



Phone Features and Setup

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Phone Features and Setup Overview

After you install Cisco IP Phones in your network, configure their network settings, and add them to Third-Party Call Control System, you must use the Third-Party Call Control System to configure telephony features, optionally modify phone templates, set up services, and assign users.

You can modify additional settings for the Cisco IP Phone from Third-Party Call Control Configuration Utility. Use this web-based application to set up phone registration criteria and calling search spaces, to configure corporate directories and services, and to modify phone button templates, among other tasks.

Cisco IP Phone User Support

If you are a system administrator, you are likely the primary source of information for Cisco IP Phone users in your network or company. It is important to provide current and thorough information to end users.

To successfully use some of the features on the Cisco IP Phone (including Services and voice message system options), users must receive information from you or from your network team or must be able to contact you for assistance. Make sure to provide users with the names of people to contact for assistance and with instructions for contacting those people.

We recommend that you create a web page on your internal support site that provides end users with important information about their Cisco IP Phones.

Consider including the following types of information on this site:

- User guides for all Cisco IP Phone models that you support
- Information on how to access the Cisco Unified Communications Self Care Portal
- List of features supported

- User guide or quick reference for your voicemail system

Telephony Features for Cisco IP Phone

After you add Cisco IP Phones to Third-Party Call Control system, you can add functionality to the phones. The following table includes a list of supported telephony features, many of which you can configure by using Third-Party Call Control system.



Note The Third-Party Call Control system also provides several service parameters that you can use to configure various telephony functions.

Feature	Description and More Information
AES 256 Encryption Support for Phones	Enhances security by supporting TLS 1.2 and new ciphers.
Alphanumeric Dialing	Allows users to place a call with alphanumeric characters. You can use these characters for alphanumeric dialing: a-z, A-Z, 0-9, -, _, ., and +.
Any Call Pickup	Allows users to pick up a call on any line in their call pickup group, regardless of how the call was routed to the phone.
Audio Settings	Configures audio settings for the phone speaker, the handset, and the headsets that are connected to the phone.
Auto Answer	Connects incoming calls automatically after a ring or two. Auto Answer works with either the speakerphone or the headset.
Blind Transfer	Blind Transfer: This transfer joins two established calls (call is in hold or in connected state) into one call and drops the feature initiator from the call. Blind Transfer does not initiate a consultation call and does not put the active call on hold. Some JTAPI/TAPI applications are not compatible with the Join and Blind Transfer feature implementation on the Cisco IP Phone and you may need to configure the Join and Direct Transfer Policy to disable join and direct transfer on the same line or possibly across lines.
Busy Lamp Field (BLF)	Allows user to monitor call state of a directory number.
Busy Lamp Field (BLF) Pickup	Allows user to pick up incoming calls to the directory number monitored through BLF.
Call Back	Provides users with an audio and visual alert on the phone when a busy or unavailable party becomes available.
Call Display Restrictions	Determines the information that will display for calling or connected lines, depending on the parties who are involved in the call. RPID and PAID caller id handling are supported.
Call Forward	Allows users to redirect incoming calls to another number. Call Forward options include Call Forward All, Call Forward Busy, Call Forward No Answer.

Feature	Description and More Information
Call Forward Notification	Allows you to configure the information that the user sees when receiving a forwarded call.
Call History for Shared Line	Allows you to view shared line activity in the phone Call History. This feature: <ul style="list-style-type: none"> • Logs missed calls for a shared line. • Logs all answered and placed calls for a shared line.
Call Park	Allows users to park (temporarily store) a call and then retrieve the call by using another phone.
Call Pickup	Allows users to redirect a call that is ringing on another phone within their pickup group to their phone. You can configure an audio and visual alert for the primary line on the phone. This alert notifies the users that a call is ringing in their pickup group.
Call Waiting	Indicates (and allows users to answer) an incoming call that rings while on another call. Incoming call information appears on the phone display.
Caller ID	Caller identification such as a phone number, name, or other descriptive text appear on the phone display.
Caller ID Blocking	Allows a user to block their phone number or name from phones that have caller identification enabled.
Calling Party Normalization	Calling party normalization presents phone calls to the user with a dialable phone number. Any escape codes are added to the number so that the user can easily connect to the caller again. The dialable number is saved in the call history and can be saved in the Personal Address Book.
Conference	Allows a user to talk simultaneously with multiple parties by calling each participant individually. Allows a noninitiator in a standard (ad hoc) conference to add or remove participants; also allows any conference participant to join together two standard conferences on the same line. Note Be sure to inform your users whether these features are activated.
Configurable RTP/sRTP Port Range	Provides a configurable port range (2048 to 65535) for Real-Time Transport Protocol (RTP) and secure Real-Time Transport Protocol (sRTP). The default RTP and sRTP port range is 16384 to 16538. You configure the RTP and sRTP port range in the SIP Profile.
Directed Call Pickup	Allows a user to pick up a ringing call on a DN directly by pressing the GPickUp softkey and entering the directory number of the device that is ringing.
Divert	Allows a user to transfer a ringing, connected, or held call directly to a voice-messaging system. When a call is diverted, the line becomes available to make or receive new calls.

Feature	Description and More Information
Do Not Disturb (DND)	When DND is turned on, either no audible rings occur during the ringing-in state of a call, or no audible or visual notifications of any type occur.
DND and Call Forward Indication on Non-selected Line Key	Displays the DND and call forward icons next the to the line key label. The line key should be enabled with feature key sync. The line key should also be enabled with DND or call forward.
Emergency Calls	Enables users to make emergency calls. The emergency services receive the phone's location and a call-back number, to use when the emergency call unexpectedly disconnects.
Executive-Assistant	Indicates shared call control for executives and their assistants.
Headset Sidetone Control	Allows an administrator to set the sidetone level of a wired headset.
Group Call Pickup	Allows a user to answer a call that is ringing on a directory number in another group.
Hold Status	Enables phones with a shared line to distinguish between the local and remote lines that placed a call on hold.
Hold/Resume	<p>Allows the user to move a connected call from an active state to a held state.</p> <ul style="list-style-type: none"> • No configurations are required unless you want to use Music On Hold. See “Music On Hold” in this table. • See “Hold Reversion” in this table.
HTTP Download	Enhances the file download process to the phone to use HTTP by default. If the HTTP download fails, the phone reverts to using the TFTP download.
HTTPS for Phone Services	<p>Increases security by requiring communication using HTTPS.</p> <p>Note When the web is in HTTPS mode, the phone is an HTTPS server.</p>
Improve Caller Name and Number Display	Improves the display of caller names and numbers. If the Caller Name is known, then the Caller Number is displayed instead of <code>Unknown</code> .
Jitter Buffer	The Jitter Buffer feature handles jitter from 10 milliseconds (ms) to 1000 ms for both audio and video streams.
Join Across Lines	<p>Allows users to combine calls that are on multiple phone lines to create a conference call.</p> <p>Some JTAPI/TAPI applications are not compatible with the Join and Direct Transfer feature implementation on the Cisco IP Phone and you may need to configure the Join and Direct Transfer Policy to disable join and direct transfer on the same line or possibly across lines.</p>
Join	Allows users to combine two calls that are on one line to create a conference call and remain on the call.
Message Waiting	Defines directory numbers for message waiting on and off indicators. A directly-connected voice-message system uses the specified directory number to set or to clear a message waiting indication for a particular Cisco IP Phone.

Feature	Description and More Information
Message Waiting Indicator	A light on the handset that indicates that a user has one or more new voice messages.
Minimum Ring Volume	Sets a minimum ringer volume level for an IP phone.
Missed Call Logging	Allows a user to specify whether missed calls will be logged in the missed calls directory for a given line appearance.
Multicasting Paging	Enables users to page some or all phones. If the phone is on an active call while a group page starts, the incoming page is ignored.
Multiple Calls Per Line Appearance	<p>Each line can support multiple calls. By default, the phone supports two active calls per line, and a maximum of ten active calls per line. Only one call can be connected at any time; other calls are automatically placed on hold.</p> <p>The system allows you to configure maximum calls/busy trigger not more than 10/6. Any configuration more than 10/6 is not officially supported.</p>
Music On Hold	Plays music while callers are on hold.
Mute	Mutes the handset or headset microphone.
No Alert Name	Makes it easier for end users to identify transferred calls by displaying the original caller's phone number. The call appears as an Alert Call followed by the caller's telephone number.
Pause in Speed Dial	Users can set up the speed-dial feature to reach destinations that require Forced Authorization Code (FAC) or Client Matter Code (CMC), dialing pauses, and additional digits (such as a user extension, a meeting access code, or a voicemail password) without manual intervention. When the user presses the speed dial, the phone establishes the call to the specified DN and sends the specified FAC, CMC, and DTMF digits to the destination and inserts the necessary dialing pauses.
Peer Firmware Sharing (PFS)	<p>Allows IP Phones located at remote sites to share the firmware files amongst them, which saves bandwidth when the upgrade process takes place. This feature uses Cisco Peer-to-Peer-Distribution Protocol (CPPDP) which is a Cisco proprietary protocol used to form a peer-to-peer hierarchy of devices. CPPDP is also used to copy firmware or other files from peer devices to the neighbouring devices.</p> <p>PFS aids in firmware upgrades in branch/remote office deployment scenarios that run over bandwidth-limited WAN links.</p> <p>Provides the following advantages over the traditional upgrade method:</p> <ul style="list-style-type: none"> • Limits congestion on TFTP transfers to centralized remote TFTP servers • Eliminates the need to manually control firmware upgrades • Reduces phone downtime during upgrades when large numbers of devices are reset simultaneously <p>The more the number of IP phones, the better it's performance compared to the traditional firmware upgrade method.</p>

Feature	Description and More Information
Plus Dialing	<p>Allows the user to dial E.164 numbers prefixed with a plus (+) sign.</p> <p>To dial the + sign, the user needs to press and hold the star (*) key for at least 1 second. This applies to dialing the first digit for an on-hook (including edit mode) or off-hook call.</p>
Power Negotiation over LLDP	Allows the phone to negotiate power using Link Level Endpoint Discovery Protocol (LLDP) and Cisco Discovery Protocol (CDP).
Problem Reporting Tool	Submits phone logs or reports problems to an administrator.
Programmable Feature Buttons	You can assign features, such as New Call, Call Back, and Forward All to line buttons.
Redial	Allows users to call the most recently dialed phone number by pressing a button or the Redial softkey.
Remote Customization (RC)	Allows a service provider to customize the phone remotely. There is no need for either the service provider to physically touch the phone or a user to configure the phone. The service provider can work with a sales engineer at the time of ordering to set this up.
Ringtone Setting	Identifies ring type used for a line when a phone has another active call.
Reverse Name Lookup	Identifies the caller name using the incoming or outgoing call number. You must configure either the LDAP Directory or the XML directory. You can enable or disable the reverse name lookup using the phone administration web page.
RTCP Hold For SIP	Ensures that held calls are not dropped by the gateway. The gateway checks the status of the RTCP port to determine if a call is active or not. By keeping the phone port open, the gateway will not end held calls.
Serviceability for SIP Endpoints	<p>Enables administrators to quickly and easily gather debug information from phones.</p> <p>This feature uses SSH to remotely access each IP phone. SSH must be enabled on each phone for this feature to function.</p>
Shared Line	Allows a user with multiple phones to share the same phone number or allows a user to share a phone number with a coworker.
Show Calling ID and Calling Number	<p>The phones can display both the calling ID and calling number for incoming calls. The IP phone LCD display size limits the length of the calling ID and the calling number that display.</p> <p>The Show Calling ID and Calling Number feature applies to the incoming call alert only and does not change the function of the Call Forward and Hunt Group features.</p> <p>See “Caller ID” in this table.</p>

Feature	Description and More Information
Show Duration for Call History	Displays the time duration of placed and received calls in the Call History details. If the duration is greater than or equal to one hour, the time is displayed in the Hour, Minute, Second (HH:MM:SS) format. If the duration is less than one hour, the time is displayed in the Minute, Second (MM:SS) format. If the duration is less than one minute, the time is displayed in the Second (SS) format.
Silence Incoming Call	Allows you to silence an incoming call by pressing Ignore softkey or by pressing the volume button down.
SIP Transport Auto-Selection	Configures the phone to select the appropriate SIP transport protocol automatically, based on the NAPTR records on the DNS server. See Configure the SIP Transport, on page 43 .
Speed Dial	Dials a specified number that has been previously stored.
Time Zone Update	Updates the Cisco IP Phone with time zone changes.
Transfer	Allows users to redirect connected calls from their phones to another number. Some JTAPI/TAPI applications are not compatible with the Join and Direct Transfer feature implementation on the Cisco IP Phone and you may need to configure the Join and Direct Transfer Policy to disable join and direct transfer on the same line or possibly across lines.
Voice/Video data priorities	Enables you to prioritise voice or video data in limited bandwidth conditions, by specifying different ToS field values for voice and video packets.
Voice Message System	Enables callers to leave messages if calls are unanswered.
Web Access Enable by Default	Web services are enabled by default.
XSI call logs display	Allows you to configure a phone to display recent call logs from either the BroadWorks server or the local phone. After you enable the feature, the Recents screen has a Display recents from menu and the user can choose the XSI call logs or the local call logs.

Feature Buttons and Softkeys

The following table provides information about features that are available on softkeys, features that are available on dedicated feature buttons, and features that you need to configure as programmable feature buttons. A “Supported” entry in the table indicates that the feature is supported for the corresponding button type or softkey. Of the two button types and softkeys, only programmable feature buttons require configuration in Cisco IP Phone administration.

Table 1: Features with Corresponding Buttons and Softkeys

Feature Name	Dedicated Feature Button	Programmable Feature Button	Softkey
Answer		Supported	Supported
Call Back		Supported	Supported
Call Forward All		Supported	Supported
Call Park		Supported	Supported
Call Park Line Status		Supported	
Call Pickup (Pick Up)		Supported	Supported
Call Pickup Line Status		Supported	
Conference	Supported		Supported (only displayed during connected call conference scenario)
Divert			Supported
Do Not Disturb		Supported	Supported
Executive - Join ongoing call			Supported
Executive - Call filtering activation and deactivation			Supported
Executive - Call transfer to self			Supported
Executive - Access to Settings > Assistant menu		Supported	
Executive Assistant - Call diversion activation and deactivation			Supported
Executive Assistant - Call filtering activation and deactivation			Supported
Executive Assistant - Call initiation on behalf of executive			Supported

Feature Name	Dedicated Feature Button	Programmable Feature Button	Softkey
Executive Assistant - Call transfer to executive			Supported (Only displayed when calls for or on behalf of executives are put on hold)
Executive Assistant - Access to Settings > Executive menu		Supported	
Group Pickup (Group Pick Up)		Supported	Supported
Hold	Supported		Supported
Hunt Groups		Supported	Supported
Intercom		Supported	
Malicious Call Identification (MCID)		Supported	Supported
Meet Me		Supported	Supported
Mobile Connect (Mobility)		Supported	Supported
Mute	Supported		
Other Pickup		Supported	Supported
PLK Support for Queue Status		Supported	Supported
Privacy		Supported	
Queue Status		Supported	
Quality Reporting Tool (QRT)		Supported	Supported
Record	Not supported	Not supported	Supported
Redial		Supported	Supported
Speed Dial		Supported	Supported
Speed Dial Line Status		Supported	
Transfer	Supported		Supported (only displayed during connected call transfer scenario)

Enable Users to Configure Features on Line Keys

You can enable users to configure these features on line keys:

- Speed dial
- Busy Lamp Field (BLF) to monitor a coworker's line, with the following options:
 - Speed dial to the monitored line
 - Call pickup from the monitored line

Users can select any available line keys to configure features. They can also select a line key that is functioning as a speed-dial key or as a BLF key. The user's configuration will override any existing configuration for the line key. Users cannot select line keys on which you have configured other features. If a user selects a BLF list key, the phone adjusts the positions of the BLF list keys using the next available line keys.

For the BLF feature options, the phone subscribes to the BLF list URI that you specify (XML parameter `BLF_List_URI`), to be notified of changes in the status of the monitored lines. If you do not specify a BLF list URI, the phone subscribes to `$USER@$PROXY`.

Procedure

- Step 1** On the phone administration web page, go to **Admin Login > Advanced, Voice** tab.
- Step 2** To allow features, go to **Att Console > General**, and configure **Customizable PLK Options** as described in [General](#).

When this step is complete, users can configure features on Key Expansion Module keys.

- Step 3** To enable feature configuration on a line key on the phone, do one of the following:
- Disable the extension function for the line key:
 1. Go to **Voice > Phone**.
 2. Set **Extension** to **Disabled** in the corresponding **Line Key number** section.
 - Disable service on the corresponding line:
 1. Go to **Voice**.
 2. Go to the corresponding **Ext number** tab.
 3. In the **General** section, set **Line Enable** to **No**.
-

Configure a Speed Dial on a Line Key

You can configure speed dial on an idle line of a user phone. The user can then use that line key to speed-dial a number. When you enable the speed dial on the line key, the user sees the speed-dial icon a name for the speed dial line key. The user presses the line key to dial the assigned extension.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

-
- Step 1** Select **Voice > Phone**.
 - Step 2** Select a Line Key on which to configure speed-dial.
 - Step 3** From the Extension pulldown menu, select **Disabled** to disable the extension.
 - Step 4** In the **Extended Function** field, enter a string in this format:

```
fnc=sd;ext=9999@$PROXY;nme=xxxx
```

If you configure a phone with alphanumeric dialing feature in which the phone can place a call with alphanumeric characters instead of the traditional digits, you can enter a string in this format:

```
fnc=sd;ext=xxxx.yyyy@$PROXY;vid=n;nme=xxxx
```

where:

- fnc= sd means function=speed dial
- ext= 9999 is the phone that the line key calls. Replace 9999 with appropriate phone number.
ext= xxxx.yyyy is the phone that the line key calls. Replace xxxx.yyyy with alphanumeric characters. You can use these characters for alphanumeric dialing: a-z, A-Z, 0-9, -, _, ., and +.
- vid=n is the line index of the phone.
- nme= XXXX is the name displayed on the phone for the speed-dial line key. Replace XXXX with a name.

You can also configure XML service with line key. Enter a string in this format:

```
fnc=xml;url=http://xml.service.url;nme=name
```

- Step 5** Click **Submit All Changes**.
-

Configure a Speed Dial with the Configuration Utility Page

You can configure speed dials on the phone with the web interface.

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 **Speed Dial** section, enter a name and number that corresponds to the speed dial entry.
- Step 3** Click **Submit All Changes**.
-

DTMF Wait and Pause Parameters

Speed dial, directory, extended function, and other strings configured in the phone can include *wait* (X) and *pause* (,) characters. These characters allow manual and automatic DTMF (Dual-Tone Multi-Frequency) signal transmission.

You can add the wait and pause character with speed-dial, extended function, or directory strings in the format:

```
{Dial_String}[ ][,|X][DTMF_string][,|X][DTMF_string]
```

where:

- Dial_String—is the number that the user is trying to reach. For example, 8537777 or 14088537777.
- [](space)—is a dial termination character that defines or delimits the end of the dial string. The space is mandatory. If the phone encounters an X or a comma (,) before the space, the characters are treated as part of dial string.
- , (comma)—is a 2-second pause that is inserted for each comma in the string.
- X (wait)—indicates that the phone is waits for user input and acknowledgement.

When the user manually enters the DTMF signal with the key pad, the user sees a message to acknowledge that the transmission of the manual entry is complete. On confirmation, the phone sends any DTMF signals defined by the *DTMF_string*. The phone executes the next parameter. If there are no more parameters in the dial string to execute, the phone exits to the main screen.

The wait prompt window does not disappear until the user confirms the wait prompt or the call is ended either by the user or ended by the remote device.

- DTMF_string—is the DTMF signals that a user sends to a remote device after the call is connected. The phone cannot send signals other than valid DTMF signals.

Example:

```
18887225555,,5552X2222
```

A speed dial entry triggers the phone to dial 18887225555. The space indicates the end of the dial string. The phone waits 4 seconds (2 commas), and then sends the DTMF signals 5552.

A message is displayed, prompting the user to manually enter digits. When the user finishes dialing the digits, the user presses **OK** to confirm the manual input is complete. The phone sends the DTMF signals 2222.

Usage Guidelines

A user can transmit digits any time, as long as the call is connected.

The maximum length of the string, including the Xs or commas (,), is limited to the length of a speed-dial entry, dial screen entry, directory entry, and other dialed strings.

When a wait is initiated, the phone displays the home screen and prompts the user to input more digits with the key pad. If this action occurs while the user is editing an entry, the edits might be lost.

If only the first part of a dial string matches a dial plan when the call is dialed, the portion of the dial string that does not match the dial string is ignored. For example:

```
85377776666,,1,23
```

If 8537777 matches a dial plan, the characters 6666 are ignored. The phone waits 4 seconds before sending DTMF 1. It then wait 2 seconds and sends DTMF 23.

When logging the call, the phone only logs the dial string; the DTMF strings are not logged.

Valid DTMF signals are 0-9, *, or #. All other characters are ignored.

Limitations

When the call is connected and immediately transferred, the phone might not be able to process the DTMF signals. This depends on the length of time that the call is connected before it is transferred.

Speed Dial

Parameter	Description
Speed Dial Name	Indicates the name given to the speed dial.
Speed Dial Number	Indicates the number allocated to the speed dial.

Configure a Speed Dial on a Key Expansion Module

You can configure speed dial on a key expansion module line. The user can then press the line key to call a frequently dialed number.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

-
- Step 1** Select **Voice > Att Console**.
 - Step 2** Select a key expansion module line key on which to enable the speed dial.
 - Step 3** Enter a string in this format:

```
fnc=sd;ext=9999@$PROXY;vid=n;nme=xxxx
```

where:

- fnc= sd means function=speed dial

- ext= 9999 is the phone that the line key calls. Replace 9999 with numbers.
- vid=n is the line index of the phone.
- nme= XXXX is the name displayed on the phone for the speed-dial line key. Replace XXXX with a name.

You can also configure an XML service on key expansion module key. Enter the string in this format:

```
fnc=xml;url=http://xml.service.url;nme=name
```

Step 4 Click **Submit All Changes**.

Enable Conference Button with a Star Code

You can add a star code to the Conference button so that your user can press the button only once to add many active calls to a conference. You can enable this feature from the phone web page.

Before you begin

- The phone server must support this feature.
- Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

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

Step 2 In the **Call Features Settings** section, select **Yes** for the **Conference Single Hardkey** field, enter a star code in the **Conference Bridge URL**, and press **Submit All Changes**. For example, you can enter *55 to represent the conference bridge URL of a telecom service provider.

You can also enable the conference button with a xml file. Enter a string in this format:

```
<Conference_Bridge_URL_1_ ua="na">*55</Conference_Bridge_URL_1_>
<Conference_Single_Hardkey_1_ ua="na">Yes</Conference_Single_Hardkey_1_>
```

Enable Dial Assistance

You can configure dial assistance so that your users can place calls more quickly. As a user dials, the phone displays a list of closely-matched phone numbers on the screen.

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 **Supplementary Services** section, set **Dial Assistance** field to **Yes**.
 - Step 3** Click **Submit All Changes**.
-

Set up Extra Line Keys

Enable this feature if you want to use the buttons on both sides of the phone screen as line keys.

Procedure

-
- Step 1** On the Configuration Utility page, click **Admin Login > Voice > Phone**.
 - Step 2** Choose a line key and select an extension to enable it.
 - Step 3** Click **Submit All Changes**.
-

Phone Configuration for Monitoring Other Phones

You can configure the phone to monitor the status of lines on other phones. This feature is useful if users routinely handle calls for colleagues and need to see if they are available to answer calls. The phone monitors each line on a separate line key. The monitoring line keys function as Busy Lamp Field (BLF) keys. A BLF is an LED that changes color to indicate the status of the monitored line:

Table 2: BLF Key LED Status

LED Color	Meaning
Green	The monitored line is available.
Red	The monitored line is busy.
Blinking red	The monitored line is ringing.
Amber	Error in BLF key configuration.

If the phone is registered to a BroadSoft server, you can set up the phone to monitor multiple users, with a single set of configurations.

Configure the Phone to Monitor Multiple Users' Lines

If the phone is registered to a BroadSoft server, you can configure the phone to monitor the entire BLF list. The phone assigns available line keys in sequence to monitor the BLF list entries, and starts showing the status of the monitored lines on the BLF keys.

Before you begin

- Make sure that the phone is registered to a BroadSoft server.
- You set up a BLF list for a user of the phone on the BroadSoft server.

Procedure

-
- Step 1** On the phone administration web page, go to **Admin Login > Advanced, Voice > Att Console > General**.
- Step 2** Configure **BLF List URI**, **BLF List**, and **Use Line Keys For BLF List** as described in [General](#).
If you allow users to configure individual BLF keys (see [Enable Users to Configure Features on Line Keys, on page 11](#)), we recommend setting **BLF List** to **Hide**.
- Step 3** Click **Submit All Changes**.
-

Configure the Busy Lamp Field in the Phone Configuration File

If the phone is registered to a BroadSoft server, you can use the phone configuration file to configure the busy lamp field.

Procedure

-
- Step 1** Edit the BLF_List_URI parameter of the phone configuration file that is available in the BroadSoft server.
- Step 2** Add the List URI: sip: parameter @ domain name.
The List URI must match the one defined in the BroadSoft server.
- Step 3** Save the changes.
-

Configure a Line Key on the Phone to Monitor a Single User's Line

You can configure busy lamp field on a phone line when a user needs to monitor a coworker's availability to handle calls.

You can configure the busy lamp field to work with any combination of speed dial or call pickup. For example, busy lamp field alone, busy lamp field and speed dial, busy lamp field and call pickup, or busy lamp field, speed dial, and call pickup can all be configured to work together. But speed dial alone requires a different configuration.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

- Step 1** Select **Voice > Phone**.
- Step 2** Select a line key on which to configure a busy lamp field.
- Step 3** Select **Disabled** to disable the extension.
- Step 4** In the **Extended Function** field, enter a string in this format:

```
fnc=blf ; sub=xxxx@$PROXY ; usr=yyyy@$PROXY
```

```
fnc=blf ; sub=xxxx@$PROXY ; ext=yyyy@$PROXY
```

Where:

- fnc=blf means function=busy lamp field
- sub=the URI to which the SUBSCRIBE message should be sent. For a BroadSoft server, this name must be identical to the name defined in the **List URI: sip:** parameter. xxxx is the name that is defined in **List URI: sip:** parameter. Replace xxxx with the exact defined name. \$PROXY is the server. Replace \$PROXY with the server address or name.
- usr/ext=the user that the busy lamp field monitors. yyyy is user id of the phone that the busy lamp field monitors. Replace yyyy with the exact user id of the monitored phone. \$PROXY is the server. Replace \$PROXY with the server address or name.

- Step 5** (Optional) You can configure the busy lamp field to work with any combination of speed dial or call pickup. To enable the busy lamp field to work with speed dial or call pickup, enter a string in the following format in the Extended Function field:

```
fnc=blf+sd+cp ; sub=xxxx@$PROXY ; usr=yyyy@$PROXY.
```

Where:

sd= speed dial

cp= call pickup

- Step 6** Click **Submit All Changes**.
-

Configure Busy Lamp Field with Other Features

You can configure busy lamp field to work with other features on your key expansion module, such as speed dial, and call pickup. Use the information in the following table as a guide when selecting the correct string format.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

- Step 1** Select **Voice > Att Console**.
- Step 2** Select a key expansion module line key.
- Step 3** Enter a string in the appropriate format.

Feature	String Format
Busy Lamp Field and Speed Dial	<code>fnc=blf+sd;sub=xxx@proxy;ext=monitored userID@proxy.</code>
Busy Lamp Field, Speed Dial, and Call Pickup	<code>fnc=blf+sd+cp;sub=xxx@proxy;ext=monitored userID@proxy.</code>
Busy Lamp Field, Speed Dial, and Park Notification	<code>fnc=blf+sd;sub=xxx@proxy;ext=monitored userID@proxy.</code> This combination cannot be configured using the extended function. This combination is supported on Broadsoft servers only and it is configured using the BLF List and related configuration on the server.
Busy Lamp Field, Speed Dial, Park Notification, and Call Pickup	<code>fnc=blf+sd+cp;sub=xxx@proxy;ext=monitored userID@proxy.</code> This combination cannot be configured using the extended function. This combination is supported on Broadsoft servers only and it is configured using the BLF List and related configuration on the server.
Busy Lamp Field and Park Notification	<code>fnc=blf;sub=xxx@proxy;ext=monitored userID@proxy.</code> This combination cannot be configured using the extended function. This combination is supported on Broadsoft servers only and it is configured using the BLF List and related configuration on the server.
Busy Lamp Field, Park Notification, and Call Pickup	<code>fnc=blf+cp;sub=xxx@proxy;ext=monitored userID@proxy.</code> This combination cannot be configured using the extended function. This combination is supported on Broadsoft servers only and it is configured using the BLF List and related configuration on the server.
Busy Lamp Field and Call Pickup	<code>fnc=blf+cp;sub=xxx@proxy;ext=monitored userID@proxy</code>

- Step 4** Click **Submit All Changes**.

Configure the Busy Lamp Field Display Label

You can configure the busy lamp field on a key expansion module or on a device to display the phone user's name, extension, or both.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

-
- Step 1** Select **Voice > Att Console**.
- Step 2** Set **BLF Label Display Mode** to one of the following:
- **Both**: Displays both the user's name and extension.
 - **Name**: Displays the user's name only.
 - **Extension**: Displays the user's extension only.
-

Configure Alphanumeric Dialing

You can configure a phone so that the user of the phone can make a call by dialing alphanumeric characters instead of dialing only digits. In the phone web page, you can configure alphanumeric dialing with speed-dial, blf, and call pickup.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

-
- Step 1** Select **Voice > Ext**.
- Step 2** In the **Enable URI Dialing 1**, select **Yes** to enable alphanumeric dialing.
- In the phone page, you can add a string on a line key in this format to enable speed dial with alphanumeric dialing capability:

```
fnc=sd;ext=xxxx.yyyy@$PROXY;nme=yyyy,xxxx
```

For example:

```
fnc=sd;ext=first.last@$PROXY;nme=Last,First
```

The above example will enable the user to dial "first.dial" to make a call.

Note The supported characters that you can use for alphanumeric dialing are a-z, A-Z, 0-9, -, _, ., and +.

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

Configure a Paging Group (Multicast Paging)

You can configure multicast paging so that users can page all the phones at once or page a group of phones without involving a server. On the Configuration Utility page, you configure a phone as a part of a paging group and can subscribe them to the same multicast address. This enables users to direct pages to specific groups of phones. When you assign each paging group with a unique number, the user dials the paging group number to start paging. All phones that are subscribed to the same multicast address (also configured on the Configuration Utility page) receive the page. The user hears a paging tone of three short beeps when there is an incoming paging call.

Keep these things in mind:

- Your network must support multicasting so that all devices in the same paging group are able to join the corresponding multicast group.
- Paging groups must use even-numbered port numbers.
- If the phone is on an active call when a group page starts, the incoming page is ignored.
- Group paging is one way and uses the G711 codec. The paged phone can only listen to the call from the originator.
- Incoming pages are ignored when DND is enabled.
- When paging occurs, the speaker on the paged phones automatically powers on unless the handset or the headset is in use.
- If the phone is on an active call when a group page starts, the incoming page is ignored. When the call ends, the page is answered, if the page is active.
- When multiple pages occur, the pages are answered in chronological order. Until the active page ends, the next page is not answered.

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 **Multiple Paging Group Parameters** section, enter a string in the **Group Paging Script** field in this format:

```
pggrp=multicast-address:port;[name=xxxx;]num=yyy;[listen={yes|no}];
```

where:

- multicast-address = Multicast IP address of the phone that listens for and receives pages.
- port = Port on which to page; you must use different, even-number ports for each paging group.

Caution Multicast paging doesn't work with odd-numbered ports.

- name (optional) = xxxx is the name of the paging group. Replace xxxx with a name. The name can consist maximum of 64 characters.
- num= yyy is a unique number that the user dials to access the paging group. Replace yyy with a number. The number can consist maximum of 64 characters and the allowed range is 1024 to 32767.
- listen = Indicates whether the phone listens on the page group. Only the first two groups with listen set to **yes** listen to group pages. If the field is not defined, the default value is **no**, so you must set this field to listen to the group pages.

You can add more paging groups by appending to the configuration string. Here is an example of several paging groups.

```
pggrp=224.168.168.168:34560;name=All;num=500;listen=yes;
pggrp=224.168.168.168:34562;name=GroupA;num=501;listen=yes;
pggrp=224.168.168.168:34564;name=GroupB;num=502;
pggrp=224.168.168.168:34566;name=GroupC;num=503;
```

This example creates four paging groups: **All**, **GroupA**, **GroupB**, and **GroupC**. Users dial 500 to send pages to all phones, 501 to send pages to phones configured as part of the **GroupA** group, 502 to send pages to phones configured as part of the **GroupB** group, and 503 to send pages to phones configured as part of the **GroupC** group. The configured phone receives pages directed to the **All** and **GroupA** groups.

Step 3 Click **Submit All Changes**.

Add Priority Paging

You can set paging priority. You no longer need to register the phone to send or receive a page and this feature is known as “Out of Band Paging” feature. You can configure maximum of five paging groups on the phone.

When a paging is initiated during an active call, your user sees an incoming page or outgoing page icons on the phone.

Priority has no impact during a regular page. Only when the phone receives a call during an active page, priority impacts the active call. Following scenarios explain how priority of an active page impacts an active call:

- PG_PRI_EMERGENT(Priority 0): If the phone receives a page with priority 0 during a call, the call will be put on hold. After the paging is complete, the call resumes.
- PG_PRI_IMPORTANT(Priority 1): If the phone receives a page with priority 1 during a call, the call and the page audio is mixed.
- PG_PRI_NORMAL (Priority 2): If the device receives a page with priority 2 during a call, the phone does not display any incoming page icon on the phone screen and the user only hears a notification tone. Once the call ends and if the page is still active, the user sees the paging notification on the phone.
- PG_PRI_MINOR (Priority 3): If the phone receives a page with priority 3 during a call, the page is ignored.

Procedure

Step 1 In the phone web page, select **Admin Login > Advanced > Voice > Phone**.

Step 2 In the **Multipaging Group Parameters** section, enter a string in this format in the **Group Paging Script** field.

```
pggrp=multicast-address:port;[name=xxxx;]num=yyy;[listen={yes|no}];pri=n
```

where:

- multicast-address = Multicast IP address of the phone that listens for and receives pages.
- port = Port on which to page; you must use different ports for each paging group.
- name (optional) = xxxx is the name of the paging group. Replace xxxx with a name. The name can consist maximum of 64 characters.
- num= yyy is a unique number that the user dials to access the paging group. Replace yyy with a number. The number can consist maximum of 64 characters and the allowed range is 1024 to 32767.
- listen = Indicates whether the phone listens on the page group. Only the first two groups with listen set to yes listen to group pages. If the field is not defined, the default value is no, so you must set this field to listen to the group pages.
- pri = n indicates the priority level of the paging. Priority level ranges from 0 to 4.

You can add more paging groups by appending to the configuration string and set the paging priority. Here is an example.

```
pggrp=224.168.168.168:34560;name=All;num=500;listen=yes;pri=0
pggrp=224.168.168.168:34562;name=GroupA;num=501;listen=yes;pri=1
pggrp=224.168.168.168:34564;name=GroupB;num=502;pri=2
pggrp=224.168.168.168:34566;name=GroupC;num=503;pri=3
```

This example creates four paging groups: All, GroupA, GroupB, and GroupC. Users dial 500 to send pages to all phones. If the phone receives a page on the “All” group during a call, the call will be put on hold.

User dials 501 to send pages to phones configured as part of the GroupA group. If the phone receives a page on the “GroupA” group during a call, the audio from page and call will be mixed.

User dials 502 to send pages to phones configured as part of the GroupB group. If the phone configured in GroupA receives a page during an active call, the paging UI will not show up on the device, and a notification tone will be played upon receiving the page. Once the active call ends, and if the page is still active, the paging UI will show up on the device.

User dials 503 to send pages to phones configured as part of the GroupC group. If the phone configured in GroupC receives a page during an active call, the page will be ignored.

Step 3 Click **Submit All Changes**.

Call Park

With call park, a call can be parked and then retrieved either from your phone or from another phone. The following LED colors display on the line key when this feature is configured:

- Green LED—Call park is successfully configured.
- Amber LED—Call park is not configured.
- Red slow blinking LED—A call is parked.

Configure Call Park with Star Codes

You can configure call park so that the user can put a call on hold and then retrieve the call from either the user's phone or another phone.

When configuring call park, the Call Park Code and the Call Unpark Code must match the Feature Access Code configured on the server.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

-
- Step 1** Select **Voice > Regional**.
 - Step 2** Enter ***68** in the **Call Park Code** field.
 - Step 3** Enter ***88** in the **Call Unpark Code** field.
 - Step 4** Click **Submit All Changes**.
-

Add Call Park to a Programmable Line Key

You can add call park to a line key to enable the user to temporarily store and retrieve calls. Call park is supported on private lines and shared lines.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

-
- Step 1** Select **Voice > Phone**.
 - Step 2** Select a line key.
 - Step 3** Select **Disabled** to disable the extension.
 - Step 4** In the **Extended Function** field, enter a string in this format:
 For a private line, enter `fnc=prk;sub=$USER@$PROXY;nme=CallPark-Slot1`.
 For a shared line, enter `fnc=prk;sub=$USER@$PROXY;nme=Call-Park1;orbit=<DN of primary line>`.
 where:

- `fnc=prk` means function=call park
- `sub=999999` is the phone to which the call parks. Replace 999999 with a numbers.
- `nme=XXXX` is the name displayed on the phone for the call park line key. Replace XXXX with a name.

Step 5 Click **Submit All Changes**.

Add Call Park on a Key Expansion Module Line Key

You can add call park to a Key Expansion Module line key to enable the user to temporarily store calls to the same phone to which the Key Expansion Module is connected or to store calls to a different phone. The user can also unpark the call from the line key.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

Step 1 Select **Voice > Att Console**.

Step 2 Select a Key Expansion Module line key on which to enable the call park.

Step 3 Enter a string in this format:

For a private line, enter `fnc=park;sub=$USER@$PROXY;nme=CallPark-Slot1`.

For a shared line, enter `fnc=prk;sub=$USER@$PROXY;nme=Call-Park1;orbit=<DN of primary line>`.

where:

- `fnc=prk` means function=call park
- `sub=999999` is the phone to which the call parks. Replace 999999 with a numbers.
- `nme=XXXX` is the name displayed on the phone for the call park line key. Replace XXXX with a name.

Step 4 Click **Submit All Changes**.

Configure the LCD Brightness for a Key Expansion Module

You can configure the brightness of the LCD display on the key expansion module from the Attendant Console.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

Step 1 Select **Voice > Att Console**.

Step 2 Set the **Attendant Console LCD Contrast** to a value between 1 and 15.

The higher the number, the greater the brightness on the key expansion module screen. If no value is entered, the LCD brightness level is equal to 1, the dimmest value.

Configuring Programmable Softkeys

You can customize the softkeys displayed on the phone. The default softkeys (when the phone is in an idle state) are Redial, Directory, Call Forward, and Do Not Disturb. Other softkeys are available during specific call states (for example, if a call is on hold, the Resume softkey displays).

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

Step 1 Select **Voice > Phone**.

Step 2 Under **Programmable Softkeys**, edit the softkeys depending on the call state that you want the softkey to display. For more information, see [Programmable Softkeys, on page 28](#).

In the Programmable Softkeys section, each phone state is displayed and the softkeys that are available to display during that state are listed. Each softkey is separated by a semicolon. Softkeys are shown in the format:

```
softkeyname [[ position ]
```

where softkeyname is the name of the key and position is where the key is displayed on the IP phone screen. Positions are numbered, with position one displayed on the lower left of the IP phone screen, followed by positions two through four. Additional positions (over four) are accessed by pressing the right arrow key on the phone. If no position is given for a softkey, the key will float and appears in the first available empty position on the IP phone screen.

Step 3 Click **Submit All Changes**.

Customize a Programmable Softkey

The phone provides sixteen programmable softkeys (fields PSK1 through PSK16). You can define the fields by a speed-dial script.

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 **Programmable Softkeys** section, set the **Programmable Softkey Enable** to **Yes**.
- Step 3** Select a programmable softkey number field on which to configure a phone feature.
- Step 4** Enter the string for the programmable soft key. See the different types of programmable softkeys described in [Configure Speed Dial on a Programmable Softkey, on page 27](#).
- Step 5** Click **Submit All Changes**.
-

Configure Speed Dial on a Programmable Softkey

You can configure programmable softkeys as speed dials. The speed dials can be extensions or phone numbers. You can also configure programmable softkeys with speed dials that perform an action that a vertical service activation code (or a star [*] code) defines. For example, if you configure a programmable softkey with a speed dial for *67, the call is placed on hold.

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 **Programmable Softkeys** section, set the **Programmable Softkey Enable** to **Yes**.
- Step 3** To configure a speed dial PSK, enter the following in the **PSK number** field:

```
fnc=sd;ext=extensionname/starcode@$PROXY;vid=n;nme=name
```

Where:

- fnc= function of the key (speed dial)
- extensionname=extension being dialed or the star code action to perform
- vid= n is the extension that the speed dial will dial out
- name is the name of the speed dial being configured

Note The **name** field displays on the softkey on the IP phone screen. We recommend a maximum of 10 characters for a phone. If more characters are used, the label might be truncated on the phone screen.

- Step 4** Edit the following:
- **Idle Key List:** Edit the field as described in the following example:

```
redial|1;newcall|2;dnd;psk1
```

If the user incorrectly configures the programmable softkey list features on the phone, the key list on the phone LCD does not update. For example:

- If a user enters `rdeial;newcall;cfwd` (redial has been misspelt), the key list is not updated and the user does not see any change on the LCD.
- If a user enters `redial;newcall;cfwd;delchar`, the user will not see a change on the LCD, as the delchar softkey is not allowed in the **Idle Key List**. Hence, this is an incorrect configuration of the programmable softkey list.

- **PSK1:**

```
fnc=sd;ext=5014@$PROXY;nme=sktest1
```

Note In this example, we are configuring a softkey on a phone as a speed dial number for extension 5014 (sktest1).

You can also configure an XML service on the programmable soft key. Enter the string in this format:

```
fnc=xml:url=http://xml.service.url;nme=name
```

Step 5 Click **Submit All Changes**.

Programmable Softkeys

Keyword	Key Label	Definition	Available Phone Status
acd_login	Agt signin	Logs user in to Automatic Call Distribution (ACD).	Idle
acd_logout	AgtSignOut	Logs user out of ACD.	Idle
answer	Answer	Answers an incoming call.	Ringing
astate	Agt Status	Checks the ACD status.	Idle
avail	Avail	Denotes that a user who is logged in to an ACD server has set his status as available.	Idle
barge	Barge	Allows another user to interrupt a shared call.	Shared-Active, Shared-Held
baresilent	BargeSilent	Allows another user to interrupt a shared call with the mic disabled.	Shared-Active
bxfer	BlindXfer	Performs a blind call transfer (transfers a call without speaking to the party to whom the call is transferred). Requires that Blind Xfer Serv is enabled.	Connected Connected Video
call (or dial)	Call	Calls the selected item in a list.	Dialing Input
call info	Call Info	Show call information	Progressing

Keyword	Key Label	Definition	Available Phone Status
calllist	Call list	Provides access to the call list while on a connected video call.	Connected, Connected Video
cancel	Cancel	Cancels a call (for example, when conferencing a call and the second party is not answering).	Off-Hook
cfwd	Forward / Clr fwd	Forwards all calls to a specified number.	Idle, Off-Hook, Shared-Active, Hold, Shared-Held
crdpause	PauseRec	Pause recording	Connected, Conferencing
crdresume	ResumeRec	Resume recording	Connected, Conferencing
crdstart	Record	Start a recording	Connected, Conferencing
crdstop	StopRec	Stop recording	Connected, Conferencing
conf	Conference	Initiates a conference call. Requires that Conf Server is enabled and there are two or more calls that are active or on hold.	Connected Connected Video
confLx	Conf line	Conferences active lines on the phone. Requires that Conf Serv is enabled and there are two or more calls that are active or on hold.	Connected Connected Video
delchar	delChar - backspace Icon	Deletes a character when entering text.	Dialing Input
dir	Dir	Provides access to phone directories.	Idle, Miss, Off-Hook (no input), Connected, Start-Xfer, Start-Conf, Conferencing, Hold, Ringing, Shared-Active, Shared-Held
disp_code	DispCode	Enter Disposition Code	Idle, Connected, Conferencing, Hold
dnd	DND / Clr Dnd	Sets Do Not Disturb to prevent calls from ringing the phone.	Idle, Off-Hook, Hold, Shared-Active, Shared-Held, Conferencing, Start-Conf, Start-Xfer, Connected video
emergency	Emergency	Enter emergency number	Connected
em_login (or signin)	Sign in	Logs user in to Extension Mobility.	Idle
em_logout (or signout)	Sign out	Logs user out of Extension Mobility.	Idle

Keyword	Key Label	Definition	Available Phone Status
endcall	End call	Ends a call.	Connected, Off-hook, Progressing, Start-Xfer, Start-Conf, Conferencing, Releasing, Hold, and Connected Video
favorites	Favorites	Provides access to "Speed Dials".	Idle, Miss, Off-Hook (no input), Connected, Start-Xfer, Start-Conf, Conferencing, Hold, Ringing, Shared-Active, Shared-Held Connected Video
gpickup	GrPickup	Allows user to answer a call ringing on an extension by discovering the number of the ringing extension.	Idle, Off-Hook
hold	Hold	Put a call on Hold.	Connected, Start-Xfer, Start-Conf, Conferencing, Connected Video
ignore	Decline	Ignores an incoming call.	Ringing
ignoresilent	Ignore	Silences an incoming call	Ringing
join	Join	Connects a conference call. If the conference host is user A and users B & C are participants, when A presses "Join", A will drop off and users B & C will be connected.	Conferencing
lcr	Call Rtn/lcr	Returns the last missed call.	Idle, Missed-Call, Off-Hook (no input)
left	Left arrow icon	Moves the cursor to the left.	Dialing Input
messages	Messages	Provides access to voicemail.	Idle, Miss, Off-Hook (no input), Connected, Start-Xfer, Start-Conf, Conferencing, Hold, Ringing, Shared-Active, Shared-Held Connected Video
miss	Miss	Displays the list of missed calls.	Missed-Call
newcall	New Call	Begins a new call.	Idle, Hold, Shared-Active, Shared-Held
option	Option	Opens a menu of input options.	Off-Hook

Keyword	Key Label	Definition	Available Phone Status
park	Park	Puts a call on hold at a designated "park" number.	Connected Connected Video
phold	PrivHold	Puts a call on hold on an active shared line.	Connected Connected Video
pickup	PickUp	Allows a user to answer a call ringing on another extension by entering the extension number.	Idle, Off-Hook
pip	PIP icon	Allows user to move PIP to one of the four corners of the screen or turn PIP off.	Connected Video
recents	Recents	Displays the All calls list from call history.	Idle, Off-Hook, Shared-Active, Shared-Held
redial	Redial	Displays the redial list.	Idle, Connected, Start-Conf, Start-Xfer, Off-Hook (no input), Hold Connected Video
resume	Resume	Resumes a call that is on hold.	Hold, Shared-Held
right	Right arrow icon	Moves the cursor to the right.	Dialing (input)
settings	Settings	Provides access to "Information and Settings".	All
showvideo	Show video	Provides access to the video session while on a connected video call and the call list is in view	Connected
starcode	Input Star Code/*code	Displays a list of star codes that can be selected.	Off-Hook, Dialing (input)
swap	Swap	Allows user to swap the remote video stream and selfview during an active video call.	Connected Video
trace	Trace	Trigger trace	Idle, Connected, Conferencing, Hold
unavail	Unavail	Denotes that a user who is logged in to an ACD server has set his status as unavailable.	Idle
unpark	Unpark	Resumes a parked call.	Idle, Off-Hook, Connected, Shared-Active Connected Video

Keyword	Key Label	Definition	Available Phone Status
xfer	Transfer	Performs a call transfer. Requires that Attn Xfer Serv is enabled and there is at least one connected call and one idle call.	Connected, Start-Xfer, Start-Conf
xferlx	Xfer line	Transfers an active line on the phone to a called number. Requires that Attn Xfer Serv is enabled and there are two or more calls that are active or on hold.	Connected Connected Video

Programmable Softkeys for Executives and Assistants

Keyword	Key Label	Definition	Available Phone Status
bridgein	Bridge in	Only available to executives who have assistants. Joins the user (executive) to an ongoing call with an assistant.	Idle, Shared-Active
callpush	Call push	Only available to executive assistants. Transfers an ongoing call from the user (assistant) to the executive.	Hold
callretrieve	Retrieve	Only available to executives who have assistants. Transfers an ongoing call from the assistant to the user (executive).	Idle, Shared-Active
divert	Clr divert	Only available to executive assistants. Deactivates call diversion for the user (assistant).	Available only when call diversion is active and you navigate the menu Settings > Executive . It is also available when you press the line key that is configured as Executive .
	Divert	Only available to executive assistants. Activates call diversion for the user (assistant). All incoming calls to the user, for executives that the user handles, are diverted to the specified destination.	Available when you navigate the menu Settings > Executive . It is also available when you press the line key that is configured as Executive .
proxycall	Proxy call	Only available to executive assistants. Initiates a call on behalf of the selected executive.	Available when you navigate the menu Settings > Executive . It is also available when you press the line key that is configured as Executive .

Configure Provisioning Authority

You can set up provisioning authority so that users can access their personalized phone settings from other phones. For example, people who work different shifts or who work at different desks during the week can share an extension, yet have their own personalized settings.

The **Sign in** softkey appears on the phone when you enable provisioning authority on the phone. Users enter their usernames and passwords to access their personal phone settings. Users can also ignore the sign-in and use the phone as a guest. After users sign in, they have access to their personal directory numbers on the phone. When the user signs out, the phone reverts to a basic profile with limited features.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

- Step 1** Select **Voice > Provisioning**.
- Step 2** In the **Configuration Profile** section, set the **Profile Rule** field to the phone configuration file's URL.
- Example:**
http://192.0.2.1:80/dms/CP-MMxx-MPP/MMxxSystem.xml
where,
MM– Cisco IP Phone MM Series with Multiplatform Firmware (68, 78, or 88)
MMxx– Cisco specific phone model (for example, 7841,7861, 8845, 8865 or 7832)
- Step 3** Select **Admin Login > advanced > Voice > Phone**.
- Step 4** Fill in the **EM Enable** and **EM User Domain** fields in the **Extension Mobility** section, based on the information provided in the phone configuration file.
- Step 5** Set the amount of time (in minutes) that the phone session will last for in the **Session Timer(m)** field. The phone signs out when the session times out.
- Step 6** Set the amount of time (in seconds) that the user has to cancel the sign-out in **Countdown Timer(s)**.
- Step 7** Choose input type of the password from the **Preferred Password Input Mode** field.
- For information on Extension Mobility fields, see [Extension Mobility](#).
- Your user can also change the password input type from the phone.
- Step 8** (Optional) If the **Programmable Softkey Enable** field in the **Programmable Softkeys** section is set to **Yes**, add **signin** to **Idle Key List**.
- Example:**
newcall|1;signin|2
- Step 9** Click **Submit All Changes**.
-

Configure Provisioning Authority in the Phone Configuration File

You can enable provisioning authority in the default configuration file for your phones, so that you don't need to set up the feature manually for each phone.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

-
- Step 1** In the phone configuration file, set the following parameters:
- Set the Provisioning Authority profile rules in the **Profile_Rule** parameters.

Example:

```
<Profile_Rule ua="na">("$EMS" eq "mobile" and "$MUID" ne "" and "$MPWD" ne "")?[--uid $MUID$PDOM --pwd $MPWD] http://10.74.121.51:80/dms/CP-8851-3PCC/8851System.xml|http://10.74.121.51:80/dms/CP-8851-3PCC/8851System.xml</Profile_Rule>
```
 - Set the **EM_Enable** parameter to **Yes**.

Example:

```
<EM_Enable ua="na">Yes</EM_Enable>
```
 - Enter the domain for the phone, or the authentication server in the **EM_User_Domain** parameter.

Example:

```
<EM_User_Domain ua="na">@10.74.121.51</EM_User_Domain>
```
- Step 2** Save the configuration file and upload it to your provisioning server.
- Step 3** Select **Voice > Provisioning**.
- Step 4** Enter the filepath to the configuration file in one of the **Profile Rule** fields.
- Example:**
- ```
http://<SERVER IP ADDRESS>:80/dms/td_8861/8861System.xml
```
- Step 5** Click **Submit All Changes**.
- 

## Enable Hoteling on a Phone

Set up the hotel feature on Broadworks and set the phone as a host or a guest.

### Procedure

- 
- Step 1** Select **Voice > Ext [n]** (where [n] is the extension number).
- Step 2** In the **Call Feature Settings** section, set **Enable Broadsoft Hoteling** to **Yes**.
- Step 3** Set the amount of time (in seconds) that the user can be signed in as a guest on the phone in **Hoteling Subscription Expires**.

- Step 4** Click **Submit All Changes**.
- 

## Set the User Password

Users can set their own password on their phones, or you can set a password for them.

### Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

### Procedure

---

- Step 1** Select **Voice > System**.
- Step 2** Set a password in the **User Password** field.
- Step 3** Click **Submit All Changes**.
- 

## Download Problem Reporting Tool Logs

Users submit problem reports to you with the Problem Reporting Tool.

If you are working with Cisco TAC to troubleshoot a problem, they typically require the logs from the Problem Reporting Tool to help resolve the issue.

To issue a problem report, users access the Problem Reporting Tool and provide the date and time that the problem occurred, and a description of the problem. You need to download the problem report from the Configuration Utility page.

### Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

### Procedure

---

- Step 1** Select **Info > Debug Info > Device Logs**.
- Step 2** In the **Problem Reports** area, click the problem report file to download.
- Step 3** Save the file to your local system and open the file to access the problem reporting logs.
- 

## Configure PRT Upload

You must use a server with an upload script to receive the problem reports that the user sends from the phone.

- If the URL specified in the **PRT Upload Rule** field is valid, users get a notification alert on the phone UI saying that they have successfully submitted the problem report.
- If the **PRT Upload Rule** field is empty or has an invalid URL, users get a notification alert on the phone UI saying that the data upload failed.

The phone uses an HTTP/HTTPS POST mechanism, with parameters similar to an HTTP form-based upload. The following parameters are included in the upload (utilizing multipart MIME encoding):

- devicename (example: "SEP001122334455")
- serialno (example: "FCH12345ABC")
- username (The user name is either the **Station Display Name** or the **User ID** of the extension. The **Station Display Name** is first considered. If this field is empty, then the **User ID** is chosen.)
- prt\_file (example: "probrep-20141021-162840.tar.gz")

You can generate PRT automatically at specific intervals and can define the PRT file name.

A sample script is shown below. This script is provided for reference only. Cisco does not provide support for the upload script installed on a customer's server.

```
<?php

// NOTE: you may need to edit your php.ini file to allow larger
// size file uploads to work.
// Modify the setting for upload_max_filesize
// I used: upload_max_filesize = 20M

// Retrieve the name of the uploaded file
$filename = basename($_FILES['prt_file']['name']);

// Get rid of quotes around the device name, serial number and username if they exist
$devicename = $_POST['devicename'];
$devicename = trim($devicename, '"\'');

$serialno = $_POST['serialno'];
$serialno = trim($serialno, '"\'');

$username = $_POST['username'];
$username = trim($username, '"\'');

// where to put the file
$fullfilename = "/var/prtuploads/".$filename;

// If the file upload is unsuccessful, return a 500 error and
// inform the user to try again

if(!move_uploaded_file($_FILES['prt_file']['tmp_name'], $fullfilename)) {
 header("HTTP/1.0 500 Internal Server Error");
 die("Error: You must select a file to upload.");
}

?>
```

### Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

## Procedure

---

- Step 1** Select **Voice > Provisioning**.
- Step 2** In the **Problem Report Tool** section, set the fields as described in the [Problem Report Tool](#).  
You can also configure the parameters in the phone configuration file with XML(cfg.xml) code. Enter the string in this format:
- ```
<PRT_Upload_Rule ua="na">  
http://64.101.234.132:8000//Users/abcd/uploads/prt/test-prt.tar.gz  
</PRT_Upload_Rule>  
<PRT_Upload_Method ua="na">POST</PRT_Upload_Method>  
<PRT_Max_Timer ua="na">20</PRT_Max_Timer>
```
- Step 3** Click **Submit All Changes**.
-

Configure a Phone to Accept Pages Automatically

The Single Paging or Intercom feature enables a user to directly contact another user by phone. If the phone of the person being paged has been configured to accept pages automatically, the phone does not ring. Instead, a direct connection between the two phones is automatically established when paging is initiated.

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 **Supplementary Services** section, choose **Yes** for the **Auto Answer Page** field.
- Step 3** Click **Submit All Changes**.
-

Server-Configured Paging

You can configure a paging group on a server so that users can page a group of phones. For more details, refer to your server documentation.

Manage Phones with TR-069

You can use the protocols and standards defined in Technical Report 069 (TR-069) to manage phones. TR-069 explains the common platform for management of all phones and other customer-premises equipment (CPE) in large-scale deployments. The platform is independent of phone types and manufacturers.

As a bidirectional SOAP/HTTP-based protocol, TR-069 provides the communication between CPEs and Auto Configuration Servers (ACS).

For TR-069 Enhancements, see [TR-069 Parameter Comparison](#).

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

- Step 1** Select **Voice > TR-069**.
 - Step 2** Set up the fields as described in [TR-069](#).
 - Step 3** Click **Submit All Changes**.
-

View TR-069 Status

When you enable TR-069 on a user phone, you can view status of TR-069 parameters on the Configuration page.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

Select **Info > Status > TR-069 Status**.

You can view status of TR-069 parameters in [TR-069](#).

Enable Electronic Hookswitch

The Electronic Hookswitch feature enables users to use headsets that electronically connect a wireless headset to a phone. Typically, the headset requires a base that plugs into the phone and communicates with the headset. Here are the supported headsets:

- Plantronics Savi 740

- Jabra PRO920
- Jabra PRO9400
- Sennheiser DW Pro I

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

- Step 1** Select **Voice > User**.
- Step 2** Set up the fields as described in [Audio Volume](#).
- Step 3** Click **Submit All Changes**.
-

Report All Phone Issues from the Phone Web Page

If you are working with Cisco TAC to troubleshoot a problem, they typically require the logs from the Problem Reporting Tool to help resolve the issue. You can generate PRT logs using the phone web page and upload them to a remote log server.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

- Step 1** Select **Info > Debug Info**.
- Step 2** In the **Problem Reports** section, click **Generate PRT**.
- Step 3** Enter the following information in the **Report Problem** screen:
- Enter the date that you experienced the problem in the **Date** field. The current date appears in this field by default.
 - Enter the time that you experienced the problem in the **Time** field. The current time appears in this field by default.
 - In the **Select Problem** drop-down list box, choose the description of the problem from the available options.
- Step 4** Click **Submit** in the **Report Problem** screen.
- The Submit button is enabled only if you select a value in the **Select Problem** drop-down list box.
- You get a notification alert on the Phone Web page that indicates if the PRT upload was successful or not.
-

Factory Reset the Phone with the Web UI Button

You can factory reset the phone from the phone web page. The reset only happens if the phone is idle. If the phone is not idle, the phone web page shows a message that the phone is busy and that you need to try again.

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

-
- Step 1** Select **Admin Login** > **advanced** > **Info** > **Debug Info**.
 - Step 2** In the **Factory Reset** section, click **Factory Reset**.
 - Step 3** Click **Confirm factory reset**.
-

Set up a Secure Extension

You can configure an extension to only accept secure calls. If the extension is configured to only accept secure calls then any calls the extension makes will be secure.

You can also configure a secured extension with XML services. Enter a string in this format:

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

```
<Secure_Call_Option_1_ ua="na">Optional</Secure_Call_Option_1_>
```

Before you begin

- Make sure that **Secure Call Serv** is enabled (set to **Yes**) in the **Supplementary Services** area on the **Voice** > **Phone** tab.
- Make sure that SIP Transport parameter of the extension is set to TLS.
- Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

-
- Step 1** Select **Voice** > **Ext(n)**.
 - Step 2** In the **Call Feature Settings** section, in the **Secure Call Option** field, choose **Optional** to retain the current secure call option for the phone, or **Required** to reject nonsecure calls from other phones.
 - Step 3** Click **Submit All Changes**.
-

Capture Packets

Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

- Step 1** Select **Info > Debug Info**.
- Step 2** In the **Problem Report Tool** section, click the **Start Packet Capture** button in the **Packet Capture** field.
- Step 3** Choose **All** to capture all packets that the phone receives and select **Host IP Address** to capture packets only when source or destination is the IP address of the phone.
- Step 4** Make phone calls to and from the selected phone.
- Step 5** When you want to stop the packet capture, click **Stop Packet Capture**.
- Step 6** Click **Submit**.
You see a file in the **Capture File** field. This file contains the filtered packets.
-

Emergency Calls

Emergency Call Support Background

Emergency call service providers can register a phone's location for each IP-based phone in a company. The location information server (LIS) transfers the emergency response location (ERL) to the phone. The phone stores its location during registration, after the phone restarts, and when a person signs in to the phone. The location entry can specify the street address, building number, floor, room, and other office location information.

When you place an emergency call, the phone transfers the location to the call server. The call server forwards the call and the location to the emergency call service provider. The emergency call service provider forwards the call and a unique call-back number (ELIN) to the emergency services. The emergency service or public safety answering point (PSAP) receives the phone location. The PSAP also receives a number to call you back, if the call disconnects.

See [Emergency Call Support Terminology, on page 42](#) for the terms used to describe emergency calls from the phone.

You insert the following parameters to obtain the phone's location for any phone extension number:

- **Company Identifier**—A Unique number (UUID) assigned to your company by the NG9-1-1 service provider.
- **Primary Request URL**—The HTTPS address of the primary server used to obtain the phone location.
- **Secondary Request URL**—The HTTPS address of a secondary server (backup) used to obtain the phone location.

- **Emergency Number**—A sequence of digits that identify an emergency call. You can specify multiple emergency numbers, by separating each emergency number with a comma.

Common emergency service numbers include:

- North America—911
- European countries—112
- Hong Kong—999

The phone requests new location information for the following activities:

- You register the phone with the call server.
- A person restarts the phone and the phone was previously registered with the call server.
- A guest signs in to the phone.
- You change the network interface used in the SIP registration. For example, change Wi-Fi to Ethernet.
- You change the IP address of the phone.

If all of the location servers do not send a location response, the phone re-sends the location request every two minutes.

Emergency Call Support Terminology

The following terms describe emergency call support for the Cisco Multiplatform Phones.

- **Emergency Location ID Number (ELIN)**—A number used to represent one or more phone extensions that locate the person who dialed emergency services.
- **Emergency Response Location (ERL)**—A logical location that groups a set of phone extensions.
- **HTTP Enabled Location Delivery (HELD)**—An encrypted protocol that obtains the PIDF-LO location for a phone from a location information server (LIS).
- **Location Information Server (LIS)**—A server that responds to a SIP-based phone HELD request and provides the phone location using a HELD XML response.
- **Emergency Call Service Provider**—The company that responds to a phone HELD request with the phone's location. When you make an emergency call (which carries the phone's location), a call server routes the call to this company. The emergency call service provider adds an ELIN and routes the call to the emergency services (PSAP). If the call is disconnected, the PSAP uses the ELIN to reconnect with the phone used to make the emergency call.
- **Public Safety Answering Point (PSAP)**—Any emergency service (for example, fire, police, or ambulance) joined to the Emergency Services IP Network.
- **Universally Unique Identifier (UUID)**—A 128-bit number used to uniquely identify a company using emergency call support.

Configure a Phone to Make Emergency Calls

Before you begin

- Obtain the E911 Geolocation Configuration URLs and the company identifier for the phone from your emergency call services provider. You can use the same Geolocation URLs and company identifier for multiple phone extensions in the same office area.
- Access the phone administration web page. See [Access the Phone Web Page](#).

Procedure

- Step 1** Click the **Voice > Ext *n***, where *n* is the phone extension number (1-10) of the phone web dialog.
- Step 2** In the **Dial Plan** area, set the **Emergency Number** to the digits that correspond to the customer emergency service numbers.
- To specify multiple emergency numbers, separate each emergency number with a comma.
- Step 3** In the **E911 Geolocation Configuration** area, set the **Company UUID** to the unique customer identifier obtained from your emergency call service provider.
- For example:
- ```
07072db6-2dd5-4aa1-b2ff-6d588822dd46
```
- Step 4** Specify the encrypted **Primary Request URL** to the main georedundant server. This location information server returns the location for this phone.
- For example:
- ```
https://prod.blueearth.com/e911Locate/held/held_request.action
```
- Step 5** Specify the encrypted **Secondary Request URL** for the backup server that can return location information.
- For example:
- ```
https://prod2.blueearth.com/e911Locate/held/held_request.action
```
- Step 6** Click **Submit All Changes**.
- 

## Configure the SIP Transport

For SIP messages, you can either specify the transport protocol of your choice, or, you can let the phone select the appropriate protocol automatically, for each extension.

When you set up automatic selection, the phone determines the transport protocol based on the Name Authority Pointer (NAPTR) records on the DNS server. The phone uses the protocol specified in the record that has the lowest order and preference. When there are multiple records with the same order and preference, the phone looks for a protocol within the records, in the following order of preference: 1. UDP, 2. TCP, and 3. TLS. The phone uses the first protocol that it finds, in that order of preference.

**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 an extension number.
- Step 2** In the **SIP Settings** section, set the **SIP Transport** parameter as described in [SIP Settings](#).
- Step 3** Click **Submit All Changes**.
- 

## Block Non-Proxy SIP Messages to a Phone

You can disable the ability of the phone to receive incoming SIP messages from a non-proxy server. When you enable this feature, the phone only accepts SIP messages from:

- proxy server
- outbound proxy server
- alternative proxy server
- alternative outbound proxy server
- IN-Dialog message from proxy server and non-proxy server. For example: Call Session dialog and Subscribe dialog

**Before you begin**

Access the phone administration web page. See [Access the Phone Web Page](#).

**Procedure**

---

- Step 1** Select **Voice > System**.
- Step 2** In the **System Configuration** section, set the **Block Nonproxy SIP** field as described in the [System Configuration](#).
- Step 3** Click **Submit All Changes**.
- 

## Configure a Privacy Header

A user privacy header in the SIP message sets user privacy needs from the trusted network.

You can set the user privacy header value for each line extension using the phone web page.

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.

### Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

### Procedure

---

- Step 1** Select **Voice > Extension**.
  - Step 2** In the **SIP Settings** section, set the **Privacy Header** field as described in the [SIP Settings](#).
  - Step 3** Click **Submit All Changes**.
- 

## Enable P-Early-Media Support

You can determine whether to include the P-Early-Media header in the SIP message of outgoing calls. The P-Early-Media header contains the status of the early media stream. If the status indicates that the network is blocking the early media stream, the phone plays the local ringback tone. Otherwise, the phone plays the early media while waiting for the call to be connected.

### Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

### Procedure

---

- Step 1** Select **Voice > Ext (n)**.
  - Step 2** In the SIP Settings section, set the **P-Early-Media Support** field as described in [SIP Settings](#).
  - Step 3** Click **Submit All Changes**.
- 

## Peer Firmware Sharing

Peer Firmware Sharing (PFS) is a firmware distribution model which allows a Cisco IP phone to find other phones of the same model or series on the subnet and share updated firmware files when you need to upgrade multiple phones all at the same time. PFS uses Cisco Peer-to-Peer-Distribution Protocol (CPPDP) which is a

Cisco proprietary protocol. With CPPDP, all the devices in the subnet form a peer-to-peer hierarchy, and then copy the firmware or the other files from peer devices to the neighboring devices. To optimize firmware upgrades, a root phone downloads the firmware image from the load server and then transfers the firmware to other phones on the subnet using TCP connections.

Peer firmware sharing:

- Limits congestion on TFTP transfers to centralized remote load servers.
- Eliminates the need to manually control firmware upgrades.
- Reduces phone downtime during upgrades when large numbers of phones are reset simultaneously.



#### Note

- Peer firmware sharing does not function unless multiple phones are set to upgrade at the same time. When a NOTIFY is sent with Event:resync, it initiates a resync on the phone. Example of an xml that can contain the configurations to initiate the upgrade:  

```
"Event:resync;profile="http://10.77.10.141/profile.xml"
```
- When you set the Peer Firmware Sharing Log server to an IP address and port, the PFS specific logs are sent to that server as UDP messages. This setting must be done on each phone. You can then use the log messages when troubleshooting issues related to PFS.

Peer\_Firmware\_Sharing\_Log\_Server specifies UDP Remote syslog server hostname and the port. The port defaults to the default syslog 514.

For example:

```
<Peer_Firmware_Sharing_Log_Server>192.168.5.5</ Peer_Firmware_Sharing_Log_Server>
```

To use this feature, enable PFS on the phones.

## Enable Peer Firmware Sharing

You can enable Peer Firmware Sharing (PFS) when you want a phone to find other phones of the same model or series on the subnet and share updated firmware files. The phones are organized into a hierarchy and one of the phones in that hierarchy acts as a root phone. After the hierarchy formation, the root phone downloads the firmware image from the load server and then transfers the firmware to other phones in the hierarchy.

### Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

### Procedure

- 
- Step 1** Select **Voice > Provisioning**.
  - Step 2** In the **Firmware Upgrade** section, set the **Peer Firmware Sharing** and the **Peer Firmware Sharing Log Server** fields as described in [Firmware Upgrade](#).
  - Step 3** Click **Submit All Changes**.
-

# Enable Profile Account

You can set up the profile account feature from the phone web page. It allows users to collect authentication information. Authentication information is required while the phone tries to resynchronize and download configuration file and gets an HTTP or HTTPS 401 authentication error for the first time. When you enable this feature, the **Profile account setup** screen is displayed automatically on the phone for the following situations:

- When the HTTP or HTTPS 401 authentication error occurs during provisioning for the first time after the phone reboots
- When the Profile Account username and password are empty
- When there are no username and password in the Profile Rule

There are two ways to trigger the **Profile account setup** screen. The screen can be triggered when there is an HTTP or HTTPS 401 authentication error for the first time. If the user misses or ignores the popup screen, it can also be triggered through the phone screen menu.

When you disable the feature, the **Profile account setup** screen isn't displayed on the phone.

Use this information to help you troubleshoot this situation.

Username and password in the **Profile Rule** have higher priority than profile account.

- When you provide a correct URL in the **Profile Rule** field without a username and password, the phone requires HTTP or HTTPS basic authentication or digest to resync the profile. With correct profile account, authentication passes. With incorrect profile account authentication fails.
- When you provide a correct URL in the **Profile Rule** field with correct username and password, the phone requires HTTP or HTTPS basic authentication or digest to resync the profile. Profile account is not used for phone resync. Sign-in is successful.
- When you provide a correct URL in the **Profile Rule** field with incorrect username and password, the phone requires HTTP or HTTPS basic authentication or digest to resync the profile. Profile account is not used for phone resync. Sign-in always fails.
- When you provide incorrect URL in the **Profile Rule** field, sign-in always fails.

## Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

## Procedure

- 
- |               |                                                                                                                                                      |
|---------------|------------------------------------------------------------------------------------------------------------------------------------------------------|
| <b>Step 1</b> | Select <b>Voice &gt; Provisioning</b> .                                                                                                              |
| <b>Step 2</b> | In the <b>Configuration Profile</b> section, set the <b>Profile Account Enable</b> field as described in the <a href="#">Configuration Profile</a> . |
| <b>Step 3</b> | Click <b>Submit All Changes</b> .                                                                                                                    |
-

# Profile Authentication

Profile Authentication allows phone users to resynchronize the provisioning profile onto the phone. Authentication information is required while the phone tries to resynchronize and download configuration file for the first time and gets an HTTP or HTTPS 401 authentication error. When you enable this feature, the **Profile account setup** screen is displayed on the phone for the following situations:

- When the HTTP or HTTPS 401 authentication error occurs during first-time provisioning after the phone reboots
- When the profile account username and password are empty
- When there are no username and password in the Profile Rule

If the **Profile account setup** screen is missed or ignored, the user can also access the setup screen through the phone screen menu, or the **Setup** softkey, which displays only when no line on the phone is registered.

When you disable the feature, the **Profile account setup** screen doesn't display on the phone.

The username and password in the **Profile Rule** field have a higher priority than the profile account.

- When you provide a correct URL in the **Profile Rule** field without a username and password, the phone requires authentication or digest to resynchronize the profile. With the correct profile account, authentication passes. With an incorrect profile account, authentication fails.
- When you provide a correct URL in the **Profile Rule** field with a correct username and password, the phone requires authentication or digest to resynchronize the profile. The profile account is not used for phone resynchronization. Sign-in is successful.
- When you provide a correct URL in the **Profile Rule** field with an incorrect username and password, the phone requires authentication or digest to resynchronize the profile. The profile account isn't used for phone resynchronization. Sign-in always fails.
- When you provide an incorrect URL in the **Profile Rule** field, sign-in always fails.

## Specify the Profile Authentication Type

You can specify the profile authentication type from the phone administration web page.

### Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

### Procedure

---

- Step 1** Select **Voice > Provisioning**.
  - Step 2** In the **Configuration Profile** section, set the **Profile Authentication Type** field as described in the [Configuration Profile](#).
  - Step 3** Click **Submit All Changes**.
-



# Add Ignore Programmable Soft Key to Silence an Incoming Call

You can add the **Ignore** softkey on the phone. User can press this softkey to silence an incoming call when busy and don't want to be disturbed. When the user presses the softkey, the phone stops ringing, but the user gets a visual alert, and, can answer the phone call.

## 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 **Programmable Softkeys** section, set the **Programmable Softkey Enable** to **Yes**.
  - Step 3** Enter the following values in the **Ring Key List** field:  
`answer|1;ignore|2;ignoresilent|3;`
  - Step 4** Click **Submit All Changes**.
- 

# Enable BroadWorks Anywhere

You can configure a phone to allow a call to seamlessly be moved from one desk phone(location) to another mobile phone or desk phone(location).

When you enable this feature, the **Anywhere** menu is added into the phone screen. The user can use this menu to add multiple phones as locations to the extension. When there is an incoming call in that extension, all the added phones will ring and the user can answer the incoming call from any location. The locations list also gets saved to the BroadWorks XSI server.

## Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

## Procedure

---

- Step 1** Select **Voice > Ext(n)**.
  - Step 2** In the **XSI Line Service** section, set the **XSI Host Server**, **XSI Authentication Type**, **Login User ID**, **Login Password**, and **Anywhere Enable** field as described in the [XSI Line Service](#).  
If you select **SIP Credentials** for **XSI Authentication Type**, you need to enter subscriber **Auth ID** and **Password** in the **Subscriber Information** section.
  - Step 3** Click **Submit All Changes**.
-

# Sync the Block Caller ID Feature with the Phone and the BroadWorks XSI Server

You can sync the **Block caller id** status on the phone and the **Line ID Blocking** status on the BroadWorks XSI server. When you enable the synchronization, the changes that the user makes in the **Block caller id** settings also changes the BroadWorks server settings.

## Before you begin

Access the phone administration web page. See [Access the Phone Web Page](#).

## Procedure

---

- Step 1** Select **Voice > Ext(n)**.
  - Step 2** In the **XSI Line Service** section, set the **Block CID Enable** field as described in the [XSI Line Service](#).
  - Step 3** Click **Submit All Changes**.
- 

# Enable Viewing BroadWorks XSI Call Logs on a Line

You can configure a phone to display recent call logs from either the BroadWorks server or the local phone. After you enable the feature, the Recents screen has a **Display recents from** menu and the user can choose the XSI call logs or the local call logs.

## 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 **XSI Phone Service** section, set the **XSI Host Server**, **XSI Authentication Type**, **Login User ID**, **Login Password**, and **Directory Enable** fields as described in [XSI Phone Service](#).  
If you select **SIP Credentials** for **XSI Authentication Type**, you need to enter **SIP Auth ID** and **SIP Password** in this section.
  - Step 3** Set the **CallLog Associated Line** and **Display Recents From** fields as described in [XSI Phone Service](#).  
**Note** The **Display recents from** menu doesn't appear in the **Recents** phone screen when you set the value of the **CallLog Enable** field to **No**.
  - Step 4** Click **Submit All Changes**.
-

# DND and Call Forwarding Status Sync

You can configure the settings on the phone administration web page to enable status synchronization of do not disturb (DND) and call forwarding between the phone and the server.

There are two ways to synchronize the feature status:

- Feature Key Synchronization (FKS)
- XSI Synchronization





**Note** Feature Key Sync must be enabled for all the executive and assistant users.

FKS uses SIP messages to communicate the feature status. XSI Synchronization uses HTTP messages. If both FKS and XSI synchronization are enabled, FKS takes precedent over XSI synchronization. See the table below for how FKS interacts with XSI synchronization.

**Table 3: Interaction Between FKS and XSI Synchronization**

| Feature Key Sync | DND Enabled | CFWD Enabled | DND Sync   | CFWD Sync  |
|------------------|-------------|--------------|------------|------------|
| Yes              | Yes         | Yes          | Yes (SIP)  | Yes (SIP)  |
| Yes              | No          | No           | Yes (SIP)  | Yes (SIP)  |
| Yes              | No          | Yes          | Yes (SIP)  | Yes (SIP)  |
| Yes              | No          | No           | Yes (SIP)  | Yes (SIP)  |
| No               | Yes         | Yes          | Yes (HTTP) | Yes (HTTP) |
| No               | No          | Yes          | No         | Yes (HTTP) |
| No               | Yes         | No           | Yes (HTTP) | No         |
| No               | No          | No           | No         | No         |

If a line key is configured with FKS or XSI synchronization and is also enabled with DND or call forwarding, the respective DND  icon or the call forwarding  icon is displayed next to the line key label. If the line key has a missed call, a voice message, or an urgent voicemail alert, the DND icon or the call forwarding icon is also displayed with the alert notification.

## Related Topics

[Enable Feature Key Sync](#), on page 52

[Enable Call Forwarding Status Sync via XSI Service](#), on page 52

[Enable DND Status Sync via XSI Service](#), on page 53

## Enable Feature Key Sync

When you enable the Feature Key Synchronization (FKS), the settings of call forwarding and do not disturb (DND) on the server are synchronized to the phone. The changes in DND and call forwarding settings made on the phone will also be synchronized to the server.

When you enable the Feature Key Synchronization (FKS), the settings of call forwarding and do not disturb (DND) on the server are synchronized to the phone. The changes in DND and call forwarding settings made on the phone will also be synchronized to the server. If configured, executives can access the **Settings > Assistant** menu on the phone. Similarly, assistants can access the **Settings > Executive** menu.



---

**Note** Feature Key Sync must be enabled for all the executive and assistant users.

---

### 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 extension number).
- Step 2** In the **Call Feature Settings** section, set the **Feature Key Sync** field to **Yes**.
- Step 3** Click **Submit All Changes**.

### Related Topics

- [DND and Call Forwarding Status Sync](#), on page 51
- [Enable Call Forwarding Status Sync via XSI Service](#), on page 52
- [Enable DND Status Sync via XSI Service](#), on page 53

## Enable Call Forwarding Status Sync via XSI Service

When call forwarding sync is enabled, the settings related to call forwarding on the server are synchronized to the phone. The changes in call forwarding settings made on the phone will also be synchronized to the server.



---

**Note** If XSI sync for call forwarding is enabled and the XSI host server or XSI account is not configured correctly, the phone user can't forward calls on the phone.

---

### Before you begin

- Access the phone administration web page. See [Access the Phone Web Page](#).
- Configure the XSI host server and the corresponding credentials on the **Voice > Ext (n)** tab.
  - When using **Login Credentials** for XSI server authentication, enter **XSI Host Server**, **Login User ID**, and **Login Password** in the **XSI Line Service** section.

- When using **SIP Credentials** for XSI server authentication, enter **XSI Host Server** and **Login User ID** in the **XSI Line Service** section, and **Auth ID** and **Password** in the **Subscriber Information** section.
- Disable Feature Key Sync (FKS) in **Call Feature Settings** section from **Voice > Ext (n)** .

### Procedure

---

- Step 1** Select **Voice > Ext [n]** (where [n] is the extension number).
- Step 2** Set the **CFWD Enable** field to **Yes**.
- Step 3** Click **Submit All Changes**.

### Related Topics

- [DND and Call Forwarding Status Sync](#), on page 51
- [Enable Feature Key Sync](#), on page 52

## Enable DND Status Sync via XSI Service

When do not disturb (DND) sync is enabled, the DND setting on the server is synchronized to the phone. The changes in DND setting made on the phone will also be synchronized to the server.



**Note** If XSI sync for DND is enabled and the XSI host server or XSI account is not configured correctly, the phone user can't turn on DND mode on the phone.

---

### Before you begin

- Access the phone administration web page. See [Access the Phone Web Page](#).
- Configure the XSI host server and the corresponding credentials on the **Voice > Ext (n)** tab.
  - When using **Login Credentials** for XSI server authentication, enter **XSI Host Server**, **Login User ID**, and **Login Password** in the **XSI Line Service** section.
  - When using **SIP Credentials** for XSI server authentication, enter **XSI Host Server** and **Login User ID** in the **XSI Line Service** section, and **Auth ID** and **Password** in the **Subscriber Information** section.
- Disable Feature Key Synchronization (FKS) in **Call Feature Settings** section from **Voice > Ext (n)**.

### Procedure

---

- Step 1** Select **Voice > Ext [n]** (where [n] is the extension number).
- Step 2** Set the **DND Enable** field to **Yes**.

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

---

**Related Topics**

[DND and Call Forwarding Status Sync](#), on page 51

[Enable Feature Key Sync](#), on page 52

## Executives and Assistants

You can set up executives and their assistants to share control of calls.

You configure users as executives and assistants in BroadWorks. The BroadWorks configuration also sets up the relationships between the executives and assistants. For more information, see the BroadWorks documentation.

After the BroadWorks configuration, you configure the following phone settings.

- Enable the synchronization of executive-assistant settings between the phone and the server.
- Update the dial plan to enable users to dial service activation codes.
- Configure a line key for the **Executive / Assistant** menu access if necessary.
- Change the service activation codes if necessary.
- Change the programmable softkeys if necessary.



---

**Important**

- Executives and assistants cannot share phones. Do not configure extensions on the same phone for an executive and an assistant.
  - The executive-assistant feature is recommended for private lines.
  - The number of calls that an assistant can initiate in parallel is limited to the **Call Appearances Per Line** setting under **Miscellaneous Line Key Settings** on the phone web page > **Voice** > **Phone** tab.
- 

## Synchronization of Executive-Assistant Settings

Executive and assistant functions require settings to be synchronized between the phones and the server via Feature Key Synchronization (FKS). Once you enable FKS, executives can access the **Settings** > **Assistant** menu on the phone. Similarly, assistants can access the **Settings** > **Executive** menu.

To enable FKS, see [Enable Feature Key Sync](#), on page 52.

## Dial Plan for Executives and Assistants

The dial plan for executives and assistants must include the following digit sequences:

- #xxx or a variant that includes the expression, to enable users to dial # codes.
- \*xxx or a variant that includes the expression, to enable users to dial \* codes.

- #xx+xxxxxxxxxxxx\*xxxxxxxxxxxx to enable users to dial # codes followed by numbers.

See [Dial Plan Overview](#) for information on the dial plan.

See [Edit the Dial Plan on the IP Phone](#) for details on how to add entries to the dial plan.

## Configure a Line Key for Executive and Assistant Menu Access

You can configure a line key for access to the **Settings > Assistant** menu on an executive's phone, and the **Settings > Executive** menu on an assistant's phone.



**Note** You can include the line key configuration in the XML Configuration file, as shown in the sample below.

```
<!-- Line Key 2 -->
<Extension_2_ ua="na">Disabled</Extension_2_>
<Short_Name_2_ ua="na">$USER</Short_Name_2_>
<Share_Call_Appearance_2_ ua="na">private</Share_Call_Appearance_2_>
<Extended_Function_2_ ua="na">fnc=bw-exec-assist</Extended_Function_2_>
```

### 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 **Line Key** section for the line key that you want to configure, specify the **Extended Function** as `fnc=bw-exec-assist`.
- Note** The **Extension** for the line key must be **Disabled**.  
We recommend setting **Share Call Appearance** to **private**.
- See [Line Key](#) for details on the fields in the **Line Key** section.
- Step 3** Click **Submit All Changes**.

## Service Activation Codes for Executives and Assistants

Executives and assistants can access most functions through a service activation code or a programmable softkey.

The following functions can be accessed only through service activation codes.



**Important** You will need to inform the users of the service activation codes for these functions in order to enable them to perform these actions.

- Opting in to and out of pools by assistants
- Joining ongoing calls by assistants
- Transferring ongoing calls to themselves by assistants

All of the service activation codes for the feature are set up by default.

You can change the service activation codes for specific functions according to your organization's requirement. See [Vertical Service Activation Codes](#) for details.



**Important** If you change a service activation code through the phone website, make sure to update the corresponding setting in BroadWorks, and vice versa.

### Executive-Assistant Service Activation Codes in the XML Configuration File



**Important** If you change a service activation code in the XML Configuration file, make sure to update the corresponding setting in BroadWorks.

The following sample XML Configuration file section shows the parameters (XML tags) and values for the executive-assistant feature service activation codes.

```
<!-- Vertical Service Activation Codes -->
<Exec_Assistant_Call_Initiate_Code ua="na">#64</Exec_Assistant_Call_Initiate_Code>
<Exec_Call_Filter_Act_Code ua="na">#61</Exec_Call_Filter_Act_Code>
<Exec_Call_Filter_Deact_Code ua="na">#62</Exec_Call_Filter_Deact_Code>
<Exec_Assistant_Call_Push_Code ua="na">#63</Exec_Assistant_Call_Push_Code>
<Exec_Call_Retrieve_Code ua="na">*11</Exec_Call_Retrieve_Code>
<Exec_Call_Bridge_Code ua="na">*15</Exec_Call_Bridge_Code>
```

The following table describes these parameters and values.

| Parameter                         | Value                                                     | Description                                                            |
|-----------------------------------|-----------------------------------------------------------|------------------------------------------------------------------------|
| Exec_Assistant_Call_Initiate_Code | The # or * code that you want to be used for the function | For assistants to initiate calls on behalf of executives               |
| Exec_Call_Filter_Act_Code         | The # or * code that you want to be used for the function | For executives to activate call filtering                              |
| Exec_Call_Filter_Deact_Code       | The # or * code that you want to be used for the function | For executives to deactivate call filtering                            |
| Exec_Assistant_Call_Push_Code     | The # or * code that you want to be used for the function | For assistants to transfer an ongoing call to an executive             |
| Exec_Call_Bridge_Code             | The # or * code that you want to be used for the function | For executives or assistants to join an ongoing call                   |
| Exec_Call_Retrieve_Code           | The # or * code that you want to be used for the function | For executives or assistants to transfer an ongoing call to themselves |



## Programmable Softkeys for Executives and Assistants

All of the programmable softkeys for executives and assistants are set up by default.

See [Programmable Softkeys, on page 28](#) and [Programmable Softkeys](#) for details on programmable softkeys.

### Executive-Assistant Programmable Softkeys in the XML Configuration File

The following sample XML Configuration file section shows the parameters (XML tags) and values for the programmable softkeys for executives and assistants.

```
<!-- Programmable Softkeys -->
<Programmable_Softkey_Enable ua="na">No</Programmable_Softkey_Enable>
<Idle_Key_List ua="na">em_login;acd_login;acd_logout;astate;avail;unavail;redial;
recents;cfwd;dnd;lcr;pickup;gpickup;unpark;em_logout;gustin;gustout;callretrieve;
bridgein;</Idle_Key_List>
<Hold_Key_List ua="na">resume|1;endcall|2;newcall|3;redial;dir;cfwd;dnd;
callpush;</Hold_Key_List>
<Shared_Active_Key_List ua="na">newcall|1;barge|2;bargesilent|3;cfwd|4;dnd|5;
callretrieve;bridgein</Shared_Active_Key_List>
<Shared_Held_Key_List ua="na">resume|1;barge|2;cfwd|3;dnd|4;</Shared_Held_Key_List>
<Exec_Assistant_Key_List ua="na">proxycall|2;divert|3;</Exec_Assistant_Key_List>
```

The following table describes these parameters and values.

| Parameters                               | Value        | Description                                                                                                |
|------------------------------------------|--------------|------------------------------------------------------------------------------------------------------------|
| Idle_Key_List,<br>Shared_Active_Key_List | bridgein     | Enables executives to join an ongoing call                                                                 |
| Idle_Key_List,<br>Shared_Active_Key_List | callretrieve | Enables executives to transfer an ongoing call to themselves                                               |
| Hold_Key_List                            | callpush     | Enables assistants to transfer an ongoing call to an executive, after putting the call on hold             |
| Exec_Assistant_Key_List                  | proxycall    | Enables assistants to initiate calls on behalf of executives, from the <b>Settings &gt; Executive</b> menu |
| Exec_Assistant_Key_List                  | divert       | Enables assistants to activate or deactivate call diversion, from the <b>Settings &gt; Executive</b> menu  |

## Configure Priorities for Voice and Video Data

You can prioritize voice or video data in limited bandwidth conditions.

You will need to configure the priorities individually on each line of a phone.

You can configure different priorities for different areas of traffic. For example, you can configure different priorities for internal and external traffic by setting up different configurations on internal and external lines. For effective traffic management, specify the same settings on all the phone lines in a group.

The Type of Service (ToS) field of a data packet determines the packet's priority in data traffic. You can configure the desired priorities by specifying appropriate values for the ToS fields of voice and video packets, for each phone line.

For voice data, the phone applies the ToS value that it receives by LLDP. When there is no ToS value available by LLDP, the phone applies the value that you specify for voice packets.

For video data, the phone always applies the ToS value that you specify for video packets.

The default values prioritize voice over video.

### 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 an extension number.
  - Step 2** In the **Network Settings** section, set the parameter values as described in [Network Settings](#).
  - Step 3** Click **Submit All Changes**.
-