Cisco IP Phone Customization

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Phone Information and Display Settings

The phone web user interface allows you to customize settings such as the phone name, background picture, logo, and screen saver.

Configure the Phone Name

Before you begin

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

Procedure

Step 1 Select **Voice** > **Phone**.

Step 2 Under **General**, enter the phone name in the **Station Display Name** field.

This name displays on the phone LCD in the top left corner.

Step 3 Click **Submit All Changes**.

Change Wallpaper from the Phone Page

Your administrator can allow you to change the default wallpaper on your phone to one of the wallpapers available.

Procedure

Step 1 On the phone web page, select **User Login** > **Voice** > **User**.

Step 2 In the **Phone Background** field of the **Screen** section, select any of the options as a phone wallpaper.

- **Default**: Phone does not have any wallpaper. If no wallpaper is added to the phone screen, the phone screen displays monochrome wallpaper.
- **Logo**: In the phone web page you can select **Logo** as your phone background option. The logo that you add in the **Logo URL** is used as the wallpaper.

  Caution Do not exceed a maximum length of 255 characters for the **Logo URL** or **Picture Download URL**.

  The logo display area is the center of the phone screen. The logo display area size of the phone is 128x128 pixels. If original logo size does not fit display area, the logo scales to fit the display area.

Step 3 Click **Submit All Changes**.

Add a Logo as the Boot Display

If you want your user to see a logo icon when the phone restarts, enable this feature from the phone web page.

Before you begin

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

Procedure

Step 1 Select **Voice** > **User**.

Step 2 In the **Screen** section, select **Logo** from the **Boot Display** field. In the **Logo URL** field, enter a URL or path for the location where the logo image is saved.

You can also download a picture and add it as a boot display: select **Download Picture** from the **Boot Display** field. In the **Picture Download URL** field, enter a URL or path for the location where the picture is saved.

The logo must be a .jpg or a .png file. The phone has a fixed display area. So, if the original logo size doesn't fit into the display area, you need to scale it to fit the screen. For the Cisco IP Phone 7811, 7821, 7841 and 7861 the logo display area is at the mid-center of the phone screen. The display area size of the Cisco IP Phone 7811 is 48x48. The display area size of the Cisco IP Phone 7821, 7841, and 7861 is 64x64.

Step 3 Click **Submit All Changes**.

Adjust Backlight Timer from Configuration Utility

You can save energy by disabling the backlight on each phone at a preset time. The phone's desktop remains visible, even with the backlight off.

User can select **User Login** > **Advanced** > **Voice** > **User** and can adjust the backlight timer.

Backlights are not supported on the Cisco IP Phone 7811.

Before you begin

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

Procedure

Step 1 Select **Voice** > **User**.

Step 2 Under Screen, select a setting for the **Back Light Timer** parameter.

Step 3 In the **LCD Contrast** field, enter a number for the desired contrast.

Configure the Number of Call Appearances Per Line

Phones that support multiple call appearances on a line can be configured to specify the number of calls to allow on the line.

Before you begin

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

Procedure

Step 1 Select **Voice** > **Phone**.

Step 2 In the **Miscellaneous Line Key Settings** section, use the **Call Appearances Per Line** drop-down list box to specify the number of calls per line to allow.

Step 3 Click **Submit All Changes**.

Reverse Name Lookup for Incoming and Outgoing Calls

Reverse name lookup searches for the name of a number in an incoming, outgoing, conference, or transfer call. The reverse name lookup acts when the phone cannot find a name using the service provider directory, Call History, or your contacts. Reverse name lookup needs a valid LDAP Directory or XML Directory configuration.

The reverse name lookup searches the phone's external directories. When a search succeeds, the name is placed in the call session and in the call history. For simultaneous, multiple phone calls, reverse name lookup

searches for a name to match the first call number. When the second call connects or is placed on hold, reverse name lookup searches for a name to match the second call.

Reverse name lookup is enabled by default.

Reverse name lookup searches the directories in the following order:

1. Phone contacts
2. Call History
3. LDAP Directory
4. XML Directory

Note The phone searches the XML directory using this format: `directory_url?n=incoming_call_number`.

Example: For a multiplatform phone using a third-party service, the phone number (1234) search query has this format, `http://your-service.com/dir.xml?n=1234`.

Enable and Disable Reverse Name Lookup

Before you begin

- Configure one of these directories before you can enable or disable the reverse name lookup:
  - LDAP Corporate Directory
  - XML Directory
- Access the phone administration web page. See Access the Phone Web Page.

Procedure

Step 1 Select **Voice** > **Phone**.

Step 2 In the **Supplementary Services** area, set the **Reverse Phone Lookup Serv** to:

- **Yes**–Enable the reverse name lookup feature.
- **No**–Disable the reverse name lookup feature.

Step 3 Click **Submit All Changes**.

Step 4 Alternative method is to use the `config.xml` file to provision the reverse name lookup feature.

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

Call Features Configuration

Enable Call Transfer

Before you begin

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

Procedure

Step 1 Select **Voice** > **Phone**.

Step 2 Under **Supplementary Services**, choose **Yes** for each of the transfer services that you want to enable:

- **Attn Transfer Serv**—Attended call transfer service. The user answers the call before transferring it.
- **Blind Transfer Serv**—Blind call transfer service. The user transfers the call without speaking to the caller.

Step 3 To disable a transfer service, set the field to **No**.

Step 4 Click **Submit All Changes**.

Call Forward

To enable call forwarding, you can enable the feature in two places: on the Voice tab and the User tab of the phone web page.

Enable Call Forwarding on Voice Tab

Perform this task if you want to enable call forward for a user.

Before you begin

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

Procedure

Step 1 Select **Voice** > **Phone**.

Step 2 Under **Supplementary Services**, choose **Yes** for each of the call forwarding services that you want to enable:

- **Cfwd All Serv**—Forwards all calls.
- **Cfwd Busy Serv**—Forwards calls only if the line is busy.
- **Cfwd No Ans Serv**—Forwards calls only if the line is not answered.

Step 3 Click **Submit All Changes**.

Related Topics

DND and Call Forwarding Status Sync
Enable Feature Key Sync
Enable Call Forwarding Status Sync via XSI Service

Enable Call Forwarding on User Tab

Perform the following task if you want to give a user the ability to modify the call forward settings from the Configuration Utility page.

Before you begin

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

Procedure

Step 1 Select **Voice** > **User**.
Step 2 Under **Call Forward**, choose **Yes** for CFWD Setting.
Step 3 Click **Submit All Changes**.

Enable Conferencing

Before you begin

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

Procedure

Step 1 Select **Voice** > **Phone**.
Step 2 Under **Supplementary Services**, choose **Yes** in the **Conference Serv** drop-down list box.
Step 3 Click **Submit All Changes**.

Enable Remote Call Recording with SIP REC

You can enable call recording on a phone so that your user can record an active call. The recording mode configured on the server controls the display of the recording softkeys for each phone.

Table 1: Recording Mode and Recording Softkeys

Recording Mode in Server	Recording Softkeys Available on the Phone
Always	No softkeys available. Your user can't control recording from the phone. Recording starts automatically when a call is connected.
Always with Pause/Resume	PauseRec ResumeRec When a call is connected, recording starts automatically and your user can control the recording.
On Demand	Record PauseRec ResumeRec When a call is connected, recording starts automatically but the recording is not saved until the user presses the **Record** softkey. Your user sees a message when recording state changes.
On Demand with User Initiated Start	Record PauseRec StopRec ResumeRec The recording only starts when your user presses the **Record** softkey. Your user sees a message when recording state changes.

During a recording, your user sees different icons which depend on the recording state. The icons are displayed on the Calls screen and also on the line key on which the user is recording a call.

Table 2: Recording Icons

Icon	Meaning
◉	Recording in progress.
◉	Recording paused

Before you begin

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

Procedure

Step 1 Select **Voice** > **Phone**.

Step 2 In the **Supplementary Services** section, click **Yes** or click **No** to enable or to disable call recording in the **Call Recording Serv** field.

Step 3 (Optional) In the **Programmable Softkeys** section, to enable softkeys, add a string in this format in the **Connected Key List** and **Conferencing Key List** fields.

```
crdstart;crdstop;crdpause;crdresume
```

Step 4 In the phone web page, click the **Ext(n)** tab that requires call recording.

Step 5 In the **SIP Settings** section, in the **Call Recording Protocol**, select **SIPREC** as the call recording protocol.

For details on the **SIP Settings** fields, see SIP Settings.

Step 6 Click **Submit All Changes**.

Enable Remote Call Recording with SIP INFO

You can enable call recording on a phone so that your user can record an active call.

During a recording, your user sees different icons which depend on the recording state. The icons are displayed on the Calls screen and also on the line key on which the user is recording a call.

Your user presses the following softkeys to control the phone recording:

- **Record**
- **StopRec**

The recording only starts when your user presses the **Record** softkey. Your user sees a message when recording state changes and the recording icon displays on the call screen.

Once a phone recording starts, the **StopRec** softkey can work. The recording stops when your user presses the **StopRec** softkey. Your user sees a message when the recording state changes.

Table 3: Recording Icons

Icon	Meaning
◉	Recording in progress.

Before you begin

- You need to set up call recording on the call control system.
- Access the phone administration web page. See Access the Phone Web Page.

Procedure

Step 1 Select **Voice** > **Phone**.

Step 2 In the **Supplementary Services** section, click **Yes** or click **No** to enable or to disable call recording in the **Call Recording Serv** field.

Step 3 (Optional) In the **Programmable Softkeys** section, to enable softkeys, add a string in this format in the **Connected Key List** and **Conferencing Key List** fields.

```
crdstart;crdstop;crdpause;crdresume
```

Step 4 In the phone web page, click the **Ext(n)** tab that requires call recording.

Step 5 In the **SIP Settings** section, in the **Call Recording Protocol**, select **SIPINFO** as the call recording protocol.

For details on **SIP Settings** fields, see SIP Settings.

Step 6 Click **Submit All Changes**.

Configure Missed Call Indication with the Configuration Utility

Before you begin

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

Procedure

Step 1 Select **Voice** > **User**.

The user can select **User Login** > **Voice** > **User**.

Step 2 Click **Submit All Changes**.

Enable Do Not Disturb

You can allow persons to turn the Do not disturb feature on or off. The caller receives a message that the person is unavailable. A person can press the **Ignore** softkey on the phone to divert an incoming call to another destination.

If the feature is enabled for the phone, users can turn the feature on or off with the DND softkey.

Before you begin

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

Procedure

Step 1 Choose **Voice** > **User**.

Step 2 In the **Supplementary Services** area, select **Yes** in the **DND Setting** drop-down list.

Step 3 Click **Submit All Changes**.

When you select a line (multiline phone), a Do Not Disturb banner displays at the top of the phone screen.

What to do next

Change another setting to ensure that multiline phones correctly display the Do not disturb (currently, a steady, green color) status for each selected or unselected line. See DND and Call Forwarding Status Sync.

Users can enable or turn off the DND feature for each phone line if you configure star codes for DND. See Configure Star Codes for DND, on page 10.

Related Topics

DND and Call Forwarding Status Sync
Enable Feature Key Sync
Enable DND Status Sync via XSI Service

Configure Star Codes for DND

You can configure star codes that a user dials to turn on or off the do not disturb (DND) feature on a phone.

Before you begin

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

Procedure

Step 1 Select **Voice** > **Regional**.
Step 2 In the **Vertical Service Activation Codes** area, enter *78 in the **DND Act Code** field.
Step 3 In the **Vertical Service Activation Codes** area, enter *79 in the **DND Deact Code** field.
Step 4 Click **Submit All Changes**.

Set Up a Call Center Agent Phone

You can enable a phone with Automatic Call Distribution (ACD) features. This phone acts as a call center agent's phone and can be used to trace a customer call, to escalate any customer call to a supervisor in emergency, to categorize contact numbers using disposition codes, and to view customer call details.

Before you begin

- Set up the phone as a call center phone on the BroadSoft server.
- Access the phone administration web page. See Access the Phone Web Page.

Procedure

Step 1 Select **Voice** > **Ext(n)**.
Step 2 In the **ACD Settings** section, set up the fields as described in ACD Settings.

Step 3 Click **Submit All Changes**.

Set Up a Phone for Presence

Before you begin

- Set up the Broadsoft server for XMPP.
- Access the phone administration web page. See Access the Phone Web Page.

Procedure

Step 1 Select **Voice** > **Phone**.
Step 2 In the **Broadsoft XMPP** section, set the fields as described in Broadsoft XMPP.
Step 3 Click **Submit All Changes**.

Shared Lines

A shared line is a directory number that appears on more than one phone. You can create a shared line by assigning the same directory number to different phones.

Incoming calls display on all phones that share a line, and anyone can answer the call. Only one call remains active at a time on a phone.

Call information displays on all phones that are sharing a line. If somebody turns on the privacy feature, you do not see the outbound calls made from the phone. However, you see inbound calls to the shared line.

All phones with a shared line ring when a call is made to the line. If you place the shared call on hold, anyone can resume the call by pressing the corresponding line key from a phone that shares the line. You can also press the **Select** button if the Resume icon is displayed.

The following shared line features are supported:

- Line Seizure
- Public Hold
- Private Hold
- Silent Barge (only through enabled programmable softkey)

The following features are supported as for a private line

- Transfer
- Conference
- Call Park / Call Retrieve

- Call Pickup
- Do Not Disturb
- Call Forward

You can configure each phone independently. Account information is usually the same for all IP phones, but settings such as the dial plan or preferred codec information can vary.

Configure a Shared Line

You can create a shared line by assigning the same directory number to different phones on the phone web page.

Before you begin

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

Procedure

Step 1 Select **Voice**.

Step 2 Click the **Ext_n** tab of the extension that is shared.

Step 3 Under **General** in the Line Enable list, choose **Yes**.

Step 4 Under **Share Line Appearance** in the Share Ext list, select **Shared**.

If you set this extension to **Private**, the extension does not share calls, regardless of the Share Call Appearance setting on the Phone tab. If you set this extension to **Shared**, calls follow the Share Call Appearance setting on the Phone tab.

Step 5 In the **Shared User ID field**, enter the user ID of the phone with the extension that is being shared.

Step 6 In the **Subscription Expires** field, enter the number of seconds before the SIP subscription expires. The default is 60 seconds.

Until the subscription expires, the phone gets NOTIFY messages from the SIP server on the status of the shared phone extension.

Step 7 In the **Restrict MWI** field, set the message waiting indicator:

- **Yes**—Lights only for messages on private lines (SIP).
- **No**—Lights for all messages.

Step 8 Under **Proxy and Registration**, enter the IP address of the proxy server in the Proxy field.

Step 9 Under **Subscriber Information,** enter a Display Name and User ID (extension number) for the shared extension.

Step 10 In the Phone tab, under **Miscellaneous Line Key Settings**, configure SCA Barge-In Enable:

- **Yes**—Allows users to take over the call on a shared line.
- **No**—Prevents users from taking over the call on a shared line.

Step 11 Click **Submit All Changes.**

Configure Voice Mail

You can configure the internal or external phone number or URL for the voice mail system. If you are using an external voice mail service, the number must include any digits required to dial out and any required area code

Before you begin

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

Procedure

Step 1 Select **Voice** > **Phone**.

Step 2 Under **General**, enter the **Voice Mail Number**.

Step 3 Click **Submit All Changes**. The phone reboots.

Configure Voice Mail for each Extension

Before you begin

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

Procedure

Step 1 Select **Voice** > **Extn**.

Step 2 Under **Call Feature Settings**, enter the **Voice Mail Server**.

Step 3 (Optional) Enter the **Voice Mail Subscribe Interval**; the expiration time in seconds, of a subscription to a voice mail server.

Step 4 Click **Submit All Changes**.

The phone reboots.

Configure the Message Waiting Indicator

You can configure the Message Waiting Indicator for separate extensions on the phone. The Message Waiting Indicator lights based on the presence of new voicemail messages in the mailbox.

You can enable the indicator at the top of your IP phone to light when voice mail is left, or display a seeing message waiting notification.

Before you begin

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

Procedure

Step 1 Select **Voice** > **Extn**.

Step 2 Under **Call Feature Settings** in the **Message Waiting**, choose **Yes** to enable.

Assign a Ringtone to an Extension

Before you begin

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

Procedure

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

Step 2 Under **Call Feature Settings**, use the **Default Ring (n)** drop-down list box to specify one of the following:

- No Ring
- Choose one of the available 12 ringtones.

Step 3 Click **Submit All Changes**.

Add Distinctive Ringtone

You can configure the characteristics of each ring tone using a ring tone script. When phone receives SIP Alert-INFO message and the message format is correct, then the phone plays the specified ringtone. Otherwise, the phone plays the default ringtone.

Procedure

In a ring tone script, assign a name for the ring tone and add the script to configure a distinctive ringtone in the format:

```
n=ring-tone-name;h=hint;w=waveform-id-or-path;c=cadence-id;b=break-time;t=total-time
```

where:

n = ring-tone-name that identifies this ring tone. This name appears on the Ring Tone menu of the phone. The same name can be used in a SIP Alert-Info header in an inbound INVITE request to tell the phone to play the corresponding ring tone. The name should contain the same characters allowed in a URL only.

h = hint used to SIP Alert-INFO rule.

w = waveform-id-or-path which is the index of the desired waveform to use in this ring tone. The built-in waveforms are:

- 1 = Classic phone with mechanical bell
- 2 = Typical phone ring
- 3 = Classic ring tone
- 4 = Wide-band frequency sweep signal

You can also enter a network path (url) to download a ring tone data file from a server. Add the path in this format:

```
w=[tftp://]hostname[:port]/path
```

c = is the index of the desired cadence to play the given waveform. 8 cadences (1–8) as defined in <Cadence 1> through <Cadence 8>. Cadence-id can be 0 If w=3,4, or an url. Setting c=0 implies the on-time is the natural length of the ring tone file.

b = break-time that specifies the number of seconds to break between two bursts of ring tone, such as b=2.5.

t = total-time that specifies the total number of seconds to play the ring tone before it times out.

Configure the Audio Settings

The user can modify volume settings by pressing the volume control button on the phone, then pressing the **Save** softkey.

Before you begin

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

Procedure

Step 1 Select **Voice** > **User**.

Step 2 In the **Audio Volume** section, configure a volume level of 1 (quiet) through 10 (loudest):

- **Ringer Volume**—Sets the ringer volume.
- **Speaker Volume**—Sets the volume for the full-duplex speakerphone.
- **Headset Volume**—Sets the headset volume.
- **Handset Volume**—Sets the handset volume.

Step 3 Click **Submit All Changes**.

Specify Audio Compliance Standard

You can specify a compliance standard for the audio tuning for the phone. When a compliance standard is specified, the acoustic parameters that conform to the specified standard are loaded to the phone.

Before you begin

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

Procedure

Step 1 Go to **Voice** > **User**.

Step 2 In the **Audio Compliance** section, choose an option from the **Compliant Standard** drop-down list as described in Audio Compliance.

Step 3 Click **Submit All Changes**.

User Access Control

The Cisco IP Phone respects only the "ua" user access attribute. For a specific parameter, the "ua" attribute defines access by the user account to the administration web server. If the "ua" attribute is not specified, the phone applies the factory default user access for the corresponding parameter. This attribute does not affect access by the admin account.

Note The value of the element attribute encloses within double quotes.

The "ua" attribute must have one of the following values:

- na – no access
- ro – read-only
- rw – read/write

Phone Web Server

The web server allows administrators and users to log in to the phone by using a phone web user interface. Administrators and users have different privileges and see different options for the phone based on their role.

Configure the Web Server from the Phone Screen Interface

Use this procedure to enable the phone web user interface from the phone screen.

Procedure

Step 1 Press **Applications** .

Step 2 Select **Network configuration** > **Web Server**.

Step 3 Select **On** to enable or **Off** to disable.

Step 4 Press **Set**.

Direct Action URL

If the Enable Direct Action URL setting is set to "Yes ", these Direct action URLs are accessible only for the admin. If Admin user is password protected, the client provides a login prompt before these are accessed. The Direct Action URLs are accessible via the phone web page via the path `/admin/<direct_action>`. The syntax is:

`http[s]://<ip_or_hostname>/admin/<direct_action>[?<url>]`

For example, **`http://10.1.1.1/admin/resync?http://server_path/config.xml`**

The following table provides a list of the different direct avtion URLs that are supported.

direct_action	Description
resync	Initiates a one-time resync of the config file specified by URL. The URL to resync is provided by appending ? followed by the URL. The URL specified here will not be saved anywhere in the phone settings. **Example** http://10.1.1.1/admin/resync?http://my_provision_server.com/cfg/device.cfg
upgrade	Initiates an upgrade of a phone to the specified load. The load is specified via the upgrade rule. the rule is specified by appending ? followed by URL path to the load. The upgrade rule specified is one time only and will not be saved in any property setting. **Example** http://10.1.1.1/admin/upgrade?http://my_upgrade_server.com/loads/sip88xx.11.0.0MP2.123.loads
updateca	Initiates a one-time install of the Custom Certificate Authority (Custom CA) specified by the URL. The URL to download is provided by appending ? followed by the URL. The URL specified here will not be saved anywhere in the phone settings. **Example** http://10.1.1.1/admin/updateca?http://my_cert_server.com/certs/myCompanyCA.pem
reboot	Initiates a reboot of the phone. Does not take any parameter with ? **Example** http://10.1.1.1/admin/reboot

direct_action	Description
cfg.xml	Downloads a snapshot of the phone configuration in XML format. The passwords are hidden for security. Most of the information here corresponds to the properties on the phone web page under **Voice** tab. **Example** http://10.1.1.1/admin/cfg.xml
status.xml	Downloads a snapshot of the phone status in XML format. Most of the information here corresponds to the **Status** tab in the phone web page. **Example** http://10.1.1.1/admin/status.xml
screendump.bmp	Downloads a screenshot of the phone LCD UI at the time when this action is initiated. **Example** http://10.1.1.1/admin/screendump.bmp
log.tar	Downloads a set of archived logs stored on the phone. **Example** http://10.1.1.1/admin/log.tar

Enable Access to Phone Web Interface

Before you begin

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

Procedure

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

Step 2 Under the **System Configuration** section, choose **Yes** from the **Enable Web Server** drop-down list box.

Step 3 In the **Enable Protocol** drop-down list box, choose Http or Https.

Step 4 In the **Web Server Port** field, enter the port to access the web server. The default is port 80 for HTTP or port 443 for HTTPS.

Step 5 In the **Enable Web Admin Access** drop-down list box, you can enable or disable local access to the **Admin Login** of the phone web user interface. Defaults to **Yes** (enabled).

Step 6 In the **Admin Password** field, enter a password if you want the system administrator to log in to the phone web user interface with a password. The password prompt appears when an administrator clicks **Admin Login**. The minimum password length can be 4 characters or the maximum password length is 127 characters.

Note The password can contain any character except the Space key.

Step 7 In the **User Password** field, enter a password if you want users to log in to the phone web user interface with a password. The password prompt appears when users click **User Login**. The minimum password length can be 4 characters or the maximum password length is 127 characters.

Note The password can contain any character except the Space key.

Step 8 Click **Submit All Changes**.

XML Services

The phones provide support for XML services, such as an XML Directory Service or other XML applications. For XML services, only HTTP and HTTPS support is available.

The following Cisco XML objects are supported:

- CiscoIPPhoneMenu
- CiscoIPPhoneText
- CiscoIPPhoneInput
- CiscoIPPhoneDirectory
- CiscoIPPhoneIconMenu
- CiscoIPPhoneStatus
- CiscoIPPhoneExecute
- CiscoIPPhoneImage
- CiscoIPPhoneImageFile
- CiscoIPPhoneGraphicMenu
- CiscoIPPhoneFileMenu
- CiscoIPPhoneStatusFile
- CiscoIPPhoneResponse
- CiscoIPPhoneError
- CiscoIPPhoneGraphicFileMenu
- Init:CallHistory
- Key:Headset
- EditDial:n

The full list of supported URIs is contained in *Cisco Unified IP Phone Services Application Development Notes for Cisco Unified Communications Manager and Multiplatform Phones*, located here:

https://www.cisco.com/c/en/us/support/collaboration-endpoints/unified-ip-phone-7800-series/products-programming-reference-guides-list.html

XML Directory Service

When an XML URL requires authentication, use the parameters **`XML UserName`** and **`XML Password`**.

The parameter **XML UserName** in XML URL is replaced by $XML UserName.

For example:

The parameter XML UserName is **cisco**. The XML Directory Service URL is **http://www.sipurash.compath?username=$XML_User_Name**.

This results in the request URL: `http://www.sipurash.com/path?username=cisco`.

XML Applications

When authentication is required for CGI/Execute URL via Post from an external application (for example, a web application) to the phones, the parameter `CISCO XML EXE Auth Mode` is used in 3 different scenarios:

- Trusted—No authentication is performed (local user password is set or not). This is the default.
- Local Credential—Authentication is based on digest authentication using the local user password, if the local user password is set. If not set, then no authentication is performed.
- Remote Credential—Authentication is based on digest authentication using the remote username/password as set in the XML application on the web page (to access an XML application server).

Macro Variables

You can use macro variables in XML URLs. The following macro variables are supported:

- User ID—UID1, UID2 to UIDn
- Display name—DISPLAYNAME1, DISPLAYNAME2 to DISPLAYNAMEn
- Auth ID—AUTHID1, AUTHID2 to AUTHIDn
- Proxy—PROXY1, PROXY2 to PROXYn
- MAC Address using lowercase hex digits—MA
- Product Name—PN
- Product Series Number—PSN
- Serial Number—SERIAL_NUMBER

The following table shows the list of macros supported on the phones:

Macro Name	Macro Expansion
$	The form $$ expands to a single $ character.
A through P	Replaced by general-purpose parameters GPP_A through GPP_P.
SA through SD	Replaced by special purpose parameters GPP_SA through GPP_SD. These parameters hold keys or passwords used in provisioning. **Note** $SA through $SD are recognized as arguments to the optional resync URL qualifier, --key.

Macro Name	Macro Expansion
MA	MAC address using lowercase hex digits (000e08aabbcc).
MAU	MAC address using uppercase hex digits (000E08AABBCC).
MAC	MAC address using lowercase hex digits with a colon to separate hex digit pairs (00:0e:08:aa:bb:cc).
PN	Product Name; for example, IP Phone 7861.
PSN	Product Series Number; for example, 7861.
SN	Serial Number string; for example, 88012BA01234.
CCERT	SSL Client Certificate status, installed or not installed.
IP	IP address of the phone within its local subnet; for example, 192.168.1.100.
EXTIP	External IP of the phone, as seen on the internet; for example, 66.43.16.52.
SWVER	Software version string; for example, 2.0.6(b). Use the software version string to compare against the current phone's firmware load, with one of the following methods: • **With quotes, "$SWVER"**–Variable acts as a string in firmware load name comparisons. For `"$SWVER" eq "sip8845_65.1-0129-18-0356dev.loads"`, the phone model number and load number are part of the comparison. • **Without quotes, $SWVER**–Variable is parsed to determine a build number, plus major, minor, and micro revision numbers. For example, when the `sip88xx.11-1-1MSR-1dev.loads` and `sip8845_65.11-1-1MSR-1dev.loads` firmware names are parsed, the result ignores the model number and load number. The result for both firmware names yields a major revision=`1`, minor revision=`1`, micro revision=`1MSR`, and build number=`1`.
HWVER	Hardware version string; for example, 1.88.1.
PRVST	Provisioning State (a numeric string): • -1 = explicit resync request • 0 = power-up resync • 1 = periodic resync • 2 = resync failed, retry attempted
UPGST	Upgrade State (a numeric string): • 1 = first upgrade attempt • 2 = upgrade failed, retry attempt
UPGERR	Result message (ERR) of previous upgrade attempt; for example, http_get failed.
PRVTMR	Seconds since last resync attempt.

Macro Name	Macro Expansion
UPGTMR	Seconds since last upgrade attempt.
REGTMR1	Seconds since Line 1 lost registration with SIP server.
REGTMR2	Seconds since Line 2 lost registration with SIP server.
UPGCOND	Legacy macro name.
SCHEME	File access scheme (TFTP, HTTP, or HTTPS, obtained after parsing resync or upgrade URL).
METH	Deprecated alias for SCHEME, do not use.
SERV	Request target server hostname.
SERVIP	Request target server IP address (following DNS lookup).
PORT	Request target UDP/TCP port.
PATH	Request target file path.
ERR	Result message of resync or upgrade attempt.
UIDn	The contents of the Line n UserID configuration parameter.
ISCUST	If unit is customized, value=1, otherwise 0. **Note** Customization status viewable on Web UI Info page.
INCOMINGNAME	Name associated with first connected, ringing, or inbound call.
REMOTENUMBER	Phone number of first connected, ringing, or inbound call. If there are multiple calls, the data associated with the first call found is provided.
DISPLAYNAMEn	The contents of the Line N Display Name configuration parameter.
AUTHIDn	The contents of the Line N auth ID configuration parameter.

Configure a Phone to Connect to an XML Directory Service

Before you begin

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

Procedure

Step 1 Select **Voice** > **Phone**.

Step 2 Enter this information:

- XML Directory Service Name—Name of the XML Directory. Displays on the user's phone as a directory choice.
- XML Directory Service URL—URL where the XML Directory is located.

Step 3 Click **Submit All Changes**.

Configure a Phone to Connect to an XML Application

Before you begin

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

Procedure

Step 1 Select **Voice** > **Phone**.

Step 2 Enter this information:

- XML Application Service Name—Name of the XML application. Displays on the user's phone as a menu item.
- XML Application Service URL—URL where the XML application is located.

If you configure an unused line button to connect to an XML application, the button connects to the URL configured above. If this is not what you want, you need to enter a different URL when you configure the line button.

Step 3 Click **Submit All Changes**.