Cisco Unified MobilityManager Troubleshooting FAQs

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This document provides questions and answers to assist in configuring, administering, and troubleshooting Cisco Unified MobilityManager.

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Contents

This document contains the following sections:

Related Documentation

For more information about Cisco Unified IP Phones or Cisco Unified CallManager, refer to the following publications:

Cisco Unified MobilityManager

Refer to the documentation set for Cisco Unified MobilityManager for detailed configuration and use information. Navigate from the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_mobmg/1_2/index.htm

Cisco Unified IP Phone Documentation

Refer to publications that are specific to your language, phone model and Cisco Unified CallManager version. Navigate from the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/index.htm

Cisco Unified CallManager Documentation

Refer to the Cisco Unified CallManager Documentation Guide and other publications specific to your Cisco Unified CallManager version. Navigate from the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_callmg/index.htm

Mobile Connect Administration Questions

This section contains questions and answers to assist in Mobile Connect configuration and operation.

Q. Why does the call drop when the user answers a call on a cellular phone?

A. The call may drop due to timer settings. If a user answers the cellular phone in less than the default of 1.5 seconds for Minimum Cell Phone Ring Timer (1500 ms), the system assumes that cellular phone voice mail picked up the call. The following steps can prevent this from occurring:

1. The user should reduce the timer value on the User page.
2. If multiple users complain about the problem, the administrator should modify the timer on the System Parameters page. This will modify the timer for all users. (Individual users can still override the system parameters, if desired.)
3. If modifying the timer does not resolve the problem, then verify that the codec transcoders are configured properly in Cisco Unified CallManager.

Q. Why is caller ID not passed to the cellular phone?

A. The following criteria are required and must be verified to ensure that caller ID can be passed to the cellular phone:

- The CTI route point must be properly configured.
- The CTI route point must be associated with the CTI shared line user.
- The Calling search space selections must be correct.
- The route pattern for calling cellular phones must have a configured partition.

- The Caller ID directory number (DN) parameter must be left empty in the Cisco Unified CallManager gateway configuration. If this parameter is configured, the setting overrides the Mobile Connect adjusted caller ID.

Q. Why are only some remote destinations ringing?

A. Each call to a remote destination requires a CTI port for the duration of the call. If some remote destinations are not ringing, it is possible that not enough CTI ports are configured.

Q. Why would a user not have enough time to answer the call on the cellular phone?

A. A user might not have enough time to answer a call if the Desk Phone Timer to Ring to Voicemail duration is too short. To resolve this problem, do both of the following in Cisco Unified CallManager:

- Increase the No Answer time duration (in Line Configuration) to be greater than Maximum Cell Phone Ring Timer (19 seconds default). This setting can be adjusted by the user.
- Modify the Delay Before Ringing Cellular Phone parameter. The default is 4 seconds (4000 ms). Only an administrator can change the value.

Q. Why doesn't the user see the Resume key on the desk phone after hanging up the cellular phone for desk pickup?

A. The problem could be due to the cellular phone service provider. When disconnecting a call for the cellular phone, some cellular providers play a tone (similar to a busy tone) instead of sending a disconnect signal to Cisco Unified CallManager. Cisco Unified CallManager cannot detect the call hangup; the other party hangs upon hearing the tone. Because the call has not been disconnected immediately, the desk pickup feature cannot work.

Q. How can a call be prevented from going to the user's cellular phone voice mail instead of desk phone voice mail?

A. To prevent a call from going to the user's cellular phone voice mail, set the Maximum Cell Phone Ring Timer to be less than the cellular phone provider's voice mail time out. The default is 19 seconds. An alternative is to reduce the No Answer Duration in the Cisco Unified CallManager line configuration.

Q. Why can't the user pick up calls using the cellular phone?

A. The device might not be subscribed to the Cisco Unified MobilityManager service or cellular pickup might not be enabled for the user. Verify that the user is correctly subscribed and that cellular pickup is enabled.

Q. Why are calls not being extended to the cellular phone?

A. Calls may not be extended to the cellular phone if more than one CTI route point is created for the mobility partition in Cisco Unified MobilityManager. Outgoing CTI ports should have a calling search space configured to include only this partition, and the CTI route point should have a calling search space to call out the remote destination Directory User settings.

Q. Why is the user not receiving calls on the cellular phone?

A. To receive calls on the cellular phone, the user must enable the cellular phone as a remote destination. To test this, the user can make a direct call to that cellular phone and verify that the configured line appearance matches the user's desk phone extension. The calling search space configuration in Cisco Unified CallManager must also be correct.

Q. How can the user ensure the security of the desk phone after hanging up on the cellular phone, preventing someone else from resuming the call using desk pickup?

A. To prevent someone else from resuming the call, the user should wait for the calling party or called party (in case of Mobile Voice Access) to hang up first.

Q. Why can't users log in to User Options web pages?

A. To access the User Options pages, directory settings must be properly configured. Verify that the directory settings precisely match the settings in the DirectoryConfiguration.ini file in Cisco Unified CallManager. This is also required for Mobile Voice Access authentication.

Q. How do the various timers interact?

A. The following timers interact with each other: Delay Before Ringing Timer and Maximum Cell Phone Ring Timer (in Cisco Unified MobilityManager) and No Answer Duration in Cisco Unified CallManager. These timers may also interact with the cellular phone provider's voice mail settings.

To avoid conflicts, follow these guidelines:

- Verify that the value of the Maximum Cell Phone Ring Timer is less than the timer set for cellular phone provider's voice mail.
- Assign settings so that Delay Before Ringing Timer + Maximum Cell Phone Ring Timer < No Answer Duration.

Q. Why does music play on calls made through a Session Initiation Protocol (SIP) trunk, even after the phone is answered?

A. To avoid having the music played, check the following:

a. Select the Media Termination Point (MTP) Required checkbox.

b. Verify that the device pool includes the Media Resource Group List (MRGL) name. MRGL has the Media Resource Group (MRG) configured, and MRG has the Media Termination Point (MTP) selected from the available media resources.

c. If you are using the hairpin method to configure Mobile Voice Access, check MTP in the H323 Gateway configuration in Cisco Unified CallManager.

Q. Why does the desk phone stay in the Remote In-Use state for up to 30 seconds after the user hangs up the cell phone?

A. After hanging up the cell phone, the calling party hears the reorder tone and the desk phone stays in the Remote In-Use state for up to 30 seconds before moving to the On-Hold state. The desk phone can then resume the call with the calling party. The delay occurs because the mobile carrier waits before sending a disconnect message to Cisco Unified Communications Manager, probably to avoid false disconnects.

Mobile Voice Access Questions

This section contains questions and answers to assist in Mobile Voice Access configuration and operation.

Q. How do I prevent a call from failing after a user accesses Mobile Voice Access, enters the User ID and password, and selects option 1 to make a call?

A. To avoid the problem of call failure, do the following:

- Enable the Enable System Remote Access parameter settings in the System page and then enable the Enable User Remote Access parameter in the Line Appearance page (under User settings).

  Note If the system level parameter is not enabled, users will not be able to place calls through Mobile Voice Access, regardless of the individual user settings.

- Verify that the calling search space configured under Inbound Calls for the H323 Gateway configuration in Cisco Unified CallManager includes the outbound CTI ports partition.
- Verify that the dial peer to transfer the call to the outbound CTI port is correct

Q. How do I prevent Mobile Voice Access from dropping after the welcome message?

A. To avoid this problem, make sure that the user's directory settings are configured properly and that the settings (including password) match the settings configured in Cisco Unified CallManager.

Q. Why is the user is locked out of Mobile Voice Access?

A. Mobile Voice Access lockout is controlled by the Mobile Voice Access User Lock Out Timer system parameter. The values can range from 0 minutes (no lockout) to up to 1440 minutes (default is 15 minutes). If a user enters the wrong PIN or remote destination three times when dialed into Mobile Voice Access, the user will be locked out for the configured period of time.

The user can wait the specified time before trying again. For urgent cases, the administrator can set this parameter to 0, ask the user to dial in once, and then change it back to previous settings.

Q. Why does the call sometimes drop without any tone when making an enterprise dial tone call using Mobile Voice Access?

A. Most of the time when a wrong number is dialed, the user hears an announcement by Cisco Unified CallManager. In rare cases, if Cisco Unified CallManager plays a reorder tone, the call will drop without any tone. Also, if a call is made to a busy user and Cisco Unified CallManager cannot divert the call to voice mail or the user doesn't have voice mail, a busy tone is played. The call will also drop if the System Remote Access Call Take Back Timer (default 120 seconds) on the System parameters page expires.

Q. Why does the call drop after the user dials the destination number in Mobile Voice Access?

A. The call may be dropping if the calling search space of inbound calls for the Gateway configuration in Cisco Unified CallManager does not include the outgoing CTI ports partition.

Q. Why does the call drop after the user selects the make call option and dials the destination number using Mobile Voice Access?

A. The call may be dropping if the VoIP dial peer for the outgoing port route, as given in the *Cisco Unified MobilityManager Installation Guide*, is not correctly configured.

Serviceability Questions

This section contains questions and answers to assist with logs, records, and measurements.

Q. What are the logs, measurements, alarm, and call detail records (CDRs) available in Cisco Unified CallManager?

A. The following items are provided in Cisco Unified CallManager:

In /var/log/active/cmm/logs:

- cmmApp.log.* - for Cisco Unified MobilityManager

- CMMJatpi* - for corresponding JTAPI logs for application logs

In /var/log/active/tomcat/logs:

- cmmAdmin.log* - for Admin web page GUI
- cmmUser.log* - for User web page GUI
- cmmIvr.log.* - for Mobile Voice Access (through Mobile Voice Access)
- tar.log - for backup
- untar.log - for Restore

In /var/log/install:

- upgrade-log-* - for upgrade

In /opt/informix/database/logs:

- bar_act*.log - for backup/restore of

In /var/log/active/cmm/serviceability/alarm:

- alarm.log* - for alarm history

In /var/log/active/cmm/serviceability/measurement:

- meas.log - for measurement counters every 15 minutes

In /var/log/active/cmm/serviceability/cdr:

- cdr.log - for CMM cdrs

Q. What is the size of the logs?

A. All the logs rotate after they reach 1MB size. This size is not configurable.

Q. What is the total number of logs generated?

A. A total of 250 files (fixed, not configurable) is created, and after that it starts rotating. For JTAPI, the maximum number of log files is 100.

Q. What is the desired log level to set?

A. We recommend that users keep the logs at the default (Info) level. Debugging can be turned on when necessary. The exception is for JTAPI logs, which are always in Debug mode.

Q. How are logs rotated?

A. The logs cmmApp.log*. cmmUser.log*, cmmAdmin.log*. cmmIvr.log.*, meas.log*, alarm.log*, cdr.log* use log4j. The following example of log rotation is for cmmApp.log. Rotation for other logs is similar.

The cmmApp.log is created first. When the log hits 1MB, cmmApp.log is renamed to cmmApp.log.1 and a new cmmApp.log is created. When this hits 1MB size, cmmApp.log.1 is renamed as cmmApp.log.2, cmmApp.log is renamed as cmmApp.log.1 and a new cmmApp.log is created.

This continues until the maximum number of 250 files is created (cmmApp.log.250 is created). After that, cmmApp.log.250 will be removed, cmmApp.log.249 will be renamed to cmmApp.log.250, cmmApp.log.248 to cmmApp.log.249,....cmmApp.log.2 to cmmApp.log.3, cmmApp.log.1 to cmmApp.log.2, cmmApp.log to cmmApp.log.1, and a new cmmApp.log is created.

General Configuration Questions

This section contains questions and answers to assist with system-level configuration issues for Cisco Unified MobilityManager and Cisco Unified CallManager.

Q. How is the JTAPI update done?

A. The jtapi.jar packaged with Cisco Unified MobilityManager (/usr/local/snr/jar directory) is compatible with the Cisco Unified CallManager release 4.1. If any other release is used, configure the JTAPI update configuration in the System Parameters page. This verifies jtapi.jar compatibility with the Cisco Unified CallManager version and downloads the correct version, if necessary. This process requires restarting Cisco Unified MobilityManager. You can reboot using the platform web page or use the Restart button in the System page.

The JTAPI parameter configuration is in the jtapi.ini file located in /usr/local/snr/jar directory. This file should not be modified by the customer. If necessary, TAC can be advised to modify the file to solve any issues. Any change to this file requires a restart of Cisco Unified MobilityManager to enable the file changes to take effect.

Q. How can CTI Link status be verified?

A. Check that CTI links (Shared line CTI user links and Outgoing CTI user link), are configured properly in Cisco Unified CallManager. If a link is out of service, their might be a problem with the user ID and password. Also verify that CTI Manager is running in Cisco Unified CallManager. Delete and add the link; it cannot be modified.

Q. Why are CTI ports and the CTI route point not being properly registered?

A. To be properly registered, it is necessary that the shared line CTI ports and CTI route point be associated to the shared line CTI User. Verify that these associations are correct. If the association is correct, check in the Cisco Unified CallManager Device/Phone page to see if the ports and route point are registered. You can also verify it is cmmApp.log*, where for each shared line CTI port and Outgoing CTI port, an INFO log is printed if the registration is successful.

Q. What commands are available to debug Gateway/VXML issues?

A. The following debug commands are required to debug any potential Gateway/VXML issues:
debug isdn q931
debug cch323 h225
debug voip dial-peer all
debug voip application vxml all

Q. How can it be determined if the VXML application is loaded properly?

A. To verify that the VXML application is loaded properly, use the following command:

sh call application voice <*app name*>

This command will show a VXML page. Look for the IP address. If the IP address is not correct or has a null value, you must set the IP address in the Host Settings page of the Platform Administration web page. Then restart Cisco Unified MobilityManager and reload the application in the Gateway.

Sample VXML page:

```
VXML Application snr3
URL=http://172.22.120.104:8080/cmmivr/pages/IVRMainpage.vxml
Security not trusted
No languages configured
```

```
It has: 0 calls active.
7 incoming calls
0 calls handed off to it
3 call transfers initiated
35 pages loaded, 35 successful
38 prompt play attempts, 38 successful
0 recorded messages
The transfer mode is 'rotary'(Default)
Interpreted by Voice Browser Version 2.0 for VoiceXML 1.0 &
2.0.
```

VXML Script:

```
<?xml version=”1.0” encoding=”iso-8859-1”?>
<vxml version=”1.0”>

<form id=”main”>
<block>

<prompt>
<audio
src=”http://172.22.120.104:8080/cmmivr/audio/english/1.au”/>
</prompt>
<var name=”callerid” />
<assign name=”callerid" expr="session.telephone.ani"/>

<var name="langdir" />
<assign name="langdir" expr="'english'"/>

<if cond="callerid !='blocked' " >
<submit
next="http://172.22.120.104:8080/cmmivr/IVRCalleridLookup.do"
method="get" namelist="callerid langdir"/>
<else />
<goto next="#Getuserid"/>
</if>

</block>

</form>
<form id="Getuserid">
<field name="userid" type="digits?minlength=3;maxlength=16">
<prompt>
<audio
src="http://172.22.120.104:8080/cmmivr/audio/english/2.au"/>
</prompt>
```

```
<noinput count="3">

<exit/>

</noinput>

<filled>
<var name="callerid" />
<assign name="callerid" expr="null"/>

<var name="langdir" />
<assign name="langdir" expr="'english'"/>
<submit
next="http://172.22.120.104:8080/cmmivr/IVRUseridLookup.do"
method="get" namelist="userid callerid langdir"/>
</filled>
</field>
</form>
</vxml>
```

Q. Why does the Cisco Unified MobilityManager service on the IP phone give an error?

A. The service may be giving an error if the service is not subscribed or the directory number (DN) is not correctly configured. In the Service configuration of Cisco Unified CallManager, be sure to check the Parameter Required checkbox. The parameter should have the value "dn" (lowercase). You can also collect the snoop logs for the http request in the Mobility server using following command:
tethereal -i eth0 -f 'port 8081' -x -d tcp.port==8081,http

If there is still a problem, provide the cmmApp.log.

Q. What Caller IDs should the user expect to see?

A. If the Caller ID DN field on the Gateway configuration in Cisco Unified CallManager is configured, that setting overrides all other Cisco Unified MobilityManager caller IDs. If the PSTN service provider allows only enterprise DID numbers, then the caller ID will be modified by the provider to be the enterprise DID, even though Cisco Unified MobilityManager sends the correct caller ID.

For example, consider the following scenario:

IP Phone A: 3025
IP phone B: 3016
PhoneB_CTI: 3016 (This is the extension configured in Cisco Unified MobilityManager)
PSTN C: 9199915659
PSTN D: 9193452615
Phone B has the Shared CTI Port that is being monitored by Cisco Unified MobilityManager.

When no CallerID option is configured on the system or on the user profile in Cisco Unified MobilityManager, then the following is seen in the following call flows:

– A Calls B
IP Phone B shows: From 3025
PSTN D shows: 3025 (Some PSTN providers may not allow to show extension DN, and they require full DID number. For those this will be shown as Unknown Number or private Caller.)

– C Calls B
IP Phone B shows: From 9199915659
PSTN D shows: 9199915659
When you enable the caller ID option on the system or the user option/profile, then caller IDs change. The caller ID for this user in Cisco Unified MobilityManager was set to 6622613016.

– A Calls B
IP Phone B shows: From 3025
PSTN D shows: 6622613016

– C Calls B
IP Phone B shows: From 9199915659
PSTN D shows: 6622613016

Q. How can the root password be changed?

A. After logging in as root, use the setrootpw.sh script to change the root password.

Q. What is the default Informix password and how can it be changed?

A. The default Informix password is a combination of `XimRoFn1=` and the host name of the Cisco Unified MobilityManager. For example, if cmm101 is the host name, then the password is `XimRoFn1=cmm101`. To change the Informix password, log in as `root` and use setdbpw.sh script.

Q. How are CDRs handled in Cisco Unified MobilityManager?

A. To handle CDRs, Cisco Unified MobilityManager forks incoming calls to the cellular phone and maintains the association between multiple calls (depends on number of remote extensions provisioned). When an incoming call is answered at a cellular phone or when a call is placed using Mobile Voice Access, multiple CDR records are logged. A unique Cisco Unified MobilityManager Caller ID is generated for each new incoming call (or outgoing call through Mobile Voice Access). This is the key that ties up all the CDRs related to the Cisco Unified MobilityManager call. In addition, debugging details can be extracted by using the Cisco Unified CallManager CallId field, and from the related CDR records.

The Following CDR fields are written for each call leg.
snrCallId - CMM CallId (unique for each call)
globalCallID_callId - Cisco Unified CallManager caller ID for that call Leg
globalCallID_callManagerId - Cisco Unified CallManager
callingPartyNumber - Calling party number
calledPartyNumber - Called party Number
snrStatus - Call Status (desk pickup, cell pickup, system remote access)
dateTimeConnect - Connect time
dateTimeDisconnect - Disconnect time

Sample:
======
daf97b48e20a22148bea5ea79742be69-2,3,1,4003,14082309859,cell pickup,0,1125440504
daf97b48e20a22148bea5ea79742be69-2,2,1,4002,5053087,cell pickup,1125440504,1125440540
daf97b48e20a22148bea5ea79742be69-2,1,1,5009,5001,desk pickup,1125440544,1125440566
daf97b48e20a22148bea5ea79742be69-2,4,1,4003,14082309859,cell pickup,0,1125440564
daf97b48e20a22148bea5ea79742be69-2,5,1,4002,5053087,cell pickup,1125440564,1125440579

daf97b48e20a22148bea5ea79742be69-2,1,1,5009,5001,desk pickup,0,1125440589

In this sample, daf97b48e20a22148bea5ea79742be69-2 is the Cisco Unified MobilityManager caller ID.

There are two remote destinations associated with the extension 5001. A call is made from 5009 to 5001.

Using 4003 pooled CTI port, a call is made to 4082309859, and using 4002 a call is made to 5053087.

The call is answered in 50503087 (the connect/disconnect time is given).

The call is then resumed at the desk phone.

Then cellular pickup is performed, and the call is answered in 5053087 again.

Because the call is not answered in 4082309859, the connect time is listed as zero.

The caller ID is in the next column followed by the Cisco Unified CallManager ID.

Cisco Unified MobilityManager considers these steps all the same call, but Cisco Unified CallManager treats it as five calls with five different CDRs.

The last record shows that the resume key was presented, but the call is not resumed, so the connect time is 0.

Sample:
======
b31087e6b3eb30e1680fb125566d6ff2-4,258,1,4082309859,4002,system remote access,1124992984,1124993011
b31087e6b3eb30e1680fb125566d6ff2-4,259,1,5002,9023382,desk pickup,0,1124993016

Q. What is the equivalent old and terminology used in Cisco Unified MobilityManager logs and documents?

A. The following table lists the old and new terminology:

Old Terminology	New Terminology
SNR (Single Number Reach)	Cisco Unified MobilityManager
Enterprise Dial tone	Mobile Voice Access
VM Pull back timer	Minimum cell phone ring timer
Call Hangup timer	Maximum call wait timer before disconnect
Call pullback timer	Maximum cell phone ring timer
Outgoing CTI ports	Outgoing CTI ports (no change)
Pooled CTI ports	Pooled CTI ports (no change)

Q. Why does the user get a fastbuy/dialtone when calling the remote destination through the FXO/FXS port configuration in the gateway?

A. This problem can occur due to a known limitation on using FXO/FXS ports to call PSTN numbers using Cisco Unified MobilityManager.

Sample:
======

```
voice-port 1/0/0
timeouts call-disconnect 2
timeouts wait-release 1
supervisory disconnect dualtone mid-call supervisory answer
dualtone no battery-reversal signal groundStart !

Obviously your FXO might be connected to another kind of Analog
line. My suggestion for you is to review the following docs:

Understanding FXO Disconnect Problem:
http://www.cisco.com/en/US/partner/tech/tk652/tk653/technologies_t
ech_note09186a00800ae2d1.shtml

FXO Answer and Disconnect Supervision:
http://www.cisco.com/en/US/partner/products/sw/iosswrel/ps1839/pro
ducts_feature_guide09186a0080087b4f.html
Remember, CMM needs to know when the PSTN called has been answered
(so it will stop calling the other devices) and when the PSTN call
has been released (so it can offer the call to the originally
called IP Phone). And it needs this information on a timely
fashion.
```

Q. Why does loading the VXML application in the gateway configuration cause an error?

A. This error occurs because some IOS images do not support VXML. In these cases, another image instead should be used instead.

The H323 Gateway IOS image from 12.2(11) and later supports VXML. The following images have issues with VXML:
12.2(15), 12.3(8), 12.3(9)

H323 platform requirements for VXML:

1751V, 1760, 2610-2613, 2620-2623, 2650-2653,3660, 2610-2613XM*, 2620- 2623XM*, 2650- 2651XM*, 2691, 3725, 3745, 28XX,38XX, 5350, 5400

* - supported on IOS image 12.4 release
http://wwwin.cisco.com/access/mce/tech/docs/vxml_session_support.ppt

Q. What are the links for VXML configuration in the gateway documentation?

A. The following links have some useful information about VXML support in Gateway:
http://www.cisco.com/en/US/partner/products/sw/iosswrel/ps1839/products_feature_guide09186a00800f2405.html
http://www.cisco.com/en/US/partner/tech/tk652/tk773/technologies_design_guide09186a0080188778.shtml
http://www.cisco.com/cgi-bin/dev_support/access_level/product_support
http://von.cisco.com/interoperability/UnifiedCommunications/vxml/channelbank.doc

Q. How are bulk calls handled?

To handle bulk calls, 3,000 IP phones and 1,500 shared lines are registered to one Cisco Unified CallManager subscriber. Two JTAPI shared line links (one link for 1,000 shared lines and another for 500 shared lines) are used to link to the same subscriber. Remote destination calls cannot last more than 50 minutes for 27,000 Busy Hours Call Attempt (BHCA).

If three JTAPI shared line links are used, remote destination calls can last for an hour with 60 seconds of hold time for 27,000 BHCA (13,500 calls for desktop and 13,500 calls for the remote destination).

Each shared line user link has 500 shared lines associated with it. Therefore, it is necessary to use three shared link user links for 1,500 users/shared lines and one for the route point to perform the load balancing for JTAPI links to Cisco Unified CallManager. One must still be maintained for the outgoing port user link.