

PBX Configuration

This page describes how to configure Cisco Unified CallManager with Imagicle Call Recording.

CuCM offers many ways to record calls: always on, on demand, media forking, network based... The configuration differs slightly depending on the recording mode. Regardless the recording mode, you need to create a SIP Trunk for Imagicle Call Recording.

SIP Trunk configuration

1. Create a new **SIP Trunk Security Profile** named "Imagicle Call Recording SIP Security Profile" with following settings.

- Incoming Transport Type: **TCP + UDP**
- Outgoing Transport Type: **TCP**
- Incoming Port: **5070**
- Enable Digest Authentication: **disabled**
- Enable Application Authorization: **disabled**
- Accept Unsolicited Notification: **enabled**
- Accept Replaces Header: **enabled**

Leave all other settings to their default value.

2. Define a new **SIP Profile** named "Imagicle Call Recording SIP Profile", with following settings:

- Timer Invite Expires: **5**
- Retry INVITE: **1**
- SIP OPTIONS PING - Enable OPTIONS Ping: **Enabled**
- SIP OPTIONS PING - Ping interval for In- service Trunks: **10**
- SIP OPTIONS PING - Ping interval for Out-of-service Trunks: **5**
- SIP OPTIONS PING - Ping Retry Timer: **500**
- SIP OPTIONS PING - Ping Retry Count: **3**

Leave all other settings to their default value.

3. Create a new **SIP Trunk** named "Imagicle Call Recording Primary SIP Trunk" with following settings:

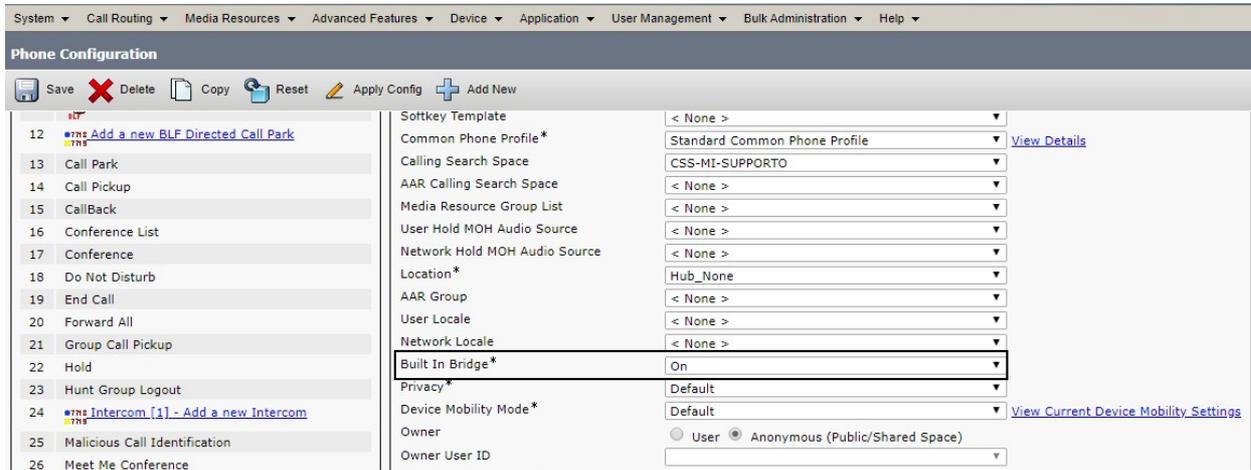
- Device name: **Imagicle_CallRecording_Primary_SIP_Trunk**
- Call Classification: **On-Net**
- Media Resource Group List: if you need to record conversations using a voice codec different from G.711 and G.729A, you need to assign a Media Resource Group List that includes at least one **hardware transcoding resource**.
- Run on all Active Unified CM Nodes: **Enabled**
- SIP Information - Destination Address: **<Imagicle server IP address>**
- SIP Information - Destination port: **5070**
- SIP Trunk Security Profile: **Imagicle Call Recording SIP Security Profile** (see above)
- SIP Profile: **Imagicle Call Recording SIP Profile** (see above)
- DTMF Signalling Method: **RFC2833**

Leave all other settings to their default value.

Configuration for Media Forking (Built-in-Bridge) Recording mode

Enable Built-in-Bridge in Phone Configuration

In the CallManager Phone Configuration **set Built-in-Bridge On** for Call Recording



Disabling unsupported codecs

Imagicle Call Recording supports G.711 and G.729A only. All other codecs must be disabled from CuCM configuration.

G.729B should be disabled at system level for all the calls, G.722 can be disabled for recorded calls only.

If the unsupported codecs can't be disabled, you must ensure to assign hardware transcoding resources to the Call Recording SIP Trunk (in the Media Resource Group List option). Otherwise, Imagicle Call Recording won't be able to record the calls that are established with an unsupported codec.

In CuCM Administration, select System, then Service Parameters. Choose your server, then "Cisco CallManager".

In the **CallManager Service Parameters** page, disable the following codecs, choosing the option "Enabled for All Devices Except Recording-Enabled devices" (CuCM 9.1 and higher) or "Disabled" (CuCM 8.x):

- G.722
- iLBC
- iSAC
- Opus

G.711 A-law Codec Enabled *	Enabled for All Devices
G.711 mu-law Codec Enabled *	Enabled for All Devices
G.722 Codec Enabled *	Enabled for All Devices Except Recording-Enabled Devic
iLBC Codec Enabled *	Enabled for All Devices Except Recording-Enabled Devic
iSAC Codec Enabled *	Enabled for All Devices Except Recording-Enabled Devic

Starting CUCM version 12, the following service parameter needs to be disabled for recorded devices only:

G.722.1 and G.722.2 Codecs Enabled *	Enabled for All Devices Except Recording-Enabled Devic
--	--

For the same reason, in the same page, remove the "G.729 Annex B (Silence Suppression) from Capabilities", by setting the parameter to **True**:

Strip G.729 Annex B (Silence Suppression) from Capabilities *	True
---	------

Alternatively, to avoid G.729B communication only on recorded phones, you can:

- Define in CuCM a **codec preference list** that includes G.729A **before** G.729.B
- Assign such codec preference list to a region used by the phones enabled for recording

How to provide a recording tone

If you want to provide the background recording tone (every 15 seconds) to the recorded parties, in the CuCM administration set the following Service Parameters for CallManager service (on all CallManager nodes).

In CuCM Administration, select System, then Service Parameters. Choose your server, then "Cisco CallManager". Locate the **Clusterwide Parameters (Feature - Call recording)** group. Set the following values:

- Play Recording Notification Tone To Observed Target: **True**
- Play Recording Notification Tone To Observed Connected Parties: **True**

The first one enables the recording tone for the agent, the second one enables the recording tone for the remote party.

Audio files format

Prior to 2021.Summer.1 release, all BiB recordings are save in Mono format. From 2021.Summer.1 onward, recorded conversations are save in Stereo format, having local party on left channel and remote party(ies) on right channel. If you need to revert this format to Mono, please contact Imagicle Tech Support.

Enable phones for recording

To leverage the recording capabilities of CuCM, some additional configurations are needed on CCM.

1. Associate the phones to the ImagicleCTI user you created. This gives several advantages and functionalities. In particular, thanks to CTI/TAPI monitoring is possible to:

- start and stop the recordings directly from the XML Phone Service, from Jabber Desktop gadget or Cisco Finesse gadget
- automate the dial-in conference calls; compared to Manual dial-in, the automated allows to resolve the remote party and call direction.

In order to be controlled via CTI correctly, please follow the procedure available [here](#).

2. Create a new **Recording Profile** (Device - Device Settings - Recording Profiles) with following settings:

- Name: **Imagicle Call Recording Profile**
- Recording CSS: **A CSS able to engage the Route Pattern described below.**
- Recording Destination Address.: **An internal, unused service phone number that will trigger the Route Pattern below defined (e.g. 8500)**

Recording Profile Configuration

Save

Status

Status: Ready

Recording Profile Information

Name*

Recording Calling Search Space

Recording Destination Address*

Save

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3. Create a **Route Pattern** to route the media forking calls to the Imagicle Call Recording SIP trunk. The pattern must match the pattern defined in the "Recording Destination Address" box of the Recording Profile.

4. Set the following settings to each **phone line** to be recorded:

- Recording Option: choose one of the following options
 - ◆ **Automatic Call Recording Enabled**, for *Always On* mode
 - ◆ **Selective Call Recording Enabled**, for *On-Demand* mode*
- Recording Profile: **Imagicle Call Recording Profile**
- Recording Media Source:
 - ◆ **Phone preferred**: if you want to use the built-in bridge recording mechanism for this phone
 - ◆ **Gateway preferred**: if you want to use the Network recording mechanism for this phone.

If you are configuring *Always On* recording mode with phone **Phone preferred**, the configuration is complete. If you are configuring *Always On* recording mode with phone **Gateway preferred**, please skip the following "Recording Control" paragraph and head directly to paragraph "Configuration for Network Based Recording mode (Gateway Recording) UCM 10.X and higher" to complete the required configuration.

If the **Extension Mobility** feature is used, you must enable for recording the line of the associated **Device Profile**.

Note: *Selective Call Recording is available since CUCM 9.1 only for some ip phone models. See the official Cisco documentation for the [list of supported phones](#) and recording modes.

Recording Control

If you are configuring the *On Demand* recording mode, there are some additional configuration steps to be performed in order to configure phone(s) to be able to invoke the recording. Imagicle Suite supports:

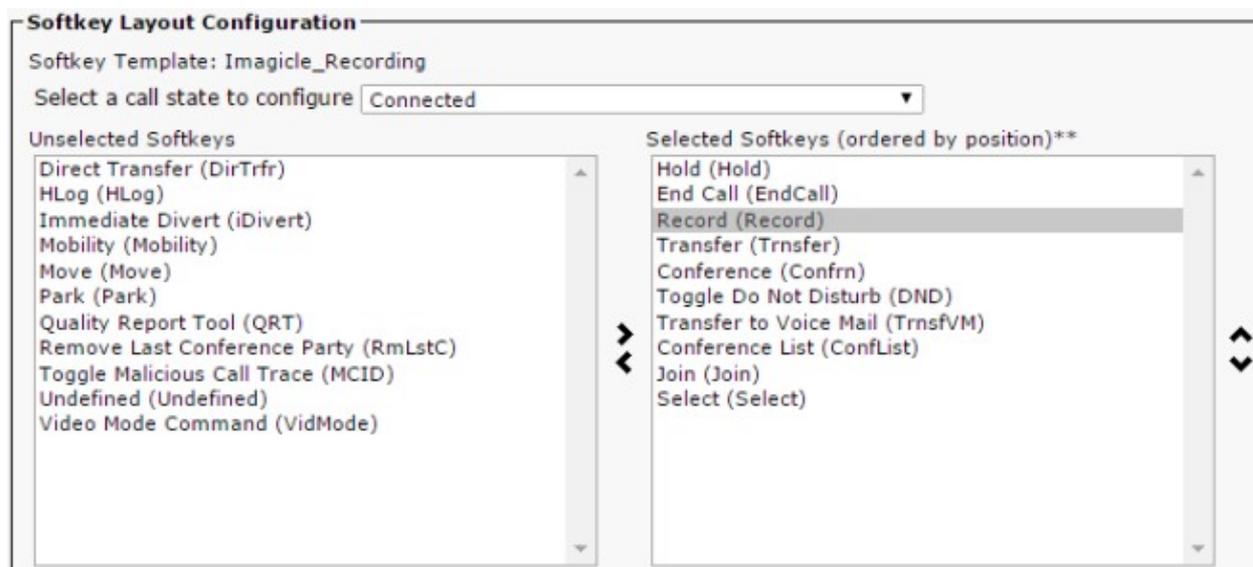
1. Standard Recording Control, with start and stop commands
2. Advanced Recording Control, with start, stop, pause and resume commands

4.1 Standard Recording Control (Start-stop) for On Demand Recording

Cisco Unified CallManager provides a build in mechanism to start the recording of a call. Basing on the phone model, you can use a SoftKey Template or a Programmable Line Key.

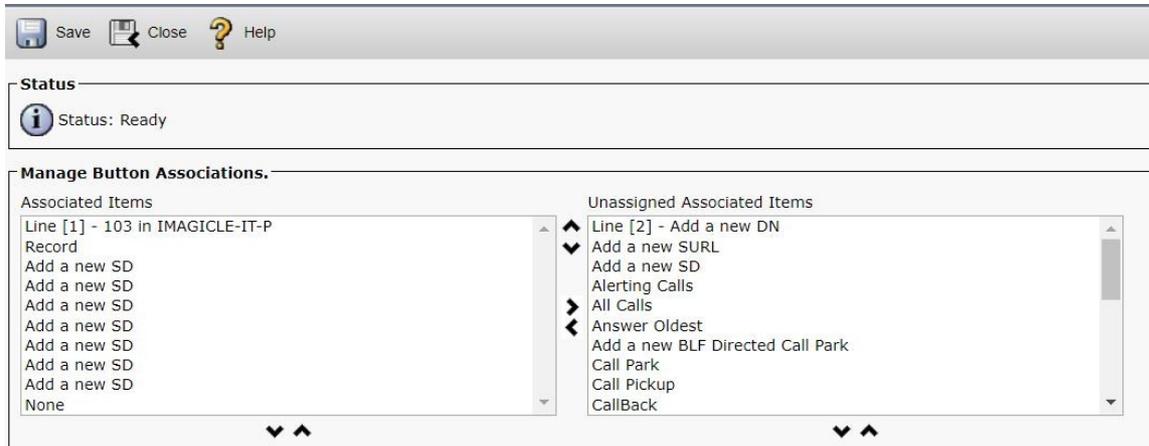
a) Adding Recording button to phones supporting **SoftKey Template** (i.e. IPC, Cisco 68xx,78xx,79xx IP Phones, ...)

Create a new **Softkey Template** (Device -> Device Settings - Softkey Templates) that includes the "Record" softkey for the *Connected* status. You can also simply add such softkey to your existing Softkey Template.



b) Adding Recording button to phones supporting **Button Template** (i.e. Cisco 99xx,88xx,89xx IP Phones, ...).

You need to configure the recording button into the Button Template, so it appears as a PLK (Programmable Line Key) on the left-hand side of the phone. From the **Device, Phone** menu, select the device you want to configure, and the click **Modify button items** on top of the button/PLK list to the right. Look for the "Record" entry among the "Unassigned Associated Items" and move it to "Associated Items" as in the screenshot below.



4.2 Advanced Recording Control (Start-stop & Pause-Resume) for On Demand Recording

Imagicle Application Suite 2018.3.1 (Spring edition) and later also support **pause** and **resume** for recordings. If pausing recordings is not a required use case, please refer to chapter 4.1 for the phones configuration alternatives. To enable pausing a recording, phones require some specific configuration.

1) Create a new Phone Service (XML)

In CuCm Web interface, go to Device, Device Settings, Phone Services. Add a new phone services with the following settings:

- Service Name: something meaningful, e.g. **Record**
- Service URL:
http://<IMAGICLE_APPLICATION_SUITE_IP_ADDRESS>/fw/Apps/Recorder/xml/OnDemand/MediaForking.aspx?name=#DEVICENAME
- Service Type: **XML Service**
- Enable: **on**

IP Phone Services Configuration

Save
 Delete
 Update Subscriptions
 Add New

Status

Status: Ready

Service Information

Service Name*	<input type="text" value="Record"/>
Service Description	<input type="text" value="Record"/>
Service URL*	<input type="text" value="http://192.168.4.35/fw/Apps/Recorder/xml/OnDemand/Me"/>
Secure-Service URL	<input type="text"/>
Service Category*	<input type="text" value="Servizio XML"/>
Service Type*	<input type="text" value="Servizio IP Phone standard"/>
Service Vendor	<input type="text" value="Imagicle"/>
Service Version	<input type="text" value="2018.2.1"/>

Enable

Note: you can copy the URL from your AppSuite web interface, from the Global Settings page, Services URL section. Ensure you copy the Media Forking URL.

- 2) Subscribe the phones you want to enable to recording, to the XML phone service you just created. You'll be able to use the new service pressing the "Services" button on the IP phone
- 3) Optionally, add the service as a new Service URL button on the phone. See the procedure at the bottom of this page.
- 4) By default, only the Start Recording option is available. To allow the users to Stop, Pause and Resume recordings, you need to change the default settings on Imagicle Application Suite. Log onto the web interface. In the Call Recording menu, select **Global Settings**. Expand the **Recording Control** section and set the relevant options.

You can also set the options per device, overriding the global setting, through the use of XML Service parameters. The procedure is detailed in the FAQ section of this guide.

Configuration for Network Based Recording mode (Gateway Recording) UCM 10.X and higher

Cisco Network Based Recording (also known as Gateway Recording) is available since UCM 10.0 and has some additional requirements if compared to the Built-In Bridge phone based recording described so far. Specific configuration is required in all the routers/CUBE's that relay incoming/outgoing calls.

Hardware and Software Requirements

Accordingly with [Cisco documentation](#), network based recording requires:

- UCM 10.0 or higher
- Supports both Voice Gateways and Unified Border Elements (CUBE) as long as they interface with Unified CM **using SIP** and the Router platform supports the **UC Services Interface** (not supported for H.323 or MGCP based calls). The word "Gateway" is used here interchangeably to refer to Voice gateways and CUBE devices
- The Gateway has to be directly connected to the Unified CM using a SIP trunk. **No support for SIP Proxy servers**
- ISR-G2 Gateways (29XX, 39XX Series) running release 15.3(3)M or later are supported. 15.3(3)M was released on CCO in July / 2013
- ASR-100X Gateways running release XE 3.10 or later are supported. XE 3.10 was released on CCO in July / 2013

- VG224 is not currently supported

Audio files format

Prior to 2021.Summer.1 release, all NBR recordings are save in Mono format. From 2021.Summer.1 onward, recorded conversations are save in Stereo format, having local party on left channel and remote party(ies) on right channel. If you need to revert this format to Mono, please contact Imagicle Tech Support.

IOS Configuration

The following IOS configuration must be applied to each CUBE/VG enabled for network based recording.

```
uc wsapi
message-exchange max-failures 100
response-timeout 1
source-address 192.168.204.52 << replace this with the actual voice LAN router IP address
probing interval negative 20
probing interval keepalive 255

provider xmf
remote-url 1 http://192.168.100.115/ucm_xmf << Create a similar URL for each CallManager node running the SIP trunk
remote-url 2 http://192.168.100.115:8090/ucm_xmf << Create a similar URL for each CallManager node running the SIP
```

Then, check the registration is successfully, by invoking the command:

```
show wsapi registration all
```

The following command allows debugging the call media forking when call recording is triggered:

```
show call media-forking
```

Ensure that the internal HTTP server of the Cisco router is enabled, by issuing the following IOS command:

```
ip http server
```

CUCM Configuration for Network Based Recording

To enable the network recording technology, you need to go through the following specific configuration steps on CUCM.

1. On all **SIP trunks** pointing to the CUBE/VG, under "Recording information" select the option "**This trunk connects to a recording-enabled gateway**".
2. Create a **Recording Profile** (Device - Device Settings - Recording Profiles) with following settings:
 - Name: **Imagicle Call Recording Profile**
 - Recording CSS: **A CSS able to engage the Route Pattern described above.**
 - Recording Destination Address.: **An internal, unused service phone number that will trigger the Route Pattern below defined (e.g. 8500)**

Recording Profile Configuration

 Save

Status
 Status: Ready

Recording Profile Information
 Name*
 Recording Calling Search Space
 Recording Destination Address *

3. On all IP phone lines enabled for call recording, select the **"Automatic Call Recording Enabled"** option in "Recording Option" and the **"Gateway Preferred"** option in the "Recording Media Source". Use the Recording Profile you created.

Recording Option*	<input type="text" value="Automatic Call Recording Enabled"/>
Recording Profile	<input type="text" value="Imagicle Recording Profile"/>
Recording Media Source*	<input type="text" value="Gateway Preferred"/>
Monitoring Calling Search Space	<input type="text" value=" < None >"/>
<input checked="" type="checkbox"/> Log Missed Calls	

4. Create a **Route Pattern** to route the recording calls to the Imagicle Call Recording SIP trunk. The pattern must match the number set in the recording profile you created.

Recording remote destinations

Network recording allows to record incoming calls answered by remote (off-cluster) devices. This allows to handle some particular scenarios, including:

- **Single Number Reach:** the user can answer his office incoming calls from his/her mobile (GSM) phone.
- On-call duty for critical services.
- Incoming calls transferred to **another CUCM cluster**.
- Incoming calls transferred to **off-net destinations** (remote call-centres, IVR, etc.).

This section describes how to configure CUCM to record this kind of calls.

Single Number Reach users

To record phone calls of users that are already enabled on CUCM for single number reach (Mobile Connect):

- 1) Configure CUCM for Network Recording, as described above.
- 2) In the Remote Destination Profile, select the associated DN and enable it for recording like a regular phone line:

Remote Destination Profile Configuration

Save Delete Copy Add New

Status
 Status: Ready

Association

1	Line [1] - 229 in IP Phones
2	Line [2] - Add a new DN

Remote Destination Profile Information

Name*	RDP Brian McAdams
Description	RDP Brian McAdams
User ID*	user1
Device Pool*	Default
Calling Search Space	CSS_AllIpPhones
AAR Calling Search Space	< None >
User Hold Audio Source	1-SampleAudioSource
Network Hold MOH Audio Source	< None >
Privacy*	Default

DN properties:

Recording Option*	Automatic Call Recording Enabled
Recording Profile	Imagicle Recording Profile
Recording Media Source*	Gateway Preferred
Monitoring Calling Search Space	< None >
<input checked="" type="checkbox"/> Log Missed Calls	

Other remote destinations

If you need to record calls answered by remote destinations that are not associated to a user for Single Number Reach, the following configuration settings must be done on CUCM.

Just as an example, suppose you have an operator pool working in a Hunt Group (or QME agent group), normally being recorded. Suppose the Hunt Group escalates the call to a remote branch (calling a landline PSTN number) in the case no operators are available.

In order to record the calls forwarded to the remote branch, you need to create a dummy user with an "ad-hoc" extension enabled for the Single Number Reach and associated to the remote destination you need to record.

If you want to record such forwarded calls, you need to go through the following configuration steps.

1) Define a new **end-user** (even a generic dummy user, like *rec.user1*) enabling the checkbox "Enable Mobility":

End User Configuration

 Save

Directory Number Associations

Primary Extension

Mobility Information

Enable Mobility

Enable Mobile Voice Access

Maximum Wait Time for Desk Pickup*

Remote Destination Limit*

Remote Destination Profiles

2) Create a **new DN**, just for recording purpose. This DN will trigger the outgoing call to the remote destination. Add the DN in the last position of the Hunt Group, for escalation purposes (e.g. 82501, in picture below).

Directory Number Configuration

 Save

Status

 Directory Number Configuration has refreshed due to a directory number change. Please click Save

Directory Number Information

Directory Number* Urgent Priority

Route Partition

Description

Alerting Name

ASCII Alerting Name

External Call Control Profile

Active

Directory Number Settings

Voice Mail Profile (Choose <Nor

Calling Search Space

3) Create a new **Remote Destination Profile** (Device => Device Settings => Remote Destination Profile) with the following properties:

- Name and description: whatever you want, for instance: "RDP Employee Mobile Phone"
- UserID: the end-user created at the step 1 (*rec.user1*)
- CSS: a valid CSS enabled to place calls to the remote destination

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- Device Pool: a convenient device pool. *This device pool should contain at least a transcoding resource if the outgoing call leg is established with an unsupported codec.*

Save the Remote Destination Profile:

Remote Destination Profile Configuration

Save Delete Copy Add New

Status
Status: Ready

Association

1	Line [1] - Add a new DN
---	---

Remote Destination Profile Information

Name* RDP Employee Mobile Phone

Description RDP Employee Mobile Phone

User ID* **rec.user1**

Device Pool* Default

Calling Search Space CSS_AllIpPhones

AAR Calling Search Space < None >

User Hold Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source < None >

Privacy* Default

Rerouting Calling Search Space CSS_AllIpPhones

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS

User Locale < None >

4) Go back to the Remote Destination Profile, then click on "Add a New Remote Destination"

Associated Remote Destinations

[Add a New Remote Destination](#)

5) Insert the remote destination number (the mobile remote branch number in the example above), including the off-net prefix, if required. Also, enable the Single Number reach checkbox and make the remote phone immediately rings as soon as the 82501 DN rings:

Remote Destination Configuration

Save

Remote Destination Information

Name: Employee Mobile Phone

Destination Number*: +441632960030

Owner User ID*: rec.user1

Enable Unified Mobility features

Remote Destination Profile*: RDP Employee Mobile Phone

Single Number Reach Voicemail Policy*: Use System Default

Enable Single Number Reach
Ring this phone and my business phone at the same time when my business line(s) is dialed.

Enable Move to Mobile
If this is a mobile phone, transfer active calls to this phone when the mobility button on your Cisco IP Phone is pressed.

Enable Extend and Connect
Allow this phone to be controlled by CTI applications (e.g. Jabber)

CTI Remote Device*: -- Not Selected --

Timer Information

Wait* 0.0 seconds before ringing this phone when my business line is dialed.*

Prevent this call from going straight to this phone's voicemail by using a time delay of* 1.5 seconds to detect when calls go straight to voicemail.*

Stop ringing this phone after* 19.0 seconds to avoid connecting to this phone's voicemail.*

6) Save the remote destination, associate it to the DN defined above (82501) and save again:

Remote Destination Configuration

Save Delete Copy Add New

Status
Add successful

Line	Line Association
Line [1] - 82501 in IP_Phones	<input checked="" type="checkbox"/>

Remote Destination Information

Name: Employee Mobile Phone

Destination Number*: +441632960030

Owner User ID*: rec.user1

Enable Unified Mobility features

Remote Destination Profile*: RDP Employee Mobile Phone

Single Number Reach Voicemail Policy*: Use System Default

Enable Single Number Reach

7) Go back to the Remote Destination Profile, select the associated DN (82501) and enable it for the network recording (Automatic Call Recording Enabled, Imagicle Recording Profile, Gateway Preferred):

Recording Option*: Automatic Call Recording Enabled

Recording Profile: Imagicle Recording Profile

Recording Media Source*: Gateway Preferred

Monitoring Calling Search Space: < None >

Log Missed Calls

Save and apply the new settings.

8) Configure in the Imagicle Application Suite a real or dummy user with the DN defined above (82501) as primary extension number and enable it for Call Recording application.

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9) Test: place a call to the DN defined above (82501). The remote phone should ring after a few seconds. As soon as the call is answered by the remote phone, the media forking should be triggered automatically and the conversation should be recorded by Imagicle Call Recording.

10) Test the real scenario you need to handle (the hunt group escalation in the example above).

Configuration for Automated Dial-In recording mode

CTI/TAPI Device association

Automated Dial-In recording requires an IP Phone to be controlled through CTI/TAPI. Associate all the devices you want to use with Automated Dial-In to the ImagicleCTI user you created during [TSP setup](#).

XML Service creation

To use Automated Dial-In recording, an XML service has to be configured once.

Log onto the CuCM web interface. Click on Device -> Device Settings -> Phone Services.

Define a new Phone Service with following parameters:

- Name: **Imagicle Call Recording Dial-In**
- Description: Imagicle Call Recording Dial-In On-Demand Service
- Service Category: **XML Service**
- Service Type: **Standard IP Phone Service**
- Service URL: **http://<IAS_ip_address>/fw/Apps/Recorder/xml/OnDemand/Start.aspx?name=#DEVICENAME#**
- Flag "Enable": set

TIP: You can automatically get the right URL to be pasted, with the right IP address, directly from the Application Suite web interface. Just open the Call Recording -> Global Setting page -> Settings panel -> Service URL sub-panel

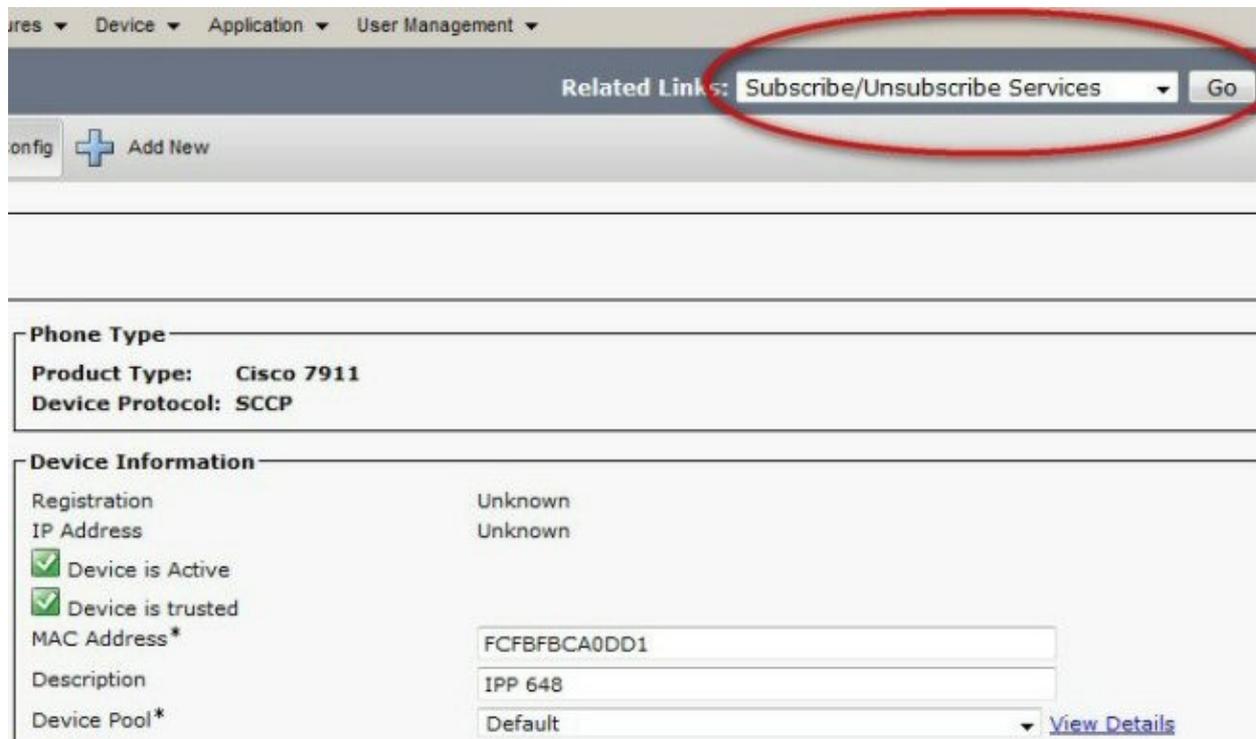
The screenshot shows a web form titled "Service Information" with the following fields and values:

Service Name*	Imagicle Call Recording Dial-In
Service Description	Imagicle Call Recording Dial-In On-Demand Service
Service URL*	http://192.168.150.127/fw/Apps/Recorder/xml/onDemand/start
Secure-Service URL	
Service Category*	XML Service
Service Type*	Standard IP Phone Service
Service Vendor	Imagicle
Service Version	2018.1.1

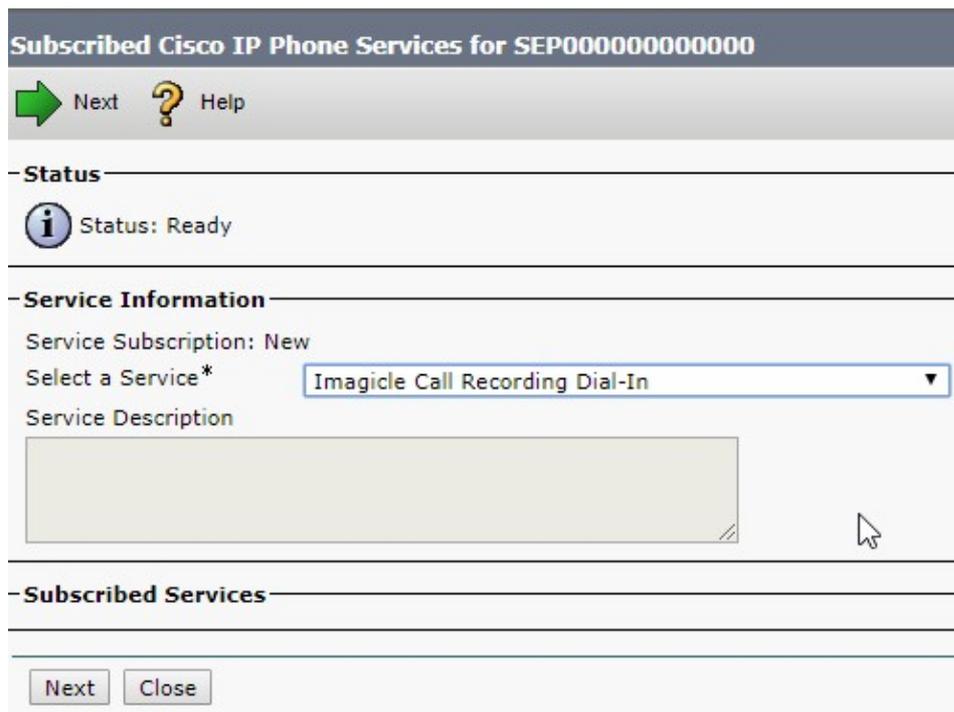
At the bottom of the form, there is a checkbox labeled "Enable" which is checked.

Service Button URL configuration

For each IP Phone where the Automated Dial-In recording has to be used, you need to configure a Service Button URL on that phone. You first need to subscribe the XML Service on the target IP Phone. Click Device -> Phone, select the target IP Phone. Then, in "Related Links" drop-down, choose "Subscribe/Unsubscribe Services" and click Go.



In "Select a Service", choose "Imagicle Call Recording Dial-In", click "Next" and then "Subscribe".



Now, you can create the Service Button URL: go back to the target phone configuration page and, in the "Association Information" panel on the left, locate and click the "Add a new SURL":

Phone Configuration

Save ✖ Delete 📄 Copy 🔄 Reset ✎ Apply Config

Status
📘 Status: Ready

Association Information

Modify Button Items

1	Line [1] - 198 (no partition)
2	Line [2] - 199 (no partition)
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
7	Add a new SD
8	Add a new SD
----- Unassigned Associated Items -----	
9	Line [3] - Add a new DN
10	Add a new SD
11	Add a new SURL

Then, in the "Service" drop-down, choose the service you want to link to the button, i.e. "Imagicle Call Recording Dial-In". Edit the "Label" field as you prefer- This is the label the operator will see on phone display. Then Save.

Configure Service URL Buttons for

Save 🗑 Close 🔍 Help

Status
📘 Status: Ready

Service URL Settings on base Phone

Button	Service	Label
1	Imagicle Call Recording Dial-In ▼	<input type="text" value="Record"/>

Save Close

Finally, click on "Modify Button Items" to associate a phone button to the Service Button URL.

CallManager Service Parameters configuration

To let a user record any conference call, the "Advanced Ad Hoc Conference" Service Parameter of the "Call Manager" service has to be enabled.

The screenshot shows the 'Service Parameter Configuration' window. At the top, there are buttons for 'Save', 'Set to Default', and 'Advanced'. Below this is a section titled 'Clusterwide Parameters (Feature - Conference)'. It contains several parameters, each with a dropdown menu or text input field. The parameter 'Advanced Ad Hoc Conference Enabled' is highlighted in orange and is set to 'True'. Other parameters include 'Suppress MOH to Conference Bridge' (True), 'Drop Ad Hoc Conference' (Never), 'Maximum Ad Hoc Conference' (4), 'Maximum MeetMe Conference Unicast' (4), 'Choose Encrypted Audio Conference Instead Of Video Conference' (True), 'Minimum Video Capable Participants To Allocate Video Conference' (2), 'Enable Click-to-Conference for Third-Party Applications' (False), and 'IMS Conference Factory URI' (cucm-conference-factory@cucm1.company.com). A 'Cluster Conferencing Prefix Identifier' field is empty. A note at the bottom states: 'There are hidden parameters in this group. Click on Advanced button'.

Route Pattern Creation

A fixed-length **Route Pattern** must be configured accordingly with the phone number prefix specified in section "Automated Recording" within "Pilot Numbers" of Imagicle Call Recording's Global Settings (e.g. 9000XXXX, in the image below).

Call Recording [All Recordings](#) [My Recordings](#) [Audit Trail](#) [Global Settings](#) [Manage Service](#)

The screenshot shows the 'Pilot numbers' configuration page. It has a 'Permissions' section with a right-pointing arrow and a 'Pilot numbers' section with a down-pointing arrow. Under 'Pilot numbers', there is a sub-section 'Automatic user identification' with a description: 'By default, users are automatically identified by their calling number. In case of overlapping dial plan, the number specified here is required to properly identify the recording user.' Below this are two fields: 'Recording pilot (prefix):' (empty) and 'User identification strategy:' with two radio button options: 'Detect user by Numeric User Id' (selected) and 'Detect user by Numeric Partition Id'. Another sub-section 'Automated Recording' has a description: 'If you need to use the CTI-controlled mechanism to automate the dial-in conference calls, please, configure here the dedicated service pilot number. The user IP phone will automatically place a call to this number, followed by further 4 digits.' Below this is a field 'CTI controlled pilot n° (prefix):' with the value '9000' entered. A note below states: '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:'. Below the note is a highlighted box containing the pattern '9000XXXX'. At the bottom right, there are 'Save' and 'Cancel' buttons.

Configuration for Live Keep recording mode

XML Service creation

To use Live Keep recording, an XML service has to be configured once.

Log onto the CuCM web interface. Click on Device -> Device Settings -> Phone Services.

Define a new Phone Service with following parameters:

- Name: **Imagicle Call Recording Live Keep**
- Description: Imagicle Call Recording Live Keep Service
- Service Category: **XML Service**
- Service Type: **Standard IP Phone Service**
- Service URL:
http://<IAS_ip_address>/fw/Apps/Recorder/xml/OnDemand/StartMediaForking.aspx?name=#DEVICENAME#
- Flag "Enable": set

TIP: You can automatically get the right URL to be pasted, with the right IP address, directly from the Application Suite web interface. Just open the Call Recording -> Global Setting page -> Settings panel -> Service URL sub-panel

Service Information	
Service Name*	Imagicle Call Recording Live Kee
Service Description	Imagicle Call Recording Live Keep Service
Service URL*	http://192.168.1.2/fw/Apps/Recorder/xml/OnDemand/Star
Secure-Service URL	
Service Category*	XML Service ▼
Service Type*	Standard IP Phone Service ▼
Service Vendor	Imagicle
Service Version	2019.1.1
<input checked="" type="checkbox"/> Enable	
<input type="checkbox"/> Enterprise Subscription	

Service Button URL configuration

For each IP Phone where Live Keep recording has to be used, you need to configure a Service Button URL on that phone. You first need to subscribe the XML Service on the target IP Phone. Click Device -> Phone, select the target IP Phone. Then, in "Related Links" drop-down, choose "Subscribe/Unsubscribe Services" and click Go.

res ▾ Device ▾ Application ▾ User Management ▾

Related Links: **Subscribe/Unsubscribe Services** ▾ Go

onfig + Add New

Phone Type

Product Type: Cisco 7911
Device Protocol: SCCP

Device Information

Registration	Unknown
IP Address	Unknown
<input checked="" type="checkbox"/> Device is Active	
<input checked="" type="checkbox"/> Device is trusted	
MAC Address*	<input type="text" value="FCFBFBCA0DD1"/>
Description	<input type="text" value="IPP 648"/>
Device Pool*	<input type="text" value="Default"/> ▾ View Details

In "Select a Service", choose "Imagicle Call Recording Live Keep", click "Next" and then "Subscribe".

Subscribed Cisco IP Phone Services for SEP000000000000

Next Help

Status

Status: Ready

Service Information

Service Subscription: New

Select a Service* ▾

Service Description

Subscribed Services

Now, you can create the Service Button URL: go back to the target phone configuration page and, in the "Association Information" panel on the left, locate and click the "Add a new SURL":

Phone Configuration

Save Delete Copy Reset Apply Config

Status
 Status: Ready

Association Information

Modify Button Items

1	Line [1] - 198 (no partition)
2	Line [2] - 199 (no partition)
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
7	Add a new SD
8	Add a new SD
----- Unassigned Associated Items -----	
9	Line [3] - Add a new DN
10	Add a new SD
11	Add a new SURL

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Then, in the "Service" drop-down, choose the service you want to link to the button, i.e. "Imagicle Call Recording Dial-In". Edit the "Label" field as you prefer- This is the label the operator will see on phone display. Then Save.

Configure Service URL Buttons for

Save Close Help

Status
 Status: Ready

Service URL Settings on base Phone

Button	Service	Label
1	Imagicle Call Rec Live Keep	Record

Save Close

Finally, click on "Modify Button Items" to associate a phone button to the Service Button URL.

Ruote Pattern Creation

A **Route Pattern** must be configured accordingly with the pilot number specified in "Live Keep pilot (prefix)" field within "Pilot Numbers" of Imagicle Call Recording's Global Settings (e.g. 9155, in the image below).

Call Recording [All Recordings](#) [My Recordings](#) [Global Settings](#) [Manage Service](#)

Settings	Data Management	Notifications	Announcements
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Recording filters >

Recording control >

Permissions >

Pilot numbers v

Automatic user identification

By default, users are automatically identified by their calling number. In case of overlapping dial plan, the number specified here is required to properly identify the recording user.

Regular pilot (prefix):

Live Keep pilot (prefix):

User identification strategy:

- Detect user by Numeric User Id
- Detect user by Numeric Partition Id

Automated Recording

If you need to use the CTI-controlled mechanism to automate the dial-in conference calls, please, configure here the dedicated service pilot number. The user IP phone will automatically place a call to this number, followed by further 4 digits.

CTI controlled pilot n° (prefix):

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:

```
9200XXXX
```

Secure recording >

Services URLs >