


# Internal Calling Party Presentation - Not Restricted

Description: This document contains the step by step procedure to restrict the Calling Party Presentation (Internal extension) display, e.g. if extension 1111 calling to extension 2222(or any other internal extension) then extension 2222 should not see the display as 1111 but instead it should be unknown.

Partition Configuration:

Call Routing → Class of Control → Partition → Add New

**Status**






 Status: Ready

**Partition Information**


To enter multiple partitions, use one line for each partition entry. You can enter up to 75 partitions; the names and descriptions can have up to a total of 1475 characters. The partition name cannot exceed 50 characters. Use a comma (,) to separate the partition name and description on each line. If a description is not entered, Cisco Unified Communications Manager uses the partition name as the description. For example:  
<< partitionName >> , << description >>  
CiscoPartition, Cisco employee partition  
DallasPartition

Name\*

**Partition Configuration**

 Save  Delete  Reset  Apply Config  Add New

**Status**

 Status: Ready


**Partition Information**

Name\*

Description

Time Schedule


Time Zone  Originating Device  Specific Time Zone

 \*- indicates required item.

Calling Search Space Configuration:


Call Routing → Class of Control → Calling Search Space → Add New

**Calling Search Space Configuration**

 Save

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**- Status**

 Status: Ready

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**- Calling Search Space Information**

Name\*

Description

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**- Route Partitions for this Calling Search Space**

Available Partitions\*\*

▼ ▲





Selected Partitions

▼ ▲

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Select the Partition configured in previous step and save the CSS.

**Calling Search Space Configuration**

 Save  Delete  Copy  Add New

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**Status**

 Status: Ready

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**Calling Search Space Information**

Name\*

Description

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**Route Partitions for this Calling Search Space**

Available Partitions\*\*

▼ ▲

Selected Partitions

▼ ▲

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## Calling Party Transformation Pattern Configuration:

Call Routing → Transformation → Transformation Pattern → Calling Party Transformation Pattern → Add New

The screenshot shows the Cisco Unified CM Administration interface. The left-hand navigation menu is expanded to show the path: **Transformation** → **Transformation Pattern** → **Calling Party Transformation Pattern**. The main content area shows the configuration page for a Calling Party Transformation Pattern, with fields for Name, Description, and various routing options. A "Save" button is visible at the bottom left of the configuration area.

Enter the Pattern as extension number (e.g. 1111) and select the Partition configured above, Select Calling Line ID Presentation "Restricted", refer below snap. Save it.

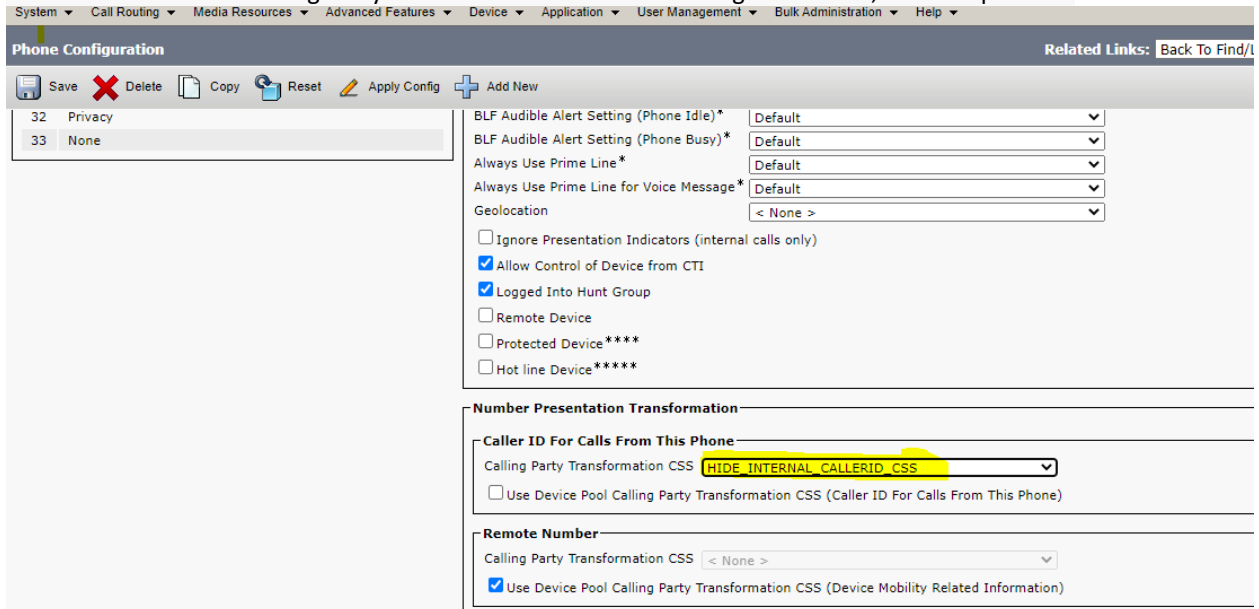
The screenshot shows the configuration page for a Calling Party Transformation Pattern. The page is titled "Calling Party Transformation Pattern Configuration" and includes a "Save" button at the top left. The configuration is organized into several sections:

- Status:** Status: Ready
- Pattern Definition:**
  - Pattern\*: 1111
  - Partition: HIDE\_INTERNAL\_CALLERID\_PT
  - Description:
  - Numbering Plan: < None >
  - Route Filter: < None >
  - Urgent Priority
  - MLPP Preemption Disabled
- Calling Party Transformations:**
  - Use Calling Party's External Phone Number Mask
  - Discard Digits: < None >
  - Calling Party Transformation Mask:
  - Prefix Digits:
  - Calling Line ID Presentation\*: Restricted
  - Calling Party Number Type\*: Cisco CallManager
  - Calling Party Numbering Plan\*: Cisco CallManager

A "Save" button is located at the bottom left of the configuration area.

Next Step is to apply Calling Party Transformation CSS in Device (Phone) where extension 1111 configured as directory number (Line)

Device→Phone→ Find your phone edit the phone configuration, got to Number Presentation Transformation section and select the Calling Party Transformation CSS which configured earlier, refer snap below.



Save the configuration of device and reset it, after device registered back make the call from this phone (extension 1111) to extension 2222 or any other extension; **Unknown** will display instead of 1111.

**Note: Configuration completed above will work with on premises CUCM setup where device directly registered with CUCM in LAN, TFTP server address manually or via DHCP scope option 150.**

**Note: If your device/endpoint is registered via Collab Edge MRA (Jabber clients, TC-based endpoints- EX/MX/C), via Expressway (MRA), then Calling Party Transformation CSS is not applied for endpoints registered via Collab Edge MRA.**

This is happening because Expressway will add P-Asserted-Identity (PAI) as VCS always done. CUCM never expects PAI coming from Endpoints and it rewrites already applied changed Calling Party Number by number presented in PAI header.

**Conditions:**

- endpoint is registered via Collab Edge MRA (Jabber clients, TC-based endpoints- EX/MX/C)
- endpoint have Calling Party Transformation CSS configured

**Workaround:**

- write SIP normalization script to remove PAI header and attach it to Endpoint SIP profile:

```
M = {}

function M.inbound_INVITE(msg)
msg:addHeader("X-MRA-Call-Phone-SIP-Profile", "true")
msg:removeHeader("P-Asserted-Identity")
end

return M
```

For more detail refer--><https://bst.cloudapps.cisco.com/bugsearch/bug/CSCur49826>

**Below is extra configuration for Device/endpoint registered via Expressway as MRA:**

SIP Normalization Script Configuration:

Device → Device Settings → SIP Normalization Scripts → ADD New

**SIP Normalization Script Configuration**

Save Delete Reset Add New Import File

**Status**  
Status: Ready

**SIP Normalization Script Info**

Name\* RemovePAIheader  
Description For Resolution:Calling Party Transformation CSS is not applied fo  
Content\*  

```
M = {}  
  
function M.inbound_INVITE(msg)  
  msg:addHeader("X-MRA-Call-Phone-SIP-Profile", "true")  
  msg:removeHeader("P-Asserted-Identity")  
end  
  
return M
```

  
Script Execution Error Recovery Action\* Message Rollback Only  
System Resource Error Recovery Action\* Disable Script  
Memory Threshold\* 50 kilobytes  
Lua Instruction Threshold\* 1000 instructions

Save Delete Reset Add New Import File

**SIP Normalization Script (1 - 10 of 10)**

Find SIP Normalization Script where Name begins with Find Clear Filter

<input type="checkbox"/>	Name ^	Description
<input type="checkbox"/>	<a href="#">CLI FOR CMS</a>	changes CLI for inbound from CMS
<input type="checkbox"/>	<a href="#">HCS-PCV-PAI-passthrough</a>	Cisco HCS platform integration with Enterprise IMS
<input type="checkbox"/>	<a href="#">RemovePAIheader</a>	For Resolution:Calling Party Transformation CSS is not applied for endpoints registered via Collab Edge MRA
<input type="checkbox"/>	<a href="#">att-header-passthrough</a>	Provides passthrough of header x-att-loop

SIP Profile Configuration:


Device → Device Settings → SIP Profile: Configure new SIP Profile for these type of Endpoint (Registered via Expressway as MRA & want to hide Calling Party Presentation (Internal extension) display)


Copy the existing Default SIP profile and change the name, description as required.

## SIP Profile Configuration

 Save  Delete  Copy  Reset  Apply Config  Add New

### Status

 Status: Ready

 All SIP devices using this profile must be restarted before any changes will take affect.

### SIP Profile Information

Name*	Video Phone SIP Profile_ RemovePAIheader
Description	Default SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agent
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and
Confidential Access Level Headers*	Disabled

Redirect by Application  
 Disable Early Media on 180  
 Outgoing T.38 INVITE include audio mline  
 Offer valid IP and Send/Receive mode only for T.38 Fax Relay  
 Use Fully Qualified Domain Name in SIP Requests  
 Assured Services SIP conformance  
 Enable External QoS\*\*

### SDP Information

Assigned Normalization SIP Scripts configured above to this SIP Profile

## SIP Profile Configuration

 Save  Delete  Copy  Reset  Apply Config  Add New

Semi Attended Transfer

Enable VAD

Stutter Message Waiting

MLPP User Authorization

### Normalization Script

Normalization Script **RemovePAIheader**

Enable Trace

<a href="#">Standard SIP Profile For TelePresence Endpoint</a>	Default SIP Profile For Cisco TelePresence Endpoint
<a href="#">Standard SIP Profile for Mobile Device</a>	Default SIP Profile for Mobile Device
<input type="checkbox"/> <a href="#">Video Phone SIP Profile</a>	Default SIP Profile
<input type="checkbox"/> <a href="#">Video Phone SIP Profile_ RemovePAIheader</a>	Default SIP Profile

## Assigned SIP Profile to Phone/Endpoint:

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Configuration > Help

**Phone Configuration** Related Links:

Save Delete Copy Reset Apply Config Add New

BLF Presence Group*	Standard Presence_group	
SIP Dial Rules	< None >	
MTP Preferred Originating Codec*	711ulaw	
Device Security Profile*	Cisco 8845 - Standard SIP Non-Secure Profile	
Rerouting Calling Search Space	< None >	
SUBSCRIBE Calling Search Space	< None >	
<b>SIP Profile*</b>	<b>Video Phone SIP Profile_ RemovePAIheader</b>	<a href="#">View Details</a>
Digest User	< None >	<b>Find</b>

Media Termination Point Required  
 Unattended Port

Save phone configuration & Reset the Phone

All set now 😊