLANs might be configured on an IP subnet basis. However, additional IP addresses might not be available to assign the phone to the same subnet as other devices that connect to the same port.
- Data traffic present on the VLAN supporting phones might reduce the quality of VoIP traffic.
- Network security may indicate a need to isolate the VLAN voice traffic from the VLAN data traffic.

You can resolve these issues by isolating the voice traffic onto a separate VLAN. The switch port to which the phone connects would be configured for separate VLANs for carrying:

- Voice traffic to and from the IP phone (auxiliary VLAN on the Cisco Catalyst 6000 series, for example)
- Data traffic to and from the PC that connects to the switch through the computer (access) port of the IP phone (native VLAN)

Isolating the phones on a separate, auxiliary VLAN increases the quality of the voice traffic and allows a large number of phones to be added to an existing network that does not have enough IP addresses for each phone.

For more information, see the documentation that is included with a Cisco switch. You can also access switch information at this URL:

http://cisco.com/en/US/products/hw/switches/index.html

External Devices

We recommend that you use good-quality external devices that are shielded against unwanted radio frequency (RF) and audio frequency (AF) signals. External devices include headsets, cables, and connectors.

Depending on the quality of these devices and their proximity to other devices, such as mobile phones or two-way radios, some audio noise may still occur. In these cases, we recommend that you take one or more of these actions:

- Move the external device away from the source of the RF or AF signals.
- Route the external device cables away from the source of the RF or AF signals.
- Use shielded cables for the external device, or use cables with a better shield and connector.
- Shorten the length of the external device cable.
- Apply ferrites or other such devices on the cables for the external device.

Cisco cannot guarantee the performance of external devices, cables, and connectors.

Caution In European Union countries, use only external speakers, microphones, and headsets that are fully compliant with the EMC Directive [89/336/EC].

SIP and NAT Configuration

SIP and the Cisco IP Phone

The Cisco IP Phone uses Session Initiation Protocol (SIP), which allows interoperation with all IT service providers that support SIP. SIP is an IETF-defined signaling protocol that controls voice communication sessions in an IP network.

SIP handles signaling and session management within a packet telephony network. *Signaling* allows call information to be carried across network boundaries. *Session management* controls the attributes of an end-to-end call.

In typical commercial IP telephony deployments, all calls go through a SIP Proxy Server. The receiving phone is called the SIP user agent server (UAS), while the requesting phone is called the user agent client (UAC).

SIP message routing is dynamic. If a SIP proxy receives a request from a UAS for a connection but cannot locate the UAC, the proxy forwards the message to another SIP proxy in the network. When the UAC is located, the response routes back to the UAS, and the two UAs connect using a direct peer-to-peer session. Voice traffic transmits between UAs over dynamically assigned ports using Real-time Protocol (RTP).

RTP transmits real-time data such as audio and video; RTP does not guarantee real-time delivery of data. RTP provides mechanisms for the sending and receiving applications to support streaming data. Typically, RTP runs on top of UDP.

SIP Over TCP

To guarantee state-oriented communications, the Cisco IP Phone can use TCP as the transport protocol for SIP. This protocol provides *guaranteed delivery* that assures that lost packets are retransmitted. TCP also guarantees that the SIP packages are received in the same order that they were sent.

TCP overcomes the problem of UDP port-blocking by corporate firewalls. With TCP, new ports do not need to be open or packets dropped, because TCP is already in use for basic activities, such as internet browsing or e-commerce.

SIP Proxy Redundancy

An average SIP Proxy Server can handle tens of thousands of subscribers. A backup server allows an active server to be temporarily switched out for maintenance. The phone supports the use of backup servers to minimize or eliminate service disruption.

A simple way to support proxy redundancy is to specify a SIP Proxy Server in the phone configuration profile. The phone sends a DNS NAPTR or SRV query to the DNS server. If configured, the DNS server returns SRV records that contain a list of servers for the domain, with their hostnames, priority, listening ports, and so forth. The phone tries to contact the servers in the order of the priority. The server with a lower number has a higher priority. Up to six NAPTR records and twelve SRV records are supported in a query.

When the phone fails to communicate with the primary server, the phone can failover to a lower-priority server. If configured, the phone can restore the connection back to the primary. Failover and failback support switches between servers with different SIP transport protocols. The phone doesn't perform failback to the primary server during an active call until the call ends and the failback conditions are met.

Example of Resource Records from the DNS Server

```
as1bsoft     3600    IN NAPTR 50   50  "s"  "SIPS+D2T"     ""  _sips._tcp.tlstest
             3600    IN NAPTR 90   50  "s"  "SIP+D2T"      ""  _sip._tcp.tcptest
             3600    IN NAPTR 100  50  "s"  "SIP+D2U"      ""  _sip._udp.udptest

_sips._tcp.tlstest  SRV 1 10 5061 srv1.sipurash.com.
                    SRV 2 10 5060 srv2.sipurash.com.
_sip._tcp.tcptest   SRV 1 10 5061 srv3.sipurash.com.
                    SRV 2 10 5060 srv4.sipurash.com.
_sip._udp.udptest   SRV 1 10 5061 srv5.sipurash.com.
                    SRV 2 10 5060 srv6.sipurash.com.

srv1     3600    IN    A   1.1.1.1
srv2     3600    IN    A   2.2.2.2
srv3     3600    IN    A   3.3.3.3
srv4     3600    IN    A   4.4.4.4
srv5     3600    IN    A   5.5.5.5
srv6     3600    IN    A   6.6.6.6
```

The following example shows the priority of the servers from the perspective of the phone.

```
Priority       IP Address     SIP Protocol    Status
1st            1.1.1.1            TLS            UP
2nd            2.2.2.2            TLS            UP
3rd            3.3.3.3            TCP            UP
4th            4.4.4.4            TCP            UP
5th            5.5.5.5            UDP            UP
6th            6.6.6.6            UDP            UP
```

The phone always sends SIP messages to the available address with the top priority and with the status UP in the list. In the example, the phone sends all the SIP messages to the address 1.1.1.1. If the address 1.1.1.1 in the list is marked with the status DOWN, the phone communicates with 2.2.2.2 instead. The phone can restore the connection back to 1.1.1.1 when the specified failback conditions are met. For more details about failover and failback, see SIP Proxy Failover, on page 12 and SIP Proxy Fallback, on page 13.

SIP Proxy Failover

The phone performs a failover in any of these cases:

- The phone sends SIP messages and doesn't get responses from the server.
- The server responds with a code that matches the specified code in **Try Backup RSC**.
- The phone gets a TCP disconnection request.

We strongly recommend that you set the **Auto Register When Failover** to **Yes** when **SIP Transport** is set to **Auto**.

You can also configure this extension-specific parameters in the configuration file:

```
<SIP_Transport_n_ ua="na">Auto</SIP_Transport_n_>
```

```
<Auto_Register_When_Failover_n_ ua="na">Yes</Auto_Register_When_Failover_n_>
```

where *n* is the extension number.

Phone Failover Behavior

When the phone fails to communicate with the currently connected server, it refreshes the server list status. The unavailable server is marked with the status DOWN in the server list. The phone tries to connect to the top-priority server with the status UP in the list.

In the following example, the addresses 1.1.1.1 and 2.2.2.2 aren't available. The phone sends SIP messages to 3.3.3.3, which has the top priority among the servers with the status UP.

```
Priority        IP Address      SIP Protocol    Status
1st             1.1.1.1             TLS          DOWN
2nd             2.2.2.2             TLS          DOWN
3rd             3.3.3.3             TCP          UP
4th             4.4.4.4             TCP          UP
5th             5.5.5.5             UDP          UP
6th             6.6.6.6             UDP          UP
```

In the following example, there are two SRV records from the DNS NAPTR response. For each SRV record, there are three A records (IP addresses).

```
Priority        IP Address      SIP Protocol    Server            Status
1st             1.1.1.1             UDP         SRV1              DOWN
2nd             1.1.1.2             UDP         SRV1              UP
3rd             1.1.1.3             UDP         SRV1              UP
4th             2.2.2.1             TLS         SRV2              UP
5th             2.2.2.2             TLS         SRV2              UP
6th             2.2.2.3             TLS         SRV2              UP
```

Let's assume that the phone failed to connect to 1.1.1.1 and then registered to 1.1.1.2. When 1.1.1.2 goes down, phone behavior depends on the setting of **Proxy Fallback Intvl**.

- When **Proxy Fallback Intvl** is set to **0**, the phone tries with the addresses in this order: 1.1.1.1, 1.1.1.3, 2.2.2.1, 2.2.2.2, 2.2.2.3.
- When **Proxy Fallback Intvl** is set to a value other than zero, the phone tries with the addresses in this order: 1.1.1.3, 2.2.2.1, 2.2.2.2, 2.2.2.3.

SIP Proxy Fallback

The proxy fallback requires a value other than zero specified in the **Proxy Fallback Intvl** field on the **Ext (n)** tab in the phone web interface. If you set this field to 0, the SIP proxy failback feature is disabled. You can also configure this extension-specific parameter in the configuration file in this format:

```
<Proxy_Fallback_Intvl_n_ ua="na">60</Proxy_Fallback_Intvl_n_>
```

where *n* is the extension number.

The time when the phone triggers a failback depends on the phone configuration and the SIP transport protocols in use.

To enable the phone to perform failback between different SIP transport protocols, set **SIP Transport** to **Auto** on the **Ext (n)** tab in the phone web interface. You can also configure this extension-specific parameter in the configuration file with the following XML string:

```
<SIP_Transport_n_ ua="na">Auto</SIP_Transport_n_>
```

where *n* is the extension number.

Failback from a UDP Connection

The failback from a UDP connection is triggered by SIP messages. In the following example, the phone first failed to register to 1.1.1.1 (TLS) at the time T1 since there's no response from the server. When SIP Timer F expires, the phone registers to 2.2.2.2 (UDP) at the time T2 (T2=T1+SIP Timer F). The current connection is on 2.2.2.2 via UDP.

```
Priority       IP Address     SIP Protocol    Status
1st            1.1.1.1            TLS          DOWN        T1 (Down time)
2nd            2.2.2.2            UDP          UP
3rd            3.3.3.3            TCP          UP
```

The phone has the following configuration:

```
<Proxy_Fallback_Intvl_n_ ua="na">60</Proxy_Fallback_Intvl_n_>
<Register_Expires_n_ ua="na">3600</Register_Expires_n_>
<SIP_Timer_F ua="na">16</SIP_Timer_F>
```

where *n* is the extension number.

The phone refreshes the registration at time T2 (T2=(3600-16)*78%). The phone checks the address list for the availability of the IP addresses and the down time. If T2-T1 > = 60, the failed server 1.1.1.1 resumes back to UP and the list is updated to the following. The phone sends SIP messages to 1.1.1.1.

```
Priority       IP Address     SIP Protocol    Status
1st            1.1.1.1            TLS           UP
2nd            2.2.2.2            UDP           UP
3rd            3.3.3.3            TCP           UP
```

Failback from a TCP or TLS Connection

The failback from a TCP or TLS connection is triggered by the parameter **Proxy Fallback Intvl**. In the following example, the phone failed to register to 1.1.1.1 (UDP) at the time T1 and thus registered to 2.2.2.2 (TCP). The current connection is on 2.2.2.2 via TCP.

```
Priority       IP Address     SIP Protocol    Status
1st            1.1.1.1            UDP          DOWN         T1 (Down time)
2nd            2.2.2.2            TCP          UP
3rd            3.3.3.3            TLS          UP
```

The phone has the following configuration:

```
<Proxy_Fallback_Intvl_n_ ua="na">60</Proxy_Fallback_Intvl_n_>
<Register_Expires_n_ ua="na">3600</Register_Expires_n_>
<SIP_Timer_F ua="na">16</SIP_Timer_F>
```

where *n* is the extension number.

The proxy fallback interval (60 seconds) counts down from T1. The phone triggers proxy failback at the time of T1+60. If you set the proxy fallback interval to 0 in this example, the phone keeps the connection on 2.2.2.2.

Dual Registration

The phone always registers to both primary (or primary outbound) and alternate (or alternate outbound) proxies. After registration, the phone sends out Invite and Non-Invite SIP messages through primary proxy first. If there is no response for the new INVITE from the primary proxy, after timeout, the phone attempts to connect with the alternate proxy. If the phone fails to register to the primary proxy, it sends an INVITE to the alternate proxy without trying the primary proxy.

Dual registration is supported on a per-line basis. Three added parameters can be configured through web user interface and remote provisioning:

- Alternate Proxy—Default is empty.
- Alternate Outbound Proxy—Default is empty.
- Dual Registration—Default is NO (turned off).

After you configure the parameters, reboot the phone for the feature to take effect.

Note Specify a value for primary proxy (or primary outbound proxy) and alternate proxy (or alternate outbound proxy) for the feature to function properly.

Dual Registration and DNS SRV Limitations

- When Dual Registration is enabled, DNS SRV Proxy Fallback or Recovery must be disabled.
- Do not use Dual Registration along with other Fallback or Recovery mechanisms. For example: Broadsoft mechanism.
- There is no recovery mechanism for feature request. However, the administrator can adjust the reregistration time for a prompt update of the registration state for primary and alternate proxy.

Dual Registration and Alternate Proxy

When the Dual Register parameter is set to **No**, Alternate Proxy is ignored.

Failover and Recovery Registration

- Failover—The phone performs a failover when transport timeout/failure or TCP connection failures; if Try Backup RSC and Retry Reg RSC values are datafilled.
- Recovery—The phone attempts to reregister with the primary proxy while registered or actively connected to the secondary proxy.

  Auto register when failover parameter controls the failover behavior when there is an error. When this parameter is set to yes, the phone re-registers upon failover or recovery.

Fallback Behavior

The fallback occurs when the current registration expires or Proxy Fallback Intvl fires.

If the Proxy Fallback Intvl is exceeded, all the new SIP messages go to primary proxy.

For example, when the value for Register Expires is 3600 seconds and Proxy Fallback Intvl is 600 seconds, the fallback triggers 600 seconds later.

When the value for Register Expires is 800 seconds and Proxy Fallback Intvl is 1000 seconds, the fallback triggers at 800 seconds.

After successful registration back to the primary server, all SIP messages go to the primary server.

RFC3311

The Cisco IP Phone supports RFC-3311, the SIP UPDATE Method.

SIP NOTIFY XML-Service

The Cisco IP Phone supports the SIP NOTIFY XML-Service event. On receipt of a SIP NOTIFY message with an XML-Service event, the phone challenges the NOTIFY with a 401 response if the message does not contain correct credentials. The client must furnish the correct credentials using MD5 digest with the SIP account password for the corresponding line of the IP phone.

The body of the message can contain the XML event Message. For example:

```
<CiscoIPPhoneExecute>
  <ExecuteItem Priority="0" URL="http://xmlserver.com/event.xml"/>
</CiscoIPPhoneExecute>
```

Authentication:

```
challenge  = MD5( MD5(A1) ":" nonce ":" nc-value ":" cnonce ":" qop-value
":" MD5(A2) )
where A1 = username ":" realm ":" passwd
and A2 = Method ":" digest-uri
```

NAT Transversal with Phones

Network Address Translation (NAT) allows multiple devices to share a single, public, routable, IP address to establish connections over the Internet. NAT is present in many broadband access devices to translate public and private IP addresses. For VoIP to coexist with NAT, NAT traversal is required.

Not all service providers provide NAT traversal. If your service provider does not provide NAT traversal, you have several options:

- **NAT Mapping with Session Border Controller**: We recommend that you choose an service provider that supports NAT mapping through a Session Border Controller. With NAT mapping provided by the service provider, you have more choices in selecting a router.
- **NAT Mapping with SIP-ALG Router**: NAT mapping can be achieved by using a router that has a SIP Application Layer Gateway (ALG). By using a SIP-ALG router, you have more choices in selecting an service provider.
- **NAT Mapping with a Static IP Address**: NAT mapping with an external (public) static IP address can be acheived to ensure interoperability with the service provider. The NAT mechanism used in the router must be symmetric. For more information, see Determine Symmetric or Asymmetric NAT.

  Use NAT mapping only if the service provider network does not provide a Session Border Controller functionality. For more information on how to configure NAT mapping with a static IP, see Configure NAT Mapping with the Static IP Address.
- **NAT Mapping with STUN**: If the service provider network does not provide a Session Border Controller functionality and if the other requirements are met, it is possible to use Session Traversal Utilities for NAT (STUN) to discover the NAT mapping. For information on how to configure NAT mapping with STUN, see Configure NAT mapping with STUN.

NAT Mapping with Session Border Controller

We recommend that you choose an service provider that supports NAT mapping through a Session Border Controller. With NAT mapping provided by the service provider, you have more choices in selecting a router.

NAT Mapping with SIP-ALG Router

NAT mapping can be achieved by using a router that has a SIP Application Layer Gateway (ALG). By using a SIP-ALG router, you have more choices in selecting an service provider.

Cisco Discovery Protocol

Cisco Discovery Protocol (CDP) is negotiation-based and determines which virtual LAN (VLAN) the Cisco IP Phone resides in. If you are using a Cisco switch, Cisco Discovery Protocol (CDP) is available and is enabled by default. CDP has these attributes:

- Obtains the protocol addresses of neighboring devices and discovers the platform of those devices.
- Shows information about the interfaces your router uses.
- Is media and protocol-independent.

If you are using a VLAN without CDP, you must enter a VLAN ID for the Cisco IP Phone.

LLDP-MED

The Cisco IP Phone supports Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED) for deployment with Cisco or other Third-Party network connectivity devices that use a Layer 2 auto discovery mechanism. Implementation of LLDP-MED is done in accordance with IEEE 802.1AB (LLDP) Specification of May 2005, and ANSI TIA-1057 of April 2006.

The Cisco IP Phone operates as a LLDP-MED Media End Point Class III device with direct LLDP-MED links to Network Connectivity Devices, according to the Media Endpoint Discovery Reference Model and Definition (ANSI TIA-1057 Section 6).

The Cisco IP Phone supports only the following limited set of Type-Length-Values (TLV) as an LLDP-MED Media Endpoint device class III:

- Chassis ID TLV
- Port ID TLV
- Time to live TLV
- Port Description TLV
- System Name TLV
- System Capabilities TLV
- IEEE 802.3 MAC/PHY Configuration/Status TLV (for wired network only)
- LLDP-MED Capabilities TLV
- LLDP-MED Network Policy TLV (for application type=Voice only)
- LLDP-MED Extended Power-Via-MDI TLV (for wired network only)
- LLDP-MED Firmware Revision TLV
- End of LLDPDU TLV

The outgoing LLDPDU contains all the preceding TLVs if applicable. For the incoming LLDPDU, the LLDPDU is discarded if any of the following TLVs are missing. All other TLVs are not validated and ignored.

- Chassis ID TLV
- Port ID TLV
- Time to live TLV
- LLDP-MED Capabilities TLV
- LLDP-MED Network Policy TLV (for application type=Voice only)
- End of LLDPDU TLV

The Cisco IP Phone sends out the shutdown LLDPDU if applicable. The LLDPDU frame contains the following TLVs:

- Chassis ID TLV
- Port ID TLV

- Time to live TLV
- End of LLDPDU TLV

There are some restrictions in the implementation of LLDP-MED on the Cisco IP Phones:

- Storage and retrieval of neighbor information are not supported.
- SNMP and corresponding MIBs are not supported.
- Recording and retrieval of statistical counters are not supported.
- Full validation of all TLVs does not take place; TLVs that do not apply to the phones are ignored.
- Protocol state machines as stated in the standards are used only for reference.

Chassis ID TLV

For the outgoing LLDPDU, the TLV supports subtype=5 (Network Address). When the IP address is known, the value of the Chassis ID is an octet of the INAN address family number followed by the octet string for the IPv4 address used for voice communication. If the IP address is unknown, the value for the Chassis ID is `0.0.0.0`. The only INAN address family supported is IPv4. Currently, the IPv6 address for the Chassis ID is not supported.

For the incoming LLDPDU, the Chassis ID is treated as an opaque value to form the MSAP identifier. The value is not validated against its subtype.

The Chassis ID TLV is mandatory as the first TLV. Only one Chassis ID TLV is allowed for the outgoing and incoming LLDPDUs.

Port ID TLV

For the outgoing LLDPDU, the TLV supports subtype=3 (MAC address). The 6 octet MAC address for the Ethernet port is used for the value of Port ID.

For the incoming LLDPDU, the Port ID TLV is treated as an opaque value to form the MSAP identifier. The value is not validated against its subtype.

The Port ID TLV is mandatory as the second TLV. Only one Port ID TLV is allowed for the outgoing and incoming LLDPDUs.

Time to Live TLV

For the outgoing LLDPDU, the Time to Live TTL value is 180 seconds. This differs from the 120-second value that the standard recommends. For the shutdown LLDPDU, the TTL value is always 0.

The Time to Live TLV is mandatory as the third TLV. Only one Time to Live TLV is allowed for the outgoing and incoming LLDPDUs.

End of LLDPDU TLV

The value is 2-octet, all zero. This TLV is mandatory and only one is allowed for the outgoing and incoming LLDPDUs.

Port Description TLV

For the outgoing LLDPDU, in the Port Description TLV, the value for the port description is the same as "Port ID TLV" for CDP. The incoming LLDPDU, the Port Description TLV, is ignored and not validated. Only one Port Description TLV is allowed for outgoing and incoming LLDPDUs.

System Name TLV

For the Cisco IP Phone, the value is SEP+MAC address.

Example: SEPAC44F211B1D0

The incoming LLDPDU, the System Name TLV, is ignored and not validated. Only one System Name TLV is allowed for the outgoing and incoming LLDPDUs.

System Capabilities TLV

For the outgoing LLDPDU, in the System Capabilities TLV, the bit values for the 2 octet system capabilities fields should be set for Bit 2 (Bridge) and Bit 5 (Phone) for a phone with a PC port. If the phone does not have a PC port, only Bit 5 should be set. The same system capability value should be set for the enabled capability field.

For the incoming LLDPDU, the System Capabilities TLV is ignored. The TLV is not validated semantically against the MED device type.

The System Capabilities TLV is mandatory for outgoing LLDPDUs. Only one System Capabilities TLV is allowed.

Management Address TLV

The TLV identifies an address associated with the local LLDP agent (that may be used to reach higher layer entities) to assist discovery by network management. The TLV allows the inclusion of both the system interface number and an object identifier (OID) that are associated with this management address, if either or both are known.

- TLV information string length—This field contains the length (in octets) of all the fields in the TLV information string.
- Management address string length—This field contains the length (in octets) of the management address subtype + management address fields.

System Description TLV

The TLV allows the network management to advertise the system description.

- TLV information string length—This field indicates the exact length (in octets) of the system description.
- System description—This field contains an alphanumeric string that is the textual description of the network entity. The system description includes the full name and version identification of the system hardware type, software operating system, and networking software. If implementations support IETF RFC 3418, the sysDescr object should be used for this field.

IEEE 802.3 MAC/PHY Configuration/Status TLV

The TLV is not for autonegotiation, but for troubleshooting purposes. For the incoming LLDPDU, the TLV is ignored and not validated. For the outgoing LLDPDU, for the TLV, the octet value autonegotiation support/status should be:

- Bit 0—Set to 1 to indicate that the autonegotiation support feature is supported.
- Bit 1—Set to 1 to indicate that autonegotiation status is enabled.
- Bit 2-7—Set to 0.

The bit values for the 2 octets PMD autonegotiation advertised capability field should be set to:

- Bit 13—10BASE-T half duplex mode
- Bit 14—10BASE-T full duplex mode
- Bit 11—100BASE-TX half duplex mode
- Bit 10—100BASE-TX full duplex mode
- Bit 15—Unknown

Bit 10, 11, 13 and 14 should be set.

The value for 2 octets operational MAU type should be set to reflect the real operational MAU type:

- 16—100BASE-TX full duplex
- 15—100BASE-TX half duplex
- 11—10BASE-T full duplex
- 10—10BASE-T half duplex

For example, usually, the phone is set to 100BASE-TX full duplex. The value 16 should then be set. The TLV is optional for a wired network and not applicable for a wireless network. The phone sends out this TLV only when in wired mode. When the phone is not set for autonegotiation but specific speed/duplexity, for the outgoing LLDPDU TLV, bit 1 for the octet value autonegotiation support/status should be clear (0) to indicate that autonegotiation is disabled. The 2 octets PMD autonegotiation advertised capability field should be set to 0x8000 to indicate unknown.

LLDP-MED Capabilities TLV

For the outgoing LLDPDU, the TLV should have the device type 3 (End Point Class III) with the following bits set for 2-octet Capability field:

Bit Position	Capability
0	LLDP-MED Capabilities
1	Network Policy
4	Extended Power via MDI-PD
5	Inventory

For the incoming TLV, if the LLDP-MED TLV is not present, the LLDPDU is discarded. The LLDP-MED Capabilities TLV is mandatory and only one is allowed for the outgoing and incoming LLDPDUs. Any other LLDP-MED TLVs will be ignored if they present before the LLDP-MED Capabilities TLV.

Network Policy TLV

In the TLV for the outgoing LLDPDU, before the VLAN or DSCP is determined, the Unknown Policy Flag (U) is set to 1. If the VLAN setting or DSCP is known, the value is set to 0. When the policy is unknown, all other values are set to 0. Before the VLAN is determined or used, the Tagged Flag (T) is set to 0. If the tagged VLAN (VLAN ID > 1) is used for the phone, the Tagged Flag (T) is set to 1. Reserved (X) is always set to 0. If the VLAN is used, the corresponding VLAN ID and L2 Priority will be set accordingly. VLAN ID valid value is range from 1-4094. However, VLAN ID=1 will never be used (limitation). If DSCP is used, the value range from 0-63 is set accordingly.

In the TLV for the incoming LLDPDU, Multiple Network Policy TLVs for different application types are allowed.

LLDP-MED Extended Power-Via-MDI TLV

In the TLV for the outgoing LLDPDU, the binary value for Power Type is set to "0 1" to indicate the power type for phone is PD Device. The Power source for the phone is set to "PSE and local" with binary value "1 1". The Power Priority is set to binary "0 0 0 0" to indicate unknown priority while the Power Value is set to maximum power value. The Power Value for the Cisco IP Phone is 12900mW.

For the incoming LLDPDU, the TLV is ignored and not validated. Only one TLV is allowed in the outgoing and incoming LLDPDUs. The phone will send out the TLV for the wired network only.

The LLDP-MED standard was originally drafted in the context of Ethernet. Discussion is ongoing for LLDP-MED for Wireless Networks. Refer to ANSI-TIA 1057, Annex C, C.3 Applicable TLV for VoWLAN, table 24. It is recommended that the TLV is not applicable in the context of the wireless network. This TLV is targeted for use in the context of PoE and Ethernet. The TLV, if added, will not provide any value for network management or power policy adjustment at the switch.

LLDP-MED Inventory Management TLV

This TLV is optional for Device Class III. For the outgoing LLDPDU, we support only Firmware Revision TLV. The value for the Firmware Revision is the version of firmware on the phone. For the incoming LLDPDU, the TLVs are ignored and not validated. Only one Firmware Revision TLV is allowed for the outgoing and incoming LLDPDUs.

Final Network Policy Resolution and QoS

Special VLANs

VLAN=0, VLAN=1, and VLAN=4095 are treated the same way as an untagged VLAN. Because the VLAN is untagged, Class of Service (CoS) is not applicable.

Default QoS for SIP Mode

If there is no network policy from CDP or LLDP-MED, the default network policy is used. CoS is based on configuration for the specific extension. It is applicable only if the manual VLAN is enabled and manual VLAN ID is not equal to 0, 1, or 4095. Type of Service (ToS) is based on configuration for the specific extension.

QoS Resolution for CDP

If there is a valid network policy from CDP:

- If the VLAN=0, 1, or 4095, the VLAN will not be set, or the VLAN is untagged. CoS is not applicable, but DSCP is applicable. ToS is based on the default as previously described.
- If the VLAN > 1 and VLAN < 4095, the VLAN is set accordingly. CoS and ToS are based on the default as previously described. DSCP is applicable.
- The phone reboots and restarts the fast start sequence.

QoS Resolution for LLDP-MED

If CoS is applicable and if CoS = 0, the default is used for the specific extension as previously described. But the value shown on L2 Priority for TLV for outgoing LLDPDU is based on the value used for extension 1. If CoS is applicable and if CoS != 0, CoS is used for all extensions.

If DSCP (mapped to ToS) is applicable and if DSCP = 0, the default is used for the specific extension as previously described. But the value show on DSCP for TLV for outgoing LLDPDU is based on value used for the extension 1. If DSCP is applicable and if DSCP != 0, DSCP is used for all extensions.

If the VLAN > 1 and VLAN < 4095, the VLAN is set accordingly. CoS and ToS are based on the default as previously described. DSCP is applicable.

If there is a valid network policy for the voice application from LLDP-MED PDU and if the tagged flag is set, the VLAN, L2 Priority (CoS), and DSCP (mapped to ToS) are all applicable.

If there is a valid network policy for the voice application from LLDP-MED PDU and if the tagged flag is not set, only the DSCP (mapped to ToS) is applicable.

The Cisco IP Phone reboots and restarts the fast start sequence.

Coexistence with CDP

If both CDP and LLDP-MED are enabled, the network policy for the VLAN determines the last policy set or changed with either one of the discovery modes. If both LLDP-MED and CDP are enabled, during startup the phone sends CDP and LLDP-MED PDUs.

Inconsistent configuration and behavior for network connectivity devices for CDP and LLDP-MED modes could result in an oscillating rebooting behavior for the phone due to switching to different VLANs.

If the VLAN is not set by CDP and LLDP-MED, the VLAN ID that is configured manually is used. If the VLAN ID is not configured manually, no VLAN is supported. DSCP is used and the network policy determines LLDP-MED if applicable.

LLDP-MED and Multiple Network Devices

You can use the same application type for network policy. However, phones receive different Layer 2 or Layer 3 QoS Network policies from multiple network connectivity devices. In such a case, the last valid network policy is accepted.

LLDP-MED and IEEE 802.X

The Cisco IP Phone does not support IEEE 802.X and does not work in a 802.1X wired environment. However, IEEE 802.1X or Spanning Tree Protocols on network devices could result in delay of fast start response from switches.