

Call Recording Announcements

Imagicle Call Recording provides an exclusive approach to play announcements for incoming/outgoing calls, both internal and external. It offers two unique methods for different recording technologies, leveraging CURRI protocol (Cisco UCM External Call Control - ECC) or TAPI-based Cisco Agent Greeting feature, without the need of any additional application.

In the following paragraphs, both methods are explained, with advices on correct usage depending on your Cisco/Imagicle environment.

Agent Greeting-based Announcement

This is the most recent recording announcement method for delivering announcements on any TAPI-enabled phone device. It offers the option to play an announcement for both incoming and outgoing calls, internal and external, and it is compatible with Media Forking, SIPREC and automated/manual Dial-In Conference recording methods.

Requirements

- Imagicle ApplicationSuite ver. 2019.Summer.1 or higher
- CUCM ver. 8.6 or higher
- Cisco TSP (see [here](#)) and relevant Application User (see [here](#))
- IP Phones should support Built-in Bridge
- IP phones to be enabled for recording announcement should be monitored via TAPI

Limitations

Currently, Agent Greeting announcements method does not work on Jabber Desktop clients older than ver. 12.9. The announcement is actually played, but it is heard by local party only.

How it works

As soon as the call recording starts, if the recording phone is TAPI controlled, the application automatically triggers a call from the recording IP phone to a particular destination number, using the built-in bridge "agent greeting" feature.

Such destination number is composed by:

- a fixed prefix, that is the configured announcement pilot number (for instance 8600)
- further 4 digits randomly generated by the application.

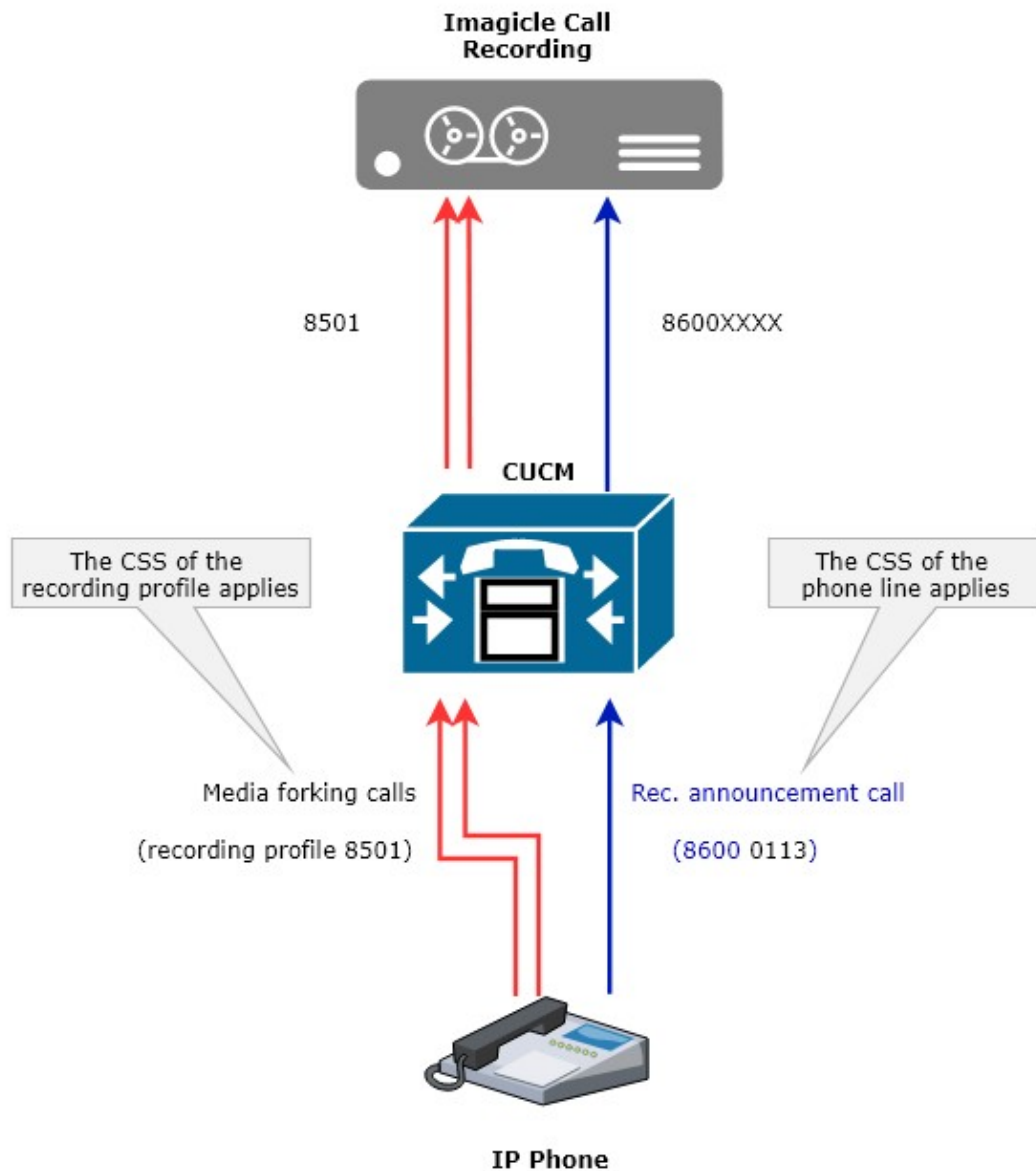
The resulting number (for instance 86000113) must be routed by the CUCM to the Imagicle Call Recording SIP Trunk.

The call recorder answers such incoming call and plays the configured recording announcement, that is heard both by the recording party and the recorded party.

Please, notice that:

- If the built-in bridge media forking recording method is adopted, a total of 3 simultaneous calls will be established between the phone and the Imagicle Call Recording: the third one (announcement call) will be automatically disconnected as soon as the announcement has been played.
- For media forking calls, the recording profile CSS applies.
- For the announcement call, the regular phone line CSS applies.

The following diagram describes such mechanism in the case of built-in bridge call recording.



IAS Configuration for External Calls Announcement

You can enable this functionality from Imagicle web portal: **Call Recording** ⌵ **Global Settings** ⌵ **Announcement**

See a screenshot sample below:

Settings	Data Management	Notifications	Announcements
Pilot If you need to use the Call Recording announcements feature, please configure here the dedicated service pilot number. Announcement pilot (prefix) <input type="text" value="8600"/> ⓘ A rule in the CUCM must be defined to route all calls matching this prefix to the Imagicle Call Recording SIP trunk. This is the pattern (a fixed length pattern) you need to define accordingly to the pilot configured in the section above: <div style="border: 1px solid #ccc; padding: 5px; background-color: #f9f9f9;">8600XXXX</div>			
Messages Configure the announcement messages that will be played when the recording starts. Different behaviors can be set according to the call direction and type. Chosen messages will be heard by all call parties. <div> External incoming calls <div> <input type="text" value="Default message"/> <input type="button" value="BuiltIn - External incoming."/> </div> </div> <div> External outgoing calls <div> <input type="text" value="Custom message"/> <input type="button" value="ThisCallIsBeingRecorded.wa"/> </div> </div> <div style="text-align: right; margin-top: 20px;"> <input type="button" value="Save"/> <input type="button" value="Cancel"/> </div>			

Announcement pilot (prefix) field should be populated with an unused DN range (10,000 numbers), which corresponds to the pilot number involved to get the announcement prompt. Once entered, same web page shows below the relevant Route Pattern to be defined in CuCM to route calls to IAS through Call Recording SIP Trunk.

External incoming/outgoing calls fields allow to choose which voice prompt to be played when Call recording is triggered. Available options are:

- *No message* â Announcement is disabled. If custom messages have been previously uploaded in the server, they will be kept for future usage.
- *Default message* â A default, factory-loaded announcement is enabled, in current IAS installation language. If custom message have been previously uploaded in the server, it will be deleted.
- *Custom message* â Custom recording announcement is enabled. You can upload any MP3 or WAV audio file.

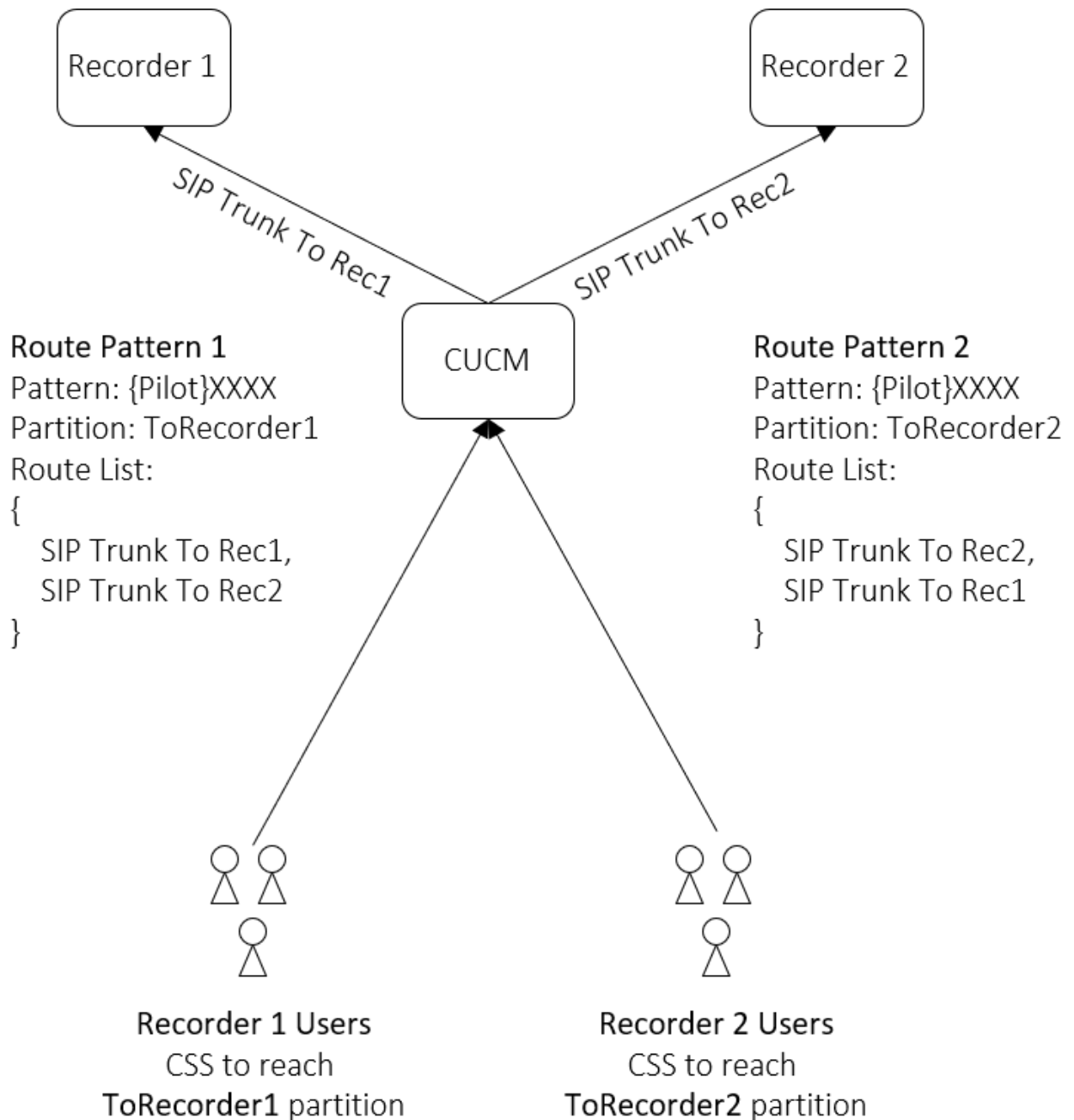
Advanced configuration - Load balancing the call announcement traffic

In a high availability configuration involving mutiple Imagicle servers, the recording announcement should normally be provided by the same Imagicle node that records the call. Therefore, the announcement call routing should follow the same node selection policy of the recording profile, that is, a route list with an active-standby selection.

However, if you deal with an Imagicle cluster and you need to balance the announcement calls among the available nodes, this is still possible. In order to do this, you need to:

- 1) create additional Route Patterns for the same Announcement Pilot pattern (for instance 8600XXXX), on different partitions. Each route pattern shuld point to a specific **Route List**, including one or more SIP Trunk(s) toward different recording nodes, ordered with chosen priority.
- 2) Create and assign different CSS to different phones in order to trigger different announcement route patterns.

See the below sample diagram:



This kind of phones partition allows to balance the overall load due to call recording announcements,

IAS Configuration for Internal Calls Announcement

Enabling Call Recording announcement for internal calls is not a common case therefore it requires an advanced configuration on the Imagicle server.

Please, access to Imagicle server using Remote Desktop and edit this file:

`C:\Program Files (x86)\StonevoiceAS\Apps\Recorder\Settings\Recorder.ini`

Set the following parameter under the [Settings] section:

`PlayAnnouncementForInternalCalls = 1`



Then save the file.

This will play a default (built-in) announcement.

If you want to customize the voice message, open this folder:

```
C:\Program Files (x86)\StonevoiceAS\Apps\Recorder\Data\AudioFiles\User
```

and copy into such folder the announcement audio file for internal calls, that must have the following file format:

- **WAV file**
- **8KHz mono sample rate**
- **G.711 A-Law or Mu-Law**

Once copied, please rename the wav file appending in front of the original filename the prefix "8049F0AA-B96C-45E3-8F29-CB9C014B4133_", so that the final filename is something like:

*8049F0AA-B96C-45E3-8F29-CB9C014B4133_***your file name**.wav

ATTENTION:

- Only one audio file with above name format is allowed in *AudioFile/User* folder.
- When a call is established between 2 phones both configured for Call Recording, the voice announcement will be played twice (one for each involved phone). This may lead to have an annoying echo in the first part of the recording, due to the voice announcement overlap.

IAS Configuration for unknown call type or unknown call direction

In those special cases where it's not possible to discern internal/external call type or incoming/outgoing call direction, Imagicle Call Recording can still play a specific announcement. This is typical of below call scenarios:

- SIPREC-based recordings, where Cisco Voice Gateway/CUBE configuration is somehow incorrect
- Media Forking-based recordings, where calls is routed through multiple CuCM nodes or through Cisco UCCX, where DN's are not TAPI-monitorable
- Manual Dial-in Conference recordings

To enable announcement in this particular scenario, you need to access Imagicle ApplicationSuite file system and amend a configuration file.

Please access to Imagicle server via RDP and edit this file:

```
C:\Program Files (x86)\StonevoiceAS\Apps\Recorder\Settings\Recorder.ini
```

Set the following parameter:

```
PlayAnnouncementForUnknownDirectionOrTypeCalls=1
```

This will play a default (built-in) announcement.

If you want to customize the voice message, open this folder:

```
C:\Program Files (x86)\StonevoiceAS\Apps\Recorder\Data\AudioFiles\User
```

Copy into such folder the announcement audio file for internal calls, that must have the following file format:

- **WAV file**
- **8KHz mono sample rate**
- **G.711 A-Law or Mu-Law**

Once copied, please rename the wav file appending in front of the original filename the prefix "4C504392-CC9F-45BD-B0C4-22BC4C46E862_", so that the final filename is something like:

*4C504392-CC9F-45BD-B0C4-22BC4C46E862_***your file name**.wav

ATTENTION:

- Only one audio file with above name format is allowed in *AudioFile/User* folder. If multiple files have the same prefix, the result is unpredictable.

CURRI-based Announcement

This is the suggested recording announcement method for delivering announcements on Jabber clients prior to ver. 12.9, where other method is not supported. It offers the option to play an announcement for incoming calls only.

Requirements

- Imagicle ApplicationSuite ver. 2016.Summer.1 or higher
- CUCM ver. 8.1.6 or higher
- PSTN provider should support "early audio cut-through", allowing to open audio streams before actual call establishment.

CUCM Configuration

1. Access to CUCM "Cisco Unified Serviceability" web portal and select Tools > Service Activation
2. Make sure that "Cisco IP Voice Media Streaming App" is **Activated**

CM Services		
	Service Name	Activation Status
<input checked="" type="checkbox"/>	Cisco CallManager	Activated
<input type="checkbox"/>	Cisco Unified Mobile Voice Access Service	Deactivated
<input checked="" type="checkbox"/>	Cisco IP Voice Media Streaming App	Activated
<input checked="" type="checkbox"/>	Cisco CTIManager	Activated
<input type="checkbox"/>	Cisco Extension Mobility	Deactivated
<input type="checkbox"/>	Cisco Extended Functionality	Deactivated

3. Access to CUCM "Cisco Unified CM Administration" web portal and go to Media Resources > Announcement

The screenshot shows the Cisco Unified CM Administration web portal. The top navigation bar includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The 'Media Resources' menu is expanded, showing a list of options including Announcer, Conference Bridge, Media Termination Point, Music On Hold Audio Source, Fixed MOH Audio Source, Music On Hold Server, Video On Hold Server, Transcoder, Media Resource Group, Media Resource Group List, MOH Audio File Management, Mobile Voice Access, and Announcement. The 'Announcement' option is highlighted with a red box. Below the menu, a table lists various announcements with their identifiers and descriptions. The table has columns for 'Announcement Identifier' and 'Description'.

Announcement Identifier	Description
System- Gone	System- Gone
System- MLPP Busy not equipped	System- MLPP Busy not equipped
System- MLPP Higher precedence	System- MLPP Higher precedence
System- MLPP Service disruption	System- MLPP Service disruption
System- MLPP Precedence access limit	System- MLPP Precedence access limit
System- MLPP Unauthorized precedence	System- MLPP Unauthorized precedence
System- Monitoring or Recording	System- Monitoring or Recording
System- Recording	System- Recording
System- Temporarily unavailable	System- Temporarily unavailable
System- Vacant number / invalid number dialed	System- Vacant number / invalid number dialed
System- Sample queued caller periodic announcement	System- Sample queued caller periodic announcement
System- Sample caller greeting	System- Sample caller greeting

4. Add New Announcement

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Announcement Configuration

Save

Status

Status: Ready

Announcement

Announcement Identifier*

Description

Default Announcement

Save

5. Save
6. Upload Audio File and select Locale

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Announcement Configuration

Save Delete Add New Upload File

Status

Status: Ready

Announcement

Announcement Identifier*

Description

Default Announcement

Save Delete Add New Upload File

*- indicates required item.

Upload File

Status: Ready

Announcement Identifier*

Locale*

Upload File

Upload File Close

*- indicates required item.

7. Add a new External Call Control Profile (ECCP)

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Find and Add

Add New

Status

15

External

Find External

Add New

Delete Selected

*- indicates required item.

Find External	Primary Web Service	Secondary Web Service	Copy
AAR Group			
Dial Rules			
Route Filter			
Route/Port			
SIP Route Pattern			
Intercom			
Class of Control			
Client Matter Codes			
Forced Authorization Codes			
Translation Pattern			
Call Park			
Directed Call Park			
Call Pickup Group			
Directory Number			
Dial Plan Installer			
Meet-Me Number/Pattern			
Route Plan Report			
Transformation			
Mobility			
Logical Partition Policy Configuration			
External Call Control Profile			
HTTP Profile			
Call Control Discovery			
Global Dial Plan Replication			
Speedy ECC 204_191			
Speedy ECC Stefano			
StoneLock ECC Stefano			

Add New Select All Clear All Delete Selected

8. Configure URL to reach Imagicle Application Suite

`http://<Imagicle_IP>:80/fw/ecc.ashx?recordingPrompt=prompt_name`

example

`http://192.168.10.10:80/fw/ecc.ashx?recordingPrompt=Recording_Announcement`

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes links for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main heading is 'External Call Control Profile Configuration'. Below this, there are icons for Save, Delete, Copy, and Add New. The 'Status' section shows 'Status: Ready'. The 'External Call Control Information' section contains the following fields:

- Name*: CURRI_Recording_TEST
- Primary Web Service*: `http://192.168.108.200:80/fw/ecc.ashx?recordingPrompt=Recording_Announcement` (highlighted with a red box)
- Secondary Web Service: (empty)
- ☐ Enable Load Balancing
- Routing Request Timer: 5000
- Diversion Rerouting Calling Search Space: < None >
- Call Treatment on Failures*: Allow Calls

At the bottom, there are buttons for Save, Delete, Copy, and Add New, and a note: '*- indicates required item.'


9. Save

10. Configure Trigger Points

Available Trigger Points in CuCM are:

- Translation Pattern trigger points are available in Unified CM 8.0(1) and later
- Route Patterns and Directory Numbers are trigger points in Unified CM 10.0 and later





Enable ECC profile in Translation Pattern


Cisco Unified CM Administration
 For Cisco Unified Communications Solutions


Navigation **Cisco Unified CM Administration**
appadmin | [Search Documentation](#)

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾

Translation Pattern Configuration
Related Links: [Back To Find/List](#) [Go](#)

 Save
  Delete
  Copy
  Add New

Status

 Status: Ready

Pattern Definition

Translation Pattern	<input type="text" value="0."/>
Partition	<input style="border: none;" type="text" value=" < None > "/>
Description	<input type="text" value="OUTGOING"/>
Numbering Plan	<input style="border: none;" type="text" value=" < None > "/>
Route Filter	<input style="border: none;" type="text" value=" < None > "/>
MLPP Precedence*	<input style="border: none;" type="text" value="Routine"/>
Resource Priority Namespace Network Domain	<input style="border: none;" type="text" value=" < None > "/>
Route Class*	<input style="border: none;" type="text" value="Default"/>
Calling Search Space	<input style="border: none;" type="text" value="ALL_IP_PHONES"/>
External Call Control Profile	<input style="border: none;" type="text" value="Imagicle Ecc-curri"/>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input style="border: none;" type="text" value="No Error"/>
<input checked="" type="checkbox"/> Provide Outside Dial Tone <input checked="" type="checkbox"/> Urgent Priority <input type="checkbox"/> Route Next Hop By Calling Party Number	

Enable ECC profile in Route Pattern Trigger Point (In Unified CM 10.0 and later)

Route Pattern Configuration

Save

Delete

Copy

Add New

Status: Ready

Route Pattern*

1XXX

Route Partition

< None >

Description

Numbering Plan

-- Not Selected --

Route Filter

< None >

MLPP Precedence*

Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

< None >

Route Class*

Default

Gateway/Route List*

SIPT-58212

(Edit)

Route Option

☒ Route this pattern
☐ Block this pattern

No Error

Call Classification*

OffNet

External Call Control Profile

< None >

Enable ECC profile in Directory Number Trigger Point (In Unified CM 10.0 and later)

Directory Number Configuration

Save

Delete

Reset

Apply Config

Add New

Status: Ready

Directory Number*

3009

Route Partition

< None >

Description

Alerting Name

ASCII Alerting Name

External Call Control Profile

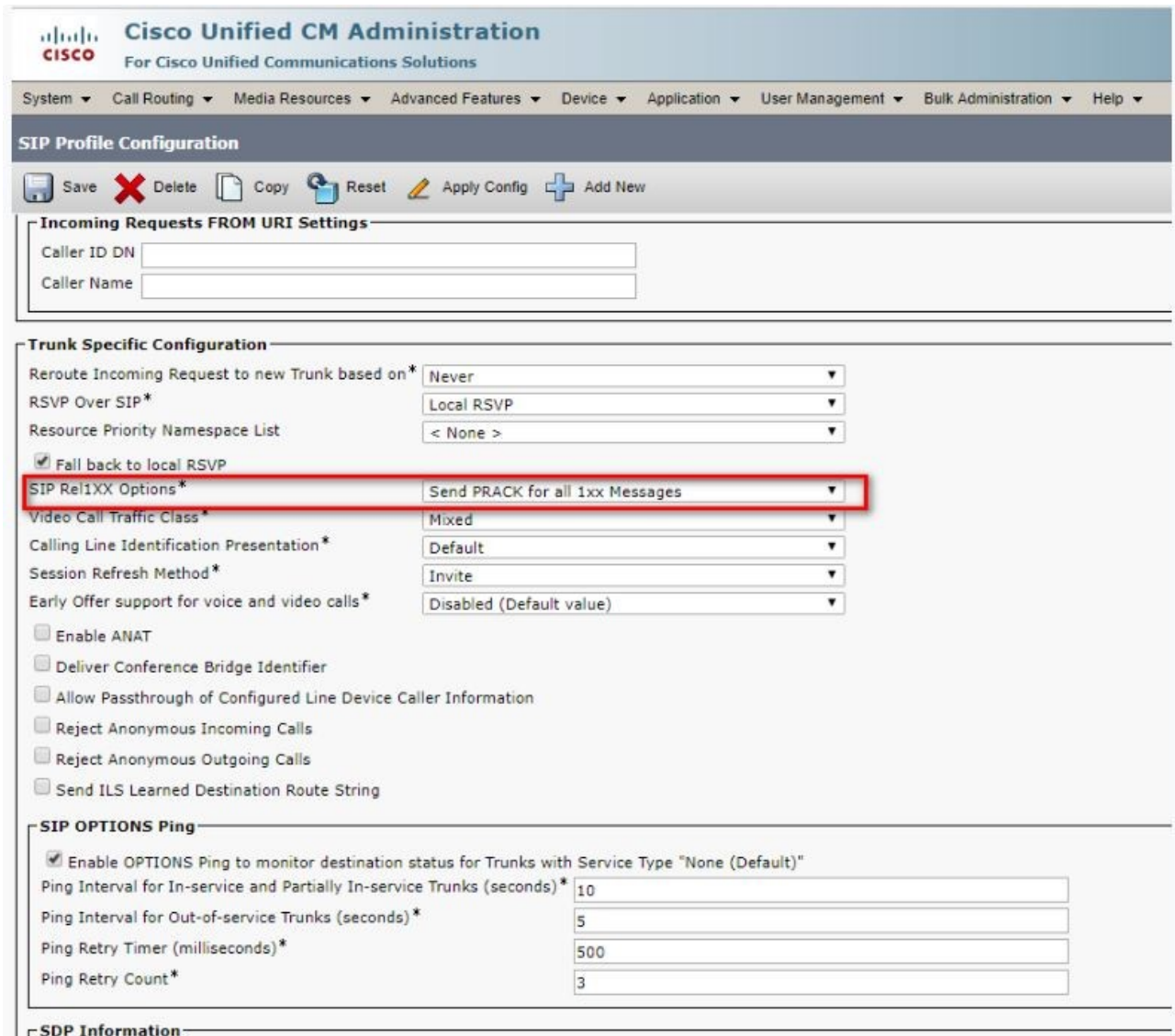
< None >

Voice Gateway Configuration

More information available in Cisco documentation [here](#)

Voicegateway needs to support **SIP Early Media**

1. Enable SIP PRACK



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

SIP Profile Configuration

Save ✖ Delete Copy Reset Apply Config + Add New

Incoming Requests FROM URI Settings

Caller ID DN
Caller Name

Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on* Never ▾
 RSVP Over SIP* Local RSVP ▾
 Resource Priority Namespace List < None > ▾
☒ Fall back to local RSVP
SIP Rel1XX Options* Send PRACK for all 1xx Messages ▾
 Video Call Traffic Class* Mixed ▾
 Calling Line Identification Presentation* Default ▾
 Session Refresh Method* Invite ▾
 Early Offer support for voice and video calls* Disabled (Default value) ▾
☐ Enable ANAT
☐ Deliver Conference Bridge Identifier
☐ Allow Passthrough of Configured Line Device Caller Information
☐ Reject Anonymous Incoming Calls
☐ Reject Anonymous Outgoing Calls
☐ Send ILS Learned Destination Route String

SIP OPTIONS Ping

☒ Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"
 Ping Interval for In-service and Partially In-service Trunks (seconds)* 10
 Ping Interval for Out-of-service Trunks (seconds)* 5
 Ping Retry Timer (milliseconds)* 500
 Ping Retry Count* 3

SDP Information

2. In the Voicegateway configuration add:

In case of a SIP Voice Gateway

GLOBAL CONFIGURATION

```
voice service voip
sip
rel1xx require 100rel
```

OR at single dial peer:



DIAL-PEER CONFIGURATION

```
dial-peer voice 1000 voip
voice-class sip rel1xx require 100rel
```

In case of a H.323 Voice Gateway VG, No configuration needed.