



# Cisco IP Conference 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

#### Procedure

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- Step 1** In the phone web user interface, navigate to **Admin Login > advanced > 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**.
-

## Customize the Startup Screen with Text and Picture

You can create a text or 128-by-48 pixel by 1-bit deep image logo to display when the Cisco IP Phone boots up. A logo displays during the boot sequence for a short period after the Cisco logo displays.

### Procedure

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**Step 1** Click **Admin Login** > **advanced** > **Voice** > **User**.

**Step 2** In the **Screen** section, select any option from the **Boot Display** field.

- **Default:** Displays a blank screen or existing screen as the startup screen.
- **Download Picture:** Displays a picture as the startup screen. Enter the path in the **Picture Download URL** field.

For example:

```
http://10.64.84.147/pictures/image04_128x48.png
```

When you enter an incorrect URL to download a new wallpaper, the phone fails to upgrade to the newer wallpaper and displays the existing downloaded wallpaper. If the phone does not have any wallpaper downloaded earlier, it displays a gray screen.

The supported phone image file attributes are: Bitmap format, one bit-per-pixel color, size 128-by-48 pixels. You can also use a TFTP server.

- **Logo:** Displays a logo as the startup screen. See [Add Logo as Boot Display](#), on page 2.
- **Text:** Displays a text as the startup screen. Enter text in the **Text Display** field. Enter up to two lines of text. Each line must be less than 32 characters. Insert a new line character (\n) and escape code (%0a) between the two lines.

For example, Super\n%0aTelecom displays:

```
Super
Telecom
```

Use the + character to add spaces for formatting. You can add multiple + characters before and after the text to center it.

**Step 3** Click **Submit All Changes**.

The phone reboots, retrieves the .png file, and displays the picture when it next boots.

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## Add Logo as Boot Display

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

### Procedure

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- Step 1** On the phone web page, select **Admin Login > Advanced > 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 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 scale to fit it. For the Cisco IP Phone 7832, the logo display area is at the mid-center of the phone screen. The display area size of the Cisco IP phone 7832 is 48x48.
- Step 3** Click **Submit All Changes**.
- 

## 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.

### Procedure

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- Step 1** Click **Admin Login > advanced > 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**.
- 

## Call Features Configuration

### Enable Call Transfer

#### Procedure

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- Step 1** Click **Admin Login > advanced > 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**.

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## 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.

#### Procedure

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**Step 1** On the Configuration Utility page, click **Admin Login > advanced > 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**.

---

### 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.

#### Procedure

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**Step 1** On the Configuration Utility page, click **Admin Login > advanced > Voice > User**.

**Step 2** Under **Call Forward**, choose **Yes** for CFWD Setting.

**Step 3** Click **Submit All Changes**.

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## Enable Conferencing

### Procedure

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- Step 1** In the phone web user interface, navigate to **Admin Login > advanced > Voice > Phone**.
  - Step 2** Under **Supplementary Services**, choose **Yes** in the **Conference Serv** drop-down list box.
  - Step 3** Click **Submit All Changes**.
- 

## Configure Missed Call Indication with the Configuration Utility

If a user is not on an active or held call and misses a call, the user needs to know about the missed call. To alert the user, configure the **Handset LED Alert** field on the Configuration Utility page. If you set this field to **Voicemail, Missed Call**, the LED on the Handset will turn on when the user has recently missed a call.

### Procedure

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- Step 1** On the Configuration Utility page, select **Admin Login > Advanced > Voice > User**.
  - Step 2** In the **Supplementary Services** section, choose **Voicemail, Missed Call** in the **Handset LED Alert** drop-down list box.  
The user can select **User Login > Voice > User**.
  - Step 3** Click **Submit All Changes**.
- 

## Enable Do Not Disturb

You can allow users to turn the do not disturb feature on or off. The caller receives a message that the user is unavailable. Users can press the **Ignore** softkey on their phones to divert a ringing call to another destination.

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

### Procedure

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- Step 1** On the Configuration Utility page, select **Admin Login > advanced > Voice > User**.
  - Step 2** In the **Supplementary Services** section, choose **Yes** in the **DND Setting** drop-down list box.
  - Step 3** Click **Submit All Changes**.
-

## Configure Synchronization of DND and Call Forward

Enable synchronization of Do Not Disturb (DND) and Call Forward to allow changes to these features that are made on the phone to be made on the server. Changes made on the server are also made on the phone.

### Procedure

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- Step 1** On the Configuration Utility page, select **Admin Login > advanced > Voice > Ext [n]** (where [n] is the extension number).
- Step 2** In the **Call Feature Settings** section, set the **Feature Key Sync** field to **Yes**.
- Step 3** Click **Submit All Changes**.
- 

## 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.

### Procedure

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- Step 1** On the Configuration Utility page, select **Admin Login > advanced > 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**.
- 

## 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.

### Procedure

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- Step 1** On the Configuration Utility page, click **Admin Login > advanced > 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**.

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## 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

### Procedure

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- Step 1** Click **Admin Login > advanced > Voice > Phone**.
- Step 2** Under **General**, enter the **Voice Mail Number**.
- Step 3** Click **Submit All Changes**. The phone reboots.
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## Configure Voice Mail for each Extension

### Procedure

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- Step 1** Click **Admin Login > advanced > 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.



### Procedure

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- Step 1** Click **Admin Login > advanced > Voice > Extn.**
- Step 2** Under **Call Feature Settings** in the **Message Waiting**, choose **Yes** to enable.
- 

## Assign a Ring Tone to an Extension

### Procedure

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- Step 1** On the Configuration Utility page, select **Admin Login > advanced > 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 ring tones.
- Step 3** Click **Submit All Changes**.
- 

## Configure the Audio Settings

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

### Procedure

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- Step 1** Click **Admin Login > advanced > Voice > User**.
- Step 2** In the **Audio Volume** section, configure a volume level between 1 and 10, with 1 being the lowest level:
- **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.
  - **Electronic HookSwitch Control**—Enables or disables the EHS feature.
- 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.




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**Note**

The value of the element attribute encloses within double quotes.

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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.

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**Procedure**

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- Step 1** Press **Settings**.
  - 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](http://10.1.1.1/admin/resync?http://server_path/config.xml)

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

direct_action	Description	Example
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.	<a href="http://10.1.1.1/admin/resync/http://my_provision_server.com/cfg/device.cfg">http://10.1.1.1/admin/resync/http://my_provision_server.com/cfg/device.cfg</a>
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.	<a href="http://10.1.1.1/admin/upgrade/http://my_upgrade_server.com/loads/88x110MP2123.tar">http://10.1.1.1/admin/upgrade/http://my_upgrade_server.com/loads/88x110MP2123.tar</a>
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.	<a href="http://10.1.1.1/admin/updateca/http://my_ca_server.com/ash/CompanyCA.pem">http://10.1.1.1/admin/updateca/http://my_ca_server.com/ash/CompanyCA.pem</a>
reboot	Initiates a reboot of the phone. Does not take any parameter with ?	<a href="http://10.1.1.1/admin/reboot">http://10.1.1.1/admin/reboot</a>
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 <b>Voice</b> tab.	<a href="http://10.1.1.1/admin/cfg.xml">http://10.1.1.1/admin/cfg.xml</a>
status.xml	Downloads a snapshot of the phone status in XML format. Most of the information here corresponds to the <b>Status</b> tab in the phone web page.	<a href="http://10.1.1.1/admin/status.xml">http://10.1.1.1/admin/status.xml</a>
screendump.bmp	Downloads a screenshot of the phone LCD UI at the time when this action is initiated.	<a href="http://10.1.1.1/admin/screendump.bmp">http://10.1.1.1/admin/screendump.bmp</a>
log.tar	Downloads a set of archived logs stored on the phone.	<a href="http://10.1.1.1/admin/log.tar">http://10.1.1.1/admin/log.tar</a>

## Enable Access to Phone Web Interface

### Procedure

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- Step 1** Click **Admin Login** > **advanced** > **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** 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** 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 are 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

There are more URIs supported which are available in the *Cisco Unified IP Phone Services Application Development Notes*.

For more information, see the *Cisco Unified IP Phone Services Application Development Notes* located here: [http://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cuipph/all\\_models/xsi/9-1-1/CUIP\\_BK\\_P82B3B16\\_00\\_phones-services-application-development-notes.html](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cuipph/all_models/xsi/9-1-1/CUIP_BK_P82B3B16_00_phones-services-application-development-notes.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.com/path?username=\\$XML\\_User\\_Name](http://www.sipurash.com/path?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 lower case hex digits—MA
- Product Name—PN
- Product Series Numbe—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. <b>Note</b> \$SA through \$SD are recognized as arguments to the optional resync URL qualifier, --key.
MA	MAC address using lower case hex digits (000e08aabbcc).
MAU	MAC address using upper case hex digits (000E08AABBCC).
MAC	MAC address using lower case hex digits with colon to separate hex digit pairs (00:0e:08:aa:bb:cc).
PN	Product Name; for example IP Phone 8861.
PSN	Product Series Number; for example 8861.
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).
HWVER	Hardware version string; for example 1.88.1.

Macro Name	Macro Expansion
PRVST	Provisioning State (a numeric string): <ul style="list-style-type: none"> <li>• -1 = explicit resync request</li> <li>• 0 = power-up resync</li> <li>• 1 = periodic resync</li> <li>• 2 = resync failed, retry attempt</li> </ul>
UPGST	Upgrade State (a numeric string): <ul style="list-style-type: none"> <li>• 1 = first upgrade attempt</li> <li>• 2 = upgrade failed, retry attempt</li> </ul>
UPGERR	Result message (ERR) of previous upgrade attempt; for example http_get failed.
PRVTMR	Seconds since last resync attempt.
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 host name.
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. <b>Note</b> Customization status viewable on Web UI Info page.
INCOMINGNAME	Name associated with first connected, ringing, or inbound call.

Macro Name	Macro Expansion
REMOTENUMBER	Phone number of first connected, ringing, or inbound call. If there are multiple calls, the data associated with the first call found will be provided.
DISPLAYNAME <sub>n</sub>	The contents of the Line N Display Name configuration parameter.
AUTHID <sub>n</sub>	The contents of the Line N auth ID configuration parameter.

## Configure a Phone to Connect to an XML Application

### Procedure

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**Step 1** In the Configuration Utility, select **Admin Login > advanced > 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**.

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## Configure a Phone to Connect to an XML Directory Service

### Procedure

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**Step 1** In the Configuration Utility, select **Admin Login > advanced > 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**.

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