Cisco IP Phone Installation

Verify the Network Setup

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

Procedure

Step 1 Configure a VoIP Network to meet the following requirements:

- VoIP is configured on your routers and gateways.

Step 2 Set up the network to support one of the following:

- DHCP support
- Manual assignment of IP address, gateway, and subnet mask

Install the Conference Phone

After the phone connects to the network, the phone startup process begins, and the phone registers with Third-party Call Control System. You need to configure the network settings on the phone if you disable the DHCP service.

If you used autoregistration, you need to update the specific configuration information for the phone such as associating the phone with a user, changing the button table, or directory number.

After the phone connects, it determines if a new firmware load should be installed on the phone.

Procedure

Step 1 Choose the power source for the phone:

- Power over Ethernet (PoE)

For more information, see Ways to Provide Power to Your Conference Phone , on page 2.

Step 2 Connect the phone to the switch.

- If you use PoE, plug the Ethernet cable into the LAN port and plug the other end into the phone.

Each phone ships with one Ethernet cable in the box.

Step 3 Monitor the phone startup process. This step verifies that the phone is configured properly.

Step 4 If you do not use autoregistration, manually configure the network settings on the phone.

See Configure the Network from the Phone, on page 3.

Step 5 Make calls with the phone to verify that the phone and features work correctly.

Step 6 Provide information to end users about how to use their phones and how to configure their phone options. This step ensures that users have adequate information to successfully use their Cisco phones.

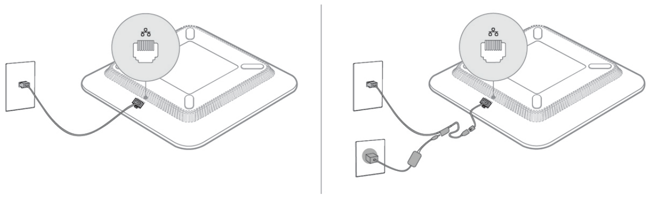
Ways to Provide Power to Your Conference Phone

Your conference phone needs power from one of these sources:

- Power over Ethernet (PoE), which your network supplies.
- Cisco IP Phone Power Injector.

The following figure shows the PoE and PoE power cable power options.

Figure 1: Conference Phone Power Options



Configure the Network from the Phone

The phone includes many configurable network settings that you may need to modify before it is functional for your users. You can access these setting through the phone menus.

The Network configuration menu provides you with options to view and configure a variety of network settings.

You can configure settings that are display-only on the phone in your Third-Party Call Control system.

Procedure

Step 1 Press **Settings**.
Step 2 Select **Network configuration**.
Step 3 Use the navigation arrows to select the desired menu and edit.
Step 4 To display a submenu, repeat step 3.
Step 5 To exit a menu, press **Back** .

Network Configuration Fields

Table 1: Network Configurations Menu Options

Field	Field Type or Choices	Default	Description
Ethernet configuration			See the following Ethernet configuration submenu table.
IP mode	Dual mode IPv4 only IPv6 only	Dual mode	Select the Internet Protocol mode for which the phone operates. In dual mode, the phone can have both IPv4 and IPv6 addresses.
IPv4 address settings	DHCP Static IP Release DHCP IP	DHCP	See the IPv4 address submenu table in the following tables.

Field	Field Type or Choices	Default	Description
IPv6 address settings	DHCP Static IP	DHCP	See the IPv6 address submenu table in the following tables.
DHCPv6 option to use		17, 160, 159	Indicates the order in which the phone uses the IPv6 addresses provided by DHCP server.
HTTP proxy settings			See the following HTTP proxy settings submenu table.
Web server	On Off	On	Indicates whether the phone has web server enabled or disabled.

Table 2: Ethernet Configuration Submenu

Field	Field Type or Choices	Default	Description
802.1x authentication	Device authentication	Off	Enables or disables the 802.1x authentication. Valid options are: • On • Off

	Transaction status	Disabled	• Transaction status—Indicates different authentication status when you turn on 802.1x in the **Device authentication** field.
			  • *Connecting*: Indicates that the authentication process is in progress.
			  • *Authenticated*: Indicates that the phone is authenticated.
			  • *Disabled*: Indicates that 802.1x authentication is disabled on the phone.
			• Protocol—Displays the protocol of the server.

Switch port config	Auto 10MB half 10MB full 100MB half 100MB full	Auto	Select speed and duplex of the network port. If the phone is connected to a switch, configure the port on the switch to the same speed/duplex as the phone, or configure both to autonegotiate.

Field	Field Type or Choices	Default	Description
CDP	On Off	On	Enable or disable 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.
LLDP-MED	On Off	On	Enable or disable LLDP-MED. LLDP-MED enables the phone to advertise itself to devices that use the discovery protocol.
Startup delay		3 seconds	Set a value that causes a delay for the switch to get to the forwarding state before the phone sends out the first LLDP-MED packet. For configuration of some switches, you might need to increase this value to a higher value for LLDP-MED to work. Configuring a delay can be important for networks that use the Spanning Tree Protocol. Default delay is 3 seconds.
VLAN	On Off	Off	Enable or disable VLAN. Permits you to enter a VLAN ID when you use VLAN without CDP or LLDP. When you use a VLAN with CDP or LLDP, that associated VLAN takes precedent over the VLAN ID you manually entered.
VLAN ID		1	Enter a VLAN ID for the IP phone when you use a VLAN without CDP (VLAN enabled and CDP disabled). Note that only voice packets are tagged with the VLAN ID. Do not use the 1 value for the VLAN ID. If VLAN ID is 1, you cannot tag voice packets with the VLAN ID.
PC port mirror	On Off	Off	Adds the ability to port mirror on the PC port. When enabled, you can see the packets on the phone. Select **On** to enable PC port mirroring and select **Off** to disable it.

Field	Field Type or Choices	Default	Description
DHCP VLAN option			Enter a predefined DHCP VLAN option to learn the voice VLAN ID. When you use a VLAN ID with CDP, LLDP, or manually select a VLAN ID, that VLAN ID takes precedent over the selected DHCP VLAN option. Valid values are: • Null • 128 to 149 • 151 to 158 • 161 to 254 Default value is null. Cisco recommends that you use DHCP Option 132.

Table 3: IPv4 Address Settings Submenu

Field	Field Type or Choices	Default	Description
Connection type	DHCP		Indicates whether the phone has DHCP enabled. • DNS1—Identifies the primary Domain Name System (DNS) server that the phone uses. • DNS2—Identifies the secondary Domain Name System (DNS) server that the phone uses. • DHCP address released—Releases the IP address that DHCP assigned. You can edit this field if DHCP is enabled. To remove the phone from the VLAN and release the IP address for reassignment, set this field to Yes and press **Set**.
	Static IP		When DHCP is disabled, you must set the Internet Protocol (IP) address of the phone. • Static IP address—Identifies the IP that you assign to the phone. The phone uses this IP address instead of acquiring an IP from the DHCP server on the network. • Subnet Mask—Identifies the subnet mask used by the phone. When DHCP is disabled, you must set the subnet mask. • Gateway address—Identifies the default router used by the phone. • DNS1—Identifies the primary Domain Name System (DNS) server that the phone uses. When DHCP is disabled, you must set this field manually. • DNS2—Identifies the primary Domain Name System (DNS) server that the phone uses. When DHCP is disabled, you must set this field manually. When you assign an IP address using this field, you must also assign a subnet mask and a gateway address. See the Subnet Mask and Default Router fields in this table.

Table 4: IPv6 Address Settings Submenu

Field	Field Type or Choices	Default	Description
Connection type	DHCP		Indicates whether the phone has Dynamic Host Configuration Protocol (DHCP) enabled. • DNS1—Identifies the primary DNS server that the phone uses. • DNS2—Identifies the secondary DNS server that the phone uses. • Broadcast Echo—Identifies if the phone responses to multicast ICMPv6 message with destination address of ff02::1. • Auto config— Identifies if the phone uses automatic configuration for the address.
	Static IP		When DHCP is disabled, you must set the Internet Protocol (IP) address of the phone and must set the values of the fields: • Static IP—Identifies the IP that you assign to the phone. The phone uses this IP address instead of acquiring an IP from the DHCP server on the network. • Prefix length—Identifies how many bits of a Global Unicast IPv6 Address are there in the network part. • Gateway—Identifies the default router used by the phone. • Primary DNS—Identifies the primary DNS server that the phone uses. When DHCP is disabled, you must set this field manually. • Secondary DNS—Identifies the primary DNS server that the phone uses. When DHCP is disabled, you must set this field manually. • Broadcast Echo—Identifies if the phone responses to multicast ICMPv6 message with destination address of ff02::1.

Table 5: HTTP Proxy Settings Submenu

<table>
<tr><th>Field</th><th>Field Type or Choices</th><th>Description</th></tr>
<tr><td rowspan="3">Proxy mode</td><td>Auto</td><td>Auto discovery (WPAD)—Enables or disables the Web Proxy Auto-Discovery protocol to retrieve a Proxy Auto-Configuration (PAC) file. Valid options are:
• On
• Off
If the value is set to Off, you need to further set the following field:
• PAC URL—Specifies the URL address for the PAC file that you want to retrieve. For example:
<code>http://proxy.department.branch.example.com</code>
The default value of Auto discovery (WPAD) is On.</td></tr>
<tr><td>Manual</td><td>• Proxy host—Specifies an IP address or hostname of the proxy server for the phone. The scheme (<code>http://</code> or <code>https://</code>) is not required.
• Proxy port—Specifies a port number of the proxy server.
• Proxy authentication—Selects an option according to the actual situation of the proxy server. If the server requires authentication credentials to grant access to the phone, then select On. Otherwise, select Off. Options are:
• Off
• On
If the value is set to On, you need to further set the following fields:
• Username—Specifies the username of a credential user on the proxy server.
• Password—Provides the specified user's password to pass the authentication of the proxy server.
The default value of Proxy authentication is Off.</td></tr>
<tr><td>Off</td><td>Disables the HTTP proxy feature on the phone.</td></tr>
</table>

Text and Menu Entry From the Phone

When you edit the value of an option setting, follow these guidelines:

- Use the arrows on the navigation pad to highlight the field that you wish to edit. Press **Select** in the navigation pad to activate the field. After the field is activated, you can enter values.
- Use the keys on the keypad to enter numbers and letters.

- To enter letters by using the keypad, use a corresponding number key. Press the key one or more times to display a particular letter. For example, press the **2** key once for “a,” twice quickly for “b,” and three times quickly for “c.” After you pause, the cursor automatically advances to allow you to enter the next letter.
- Press the softkey ⌫ if you make a mistake. This softkey deletes the character to the left of the cursor.
- Press **Back** before pressing **Set** to discard any changes that you made.
- To enter a period (for example, in an IP address), press * on the keypad.

Note The Cisco IP Phone provides several methods to reset or restore option settings, if necessary.

Verify Phone Startup

After the Cisco IP Phone has power connected to it, the phone automatically cycles through a startup diagnostic process.

Procedure

Step 1 If you are using Power over Ethernet, plug the LAN cable into the Network port.

Step 2 If you are using the power cube, connect the cube to the phone and plug the cube into an electrical outlet.

The buttons flash amber and then green in sequence during the various stages of bootup as the phone checks the hardware.

If the phone completes these stages successfully, it has started up properly.

Disable or Enable DF Bit

You can disable or enable Don't Fragment (DF) bit in the TCP, UDP, or ICMP messages to determine whether a packet is allowed to be fragmented.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **System**.

Step 2 In the **Network Settings** section, configure the parameter **Disable DF**.

- If you set the **Disable DF** to **Yes**, the Don't Fragment (DF) bit is disabled. In this case, the network can fragment an IP packet. This is the default behaviour.
- If you set the **Disable DF** to **No**, the Don't Fragment (DF) bit is enabled. In this case, the network can't fragment an IP packet. This setting doesn't allow fragmentation in cases where the receiving host doesn't have sufficient resources to reassemble internet fragments.

Step 3 Click **Submit All Changes**.

You can also configure the parameter in the phone configuration file (cfg.xml) with the following XML string:

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

Allowed values: Yes and No

Default: Yes

Configure Internet Connection Type

You can choose how your phone receives an IP address. Set the connection type to one of the following:

- Static IP—A static IP address for the phone.
- Dynamic Host Configuration Protocol (DHCP)—Enables the phone to receive an IP address from the network DHCP server.

The Cisco IP phone typically operates in a network where a DHCP server assigns IP addresses to devices. Because IP addresses are a limited resource, the DHCP server periodically renews the phone lease on the IP address. If a phone loses the IP address, or if the IP address is assigned to another device on the network, the following occurs:

- Communication between the SIP proxy and the phone is severed or degraded.

The DHCP Timeout on Renewal parameter causes the phone to request renewal of its IP address if the following occurs:

- The phone doesn't receive an expected SIP response within programmable length of time after it sends a SIP command.

If the DHCP server returns the IP address that it originally assigned to the phone, the DHCP assignment is presumed to be operating correctly. Otherwise, the phone resets to try to fix the issue.

Before you begin

Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **System**.

Step 2 In the **IPv4 Settings** section, use the **Connection Type** drop-down list to choose the connection type:

- Dynamic Host Configuration Protocol (DHCP)
- Static IP

Step 3 In the **IPv6 Settings** section, use the **Connection Type** drop-down list to choose the connection type:

- Dynamic Host Configuration Protocol (DHCP)
- Static IP

Step 4 If you choose Static IP, configure these settings in the **Static IP Settings** section:

- **Static IP**—Static IP address of the phone
- **NetMask**—Netmask of the phone (IPv4, only)
- **Gateway**—Gateway IP address

Step 5 Click **Submit All Changes**.

In the phone configuration XML file (cfg.xml), enter a string in this format:

```
<Connection_Type ua="rw">DHCP</Connection_Type>
<!-- available options: DHCP|Static IP -->
<Static_IP ua="rw"/>
<NetMask ua="rw"/>
<Gateway ua="rw"/>
```

Configure VLAN Settings

The software tags your phone voice packets with the VLAN ID when you use a virtual LAN (VLAN).

In the VLAN Settings section of the **Voice** > **System** window, you can configure the different settings:

- LLDP-MED
- Cisco Discovery Protocol (CDP)
- Network Startup Delay
- VLAN ID (manual)
- DHCP VLAN Option

The multiplatform phones support these four methods to obtain VLAN ID information. The phone attempts to obtain the VLAN ID information in this order:

1. LLDP-MED
2. Cisco Discovery Protocol (CDP)
3. VLAN ID (manual)
4. DHCP VLAN Option

Before you begin

- Access the phone administration web page. See Access the Phone Web Interface.
- Disable CDP/LLDP and manual VLAN.

Procedure

Step 1 Select **Voice** > **System**.

Step 2 In the **VLAN Settings** section, configure the parameters as defined in the VLAN Settings Parameters, on page 13 table.

Step 3 Click **Submit All Changes**.

You can also configure the parameters in the phone configuration file with XML(cfg.xml) code. To configure each parameter, see the syntax of the string in the VLAN Settings Parameters, on page 13 table.

VLAN Settings Parameters

The following table defines the function and usage of each parameter in the **VLAN Settings Parameters** section under the **System** tab in the phone web page. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Parameter Name	Description and Default Value
Enable VLAN	Controls the VLAN feature. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Enable_VLAN ua="rw">No</Enable_VLAN> • In the phone web interface, set to **Yes** to enable VLAN. The default value is **Yes**.
VLAN ID	If you use a VLAN without CDP (VLAN enabled and CDP disabled), enter a VLAN ID for the IP phone. Note that only voice packets are tagged with the VLAN ID. Do not use 1 for the VLAN ID. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <VLAN_ID ua="rw">1</VLAN_ID> • In the phone web interface, enter an appropriate value. Valid values: An integer ranging from 0 through 4095 Default: 1

Parameter Name	Description and Default Value
Enable CDP	Enable CDP only if you are using a switch that has Cisco Discovery Protocol. CDP is negotiation based and determines which VLAN the IP phone resides in. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Enable_CDP ua="na">Yes</Enable_CDP> • In the phone web page: set to **Yes** to enable CDP. Valid values: Yes/No Default: Yes
Enable LLDP-MED	Choose **Yes** to enable LLDP-MED for the phone to advertise itself to devices that use that discovery protocol. When the LLDP-MED feature is enabled, after the phone has initialized and Layer 2 connectivity is established, the phone sends out LLDP-MED PDU frames. If the phone receives no acknowledgment, the manually configured VLAN or default VLAN will be used if applicable. If the CDP is used concurrently, the waiting period of 6 seconds is used. The waiting period will increase the overall startup time for the phone. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Enable_LLDP-MED ua="na">Yes</Enable_LLDP-MED> • In the phone web interface, set to **Yes** to enable LLDP-MED. Valid values: Yes/No Default: Yes

Parameter Name	Description and Default Value
Network Startup Delay	Setting this value causes a delay for the switch to get to the forwarding state before the phone will send out the first LLDP-MED packet. The default delay is 3 seconds. For configuration of some switches, you might need to increase this value to a higher value for LLDP-MED to work. Configuring a delay can be important for networks that use Spanning Tree Protocol. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Network_Startup_Delay ua="na">3</Network_Startup_Delay>` • In the phone web interface, enter the delay in seconds. Valid values: An integer ranging from 1 through 300 Default: 3
DHCP VLAN Option	A predefined DHCP VLAN option to learn the voice VLAN ID. You can use the feature only when no voice VLAN information is available by CDP/LLDP and manual VLAN methods. CDP/LLDP and manual VLAN are all disabled. Set the value to Null to disable DHCP VLAN option. Cisco recommends that you use DHCP Option 132. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<DHCP_VLAN_Option ua="na">132</DHCP_VLAN_Option>` • In the phone web page: specify the DHCP VLAN option.

SIP Configuration

SIP settings for the Cisco IP Phone are configured for the phone in general and for the extensions.

Configure the Basic SIP Parameters

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **SIP**.

Step 2 In the **SIP Parameters** section, set the parameters as described in the SIP Parameters, on page 16 table.

Step 3 Click **Submit All Changes**.

SIP Parameters

Parameter	Description
Max Forward	Specifies SIP Max Forward value. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Max_Forward ua="na">70</Max_Forward>` • In the phone web page, enter an appropriate value. Value range: 1 to 255 Default: 70
Max Redirection	Specifies number of times an invite can be redirected to avoid an infinite loop. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Max_Redirection ua="na">5</Max_Redirection>` • In the phone web page, enter an appropriate value. Default: 5
Max Auth	Specifies the maximum number of times (from 0 to 255) a request can be challenged. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Max_Auth ua="na">2</Max_Auth>` • In the phone web page, enter an appropriate value. Allowed value: 0 to 255 Default: 2

Parameter	Description
SIP User Agent Name	Used in outbound requests. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <SIP_User_Agent_Name ua="na">$VERSION</SIP_User_Agent_Name> • In the phone web page, enter an appropriate name. Default: $VERSION If empty, the header is not included. Macro expansion of $A to $D corresponding to GPP_A to GPP_D allowed
SIP Server Name	Server header used in responses to inbound responses. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <SIP_Server_Name ua="na">$VERSION</SIP_Server_Name> • In the phone web page, enter an appropriate name. Default: $VERSION
SIP Reg User Agent Name	User-Agent name to be used in a REGISTER request. If this is not specified, the SIP User Agent Name is also used for the REGISTER request. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <SIP_Reg_User_Agent_Name ua="na">agent name</SIP_Reg_User_Agent_Name> • In the phone web page, enter an appropriate name. Default: Blank
SIP Accept Language	Accept-Language header used. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <SIP_Accept_Language ua="na">en</SIP_Accept_Language> • In the phone web page, enter an appropriate language. There is no default. If empty, the header is not included.

Parameter	Description
DTMF Relay MIME Type	MIME Type used in a SIP INFO message to signal a DTMF event. This field must match that of the Service Provider. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <DTMF_Relay_MIME_Type ua="na">application/dtmf-relay</DTMF_Relay_MIME_Type> • In the phone web page, enter an appropriate MIME type. Default: application/dtmf-relay
Hook Flash MIME Type	MIME Type used in a SIPINFO message to signal a hook flash event. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Hook_Flash_MIME_Type ua="na">application/hook-flash</Hook_Flash_MIME_Type> • In the phone web page, enter an appropriate MIME type for a SIPINFO message. Default:
Remove Last Reg	Enables you to remove the last registration before registering a new one if the value is different. Set to Yes to remove the last registration. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Remove_Last_Reg ua="na">No</Remove_Last_Reg> • In the phone web page, Select Yes or No. Allowed values: Yes or No Default: No

Parameter	Description
Use Compact Header	If set to yes, the phone uses compact SIP headers in outbound SIP messages. If inbound SIP requests contain normal headers, the phone substitutes incoming headers with compact headers. If set to no, the phones use normal SIP headers. If inbound SIP requests contain compact headers, the phones reuse the same compact headers when generating the response, regardless of this setting. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Use_Compact_Header ua="na">No</Use_Compact_Header> • In the phone web page, select Yes or No. Allowed values: Yes or No Default: No
Escape Display Name	Enables you to keep the Display Name private. Set to Yes if you want the IP phone to enclose the string (configured in the Display Name) in a pair of double quotes for outbound SIP messages. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Escape_Display_Name ua="na">No</Escape_Display_Name> • In the phone web page, select Yes or No. Allowed values: Yes or No Default: Yes.
Talk Package	Enables support for the BroadSoft Talk Package that lets users answer or resume a call by clicking a button in an external application. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Talk_Package ua="na">No</Talk_Package> • In the phone web page, select Yes to enable the Talk Package. Allowed values: Yes or No Default: No

Parameter	Description
Hold Package	Enables support for the BroadSoft Hold Package, which lets users place a call on hold by clicking a button in an external application. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Hold_Package ua="na">No</Hold_Package>` • In the phone web page, select Yes to enable support for the Hold Package. Allowed values: Yes or No Default: No
Conference Package	Enables support for the BroadSoft Conference Package that enables users to start a conference call by clicking a button in an external application. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Conference_Package ua="na">No</Conference_Package>` • In the phone web page, select Yes or No. Allowed values: Yes or No Default: No
RFC 2543 Call Hold	If set to yes, unit includes c=0.0.0.0 syntax in SDP when sending a SIP re-INVITE to the peer to hold the call. If set to no, unit will not include the c=0.0.0.0 syntax in the SDP. The unit will always include a=sendonly syntax in the SDP in either case. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<RFC_2543_Call_Hold ua="na">Yes</RFC_2543_Call_Hold>` • In the phone web page, Yes or No. Allowed values: Yes or No Default: Yes

Parameter	Description
Random REG CID on Reboot	If set to yes, the phone uses a different random call-ID for registration after the next software reboot. If set to no, the Cisco IP phone tries to use the same call-ID for registration after the next software reboot. The Cisco IP phone always uses a new random Call-ID for registration after a power-cycle, regardless of this setting. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Random_REG_CID_on_Reboot ua="na">No</Random_REG_CID_on_Reboot> • In the phone web page, select Yes or No. Default: No.
SIP TCP Port Min	Specifies the lowest TCP port number that can be used for SIP sessions. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <SIP_TCP_Port_Min ua="na">5060</SIP_TCP_Port_Min> • In the phone web page, enter an appropriate value. Default: 5060
SIP TCP Port Max	Specifies the highest TCP port number that can be used for SIP sessions. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <SIP_TCP_Port_Max ua="na">5080</SIP_TCP_Port_Max> • In the phone web page, enter an appropriate value. Default: 5080

Parameter	Description
Caller ID Header	Provides the option to take the caller ID from PAID-RPID-FROM, PAID-FROM, RPID-PAID-FROM, RPID-FROM, or FROM header. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Caller_ID_Header ua="na">PAID-RPID-FROM</Caller_ID_Header> • In the phone web page, select an option. Allowed values: PAID-RPID-FROM, AID-FROM, RPID-PAID-FROM, RPID-FROM, and FROM Default: PAID-RPID-FROM
Hold Target Before Refer	Controls whether to hold call leg with transfer target before sending REFER to the transferee when initiating a fully-attended call transfer (where the transfer target has answered). Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Hold_Target_Before_Refer ua="na">No</Hold_Target_Before_Refer> • In the phone web page, select Yes or No. Default: No
Dialog SDP Enable	When enabled and the Notify message body is too big causing fragmentation, the Notify message xml dialog is simplified; Session Description Protocol (SDP) is not included in the dialog xml content. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Dialog_SDP_Enable ua="na">No</Dialog_SDP_Enable> • In the phone web page, select **Yes** or **No**. Allowed values: Yes or No Default: No

Parameter	Description
Keep Referee When Refer Failed	If set to yes, it configures the phone to immediately handle NOTIFY sipfrag messages. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Keep_Referee_When_Refer_Failed ua="na">No</Keep_Referee_When_Refer_Failed>` • In the phone web page, select **Yes** or **No**. Allowed values: Yes or No Default: No
Display Diversion Info	Display the Diversion info included in SIP message on LCD or not. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Display_Diversion_Info ua="na">No</Display_Diversion_Info>` • In the phone web page, select **Yes** or **No**. Allowed values: Yes or No
Display Anonymous From Header	Show the caller ID from the SIP INVITE message “From” header when set to Yes, even if the call is an anonymous call. When the parameter is set to no, the phone displays "Anonymous Caller" as the caller ID. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Display_Anonymous_From_Header ua="na">No</Display_Anonymous_From_Header>` • In the phone web page, select **Yes** or **No**. Allowed values: Yes or No Default: No

Parameter	Description
Sip Accept Encoding	Supports the content-encoding gzip feature. If gzip is selected, the SIP message header contains the string “Accept-Encoding: gzip”, and the phone is able to process the SIP message body, which is encoded with the gzip format. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Sip_Accept_Encoding ua="na">none</Sip_Accept_Encoding>` • In the phone web page, enter an appropriate MIME type for a SIPINFO message. Allowed values: none and gzip Default: none
SIP IP Preference	Sets if the phone uses IPv4 or IPv6. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<SIP_IP_Preference ua="na">IPv4</SIP_IP_Preference>` • In the phone web page, select IPv4 or IPv6. Allowed values: IPv4/IPv6 Default: IPv4.
Disable Local Name To Header	Controls the display name in “Directory”, “Call History”, and in the “To” header during an outgoing call. Perform one of the following. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Disable_Local_Name_To_Header ua="na">No</Disable_Local_Name_To_Header>` • In the phone web page, select **Yes** to disable the display name. Allowed values: Yes/No Default: No

Configure the SIP Timer Values

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **SIP**.

Step 2 In the **SIP Timer Values** section, set the SIP timer values in seconds as described in SIP Timer Values (sec), on page 25.

Step 3 Click **Submit All Changes**.

SIP Timer Values (sec)

Parameter	Description
SIP T1	RFC 3261 T1 value (RTT estimate) that can range from 0 to 64 seconds. Default: 0.5 seconds
SIP T2	RFC 3261 T2 value (maximum retransmit interval for non-INVITE requests and INVITE responses) that can range from 0 to 64 seconds. Default: 4 seconds
SIP T4	RFC 3261 T4 value (maximum duration a message remains in the network), which can range from 0 to 64 seconds. Default: 5 seconds.
SIP Timer B	INVITE time-out value, which can range from 0 to 64 seconds. Default: 16 seconds.
SIP Timer F	Non-INVITE time-out value, which can range from 0 to 64 seconds. Default: 16 seconds.
SIP Timer H	INVITE final response, time-out value, which can from 0 to 64 seconds. Default: 16 seconds.
SIP Timer D	ACK hang-around time, which can range from 0 to 64 seconds. Default: 16 seconds.
SIP Timer J	Non-INVITE response hang-around time, which can range from 0 to 64 seconds. Default: 16 seconds.

Parameter	Description
INVITE Expires	INVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Ranges from 0 to 2000000. Default: 240 seconds
ReINVITE Expires	ReINVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Ranges from 0 to 2000000. Default: 30
Reg Min Expires	Minimum registration expiration time allowed from the proxy in the Expires header or as a Contact header parameter. If the proxy returns a value less than this setting, the minimum value is used.
Reg Max Expires	Maximum registration expiration time allowed from the proxy in the Min-Expires header. If the value is larger than this setting, the maximum value is used.
Reg Retry Intv	Interval to wait before the Cisco IP Phone retries registration after failing during the last registration.The range is from 1 to 2147483647 Default: 30 See the note below for additional details.
Reg Retry Long Intvl	When registration fails with a SIP response code that does not match<Retry Reg RSC>, the Cisco IP Phone waits for the specified length of time before retrying. If this interval is 0, the phone stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. Default: 1200 See the note below for additional details.
Reg Retry Random Delay	Random delay range (in seconds) to add to <Register Retry Intvl> when retrying REGISTER after a failure. Minimum and maximum random delay to be added to the short timer. The range is from 0 to 2147483647. Default: 0
Reg Retry Long Random Delay	Random delay range (in seconds) to add to <Register Retry Long Intvl> when retrying REGISTER after a failure. Default: 0

Parameter	Description
Reg Retry Intvl Cap	Maximum value of the exponential delay. The maximum value to cap the exponential backoff retry delay (which starts at the Register Retry Intvl and doubles every retry). Defaults to 0, which disables the exponential backoff (that is, the error retry interval is always at the Register Retry Intvl). When this feature is enabled, the Reg Retry Random Delay is added to the exponential backoff delay value. The range is from 0 to 2147483647. Default: 0
Sub Min Expires	Sets the lower limit of the REGISTER expires value returned from the Proxy server.
Sub Max Expires	Sets the upper limit of the REGISTER minexpires value returned from the Proxy server in the Min-Expires header. Default: 7200.
Sub Retry Intvl	This value (in seconds) determines the retry interval when the last Subscribe request fails. Default: 10.

Note The phone can use a RETRY-AFTER value when it is received from a SIP proxy server that is too busy to process a request (503 Service Unavailable message). If the response message includes a RETRY-AFTER header, the phone waits for the specified length of time before to REGISTER again. If a RETRY-AFTER header is not present, the phone waits for the value specified in the Reg Retry Interval or the Reg Retry Long Interval.

Configure the Response Status Code Handling

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **SIP**.

Step 2 In the **Response Status Code Handling** section, set the values as specified in the Response Status Code Handling Paramters, on page 28 table.

Step 3 Click **Submit All Changes**.

Response Status Code Handling Paramters

The following table defines the function and usage of the parameters in the Response Status Code Handling section under the SIP tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 6: Response Status Code Handling Paramters

Parameter	Description
Try Backup RSC	This parameter may be set to invoke failover upon receiving specified response codes. For example, you can enter numeric values 500 or a combination of numeric values plus wild cards if multiple values are possible. For the later, you can use 5?? to represent all SIP Response messages within the 500 range. If you want to use multiple ranges, you can add a comma "," to delimit values of 5?? and 6?? Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Try_Backup_RSC ua="na"/> • In the phone web page, enter an appropriate value. Default: Blank
Retry Reg RSC	Interval to wait before the phone retries registration after failing during the last registration. For example, you can enter numeric values 500 or a combination of numeric values plus wild cards if multiple values are possible. For the later, you can use 5?? to represent all SIP Response messages within the 500 range. If you want to use multiple ranges, you can add a comma "," to delimit values of 5?? and 6?? Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Retry_Reg_RSC ua="na"/> • In the phone web page, enter an appropriate value. Default: Blank

Configure NTP Server

You can configure NTP servers with IPv4 and IPv6. You can also configure NTP server with DHCPv4 option 42 or DHCPv6 option 56. Configuring NTP with Primary NTP Server and Secondary NTP server parameters has higher priority over configuring NTP with DHCPv4 option 42 or DHCPv6 option 56.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **Systems**.

Step 2 In the **Optional Network Configuration** section, set the IPv4 or IPv6 address as described in the NTP Server Parameters, on page 29 table.

Step 3 Click **Submit All Changes**.

NTP Server Parameters

The following table defines the function and usage of NTP server parameters in the Optional Network Configuration section under the System tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 7: NTP Server Parameters

Parameter	Description
Primary NTP Server	IP address or name of the primary NTP server used to synchronize its time. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Primary_NTP_Server ua="rw"/> • In the phone web page, enter the IP address of the priamry NTP server. Default: Blank

Parameter	Description
Secondary NTP Server	IP address or name of the secondary NTP server used to synchronize its time. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Secondary_NTP_Server ua="rw"/>` • In the phone web page, enter the IP address of the secondary NTP server. Default: Blank

Configure the RTP Parameters

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **SIP**.

Step 2 In the **RTP Parameters** section, set the Real-Time Transport Protocol (RTP) parameter values as described in RTP Parameters, on page 31.

Step 3 Click **Submit All Changes**.

RTP Parameters

The following table defines the function and usage of the parameters in the RTP Parameters section under the SIP tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 8: RTP Parameters

Parameter	Description
RTP Port Min	Minimum port number for RTP transmission and reception. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<RTP_Port_Min ua="na">16384</RTP_Port_Min>` • In the phone web page, enter an appropriate port number. Allowed values: 2048 to 49151 If the value range (**RTP Port Max** - **RTP Port Min**) is less than 16 or you configure the parameter incorrectly, the RTP port range (16382 to 32766) is used instead. Default: 16384
RTP Port Max	Maximum port number for RTP transmission and reception. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<RTP_Port_Max ua="na">16482</RTP_Port_Max>` • In the phone web page, enter an appropriate port number. Allowed values: 2048 to 49151 If the value range (**RTP Port Max** - **RTP Port Min**) is less than 16 or you configure the parameter incorrectly, the RTP port range (16382 to 32766) is used instead. Default: 16482

Parameter	Description
RTP Packet Size	Specifies packet size in seconds. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<RTP_Packet_Size ua="na">0.02</RTP_Packet_Size>` • In the phone web page, enter an appropriate value to specify the packet size. Allowed values: Ranges from 0.01 to 0.13. Valid values must be a multiple of 0.01 seconds. Default: 0.02
Max RTP ICMP Err	Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the phone terminates the call. If value is set to 0, the phone ignores the limit on ICMP errors. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Max_RTP_ICMP_Err ua="na">0</Max_RTP_ICMP_Err>` • In the phone web page, enter an appropriate value. Default: 0
RTCP Tx Interval	Interval for sending out RTCP sender reports on an active connection. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<RTCP_Tx_Interval ua="na">5</RTCP_Tx_Interval>` • In the phone web page, enter an appropriate value. Allowed values: 0 to 255 seconds Default: 0

Parameter	Description
Call Statistics	Specifies whether the phone sends end-of-call statistics within SIP messages when a call terminates or is put on hold. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Call_Statistics ua="na">No</Call_Statistics> • In the phone web page, select **Yes** to enable this feature. Allowed values: Yes and No Default: No
SDP IP Preferences	Select the preferred IP that the phone uses as RTP address. If the phone is in dual-mode and has both ipv4 and ipv6 addresses, it will always include both addresses in SDP by attributes "a=altc … If IPv4 address is selected, then ipv4 address has higher priority than ipv6 address in SDP and indicates that phone prefers using ipv4 RTP address. If the phone has only ipv4 address or ipv6 address, SDP does not have ALTC attributes and RTP address is specified in “c=” line. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <SDP_IP_Preference ua="na">IPv4</SDP_IP_Preference> • In the phone web page, select the preferred IP . Allowed values: IPv4 and IPv6 Default: IPv4

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>RTP Before ACK</td><td>Allows you to specify whether an RTP session starts before or after an ACK is received from the calling party.

Perform one of the following:
In the phone configuration file with XML(cfg.xml), enter a string in this format:
<code><RTP_Before_ACK ua="na">No</RTP_Before_ACK></code>
In the phone web page select:
Yes: An RTP session doesn't await an ACK, but starts after a 200 OK message is sent.
No: An RTP session doesn't start until an ACK is received from the calling party.
Allowed values: Yes and No

Default: No</td></tr>
<tr><td>SSRC Reset on RE-INVITE</td><td>Controls whether to reset the Synchronization Source (SSRC) for the new RTP and SRTP sessions.

Perform one of the following:
In the phone configuration file with XML(cfg.xml), enter a string in this format:
<code><SSRC_Reset_on_RE-INVITE ua="na">Yes</SSRC_Reset_on_RE-INVITE></code>
In the phone web page select:
Yes: the phone can avoid the call transfer error, where only one person on the call hears the audio. This occurs on calls of 30 minutes or longer, and often on three-way calls.
No: the SSRC still remains during a long duration call. In this case, this error might occur.
Allowed values: Yes and No

Default: No</td></tr>
</table>

Enable SSRC Reset for the New RTP and SRTP Sessions

You can enable the **SSRC Reset on RE-INVITE** to avoid a call transfer error, where only one person on the call hears the audio. This error occurs on calls of 30 minutes or longer, and often on three-way calls.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **SIP**.

Step 2 In the **RTP Parameters** section, set the parameter **SSRC Reset on RE-INVITE** to **Yes**.

You can also configure this parameter in the configuration file:

```
<SSRC_Reset_on_RE-INVITE ua="na">Yes</SSRC_Reset_on_RE-INVITE>
```

Allowed values: Yes and No.

Default: No

Note If you set the parameter to **No**, the SSRC remains for the new RTP and SRTP sessions (SIP re-INVITEs). The call transfer error might occur during a long duration call.

Step 3 Click **Submit All Changes**.

Control SIP and RTP Behaviour in Dual Mode

You can control SIP and RTP parameters with SIP IP Preference and SDP IP Preference fields when phone is in dual mode.

SIP IP Preference parameter defines which IP address phone tries first when it is in dual mode.

Table 9: SIP IP Preference and IP Mode

IP Mode	SIP IP Preference	Address List from DNS, Priority, Result P1 - First Priority Address P2 - Second Priority Address	Failover Sequence
Dual Mode	IPv4	P1- 1.1.1.1, 2009:1:1:1::1 P2 - 2.2.2.2, 2009:2:2:2::2 **Result**: Phone will send the SIP messages to 1.1.1.1 first.	1.1.1.1 ->2009:1:1:1:1 -> 2.2.2.2 -> 2009:2:2:2:2
Dual Mode	IPv6	P1- 1.1.1.1, 2009:1:1:1::1 P2 - 2.2.2.2, 2009:2:2:2::2 **Result**: Phone will send the SIP messages to 2009:1:1:1::1 first.	2009:1:1:1:1 -> 1.1.1.1 -> 2009:2:2:2:2 -> 2.2.2.2

IP Mode	SIP IP Preference	Address List from DNS, Priority, Result P1 - First Priority Address P2 - Second Priority Address	Failover Sequence
Dual Mode	IPv4	P1- 2009:1:1:1::1 P2 - 2.2.2.2, 2009:2:2:2::2 **Result**: Phone will send the SIP messages to 2009:1:1:1::1 first.	2009:1:1:1:1 -> 2.2.2.2 -> 2009:2:2:2:2
Dual Mode	IPv6	P1- 2009:1:1:1::1 P2 - 2.2.2.2, 2009:2:2:2::2 **Result**: Phone will send the SIP messages to 1.1.1.1 first.	2009:1:1:1:1 -> 2009:2:2:2:2 ->2.2.2.2
IPv4 Only	IPv4 or IPv6	P1 - 1.1.1.1, 2009:1:1:1::1 P2 - 2.2.2.2, 2009:2:2:2::2 **Result**: Phone will send the SIP messages to 1.1.1.1 first.	1.1.1.1 -> 2.2.2.2
IPv6 Only	IPv4 or IPv6	P1 - 1.1.1.1, 2009:1:1:1::1 P2 - 2.2.2.2, 2009:2:2:2::2 **Result**: Phone will send the SIP messages to 2009:1:1:1::1 first.	2009:1:1:1:1 -> 2009:2:2:2::2

SDP IP Preference - ALTC helps peers in dual-mode negotiate RTP address family.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **SIP**.

Step 2 In the **SIP Parameters** section, select **IPv4** or **IPv6** in the **SIP IP Preference** field.

For details, see **SDP IP Preference** field in the SIP Parameters, on page 16 table.

Step 3 In the **RTP Parameters** section, select **IPv4** or **IPv6** in the **SDP IP Preference** field.

For details, see **SDP IP Preference** in the RTP Parameters, on page 31 table.

Configure the SDP Payload Types

Your Cisco IP Phone supports RFC4733. You can choose from three audio-video transport (AVT) options to send DTMF pulses to the server.

Configured dynamic payloads are used for outbound calls only when the Cisco IP Phone presents a Session Description Protocol (SDP) offer. For inbound calls with an SDP offer, the phone follows the caller's assigned dynamic payload type.

The Cisco IP Phone uses the configured codec names in outbound SDP. For incoming SDP with standard payload types of 0-95, the phone ignores the codec names. For dynamic payload types, the phone identifies the codec by the configured codec names. The comparison is case-sensitive, so you need to set the name correctly.

You can also configure the parameters in the phone configuration file (cfg.xml). To configure each of the parameters, see the syntax of the string in SDP Payload Types, on page 38.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **SIP**.

Step 2 In the **SDP Payload Types** section, set the value as specified in SDP Payload Types, on page 38.

- **AVT Dynamic Payload**—is any nonstandard data. Both sender and receiver must agree on a number. The range is from 96 to 127. The default is 101.
- **AVT 16kHz Dynamic Payload** —is any nonstandard data. Both sender and receiver must agree on a number. The range is from 96 to 127. The default is 107.
- **AVT 48kHz Dynamic Payload** —is any nonstandard data. Both sender and receiver must agree on a number. The range is from 96 to 127. The default is 108.

Step 3 Click **Submit All Changes**.

SDP Payload Types

Parameter	Description
G722.2 Dynamic Payload	G722 Dynamic Payload type. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Try_Backup_RSC ua="na"/>` • In the phone web page, enter an appropriate value. Allowed values: Default: 96
iLBC Dynamic Payload	iLBC Dynamic Payload type. Default: 97
OPUS Dynamic Payload	OPUS Dynamic Payload type. Default: 99
AVT Dynamic Payload	AVT dynamic payload type. Ranges from 96-127. Default: 101
INFOREQ Dynamic Payload	INFOREQ Dynamic Payload type.
H264 BP0 Dynamic Payload	H264 BPO Dynamic Payload type. Default: 110
H264 HP Dynamic Payload	H264 HP Dynamic Payload type. Default: 110
G711u Codec Name	G711u codec name used in SDP. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<G711u_Codec_Name ua="na">PCMU</G711u_Codec_Name>` • In the phone web page, enter an appropriate codec name. Allowed values: Default: PCMU

Parameter	Description
G711a Codec Name	G711a codec name used in SDP. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<G711a_Codec_Name ua="na">PCMU</G711a_Codec_Name>` • In the phone web page, enter an appropriate codec name. Allowed values: Default: PCMA
G729a Codec Name	G729a codec name used in SDP. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<G729a_Codec_Name ua="na">PCMU</G729a_Codec_Name>` • In the phone web page, enter an appropriate codec name. Allowed values: Default: G729a
G729b Codec Name	G729b codec name used in SDP. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<G729b_Codec_Name ua="na">PCMU</G729b_Codec_Name>` • In the phone web page, enter an appropriate codec name. Allowed values: Default: G729b

Parameter	Description
G722 Codec Name	G722 codec name used in SDP. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<G722_Codec_Name ua="na">PCMU</G722_Codec_Name>` • In the phone web page, enter an appropriate codec name. Allowed values: Default: G722
G722.2 Codec Name	G722.2 codec name used in SDP. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<G722.2_Codec_Name ua="na">PCMU</G722.2_Codec_Name>` • In the phone web page, enter an appropriate codec name. Allowed values: Default: G722.2
iLBC Codec Name	iLBC codec name used in SDP. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<iLBC_Codec_Name ua="na">iLBC</iLBC_Codec_Name>` • In the phone web page, enter an appropriate codec name. Allowed values: Default: iLBC

Parameter	Description
OPUS Codec Name	OPUS codec name used in SDP. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<OPUS_Codec_Name ua="na">OPUS</OPUS_Codec_Name>` • In the phone web page, enter an appropriate codec name. Allowed values: Default: OPUS
AVT Codec Name	AVT codec name used in SDP. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<AVT_Codec_Name ua="na">telephone-event</AVT_Codec_Name>` • In the phone web page, enter an appropriate codec name. Allowed values: Default: telephone-event
AVT 16 kHz Dynamic Payload	AVT dynamic payload type for the 16 kHz clock rate. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<AVT_16kHz_Dynamic_Payload ua="na">107</AVT_16kHz_Dynamic_Payload>` • In the phone web page, enter the payload. Range: 96-127 Default: 107

Parameter	Description
AVT 48 kHz Dynamic Payload	AVT dynamic payload type for the 48 kHz clock rate. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<AVT_48kHz_Dynamic_Payload ua="na">108</AVT_48kHz_Dynamic_Payload>` • In the phone web page, enter the payload. Range: 96-127 Default: 108

Configure the SIP Settings for Extensions

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **Ext(n)**, where n is an extension number.

Step 2 In the **SIP Settings** section, set the parameter values as described in the Parameters for SIP Settings on Extensions, on page 43 table.

Step 3 Click **Submit All Changes**.

Parameters for SIP Settings on Extensions

The following table defines the function and usage of the parameters in the SIP Settings section under the Ext(n) tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 10: SIP Settings in Extensions

Parameter	Description

SIP Transport	Specifies the transport protocol for SIP messages.
	Perform one of the following:
	• In the phone configuration file with XML(cfg.xml), enter a string in this format:
	`<SIP_Transport_1_ ua="na">UDP</SIP_Transport_1_>`
	• In the phone web page, select the transport protocol type.
	  • UDP
	  • TCP
	  • TLS
	  • AUTO
	AUTO allows the phone to select the appropriate protocol automatically, based on the NAPTR records on the DNS server. See Configure the SIP Transport for more details.
	Default: UDP

Parameter	Description
SIP Port	The phone's port number for SIP message listening and transmission. **Note** Specify the port number here only when you are using UDP as the SIP transport protocol. If you are using TCP, the system uses a random port within the range specified in **SIP TCP Port Min** and **SIP TCP Port Max** on the **Voice** > **SIP** tab. If you need to specify a port of SIP proxy server, you can specify it using the **Proxy** field or the **XSI Host Server** field. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<SIP_Port_1_ ua="na">5060</SIP_Port_1_>` • In the phone web page, enter an appropriate port number. Default: 5060
SIP 100REL Enable	Individually enables the SIP 100REL feature. When enabled, the phone supports the 100REL SIP extension for reliable transmission of provisional responses (18x) and uses the PRACK requests. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<SIP_100REL_Enable_1_ ua="na">Yes</SIP_100REL_Enable_1_>` • In the phone web page, select Yes to enable the feature. Allowed values: Yes and No Default: No

Parameter	Description
Precondition Support	Determines whether the phone includes the precondition tag (defined in RFC 3312) in the Supported header field. • **Disabled**: The phone doesn't include the precondition tag in the Supported header filed. And the phone doesn't return the 183 response when it receives the INVITE request that contains the QoS precondition in the SDP description. • **Enabled**: The phone includes the precondition tag in the Supported header field. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Precondition_Support_1_ ua="na">Enabled</Precondition_Support_1_> • In the phone web page, select **Enabled** to enable the feature. Allowed values: Disabled and Enabled Default: Disabled
EXT SIP Port	The external SIP port number. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <EXT_SIP_Port_1_ ua="na">5060</EXT_SIP_Port_1_> • In the phone web page, enter a port number. Allowed values: Default: 5060

Parameter	Description
Auth Resync-Reboot	The Cisco IP Phone authenticates the sender when it receives a NOTIFY message with the following requests: • resync • reboot • report • restart • XML-service Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Auth_Resync-Reboot_1_ ua="na">No</Auth_Resync-Reboot_1_> • In the phone web page, select Yes to enable the feature. Allowed values: Yes and No Default: Yes
SIP Proxy-Require	The SIP proxy can support a specific extension or behavior when it receives the Proxy-Require header from the user agent. If this field is configured and the proxy does not support it, it responds with the message, unsupported. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <SIP_Proxy-Require_1_ ua="na">header<SIP_Proxy-Require_1_> • In the phone web interface, enter the appropriate header in the field provided. Default: Blank
SIP Remote-Party-ID	The Remote-Party-ID header to use instead of the From header. Select **Yes** to enable. Default: Yes

Parameter	Description
Referor Bye Delay	Controls when the phone sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referor, Refer Target, Referee, and Refer-To Target) are configured on this screen. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Referor_Bye_Delay_1_ ua="na">4</Referor_Bye_Delay_1_> • In the phone web page, enter the appropriate period of time in seconds. Allowed values: An integer from 0 through 65535 Default: 4
Refer-To Target Contact	Indicates the refer-to target. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Refer-To_Target_Contact_1_ ua="na">No</Refer-To_Target_Contact_1_> • In the phone web page, select **Yes** to send the **SIP Refer** to the contact. Allowed values: Yes and No Default: No
Referee Bye Delay	Specifies the Referee Bye Delay time in seconds. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Referee_Bye_Delay_1_ ua="na">0</Referee_Bye_Delay_1_> • In the phone web page, enter the appropriate period of time in seconds. Allowed values: An integer from 0 through 65535 Default: 0

Parameter	Description
Refer Target Bye Delay	Specifies the Refer Target Bye Delay time in seconds. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Refer_Target_Bye_Delay_1_ ua="na">0</Refer_Target_Bye_Delay_1_>` • In the phone web page, enter the appropriate period of time in seconds. Allowed values: An integer from 0 through 65535 Default: 0
Sticky 183	Controls the first 183 SIP response for an outbound INVITE. To enable this feature, Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Sticky_183_1_ ua="na">No</Sticky_183_1_>` • In the phone web page, select **Yes** to enable this feature. When enabled, the IP telephony ignores further 180 SIP responses after receiving the first 183 SIP response for an outbound INVITE. Allowed values: Yes and No Default: No

Parameter	Description
Auth INVITE	Controls if authorization is required for initial incoming INVITE requests from the SIP proxy. To enable this feature. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Auth_INVITE_1_ ua="na">No</Auth_INVITE_1_>` • In the phone web page, select **Yes** to enable this feature. When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy. Allowed values: Yes and No Default: No
Ntfy Refer On 1xx-To-Inv	If set to **Yes**, as a transferee, the phone will send a NOTIFY with Event:Refer to the transferor for any 1xx response returned by the transfer target, on the transfer call leg. If set to **No**, the phone will only send a NOTIFY for final responses (200 and higher). Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Ntfy_Refer_On_1xx-To-Inv_1_ ua="na">Yes</Ntfy_Refer_On_1xx-To-Inv_1_>` • In the phone web page, select **Yes** to enable this feature. Allowed values: Yes and No Default: Yes

Parameter	Description
Set G729 annexb	Configure G.729 Annex B settings. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Set_G729_annexb_1_ ua="na">Yes</Set_G729_annexb_1_>` • In the phone web page, select **Yes** to enable this feature. Allowed values: • None • No • Yes • Follow silence supp setting Default: Yes
User Equal Phone	When a tel URL is converted to a SIP URL and the phone number is represented by the user portion of the URL, the SIP URL includes the optional: user=phone parameter (RFC3261). For example: To: sip:+12325551234@example.com; user=phone Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<User_Equal_Phone_1_ ua="na">Yes</User_Equal_Phone_1_>` • In the phone web page, select **Yes** to enable this feature. Allowed values: Yes and No Default: No

Parameter	Description
Call Recording Protocol	Determines the type of recording protocol that the phone uses. Options are: • SIPINFO • SIPREC Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Call_Recording_Protocol_1_ ua="na">SIPREC</Call_Recording_Protocol_1_>` • In the phone web page, select a protocol from the list. Allowed values: SIPREC|SIPINFO Default: SIPREC

Parameter	Description
Privacy Header	Sets user privacy in the SIP message in the trusted network. The privacy header options are: • Disabled (default) • none—The user requests that a privacy service applies no privacy functions to this SIP message. • header—The user needs a privacy service to obscure headers which cannot be purged of identifying information. • session—The user requests that a privacy service provide anonymity for the sessions. • user—The user requests a privacy level only by intermediaries. • id—The user requests that the system substitute an id that doesn't reveal the IP address or host name. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Privacy_Header_1_ ua="na">Disabled</Privacy_Header_1_>` • In the phone web page, select an option from the list. Allowed values: Disabled|none|header|session|user|id Default: Disabled
P-Early-Media Support	Controls whether the P-Early-Media header is included in the SIP message for an outgoing call. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<P-Early-Media_Support_1_ ua="na">No</P-Early-Media_Support_1_>` • In the phone web interface, to include the P-Early-Media header, select **Yes**. Allowed values: Yes and No Default: No

Configure the SIP Proxy Server

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **Ext(n)**, where n is an extension number.

Step 2 In the **Proxy and Registration** section, set the parameter values as described in the SIP Proxy and Registration for Extension Parameters, on page 53 table.

Step 3 Click **Submit All Changes**.

SIP Proxy and Registration for Extension Parameters

The following table defines the function and usage of the parameters in the Proxy and Registration section under the Ext(n) tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 11: SIP Proxy and Registration for Extension

Parameter	Description

Proxy	SIP proxy server and port number set by the service provider for all outbound requests. For example: 192.168.2.100:6060.
	• In the phone configuration file with XML(cfg.xml), enter a string in this format:
	`<Proxy_1_ ua="na">64.101.154.134</Proxy_1_>`
	`<RTP_Port_Max ua="na">16482</RTP_Port_Max>`
	• In the phone web page, enter SIP proxy server and port number.
	When you need to refer to this proxy in another setting, for example, the speed dial line key configuration, use the `$PROXY` macro variable.
	Default: The port number is optional. If you don't specify a port, the default port 5060 is used for UDP, and the default port 5061 is used for TLS.

Parameter	Description
Outbound Proxy	Specifies an IP address or domain name. All outbound requests are sent as the first hop. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Outbound_Proxy_1_ ua="na">10.79.78.45</Outbound_Proxy_1_> • In the phone web page, enter an IP address and a domain name. Default: Empty
Proxy Outbound Proxy **For Survivable Remote Site Telephony (SRST) support**	These parameters can be configured with an extension that includes a statically-configured DNS SRV record or DNS A record. This allows for failover and fallback functionality with a secondary proxy server. The format for the parameter value is as follows: FQDN format: hostname[:port][:SRV=host-list OR :A=ip-list] Where: • host-list: srv[|srv[|srv...]] • srv: hostname[:port][:p=priority][:weight][:A=ip-list] • ip-list: ip-addr[,ip-addr[,ip-addr...]] Default: • Priority is 0. • Weight is 1. • Port is 5060 and 5061 for UDP and TLS respectively.

Parameter	Description
Alternate Proxy Alternate Outbound Proxy	This feature provides fast fall back when there is network partition at the Internet or when the primary proxy (or primary outbound proxy) is not responsive or available. The feature works well in a Verizon deployment environment as the alternate proxy is the Integrated Service Router (ISR) with analog outbound phone connection. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Alternate_Proxy_1_ ua="na">10.74.23.43</Alternate_Proxy_1_><Alternate_Outbound_Proxy_1_ ua="na">10.74.23.44</<Alternate_Outbound_Proxy_1_>` • In the phone web page, enter the proxy server addresses and port numbers in these fields. After the phone is registered to the primary proxy and the alternate proxy (or primary outbound proxy and alternate outbound proxy), the phone always sends out INVITE and Non-INVITE SIP messages (except registration) via the primary proxy. The phone always registers to both the primary and alternate proxies. If there is no response from the primary proxy after timeout (per the SIP RFC spec) for a new INVITE, the phone attempts to connect with the alternate proxy. The phone always tries the primary proxy first, and immediately tries the alternate proxy if the primary is unreachable. Active transactions (calls) never fall back between the primary and alternate proxies. If there is fall back for a new INVITE, the subscribe/notify transaction will fall back accordingly so that the phone's state can be maintained properly. You must also set Dual Registration in the Proxy and Registration section to Yes. Default: Empty
Use OB Proxy In Dialog	Determines whether to force SIP requests to be sent to the outbound proxy within a dialog. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Use_OB_Proxy_In_Dialog_1_ ua="na">Yes</Use_OB_Proxy_In_Dialog_1_>` • In the phone web page, select **Yes** or **No**. The request is ignored if the **Use Outbound Proxy** field is set to **No** or if the **Outbound Proxy** field is empty. Valid values: Yes and No Default: Yes

Parameter	Description
Register	Enables periodic registration with the proxy. This parameter is ignored if a proxy is not specified. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Register_1_ ua="na">Yes</Register_1_> • In the phone web page, To enable this feature, select **Yes**. Valid values: Yes and No Default: Yes
Make Call Without Reg	Enables making outbound calls without successful (dynamic) registration by the phone. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Make_Call_Without_Reg_1_ ua="na">No</Make_Call_Without_Reg_1_> • In the phone web page, To enable this feature, select **Yes**. If set to **No**, the dial tone plays only when registration is successful. Valid values: Yes and No Default: No
Register Expires	Defines how often the phone renews registration with the proxy. If the proxy responds to a REGISTER with a lower expires value, the phone renews registration based on that lower value instead of the configured value. If registration fails with an “Expires too brief” error response, the phone retries with the value specified in the Min-Expires header of the error. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Register_Expires_1_ ua="na">3600</Register_Expires_1_> • In the phone web page, enter a value in seconds to define how often the phone renews registration with the proxy. Valid values: Numeric. The range is from 32 seconds to 2000000 seconds. Default: 3600 seconds

Parameter	Description
Ans Call Without Reg	If enabled, the user does not have to be registered with the proxy to answer calls. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Ans_Call_Without_Reg_1_ ua="na">No</Ans_Call_Without_Reg_1_>` • In the phone web page, To enable this feature, select **Yes**. Valid values: Yes and No Default: No
Use DNS SRV	Enables DNS SRV lookup for the proxy and outbound proxy. • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Use_DNS_SRV_1_ ua="na">Yes</Use_DNS_SRV_1_>` • In the phone web page, To enable this feature, select **Yes**. Valid values: Yes and No Default: No
DNS SRV Auto Prefix	Enables the phone to automatically append a prefix to the proxy or outbound proxy name when performing a DNS SRV lookup on that name. The prefix to be appended varies with SIP transport protocols. • `_sip._udp.` for UDP protocol • `_sip._tcp.` for TCP protocol • `_sips._tcp.` for TLS protocol Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<DNS_SRV_Auto_Prefix_1_ ua="na">Yes</DNS_SRV_Auto_Prefix_1_>` • In the phone web page, to enable this feature, select **Yes**. Valid values: Yes and No Default: No

Parameter	Description
Proxy Fallback Intvl	Sets the delay after which the phone retries from the highest priority proxy (or outbound proxy) after it has failed over to a lower priority server. The phone should have the primary and backup proxy server list from a DNS SRV record lookup on the server name. It needs to know the proxy priority; otherwise, it does not retry. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Proxy_Fallback_Intvl_1_ ua="na">3600</Proxy_Fallback_Intvl_1_> • In the phone web page, enter a value in seconds to set the duration in seconds after which the phone retries. Valid values: Numeric. The range is from 0 seconds to 65535 seconds. Default: 3600 seconds
Proxy Redundancy Method	The phone creates an internal list of proxies returned in the DNS SRV records. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Proxy_Redundancy_Method_1_ ua="na">Normal</Proxy_Redundancy_Method_1_> • In the phone web page, select **Normal** and **Based on SRV Port**. If you set to **Normal**, the list contains proxies ranked by weight and priority. If you set to **Based on SRV Port**, the phone uses normal, then inspects the port number based on the first-listed proxy port. Valid values: Normal|Based on SRV Port Default: Normal
Dual Registration	Controls both the dual registration and the fast fall back feature. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Dual_Registration_1_ ua="na">No</Dual_Registration_1_> • In the phone web page, set to **Yes** to enable the Dual registration/Fast Fall back feature. To enable the feature you must also configure the alternate proxy/alternate outbound proxy fields in the Proxy and Registration section. Valid values: Yes and No Default: No

Parameter	Description
Auto Register When Failover	Controls the fallback duration. • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Auto_Register_When_Failover_1_ ua="na">Yes</Auto_Register_When_Failover_1_> • In the phone web page, If set to No, the fallback happens immediately and automatically. If the Proxy Fallback Intvl is exceeded, all the new SIP messages go to the primary proxy. If set to Yes, the fallback happens only when current registration expires, which means only a REGISTER message can trigger fallback. For example, when the value for Register Expires is 3600 seconds and Proxy Fallback Intvl is 600 seconds, the fallback is triggered 3600 seconds later and not 600 seconds later. When the value for Register Expires is 600 seconds and Proxy Fallback Intvl is 1000 seconds, the fallback is triggered at 1200 seconds. After successfully registering back to primary server, all the SIP messages go to primary server. Valid values: Yes and No Default: Yes
TLS Name Validate	This field works only when **SIP Transport** is set to **TLS** for the phone line. Specifies whether hostname verification is required when the phone line uses SIP over TLS. The options are: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <TLS_Name_Validate_1_ ua="na">Yes</TLS_Name_Validate_1_> • In the phone web page, select **Yes** when hostname verification is required. Select **No** to bypass the hostname verification. Valid values: Yes and No Default: Yes

Configure the Subscriber Information Parameters

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **Ext(n)**, where n is an extension number.

Step 2 In the **Subscriber Information** section, set the parameter values as described in the Subscriber Information Parameters, on page 60 table.

Step 3 Click **Submit All Changes**.

Subscriber Information Parameters

The following table defines the function and usage of the parameters in the RTP Parameters section under the SIP tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 12: Subscriber Information

Parameter	Description
Display Name	Name displayed as the caller ID. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Display_Name_1_ ua="na"/>` • In the phone web page, enter a name that represents the caller ID.
User ID	Extension number for this line. When you need to refer to this user ID in another setting, for example, the short name for a line key, use the `$USER` macro variable. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<User_ID_1_ ua="na">7001</User_ID_1_>` • In the phone web page, enter an extension number

Parameter	Description
Password	Password for this line. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Password_1_ ua="na">*************</Password_1_> • In the phone web page, enter a value to add password for the line. Default: Blank (no password required)
Auth ID	Authentication ID for SIP authentication. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Auth_ID_1_ ua="na"/> • In the phone web page, enter a value for an authentication ID. Default: Blank
Reversed Auth Realm	The IP address for an authentication realm other than the proxy IP address. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Reversed_Auth_Realm_1_ ua=”na”> </Reversed_Auth_Realm_1_> The parameter for extension 1 appears as defined in the phone configuration file. • In the phone web page, enter proxy IP address. Default: Blank. The proxy IP address is used as the authentication realm.

Parameter	Description
SIP URI	The parameter by which the user agent will identify itself for this line. If this field is blank, the actual URI used in the SIP signaling should be automatically formed as: sip:UserName@Domain where UserName is the username given for this line in the User ID, and Domain is the domain given for this profile in the User Agent Domain. If the User Agent Domain is an empty string, then the IP address of the phone should be used for the domain. If the URI field is not empty, but if a SIP or SIPS URI contains no @ character, the actual URI used in the SIP signaling should be automatically formed by appending this parameter with an @ character followed by the IP address of the device.

Set Up Your Phone to Use OPUS Codec Narrowband

To improve bandwidth in your network, you can set up your phones to use the narrowband OPUS codec. The narrowband codec won't conflict with the wideband codec.

Before you begin

Access the Phone Web Interface

Procedure

Step 1 Select **Voice** > **Ext <n>** where **(n)** is the number of the extension to configure.

Step 2 In the **SIP Settings** section, set **Use low-bandwidth OPUS** to **Yes**.

Step 3 Click **Submit All Changes**.

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, on page 70.

  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 , on page 65.
- **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, on page 69.

Enable NAT Mapping

You must enable NAT mapping to set NAT parameters.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **Ext(n)**.

Step 2 Set up the fields as described in NAT Mapping Parameters, on page 64.

Step 3 Click **Submit All Changes**.

NAT Mapping Parameters

The following table defines the function and usage of NAT Mapping parameters in the NAT Settings section under the Voice>Ext(n) tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 13: NAT Mapping Parameters

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>NAT Mapping Enable</td><td>To use externally mapped IP addresses and SIP/ RTP ports in SIP messages, select yes. Otherwise, select no.

Perform one of the following:
In the phone configuration file with XML(cfg.xml), enter a string in this format:
<code><NAT_Mapping_Enable_1_ ua="na">Yes</NAT_Mapping_Enable_1_></code> In the phone web page, set the parameter to Yes. Allowed values: Yes|No

Default: No</td></tr>
<tr><td>NAT Keep Alive Enable</td><td>To send the configured NAT keep alive message periodically, select yes. Otherwise, select no.

Perform one of the following:
In the phone configuration file with XML(cfg.xml), enter a string in this format:
<code><NAT_Keep_Alive_Enable_1_ ua="na">Yes</NAT_Keep_Alive_Enable_1_></code> In the phone web page, set the parameter to Yes. Allowed values: Yes|No

Default: No</td></tr>
</table>

Parameter	Description
NAT Keep Alive Msg	Enter the keep alive message that should be sent periodically to maintain the current NAT mapping. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<NAT_Keep_Alive_Msg_1_ ua="na">$NOTIFY</NAT_Keep_Alive_Msg_1_>` • In the phone web page, set the parameter to **$NOTIFY** or to **$REGISTER**. If the value is $NOTIFY, a NOTIFY message is sent. If the value is $REGISTER, a REGISTER message without contact is sent. Allowed values: $NOTIFY and $REGISTER. Default: $NOTIFY
NAT Keep Alive Dest	Destination that should receive NAT keep alive messages. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<NAT_Keep_Alive_Dest_1_ ua="na">$PROXY</NAT_Keep_Alive_Dest_1_>` • In the phone web page, set the parameter to **$PROXY** or specify a proxy server. If the value is $PROXY, the messages are sent to the current or outbound proxy. Allowed values: $PROXY or a proxy server IP address Default: $PROXY

Configure NAT Mapping with the Static IP Address

You can configure NAT mapping on the phone to ensure interoperability with the service provider.

Before you begin

- Access the phone administration web page. See Access the Phone Web Interface.
- You must have an external (public) IP address that is static.
- The NAT mechanism used in the router must be symmetric.

Procedure

Step 1 Select **Voice** > **SIP**.

Step 2 In the **NAT Support Parameters** section, set the parameters as described in the NAT Mapping with Static IP Parameters, on page 66 table.

Step 3 Click the **Ext(n)** tab.

Step 4 In the **NAT Settings** section, set the parameters as described in the NAT Mapping from Ext Tab with Static IP Parameters table.

Step 5 Click **Submit All Changes**.

What to do next

Configure the firewall settings on your router to allow SIP traffic.

NAT Mapping with Static IP Parameters

The following table defines the function and usage of NAT mapping with Static IP parameters in the NAT Support Parameters section under the Voice>SIP tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 14: NAT Mapping with Static IP Parameters

Parameter	Description
Handle VIA received	Enables the phone to process the received parameter in the VIA header. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Handle_VIA_received ua="na">Yes</Handle_VIA_received> • In the phone web page, set to **Yes**. Default: No
Handle VIA rport	Enables the phone to process the rport parameter in the VIA header. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Handle_VIA_rport ua="na">Yes</Handle_VIA_rport> • In the phone web page, set to **Yes**. Default: No

Parameter	Description
Insert VIA received	Enables to insert the received parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Insert_VIA_received ua="na">Yes</Insert_VIA_received> • In the phone web page, set to **Yes**. Default: No
Insert VIA rport	Enables to insert the rport parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Insert_VIA_rport ua="na">Yes</Insert_VIA_rport> • In the phone web page, set to **Yes**. Default: No
Substitute VIA Addr	Enables the user to use NAT-mapped IP:port values in the VIA header. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Substitute_VIA_Addr ua="na">Yes</Substitute_VIA_Addr> • In the phone web page, set to **Yes**. Default: No
Send Resp To Src Port	Enables to send responses to the request source port instead of the VIA sent-by port. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Send_Resp_To_Src_Port ua="na">Yes</Send_Resp_To_Src_Port> • In the phone web page, set to **Yes**. Default: No

Parameter	Description
NAT Keep Alive Intvl	Interval between NAT-mapping keep alive messages. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<NAT_Keep_Alive_Intvl ua="na">15</NAT_Keep_Alive_Intvl>` • In the phone web page, enter an appropriate value. Allowed values: Numeric ranges from 0 through 65535 Default: 15
EXT IP	External IP address to substitute for the actual IP address of phone in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed. If this parameter is specified, phone assumes this IP address when generating SIP messages and SDP (if NAT Mapping is enabled for that line). Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<EXT_IP ua="na">10.23.31.43</EXT_IP>` • In the phone web page, enter an external static IP address. Default: Blank

The following table defines the function and usage of NAT mapping with Static IP parameters in the NAT Support Parameters section under the Voice>Ext tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 15: NAT Mapping from Ext Tab

Parameter	Description
NAT Mapping Enable	Controls the use of externally mapped IP addresses and SIP/ RTP ports in SIP messages. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<NAT_Mapping_Enable_1_ ua="na">Yes</NAT_Mapping_Enable_1_>` • In the phone web page, set to **Yes** to use externally mapped IP addresses. Allowed values: Yes and No. Default: No

Parameter	Description
NAT Keep Alive Enable (Optional)	Configured NAT keep alive message periodically. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<NAT_Keep_Alive_Enable_1_ ua="na">Yes</NAT_Keep_Alive_Enable_1_>` • In the phone web page, set to **Yes** to configure periodic NAT keep alive messages. **Note** The service provider might require the phone to send NAT keep alive messages to keep the NAT ports open. Check with your service provider to determine the requirements. Allowed values: Yes and No. Default: No

Configure 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. The STUN protocol allows applications operating behind a network address translator (NAT) to discover the presence of the network address translator and to obtain the mapped (public) IP address (NAT addresses) and the port number that the NAT has allocated for the User Datagram Protocol (UDP) connections to remote hosts. The protocol requires assistance from a third-party network server (STUN server) located on the opposing (public) side of the NAT, usually the public Internet. This option is considered a last resort and should be used only if the other methods are not available. To use STUN:

- The router must use asymmetric NAT. See Determine Symmetric or Asymmetric NAT, on page 70.
- A computer running STUN server software is available on the network. You can also use a public STUN server or set up your own STUN server.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **SIP**.

Step 2 In the **NAT Support Parameters** section, set the **Handle VIA received**, **Insert VIA received**, **Substitute VIA Addr**, **Handle VIA rport**, **Insert VIA rport**, and **Send Resp To Src Port** parameters as described in the NAT Mapping with Static IP Parameters, on page 66 table.

Step 3 Set the parameters as described in the NAT Mapping with STUN Parameters table.

Step 4 Click the **Ext(n)** tab.

Step 5 In the **NAT Settings** section, set the parameters as described in the NAT Mapping from Ext Tab with Static IP Parameters table.

Step 6 Click **Submit All Changes**.

What to do next

Configure the firewall settings on your router to allow SIP traffic.

NAT Mapping with STUN Parameters

The following table defines the function and usage of NAT mapping with STUN parameters in the NAT Support Parameters section under the Voice>SIP tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 16: NAT Mapping with STUN Parameters

Parameter	Description
STUN Enable	Enables the use of STUN to discover NAT mapping. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<STUN_Enable ua="na">Yes</STUN_Enable>` • In the phone web page, set to **Yes** to enable the feature. Allowed values: Yes and No. Default: No
STUN Server	IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery. You can use a public STUN server or set up your own STUN server. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<STUN_Server ua="na"/>` • In the phone web page, enter an IP address or fully-qualified domain name of the STUN server. Allowed values: Default: Blank

Determine Symmetric or Asymmetric NAT

STUN does not work on routers with symmetric NAT. With symmetric NAT, IP addresses are mapped from one internal IP address and port to one external, routable destination IP address and port. If another packet is sent from the same source IP address and port to a different destination, a different IP address and port number combination is used. This method is restrictive because an external host can send a packet to a particular port on the internal host only if the internal host first sent a packet from that port to the external host.

This procedure assumes that a syslog server is configured and is ready to receive syslog messages.

To Determine Whether the Router Uses Symmetric or Asymmetric NAT:

Before you begin

- Verify that the firewall is not running on your PC. (It can block the syslog port.) By default, the syslog port is 514.
- Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **System** and navigate to **Optional Network Configuration** section.

Step 2 Enter the IP address for the **Syslog Server**, if the port number is anything other than the default, 514. It is not necessary to include the port number if it is the default.

The address and port number must be reachable from the Cisco IP phone. The port number appears on the output log file name. The default output file is `syslog.514.log` (if port number was not specified).

Step 3 Set the **Debug Level** to **Error**, **Notice**, or **Debug**.

Step 4 To capture SIP signaling messages, click the **Ext** tab and navigate to **SIP Settings**. Set the **SIP Debug Option** to **Full**.

Step 5 To collect information about what type of NAT your router uses click the **SIP** tab and navigate to **NAT Support Parameters**.

Step 6 Click **Voice** > **SIP** and navigate to **NAT Support Parameters**.

Step 7 Set **STUN Test Enable** to **Yes**.

Step 8 Determine the type of NAT by viewing the debug messages in the log file. If the messages indicate that the device is using symmetric NAT, you cannot use STUN.

Step 9 Click **Submit All Changes**.

Dial Plan

Dial Plan Overview

Dial plans determine how digits are interpreted and transmitted. They also determine whether the dialed number is accepted or rejected. You can use a dial plan to facilitate dialing or to block certain types of calls such as long distance or international.

Use the phone web user interface to configure dial plans on the IP phone.

This section includes information that you must understand about dial plans, and procedures to configure your own dial plans.

The Cisco IP Phone has various levels of dial plans and processes the digits sequence.

When a user presses the speaker button on the phone, the following sequence of events begins:

1. The phone begins to collect the dialed digits. The interdigit timer starts to track the time that elapses between digits.
2. If the interdigit timer value is reached, or if another terminating event occurs, the phone compares the dialed digits with the IP phone dial plan. This dial plan is configured in the phone web user interface in **Voice** > **Ext(n)** under the **Dial Plan** section.

Digit Sequences

A dial plan contains a series of digit sequences, separated by the **|** character. The entire collection of sequences is enclosed within parentheses. Each digit sequence within the dial plan consists of a series of elements that are individually matched to the keys that the user presses.

White space is ignored, but can be used for readability.

Digit Sequence	Function
0 1 2 3 4 5 6 7 8 9 0 * #	Characters that represent a key that the user must press on the phone keypad.
x	Any character on the phone keypad.
[sequence]	Characters within square brackets create a list of accepted key presses. The user can press any one of the keys in the list. A numeric range, for example, `[2-9]` allows a user to press any one digit from `2` through `9`. A numeric range can include other characters. For example, `[35-8*]` allows a user to press 3, 5, 6, 7, 8, or *.
. (period)	A period indicates element repetition. The dial plan accepts `0` or more entries of the digit. For example, `01.` allows users to enter 0, 01, 011, 0111, and so forth.

<dialed:substituted>	This format indicates that certain *dialed* digits are replaced by the *substituted* characters when the sequence is transmitted. The *dialed* digits can be zero to 9. For example:
	`<8:1650>xxxxxxx`
	When the user presses `8` followed by a seven-digit number, the system automatically replaces the dialed `8` with the sequence `1650`. If the user dials **85550112**, the system transmits **16505550112**.
	If the *dialed* parameter is empty and there is a value in the *substituted* field, no digits are replaced and the *substituted* value is always prepended to the transmitted string. For example:
	`<:1>xxxxxxxxxx`
	When the user dials **9725550112**, the number `1` is added at the beginning of the sequence; the system transmits **19725550112**.

Digit Sequence	Function
, (comma)	An intersequence tone played (and placed) between digits plays an outside line dial tone. For example: `9, 1xxxxxxxxxx` An outside line dial tone plays after the user presses `9`. The tone continues until the user presses `1`.
! (exclamation point)	Prohibits a dial sequence pattern. For example: `1900xxxxxxx!` Rejects any 11-digit sequence that begins with 1900.
*xx	Allows a user to enter a 2-digit star code.
S0 or L0	For Interdigit Timer Master Override, enter `S0` to reduce the short interdigit timer to 0 seconds, or enter `L0` to reduce the long interdigit timer to 0 seconds.
P	To pause, enter `P`, the number of seconds to pause, and a space. This feature is typically used for implementation of a hotline and warm line, with a 0 delay for the hot line, and a nonzero delay for a warm line. For example: `P5` A pause of 5 seconds is introduced.

Digit Sequence Examples

The following examples show digit sequences that you can enter in a dial plan.

In a complete dial plan entry, sequences are separated by a pipe character (|), and the entire set of sequences is enclosed within parentheses:

```
( [1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx
| 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11 )
```

- Extensions on your system:

  ```
  ( [1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx
   | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11 )
  ```

  `[1-8]xx` Allows a user to dial any three-digit number that starts with the digits 1 to 8. If your system uses four-digit extensions, enter the following string: `[1-8]xxx`

- Local dialing with seven-digit number:

  ```
  ( [1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx
   | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]111)
  ```

  `9, xxxxxxx` After a user presses 9, an external dial tone sounds. The user can enter any seven-digit number, as in a local call.

- Local dialing with 3-digit area code and a 7-digit local number:

```
( [1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx
| 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11 )
```

9, <:1>[2-9]xxxxxxxxx This example is useful where a local area code is required. After a user presses 9, an external dial tone sounds. The user must enter a 10-digit number that begins with a digit 2 through 9. The system automatically inserts the 1 prefix before it transmits the number to the carrier.

- Local dialing with an automatically inserted 3-digit area code:

```
( [1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx
| 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11 )
```

8, <:1212>xxxxxxx This example is useful where a local area code is required by the carrier but most calls go to one area code. After the user presses 8, an external dial tone sounds. The user can enter any seven-digit number. The system automatically inserts the 1 prefix and the 212 area code before it transmits the number to the carrier.

- U.S. long-distance dialing:

```
( [1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx
| 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11 )
```

9, 1 [2-9] xxxxxxxxx After the user presses 9, an external dial tone sounds. The user can enter any 11-digit number that starts with 1 and is followed by a digit 2 through 9.

- Blocked number:

```
( [1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx
| 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11 )
```

9, 1 900 xxxxxxx ! This digit sequence is useful if you want to prevent users from dialing numbers that are associated with high tolls or inappropriate content, such as 1-900 numbers in the U.S. After the user presses 9, an external dial tone sounds. If the user enters an 11-digit number that starts with the digits 1900, the call is rejected.

- U.S. international dialing:

```
( [1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx
| 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11 )
```

9, 011xxxxxx After the user presses 9, an external dial tone sounds. The user can enter any number that starts with 011, as in an international call from the U.S.

- Informational numbers:

```
( [1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx
| 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11 )
```

0 | [49]11 This example includes two-digit sequences, separated by the pipe character. The first sequence allows a user to dial 0 for an operator. The second sequence allows the user to enter 411 for local information or 911 for emergency services.

Acceptance and Transmission of the Dialed Digits

When a user dials a series of digits, each sequence in the dial plan is tested as a possible match. The matching sequences form a set of candidate digit sequences. As the user enters more digits, the set of candidates diminishes until only one or none is valid. When a terminating event occurs, the IP PBX either accepts the user-dialed sequence and initiates a call, or else rejects the sequence as invalid. The user hears the reorder (fast busy) tone if the dialed sequence is invalid.

The following table explains how terminating events are processed.

Terminating Event	Processing
Dialed digits have not matched any sequence in the dial plan.	The number is rejected.
Dialed digits exactly match one sequence in the dial plan.	If the dial plan allows the sequence, the number is accepted and is transmitted according to the dial plan. If the dial plan blocks the sequence, the number is rejected.
A timeout occurs.	The number is rejected if the dialed digits are not matched to a digit sequence in the dial plan within the time that the applicable interdigit timer specifies. The Interdigit Long Timer applies when the dialed digits do not match any digit sequence in the dial plan. Default: 10 seconds. The Interdigit Short Timer applies when the dialed digits match one or more candidate sequences in the dial plan. Default: 3 seconds.
A user presses the # key or the dial softkey on the IP phone screen.	If the sequence is complete and is allowed by the dial plan, the number is accepted and is transmitted according to the dial plan. If the sequence is incomplete or is blocked by the dial plan, the number is rejected.

Dial Plan Timer (Off-Hook Timer)

You can think of the Dial Plan Timer as the off-hook timer. This timer starts when the phone goes off hook. If no digits are dialed within the specified number of seconds, the timer expires and the null entry is evaluated. Unless you have a special dial plan string to allow a null entry, the call is rejected.

Note The timer before a number is dialed is whichever shorter of the dial plan default timer and the dial tone timer set in the **Dial Tone** field on the **Regional** tab.

Syntax for the Dial Plan Timer

SYNTAX: (Ps<:n> | dial plan)

- **s:** The number of seconds; The timer before a number is dialed is whichever shorter of the dial plan default timer and the dial tone timer set in the **Dial Tone** field. With the timer set to 0 seconds, the call transmits automatically to the specified extension when the phone goes off hook.
- **n:** (optional): The number to transmit automatically when the timer expires; you can enter an extension number or a DID number. No wildcard characters are allowed because the number is transmitted as shown. If you omit the number substitution, <:n>, the user hears a reorder (fast busy) tone after the specified number of seconds.

Examples for the Dial Plan Timer

Note The actual timer before a number is dialed is whichever shorter of the dial plan default timer and the dial tone timer set in the **Dial Tone** field. In the following examples, the dial tone timer is assumed to be longer than the dial plan timer.

Allow more time for users to start dialing after taking a phone off hook:

```
(P9 | (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)
```

P9 means that after taking a phone off hook, a user has 9 seconds to begin dialing. If no digits are pressed within 9 seconds, the user hears a reorder (fast busy) tone. By setting a longer timer, you allow more time for users to enter digits.

To create a hotline for all sequences on the System Dial Plan:

```
(P9<:23> | (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)
```

P9<:23> means that after taking the phone off hook, a user has 9 seconds to begin dialing. If no digits are pressed within 9 seconds, the call is transmitted automatically to extension 23.

To create a hotline on a line button for an extension:

```
(P0 <:1000>)
```

With the timer set to 0 seconds, the call is transmitted automatically to the specified extension when the phone goes off hook. Enter this sequence in the Phone Dial Plan for Ext 2 or higher on a client phone.

Interdigit Long Timer (Incomplete Entry Timer)

You can think of this timer as the incomplete entry timer. This timer measures the interval between dialed digits. It applies as long as the dialed digits do not match any digit sequences in the dial plan. Unless the user enters another digit within the specified number of seconds, the entry is evaluated as incomplete, and the call is rejected. The default value is 10 seconds.

This section explains how to edit a timer as part of a dial plan. Alternatively, you can modify the Control Timer that controls the default interdigit timers for all calls.

Syntax for the Interdigit Long Timer

SYNTAX: L:s, (dial plan)

- **s:** The number of seconds; if no number is entered after L:, the default timer is 5 seconds. With the timer set to 0 seconds, the call is transmitted automatically to the specified extension when the phone goes off hook.
- Note that the timer sequence appears to the left of the initial parenthesis for the dial plan.

Example for the Interdigit Long Timer

```
L:15, (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)
```

L:15 means that this dial plan allows the user to pause for up to 15 seconds between digits before the Interdigit Long Timer expires. This setting is especially helpful to users such as sales people, who are reading the numbers from business cards and other printed materials while dialing.

Interdigit Short Timer (Complete Entry Timer)

You can think of this timer as the complete entry timer. This timer measures the interval between dialed digits. The timer applies when the dialed digits match at least one digit sequence in the dial plan. Unless the user enters another digit within the specified number of seconds, the entry is evaluated. If the entry is valid, the call proceeds. If the entry is invalid, the call is rejected.

Default: 3 seconds.

Syntax for the Interdigit Short Timer

SYNTAX 1: S:s, (dial plan)

Use this syntax to apply the new setting to the entire dial plan within the parentheses.

SYNTAX 2: *sequence* Ss

Use this syntax to apply the new setting to a particular dialing sequence.

s: The number of seconds; if no number is entered after S, the default timer of 5 seconds applies.

Examples for the Interdigit Short Timer

To set the timer for the entire dial plan:

```
S:6, (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)
```

S:6 means that while the user enters a number with the phone off hook, the user can pause for up to 15 seconds between digits before the Interdigit Short Timer expires. This setting is especially helpful to users such as sales people, who are reading the numbers from business cards and other printed materials while dialing.

Set an instant timer for a particular sequence within the dial plan:

```
(9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxxS0 | 9,8,011xx. | 9,8,xx.|[1-8]xx)
```

9,8,1[2-9]xxxxxxxxxS0 means that with the timer set to 0, the call is transmitted automatically when the user dials the final digit in the sequence.

Edit the Dial Plan on the IP Phone

Note You can edit the dial plan in the XML configuration file. Locate the `Dial_Plan_n_` parameter in the XML configuration file, where n denotes the extension number. Edit the value of this parameter. The value must be specified in the same format as in the **Dial Plan** field on the phone administration web page, described below.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **Ext(n)**, where n is an extension number.

Step 2 Scroll to the **Dial Plan** section.

Step 3 Enter the digit sequences in the **Dial Plan** field.

The default (US-based) systemwide dial plan appears automatically in the field.

Step 4 You can delete digit sequences, add digit sequences, or replace the entire dial plan with a new dial plan.

Separate each digit sequence with a pipe character, and enclose the entire set of digit sequences within parentheses. Example:

```
(9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)
```

Step 5 Click **Submit All Changes**.

The phone reboots.

Step 6 Verify that you can successfully complete a call with each digit sequence that you entered in the dial plan.

Note If you hear a reorder (fast busy) tone, review your entries and modify the dial plan appropriately.

Regional Parameters Configuration

Regional Parameters

In the phone web user interface, use the **Regional** tab to configure regional and local settings, such as control timer values, dictionary server script, language selection, and locale to change localization. The Regional tab includes these sections:

- Call Progress Tones—Displays values of all ringtones.
- Distinctive Ring Patterns—Ring cadence defines the ringing pattern that announces a telephone call.

- Control Timer Values—Displays all values in seconds.
- Vertical Service Activation Codes—Includes Call Back Act Code and Call Back Deact Code.
- Outbound Call Codec Selection Codes—Defines the voice quality.
- Time—Includes local date, local time, time zone, and Daylight Saving Time.
- Language—Includes Dictionary Server Script, Language Selection, and Locale.

Set the Control Timer Values

If you need to edit a timer setting only for a particular digit sequence or type of call, you can edit the dial plan.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **Regional**.

Step 2 Set the **Reorder Delay**, **Interdigit Long Timer**, and **Interdigit Short Timer** parameters as described in the **Control Timer Values (sec)** table.

Step 3 Click **Submit All Changes**.

Parameters for Control Timer Values (sec)

The following table defines the function and usage of Control Timer Values parameters in the Control Timer Values(s) Parameters section under the Voice>Regional tab in the phone web interface. It also defines the syntax of the string that is added in the phone configuration file with XML(cfg.xml) code to configure a parameter.

Table 17: Parameters for Control Timer Values (sec)

Parameter	Description
Reorder Delay	Delay after far end hangs up before reorder (busy) tone is played. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Reorder_Delay ua="na">255</Reorder_Delay> • In the phone web page, set a value in seconds ranges from 0-255 secs. 0 = plays immediately, inf = never plays. Set to 255 to return the phone immediately to on-hook status and to not play the tone. Allowed values: 0–255 seconds Default: 255
Interdigit Long Timer	Long timeout between entering digits when dialing. The interdigit timer values are used as defaults when dialing. The Interdigit_Long_Timer is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: <Interdigit_Long_Timer ua="na">10</Interdigit_Long_Timer> • In the phone web page, set a value in seconds ranges from 0-64 seconds. Allowed values: 0–64 seconds Default: 10

Parameter	Description
Interdigit Short Timer	Short timeout between entering digits when dialing. The Interdigit_Short_Timer is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences. Perform one of the following: • In the phone configuration file with XML(cfg.xml), enter a string in this format: `<Interdigit_Short_Timer ua="na">3</Interdigit_Short_Timer>` • In the phone web page, set a value in seconds ranges from 0-64 seconds. Allowed values: 0–64 seconds Default: 3

Localize Your Cisco IP Phone

Before you begin

Access the phone administration web page. See Access the Phone Web Interface.

Procedure

Step 1 Select **Voice** > **Regional**.

Step 2 Configure the values in the fields in the **Time** and **Language** sections.

Step 3 Click **Submit All Changes**.

Phone Display Language

The Cisco IP Phone supports multiple languages for the phone display.

By default, the phone is set up for English. To enable the use of another language, you must set up the dictionary for the language. For some languages, you must also set up the font for the language.

After the setup is complete, you or your users can specify the desired language for the phone display.

Supported Languages for the Phone Display

On the phone administration web page, go to **Admin Login** > **Advanced** > **Voice** > **Regional**. In the **Language** section, click the **Locale** drop-down list box to see the supported languages for the phone display.

- ar-SA (Arabic)
- bg-BG (Bulgarian)
- ca-ES (Catalan)
- cs-CZ (Czech)
- da-DK (Danish)
- de-DE (German)
- el-GR (Greek)
- en-GB (English-Great Britain)
- en-US (English-United States)
- es-CO (Spanish-Colombia)
- es-ES (Spanish-Spain)
- fi-FI (Finnish)
- fr-CA (French-Canada)
- fr-FR (French)
- he-IL (Hebrew)
- hr-HR (Croatian)
- hu-HU (Hungarian)
- it-IT (Italian)
- ja-JP (Japanese)
- ko-KR (Korean)
- nl-NL (Dutch)
- nn-NO (Norwegian)
- pl-PL (Polish)
- pt-PT (Portuguese)
- ru-RU (Russian)
- sk-SK (Slovak)
- sl-SI (Slovenian)
- sv-SE (Swedish)
- tr-TR (Turkish)
- zh-CN (Chinese)
- zh-HK (Chinese-Hong Kong SAR)

Set Up Dictionaries and Fonts

Languages other than English require dictionaries. Some languages also require a font.

Note To enable Latin and Cyrillic languages, you must not add a font file.

Procedure

Step 1 Download the locale zip file for your firmware version, from cisco.com. Place the file on your server, and unzip the file.

Dictionaries and fonts for all the supported languages are included in the zip file. Dictionaries are XML scripts. Fonts are standard TTF files.

Step 2 On the phone administration web page, go to **Admin Login** > **Advanced** > **Voice** > **Regional**. In the **Language** section, specify the necessary parameters and values in the **Dictionary Server Script** field as described below. Use a semicolon (;) to separate multiple parameter and value pairs.

- Specify the location of the dictionary and font files with the `serv` parameter.

  For example: `serv=http://server.example.com/Locales/`

Make sure to include the IP address of the server, the path, and folder name.

Example: `serv=http://10.74.128.101/Locales/`

- For each language that you want to set up, specify a set of parameters as described below.

Note In these parameter specifications, *n* denotes a serial number. This number determines the sequential order in which the language options are displayed in the **Settings** menu of the phone.

0 is reserved for US-English, which has a default dictionary. You can use it optionally, to specify your own dictionary.

Use numbers starting with 1 for other languages.

  - Specify the language name with the `dn` parameter.

    Example for language name for Asian language: `d1=Chinese-Simplified`

    Example for language name for German (Latin and Cyrillic): `d2=German`

    Example for language name for French (Latin and Cyrillic): `d1=French`

    Example for language name for French (Canada) (Latin and Cyrillic) language: `d1=French-Canada`

    Example for language name for Hebrew (RTL language): `d1=Hebrew`

    Example for language name for Arabic (RTL language): `d1=Arabic`

    This name is displayed as a language option in the **Settings** menu of the phone.

  - Specify the name of the dictionary file with the `xn` parameter.

    Example for Asian language:

    Example for French (Latin and Cyrillic) languages:

    Example for Arabic (RTL language) language:

    Example for French (Canada) language: `x1=fr-CA_78xx_68xx-11.3.6.0006.xml;`

    Ensure to specify the correct file for the language and phone model that you use.

  - If a font is required for the language, specify the name of the font file with the `fn` parameter.

    For example:

    Make sure to specify the correct file for the language and phone model that you use.

See Setup for Latin and Cyrillic Languages, on page 83 for specific details on setting up Latin languages.

See Setup for an Asian Language, on page 85 for specific details on setting up an Asian language.

See Setup for RTL Languages, on page 85 for specific details on setting up RTL languages.

Step 3 Click **Submit All Changes**.

Setup for Latin and Cyrillic Languages

If you use Latin and Cyrillic languages such as French or German, you can configure up to four language options for the phone. List of Latin and Cyrillic languages:

- Bulgarian
- Catalan
- Croatian
- Czech
- Danish
- Dutch
- English (UK)
- Finnish
- French (France)
- French (Canada)
- German
- Greek
- Hungarian
- Italian
- Portuguese (Portugal)
- Norwegian
- Polish
- Russian
- Slovak
- Slovenian
- Spanish (Columbia)
- Spanish (Spain)
- Swedish
- Turkish
- Ukraine

To enable the options, set up a dictionary for each language that you want to include. To enable the language, specify a pair of `dn` and `xn` parameters and values in the **Dictionary Server Script** field, for each language that you want to include.

Example for including French and German:

Example for including French (Canada):

```
serv=http://10.74.128.101/Locales/;d1=French-Canada;x1=fr-CA_78xx_68xx-11.3.6.0006xml;
```

```
serv=http://10.74.128.101/Locales/;d1=French-Canada;x1=fr-CA_88xx-11.3.6.0006xml;
```

Note In the above examples **`http://10.74.128.101/Locales/`** is a web folder. The dictionary files are extracted in this web folder and are used in the examples.

To configure this option in the phone configuration XML file (cfg.xml), enter a string in this format:

```
<!-- Language -->
<Dictionary_Server_Script ua="na">serv=http://10.74.10.215/lodapi/resync_files/;d1=French-Canada;x1=fr-CA_88xx-11.3.6.0006.xml;</Dictionary_Server_Script>

<Language_Selection ua="na">French-Canada</Language_Selection>
<Locale ua="na">fr-CA</Locale>
```

Add values for:

- **Language Selection** Parameter as appropriate

  For French: **`French`**

  For French (Canada): **`French-Canada`**

For German: `German`

- **Locale** parameter list as appropriate

  For French: **fr-FR**

  For French (Canada): **fr-CA**

  For German: **de-DE**

After the successful configuration, the user can see the configured language option on the phone under the **Language** menu. User can access the **Language** menu from **Applications** > **Device administration**.

Setup for an Asian Language

If you use an Asian language such as Chinese, Japanese, or Korean, you can only set up one language option for the phone.

You must set up the dictionary and the font for the language. To do this, specify the `d1`, `x1` and `f1` parameters and values in the **Dictionary Server Script** field.

Example for setting up Chinese-Simplified:

Setup for RTL Languages

If you use a Right-to-Left (RTL) language such as Arabic and Hebrew, you can only set up one language option for the phone.

You must set up the dictionary and the font for the language. To do this, specify the d1, x1, and f1 parameters and values in the **Dictionary Server Script** field.

Example for Arabic:

```
serv=http://server.example.com/Locales;d1=Arabic;x1=ar-SA_88xx-11.3.4.xml;f1=ar-SA_88xx-11.3.4.ttf
```

Example for Hebrew:

```
serv=http://server.example.com/Locales;d1=Hebrew;x1=he-IL_88xx-11.3.4.xml;f1=he-IL_88xx-11.3.4.ttf
```

Values for **Language Selection** parameter must be **Arabic** or **Hebrew** as appropriate.

Values for **Locale** parameter must be **ar-SA** for Arabic and **he-IL** for Hebrew.

Specify a Language for the Phone Display

Note Your users can select the language on the phone, from **Settings** > **Device Administration** > **Language**.

Before you begin

The dictionaries and fonts required for the language are set up. See Set Up Dictionaries and Fonts, on page 82 for details.

Procedure

Step 1 On the phone administration web page, go to **Admin Login** > **Advanced** > **Voice** > **Regional**, **Language** section. In the **Language Selection** field, specify the value of the appropriate `dn` parameter value from the **Dictionary Server Script** field, for the language of your choice.

Step 2 Click **Submit All Changes**.

Vertical Service Activation Codes

Parameter	Description
Call Return Code	This code calls the last caller. Defaults to *69.
Blind Transfer Code	Begins a blind transfer of the current call to the extension specified after the activation code. Defaults to *95.
Cfwd All Act Code	Forwards all calls to the extension specified after the activation code. Defaults to *72.
Cfwd All Deact Code	Cancels call forward of all calls. Defaults to *73.
Cfwd Busy Act Code	Forwards busy calls to the extension specified after the activation code. Defaults to *90.
Cfwd Busy Deact Code	Cancels call forward of busy calls. Defaults to *91.
Cfwd No Ans Act Code	Forwards no-answer calls to the extension specified after the activation code. Defaults to *92.
Cfwd No Ans Deact Code	Cancels call forward of no-answer calls. Defaults to *93.
CW Act Code	Enables call waiting on all calls. Defaults to *56.
CW Deact Code	Disables call waiting on all calls. Defaults to *57.

Parameter	Description
CW Per Call Act Code	Enables call waiting for the next call. Defaults to *71.
CW Per Call Deact Code	Disables call waiting for the next call. Defaults to *70.
Block CID Act Code	Blocks caller ID on all outbound calls. Defaults to *61.
Block CID Deact Code	Removes caller ID blocking on all outbound calls. Defaults to *62.
Block CID Per Call Act Code	Removes caller ID blocking on the next inbound call. Defaults to *81.
Block CID Per Call Deact Code	Removes caller ID blocking on the next inbound call. Defaults to *82.
Block ANC Act Code	Blocks all anonymous calls. Defaults to *77.
Block ANC Deact Code	Removes blocking of all anonymous calls. Defaults to *87.
DND Act Code	Enables the do not disturb feature. Defaults to *78.
DND Deact Code	Disables the do not disturb feature. Defaults to *79.
Secure All Call Act Code	Makes all outbound calls secure. Defaults to *16.
Secure No Call Act Code	Makes all outbound calls not secure. Defaults to *17.
Secure One Call Act Code	Makes a secure call. Default: *18.
Secure One Call Deact Code	Disables secure call feature. Default: *19.
Paging Code	The star code used for paging the other clients in the group. Defaults to *96.

Parameter	Description
Call Park Code	The star code used for parking the current call. Defaults to *68.
Call Pickup Code	The star code used for picking up a ringing call. Defaults to *97.
Call Unpark Code	The star code used for picking up a call from the call park. Defaults to *88.
Group Call Pickup Code	The star code used for picking up a group call. Defaults to *98.
Referral Services Codes	These codes tell the IP phone what to do when the user places the current call on hold and is listening to the second dial tone. One or more *code can be configured into this parameter, such as *98, or *97|*98|*123, and so on. Max total length is 79 chars. This parameter applies when the user places the current call on hold (by Hook Flash) and is listening to second dial tone. Each *code (and the following valid target number according to current dial plan) entered on the second dial-tone triggers the phone to perform a blind transfer to a target number that is prepended by the service *code. For example, after the user dials *98, the IP phone plays a special dial tone called the Prompt Tone while waiting for the user the enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the phone sends a blind REFER to the holding party with the Refer-To target equals to *98<target_number>. This feature allows the phone to hand off a call to an application server to perform further processing, such as call park. The *codes should not conflict with any of the other vertical service codes internally processed by the IP phone. You can empty the corresponding *code that you do not want to the phone to process.

Parameter	Description
Feature Dial Services Codes	These codes tell the phone what to do when the user is listening to the first or second dial tone. One or more *code can be configured into this parameter, such as *72, or *72|*74|*67|*82, and so forth. The maximum total length is 79 characters. This parameter applies when the user has a dial tone (first or second dial tone). Enter *code (and the following target number according to current dial plan) entered at the dial tone triggers the phone to call the target number prepended by the *code. For example, after user dials *72, the phone plays a prompt tone awaiting the user to enter a valid target number. When a complete number is entered, the phone sends a INVITE to *72<target_number> as in a normal call. This feature allows the proxy to process features like call forward (*72) or BLock Caller ID (*67). The *codes should not conflict with any of the other vertical service codes internally processed by the phone. You can empty the corresponding *code that you do not want to the phone to process. You can add a parameter to each *code in Features Dial Services Codes to indicate what tone to play after the *code is entered, such as *72‘c‘|*67‘p‘. Below are a list of allowed tone parameters (note the use of back quotes surrounding the parameter without spaces) • c = Cfwd Dial Tone • d = Dial Tone • m = MWI Dial Tone • o = Outside Dial Tone • p = Prompt Dial Tone • s = Second Dial Tone • x = No tones are place, x is any digit not used above If no tone parameter is specified, the phone plays Prompt tone by default. If the *code is not to be followed by a phone number, such as *73 to cancel call forward, do not include it in this parameter. In that case, simple add that *code in the dial plan and the phone sends INVITE *73@..... as usual when user dials *73.

Cisco IP Conference Phone 7832 Series Documentation

Refer to publications that are specific to your language and phone model, and phone firmware release. Navigate from the following documentation URL:

https://www.cisco.com/c/en/us/support/collaboration-endpoints/ip-phone-7800-series-multiplatform-firmware/tsd-products-support-series-home.html