



Enterprise One Number

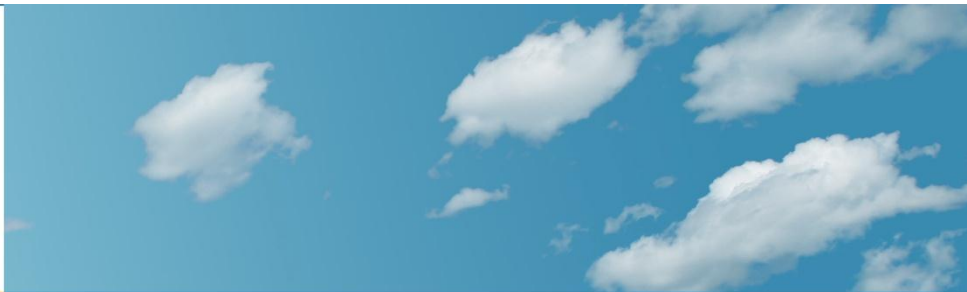
## Cisco CallManager and QuesCom GSM Gateway configuration

[info@quescom.com](mailto:info@quescom.com)

December 2010



+33 (0)4 97 23 48 48



## Table of content

<b>1. PURPOSE OF DOCUMENT .....</b>	<b>3</b>
<b>2. OVERVIEW .....</b>	<b>3</b>
<b>3. H.323 TRUNK .....</b>	<b>4</b>
3.1. CISCO CCM TRUNK CONFIGURATION .....	4
3.2. QUESCOM GATEWAY CONFIGURATION .....	6
<b>4. SIP TRUNK .....</b>	<b>8</b>
4.1. CISCO CCM TRUNK CONFIGURATION .....	8
4.2. QUESCOM GATEWAY CONFIGURATION .....	9

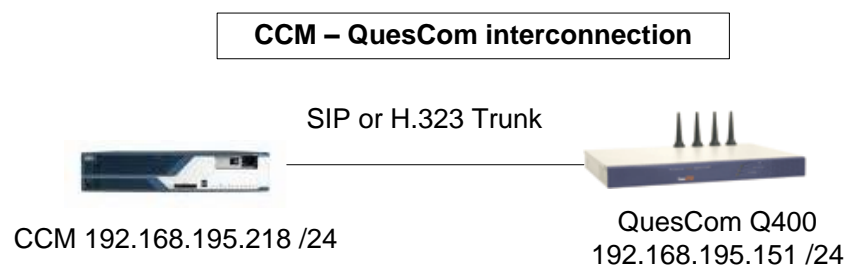
[www.quescom.com](http://www.quescom.com)

## 1. Purpose of document

The purpose of this document is to go through the different stages to configure manually CISCO CCM and QuesCom gateway interconnection through SIP or H.323 Trunk.

Note that this document shows in details the setup parameters, which are by default automatically configured by QuesCom Gateway wizard.

## 2. Overview



## 3. H.323 Trunk

### 3.1. Cisco CCM Trunk configuration

Select a Trunk type Inter-cluster Trunk (Non-Gatekeeper-Controlled)

The screenshot displays the Cisco Unified CM Administration web interface for configuring a Trunk. The page title is "Trunk Configuration" and the status is "Ready".

**Device Information**

Product:	Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol:	Inter-Cluster Trunk
Device Name*	Mobily_H323
Description	
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
Packet Capture Mode*	None
Packet Capture Duration	0

Media Termination Point Required  
 Retry Video Call as Audio  
 Path Replacement Support  
 Transmit UTF-8 for Calling Party Name  
 Unattended Port  
 SRTP Allowed - When this flag is checked, IPsec needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.  
Use Trusted Relay Point\* Default

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Incoming Calling Party National Number Prefix	Default
Incoming Calling Party International Number Prefix	Default
Incoming Calling Party Unknown Number Prefix	Default
Incoming Calling Party Subscriber Number Prefix	Default

**Multilevel Precedence and Preemption (MLPP) Information**

MLPP Domain	< None >
-------------	----------

# Enterprise One Number

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go

rouaud | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

### Trunk Configuration

Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

Prefix DN:

Redirecting Number IE Delivery - Inbound  
 Enable Inbound FastStart

#### Outbound Calls

Called Party Transformation CSS: < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection\*: Originator

Calling Line ID Presentation\*: Default

Called Party IE Number Type Unknown\*: Cisco CallManager

Calling Party IE Number Type Unknown\*: Cisco CallManager

Called Numbering Plan\*: Cisco CallManager

Calling Numbering Plan\*: Cisco CallManager

Caller ID DN:

Display IE Delivery  
 Redirecting Number IE Delivery - Outbound  
 Enable Outbound FastStart  
Codec For Outbound FastStart: G711 u-law 64K

#### Remote Cisco Unified Communications Manager Information

Server 1 IP Address/Host Name\*: 192.168.195.151  
Server 2 IP Address/Host Name:   
Server 3 IP Address/Host Name:

#### UUIE Configuration

Passing Precedence Level Through UUIE  
Security Access Level: 2

Save | Delete | Reset | Add New

\* - indicates required item.  
\*\* - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Create adequate route pattern

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go

rouaud | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

### Find and List Route Patterns

+ Add New | Select All | Clear All | Delete Selected

Status: 2 records found

Route Patterns (1 - 2 of 2) Rows per Page 50

Find Route Patterns where: Pattern begins with XXXX Find Clear Filter

Pattern	Description	Partition	Route Filter	Associated Device	Copy
XXXXX!		Quescom153		Mobilv H323	
XXXXX!	to Quescom152	Quescom152		Qlinux mobilv152	

Add New | Select All | Clear All | Delete Selected

## 3.2. QuesCom Gateway configuration

In this example we will use a QuesCom gateway based on v6.20. Please note that presentation differs from the next example of SIP trunk configuration which is based on QuesCom v5.20.

Create H323 trunk

The screenshot shows the QuesCom web interface for configuring a Foreign Gatekeeper. The interface includes a sidebar with navigation options such as Users & Mobility, Objects, and Services. The main content area displays the 'Foreign Gatekeeper' configuration form, which is divided into several sections:

- Settings:** ID (CCMH323), Gatekeeper Type (H323 selected), Name (CCMH323), IP Address (192.168.195.218), Host name, Listen Port (1720), No Resp. delay (ms) (0), Default SmartAD (Q401-CS-00020313), and VoIP Profile (ND\_RAS\_H323).
- SIP registration and authentication:** Registration user name, Dialing domain, User name, Password, Localisation (International Prefix, Country Code, Area Code, National Prefix).
- Supported prefixes:** A table with columns for Prefix and Number, and a 'Prefix' checkbox.

At the bottom of the form, there are buttons for Add, Remove, Update, and Clear All, along with a green 'SAVE' button and a red 'CANCEL' button.

# Enterprise One Number

Create a service from CCMH323 \* to GSM Pool

The screenshot shows the 'Service' configuration page in QuesCom. The 'Service enabled' checkbox is checked. Under 'Origin', 'Origin Type' is set to 'Foreign GK', 'Origin' is 'CCM323', and 'Call Type' is 'Foreign Gatekeeper'. 'Service associated' is set to 'GSM'. 'Call Server Operations' shows 'LCR Support' as 'None', 'CDR Support' as 'Yes', and 'Cost Support' as 'No'. 'VoIP Service' has 'Law Transcoding' checked and 'Quality of service(HEX)' set to 'Min Delay'. 'Switch service' is set to 'GSM' with 'Slot Balancing' checked. 'Destination' is set to 'Device Group', 'IP Address' is 'GSM\_POOL(SmartAD)', and 'Device' is 'GSM\_POOL(SmartAD)'. 'Balancing Mode' is 'Local'. 'Backup Mode' has 'DSP', 'Relay', 'RSVP', and 'Other' checked. 'Fax / Voice Service' has 'Max Duration (s)' set to 120.

Add the following service to route incoming GSM calls to CCM

The screenshot shows the 'Service' configuration page in QuesCom. The 'Service enabled' checkbox is checked. Under 'Origin', 'Origin Type' is 'Device', 'Origin' is '0401-C6-00020313(SmartAD)', and 'Call Type' is 'GSM Incoming'. 'Service associated' is set to 'VoIP'. 'Call Server Operations' shows 'LCR Support' as 'No', 'CDR Support' as 'Yes', and 'Cost Support' as 'No'. 'VoIP Service' has 'Law Transcoding' checked and 'Quality of service(HEX)' set to 'Min Delay'. 'Switch service' is set to 'GSM' with 'Slot Balancing' checked. 'Destination' is set to 'Foreign GK', 'IP Address' is 'CCMH323', and 'Device' is 'CCMH323'. 'Balancing Mode' is 'Local'. 'Backup Mode' has 'DSP', 'Relay', 'RSVP', and 'Other' checked. 'Fax / Voice Service' has 'Max Duration (s)' set to 120.

## 4. SIP TRUNK

### 4.1. Cisco CCM Trunk configuration

#### Define SIP Trunk

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go

rouaud | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

**Trunk Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

**Status**  
Update successful

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	Qlinux_mobili152
Description	
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Packet Capture Mode*	None
Packet Capture Duration	0

Media Termination Point Required  
 Retry Video Call as Audio  
 Transmit UTF-8 for Calling Party Name  
 Unattended Port

Done | Internet | 100%

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | Go

rouaud | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

**Trunk Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

Caller Name

Redirecting Diversion Header Delivery - Outbound

**SIP Information**

Destination Address	192.168.195.151
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	5060
MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	RFC 2833

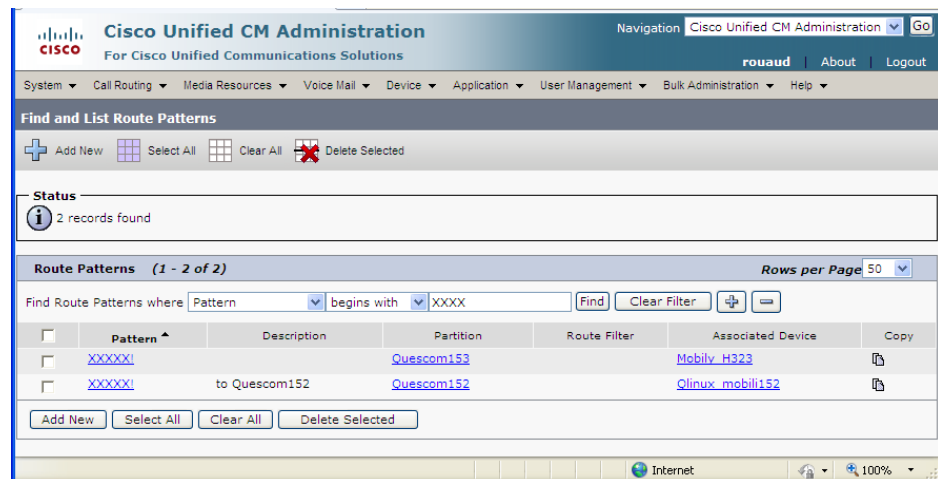
Save | Delete | Reset | Add New

**\*** - indicates required item.  
**\*\*** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Done | Internet | 100%



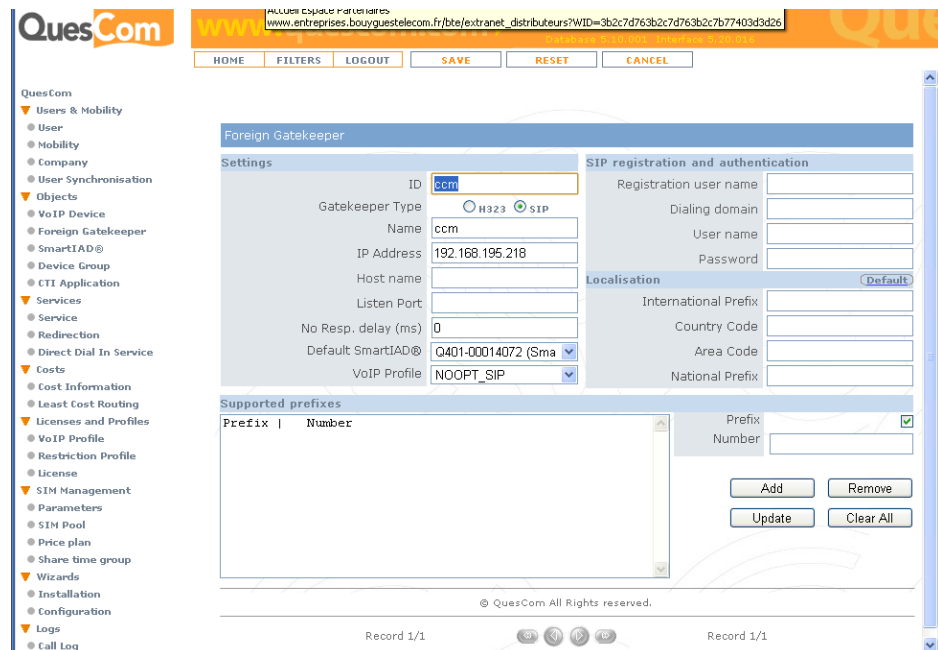
## Define adequate Route pattern



## 4.2. QuesCom Gateway configuration

This example is based on QuesCom gateway v6.20. Please note that presentation differs from the previous example of H.323 trunk configuration which is based on QuesCom v6.20 is used.

### Define SIP Trunk with CCM



## Create a service from CCMH323 \* to GSM Pool

The screenshot displays the QuesCom web interface for configuring a service. The page title is "Service" and the URL is "www.quescom.com". The interface includes a navigation menu on the left and a main configuration area with several sections:

- Service:** Service enabled
- Origin:** Origin Type:  Device  Foreign GK  CTI  Device Group. Origin: ccm. Called Prefix Number: [empty]. Call Type: Foreign Gatekeeper. Enabled for:  H323  QGP  SIP. Fax detection:
- Destination:** Destination Type:  IP Address (H323)  Foreign GK  CTI  Device Group. IP Address: [empty]. Device: GSM\_POOL (SmartIAD). Balancing mode:  None  Bal.  Cycling  Sim. Called number lookup:  Local  Nquire  External.
- Service associated:** Service type: GSM. Authentication Type: None. Called Number Type: Dialed Number. Voice Fax Mode:  Switch  VoIP  CTI Application.
- Call Server Operations:** LCR Support: None. CDR Support: Yes. Cost Support: No.
- VoIP Service:** Law Transcoding: . Quality of service (HEX): Min Delay (10).
- Switch service:** Slot:  PRI  GSM.  Any  Slot Balancing . Port Number: [grid of checkboxes].  Any  Port Balancing . IVR Profile: [empty]. IVR Called Number: [empty].
- Backup Mode:** Enabled for:  DSP  Relay  RSVP  Other.
- Fax /Voice Service:** Voice Fax Type: Store & Forward. Store & Forward Type: FAX TO EMAIL. Called Number: [empty]. Notify Receipt Type: None. Notify receipt to: [empty]. Send To: [empty]. Acknowledge Type: None. Acknowledge to: [empty]. Max Duration (s): 0. Stop on DTMF: No.

## Create a service top route Incoming GSM to CCM Trunk

QuesCom [www.quescom.com](http://www.quescom.com)

HOME
FILTERS
LOGOUT
SAVE
RESET
CANCEL

**QuesCom**

- ▼ Users & Mobility
  - User
  - Mobility
  - Company
  - User Synchronisation
- ▼ Objects
  - VoIP Device
  - Foreign Gatekeeper
  - SmartIAD@
  - Device Group
  - CTI Application
- ▼ Services
  - Service
  - Redirection
  - Direct Dial In Service
- ▼ Costs
  - Cost Information
  - Least Cost Routing
- ▼ Licenses and Profiles
  - VoIP Profile
  - Restriction Profile
  - License
- ▼ SIM Management
  - Parameters
  - SIM Pool
  - Price plan
  - Share time group
- ▼ Wizards
  - Installation
  - Configuration
- ▼ Logs
  - Call Log
  - Call Failure Log
  - Fax Log
  - SMS Log
  - Last Caller

**Service**

Service enabled

**Origin**

Origin Type  Device  Foreign GK  CTI

Device Group: Q401-00014072 (SmartIAD)

Called Prefix Number: \*

Call Type: GSM Incoming

Enabled for:  H323  QGP  SIP

Fax detection:

**Destination**

Destination Type:  IP Address (H323)  Device  Foreign GK  CTI  Device Group

IP Address: \_\_\_\_\_

Device: ccm

Balancing mode:  None  Bal.  Cycling  Sim

Called number lookup:  Local  Nquire  External

**Service associated**

Service type: VoIP

Authentication Type: None

Called Number Type: ISDN

Voice Fax Mode:  Switch  VoIP  CTI Application

**Call Server Operations**

LCR Support: No

CDR Support: Yes

Cost Support: No

**VoIP Service**

Law Transcoding:

Quality of service(HEX): Min Delay (10)

**Switch service**

PRI PRI GSM

Slot:  Any  Slot Balancing

Port Number: 0 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15

Any  Port Balancing

IVR Profile: \_\_\_\_\_

IVR Called Number: \_\_\_\_\_

**Backup Mode**

Enabled for:  DSP  Relay  RSVP  Other

**Fax /Voice Service**

Voice Fax Type: Store & Forward

Store & Forward Type: FAX TO EMAIL

Called Number: \_\_\_\_\_

Notify Receipt Type: None

Notify receipt to: \_\_\_\_\_

Send To: \_\_\_\_\_

Acknowledge Type: None


Acknowledge to: \_\_\_\_\_

Max Duration (s): 120

Stop on DTMF: No

this tree only expands or contra

www.quescom.com

  
**ONE NUMBER**  
 +33 [0]4 97 23 48 48