Revision Document FCM Emergency Call Support MPP 11.1.2

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CHAPTER 1

Administration Guide Information

The following sections describe the new and updated feature information that is inserted into these books:

- *Cisco IP Phone 6800 Series Multiplatform Phones Administration Guide*
- *Cisco IP Phone 7800 Series Multiplatform Phones Administration Guide*
- *Cisco IP Conference Phone 7832 Multiplatform Phones Administration Guide*
- *Cisco IP Phone 8800 Series Multiplatform Phones Administration Guide*

New Administration Guide Information

New for Multiplatform Firmware Release 11.1(2)

Documentation is adding the following feature topics to the Administration Guides.

New Topics Added to the Administration Guides

- Phone Administration > Phone Features and Setup > *Emergency Calls*
  - *Emergency Call Support*
  - *Emergency Call Terminology*
  - *Configure a Phone to Make Emergency Calls*
- Cisco IP Phone Troubleshooting > Monitoring Phone System > Cisco IP Phone Web Page > Voice > Extension > *E911 Geolocation Configuration*
- Cisco IP Phone Troubleshooting > Troubleshooting > Feature Troubleshooting > *Emergency Call Doesn't Connect to Emergency Services*

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 2 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**.

E911 Geolocation Configuration

E911 Geolocation Configuration

Parameter	Description
Company UUID	The Universally Unique Identifier (UUID) assigned to the customer by the emergency call services provider. Maximum identifier length is 128 characters. Defaults to blank.
Primary Request URL	Encrypted HTTPS phone location request. The request uses the phone IP addresses, MAC address, Network Access Identifier (NAI), and chassis ID and port ID assigned by the network switch manufacturer. The request also includes the location server name and the customer identifier. The server used by the emergency call services provider responds with an Emergency Response Location (ERL) that has a location Uniform Resource Identifier (URI) tied to the user phone IP address. Defaults to blank.
Secondary Request URL	Encrypted HTTPS request sent to the emergency call services provider's backup server to obtain the user's phone location. Defaults to blank.

See Emergency Call Support Terminology, on page 2 for terms that describe emergency call support for phones.

An Emergency Call Doesn't Connect to Emergency Services

Problem

A user tries to place an emergency call, but the call doesn't connect to the emergency services (fire, police, or emergency services operator).

Solution

Check the emergency call configuration:

- Company Identifier or location request URL setup is incorrect. See Configure a Phone to Make Emergency Calls, on page 3.

- An incorrect or blank emergency number exists in the Dial Plan setup. See Dial Plan, on page 5

The location request servers (emergency call service provider) did not respond with a phone location, after multiple attempts.

Updated Administration Guide Information

Documentation is changing related feature information within the following topics that are found in the Administration Guides.

New and Changed for Firmware Release 11.1(2)

Feature	New or Changed Sections
Emergency Call Support	Dial Plan, on page 5 Telephony Features for Cisco IP Phone, on page 6 Basic Reset, on page 10

Dial Plan

Parameter	Description
Dial Plan	Dial plan script for the selected extension. The dial plan syntax allows the designation of three parameters for use with a specific gateway: • uid – The authentication user-id • pwd – The authentication password • nat – If this parameter is present, use NAT mapping. Separate each parameter with a semi-colon (;).
Caller ID Map	Inbound caller ID numbers can be mapped to a different string. For example, a number that begins with +44xxxxxx can be mapped to 0xxxxxx. This feature has the same syntax as the Dial Plan parameter. With this parameter, you can specify how to map a caller ID number for display on screen and recorded into call logs.
Enable URI Dialing	Enables or disables URI dialing.

Parameter	Description
Emergency Number	Enter a comma-separated list of emergency numbers. When one of these numbers is dialed, the unit disables processing of CONF, HOLD, and other similar softkeys or buttons to avoid accidentally putting the current call on hold. The phone also disables hook flash event handling. Only the far end can terminate an emergency call. The phone is restored to normalcy after the call is terminated and the receiver is back on-hook. Maximum number length is 63 characters. Defaults to blank (no emergency number).

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.
Auto Answer	Connects incoming calls automatically after a ring or two. Auto Answer works with the speakerphone.
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: • 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.
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.
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	Allows the user to move a connected call from an active state to a held state. • 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	Increases security by requiring communication using HTTPS. **Note** When the web is in HTTPS mode, the phone is an HTTPS server.
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 `Unknown`.
Jitter Buffer	The Jitter Buffer feature handles jitter from 10 milliseconds (ms) to 1000 ms for both audio and video streams.
Join Across Lines	Allows users to combine calls that are on multiple phone lines to create a conference call. 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.
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.
Message Waiting Indicator	When you have a message, a message displays on the phone screen. Your phone also provides an audible message-waiting indicator.
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.

Feature	Description and More Information
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	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. 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.
Music On Hold	Plays music while callers are on hold.
Mute	Mutes the phone 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.
Plus Dialing	Allows the user to dial E.164 numbers prefixed with a plus (+) sign. 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.
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.
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.
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	Enables administrators to quickly and easily gather debug information from phones. This feature uses SSH to remotely access each IP phone. SSH must be enabled on each phone for this feature to function.

Feature	Description and More Information
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	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. 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. See “Caller ID” in this table.
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.
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 Message System	Enables callers to leave messages if calls are unanswered.
Web Access Enable by Default	Web services are enabled by default.

Basic Reset

Performing a basic reset of a Cisco IP Phone provides a way to recover when the phone experiences an error. The reset provides a way to reset or restore various configuration and security settings.

Note When you set up emergency calls, the phone requests an updated location whenever a person restarts the phone.

The following table describes the ways to perform a basic reset. You can reset a phone with any of these operations after the phone has started up. Choose the operation that is applicable for your situation.

Table 1: Basic Reset Methods

Operation	Action	Explanation
Restart phone	Press **Applications** and choose **Admin Settings** > **Reset settings** > **Cold Reboot**. Press **Settings** and choose **Device Administration** > **Restart**.	Resets any user and network setup changes that you have made, but that the phone has not written to its Flash memory, to previously saved settings, then restarts the phone.
Reset settings	To reset settings, press **Applications** and choose **Admin Settings** > **Reset settings** > **Factory Reset**. Press **Settings** and choose **Device Administration** > **Factory Reset**.	Restores phone configuration or settings to factory default.

Note When you set up emergency calls, the phone requests an updated location whenever the you do the following actions:

- Registers the phone with the call server.
- Restarts the phone (phone is registered).
- Changes the network interface that is used for the SIP registration.
- Changes the IP address of the phone.

User Guide Information

The following sections describe the new and updated feature information that is inserted into these books:

- *Cisco IP Phone 6800 Series Multiplatform Phones User Guide*
- *Cisco IP Phone 7800 Series Multiplatform Phones User Guide*
- *Cisco IP Conference Phone 7832 Multiplatform Phones User Guide*
- *Cisco IP Phone 8800 Series Multiplatform Phones User Guide*

New User Guide Information

Documentation is adding the following feature topics to the User Guides for this release.

New for Multiplatform Firmware Release 11.1(2)

New Topics Added to the User Guides

Calls > Make Calls > *Make an Emergency Call*

Make an Emergency Call

Use your phone to make an emergency call, similar to any other call. When you dial the emergency number, your emergency services get your phone number and location so that they can assist you.

Note If your call disconnects, the emergency services can call you back.

Before you begin

Your phone must be set up to obtain your physical location. Emergency services personnel need your location to find you when you make an emergency call.

Procedure

Enter the emergency number and press **Call**.

Updated User Guide Information

Documentation is updating related feature information within the following topics that are found in the User Guides.

Updated for Multiplatform Firmware Release 11.1(2)

Sign in to a Phone as a Guest

Your phone has a guest account when your administrator enables hoteling on your phone. You can then sign in to a different phone in your network as a guest.

Procedure

Step 1 Press **Sign in**.

Step 2 Enter your user ID and password.

The password field uses two types of input methods; alphanumeric and numeric. While you type in the password, you see **Options** softkey on the phone. You can use this softkey to change the current password input type. Select **Input all** for alphanumeric input and select **Input num** for numeric entry.

Step 3 Press **Save**.

Note An administrator can set up a phone to make emergency calls. Whenever you sign in as a guest to a registered phone, the phone transfers a request to obtain the location of the phone. The location is sent to the emergency services when you make an emergency call.

Release Notes

Emergency 911 Support

You can register each IP-based phone with an emergency call service provider by supplying the E911 Geolocation information. Registration obtains the phone's location. The location can specify the street address, building number, floor, room, and other office location information. When you dial an emergency number, the emergency service receives the phone location and a call-back number. If an emergency call disconnects, the emergency service uses the call-back number to reconnect to the caller.

Where to Find More Information

- *Cisco IP Phone 6800 Series Multiplatform Phones Administration Guide*
- *Cisco IP Phone 6800 Series Multiplatform Phones User Guide*
- *Cisco IP Phone 7800 Series Multiplatform Phones Administration Guide*
- *Cisco IP Phone 7800 Series Multiplatform Phones User Guide*
- *Cisco IP Conference Phone 7832 Multiplatform Phones Administration Guide*
- *Cisco IP Conference Phone 7832 Multiplatform Phones Users Guide*
- *Cisco IP Phone 8800 Series Multiplatform Phones Administration Guide*
- *Cisco IP Phone 8800 Series Multiplatform Phones User Guide*

Author Notes

Firmware Release

Multiplatform Firmware Release 11.1(2)

Feature IDs

Cisco uses this feature identifier to insert emergency call information into its Multiplatform Phone documentation.

- `tpcc-f1940-emergencycallsupport-1112`

Phones Supported by This Feature

- Cisco IP Phone 6800 Series Multiplatform Phones
- Cisco IP Phone 7800 Series Multiplatform Phones
- Cisco IP Conference Phone 7832 Multiplatform Phones
- Cisco IP Phone 8800 Series Multiplatform Phones

Feature Content References

- **F1940**–https://rally1.rallydev.com/#/14039422974d/detail/portfolioitem/feature/129235490612?fdp=true

Revision History

Date	Description
February 22, 2018	Insert all added Administration Guide information.
February 27, 2018	Simplify terminology for E911 introduction and configuration task.
March 2, 2018	Show where in each document the content is added as a new topic (feature ID used in bookmap). Also show content that is inserted into topics (using the feature ID attribute).
March 21, 2018	Insert context for all topics, internationalize feature, and develop a content outline for the author's notes.
March 22, 2018	Insert revisions that are based on version two peer comments.