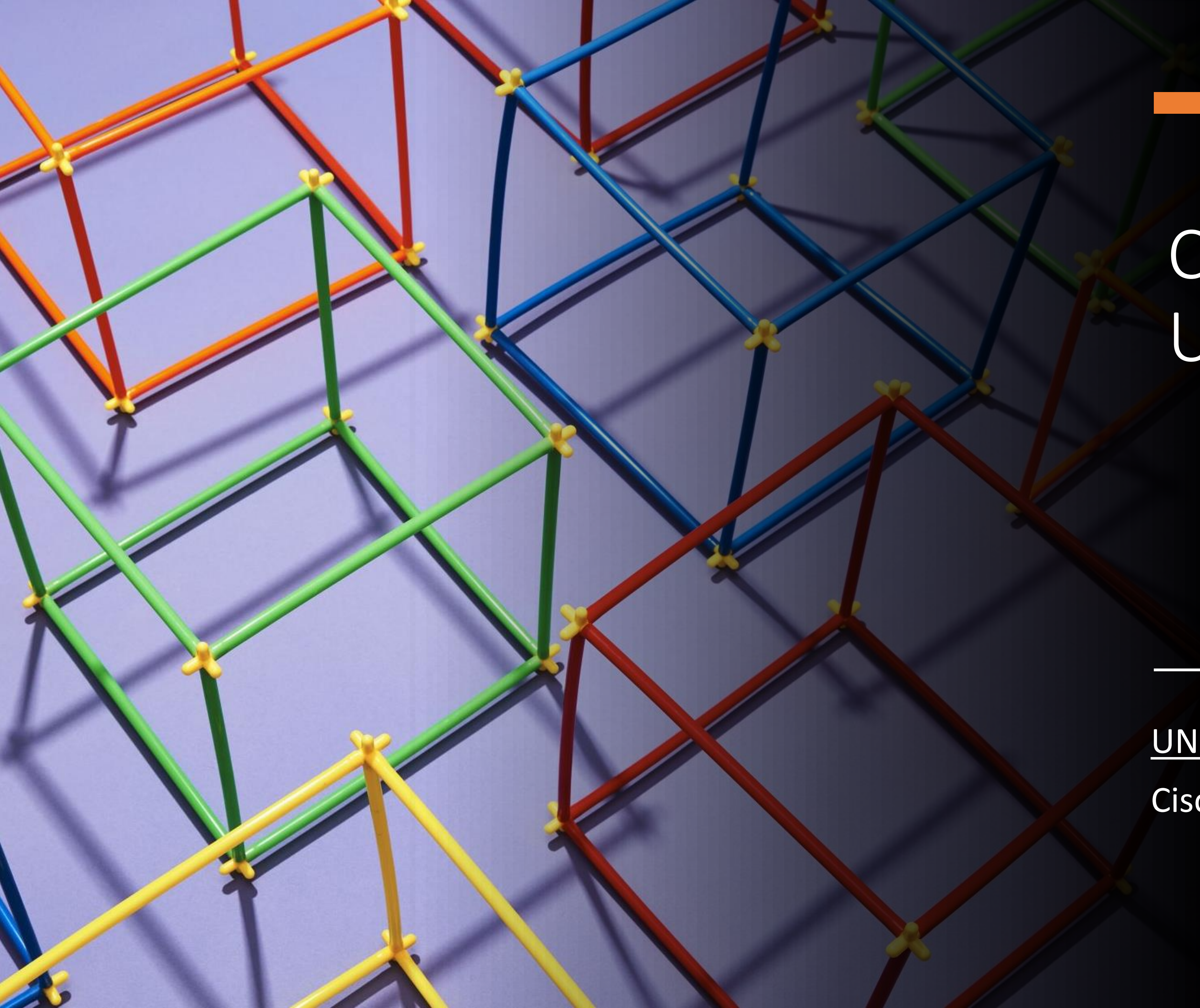


COMUNIACIONES UNIFICADAS

UNIDAD 5 - CUCM

Cisco Unified Communications Manager

CUCM BASICS
Partitions & CSS

CUCM - Partition

Una partición comprende una agrupación lógica con características de accesibilidad similares

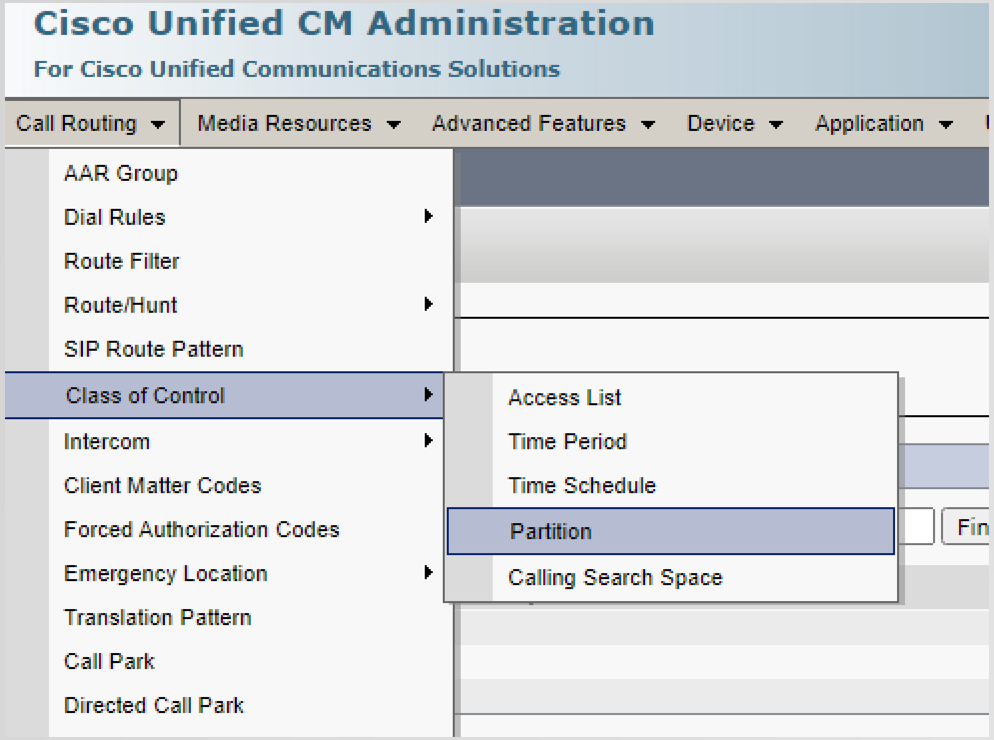
- **números de directorio** (DN)
- **patrones de ruta**(route patterns)
- patrones de traducción (translation patterns)
- puertos de casilla de voz (voice-mail ports)

Para simplificar, los nombres de las particiones suelen reflejar sus características, como

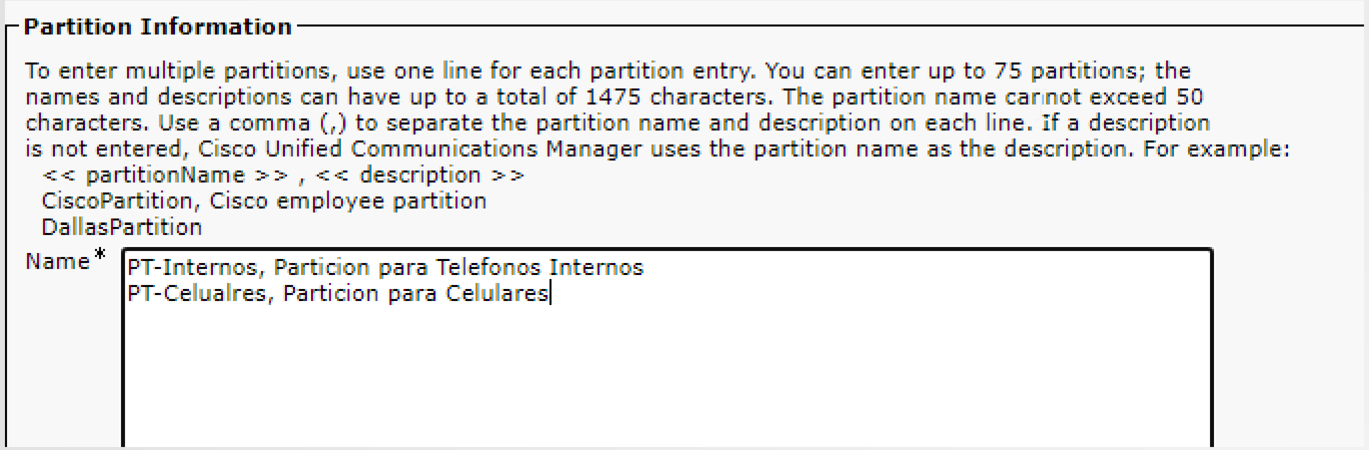
- PT-Celulares ,
- PT-Nacionales,
- PT-Interno,
- PT-Secretaria

¿Quién Puede llamarme?

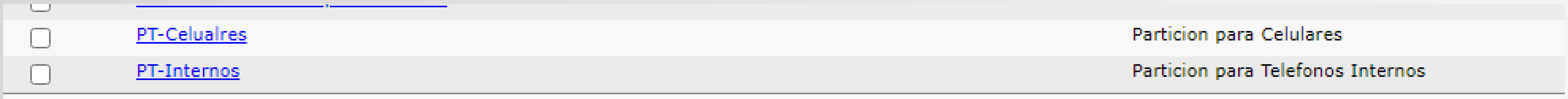
CUCM -



Cisco Unified CM Administration
For Cisco Unified Communications Solutions
Call Routing
Media Resources
Advanced Features
Device
Application
AAR Group
Dial Rules
Route Filter
Route/Hunt
SIP Route Pattern
Class of Control
Intercom
Client Matter Codes
Forced Authorization Codes
Emergency Location
Translation Pattern
Call Park
Directed Call Park
Access List
Time Period
Time Schedule
Partition
Calling Search Space

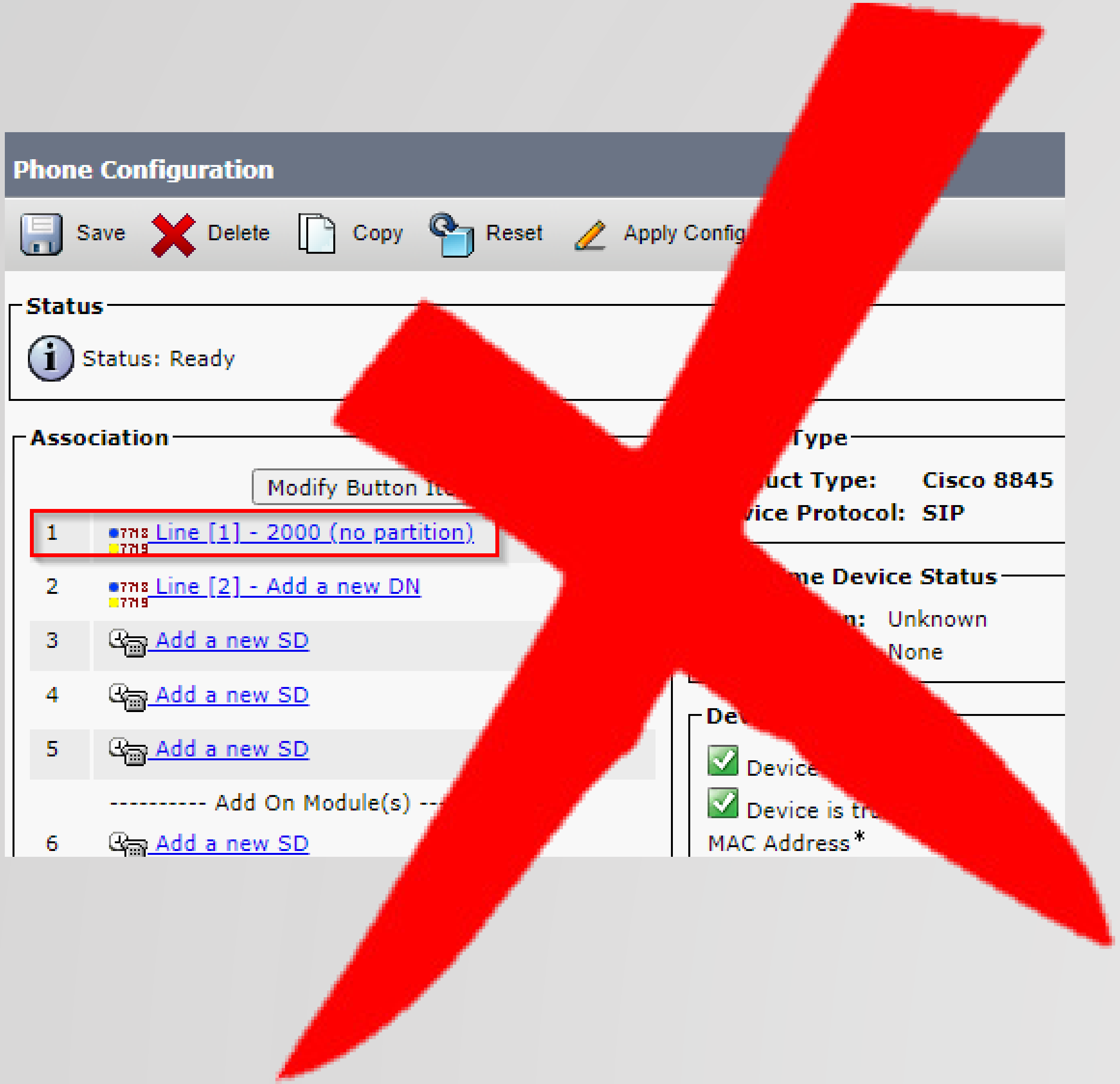


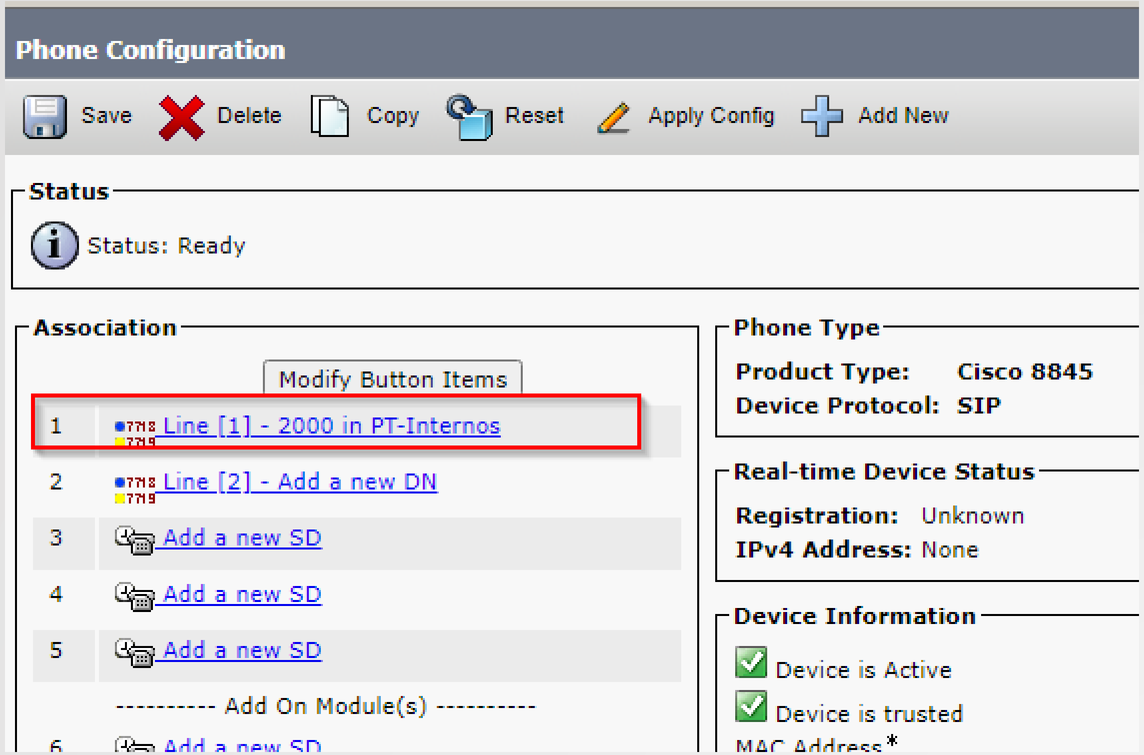
Partition Information
To enter multiple partitions, use one line for each partition entry. You can enter up to 75 partitions; the names and descriptions can have up to a total of 1475 characters. The partition name cannot exceed 50 characters. Use a comma (,) to separate the partition name and description 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*
PT-Internos, Particion para Telefonos Internos
PT-Celualres, Particion para Celulares



PT-Celualres
Particion para Celulares
PT-Internos
Particion para Telefonos Internos

CUCM - NO PARTITION / PARTITION





CUCM – Calling Search Space - CSS

Un espacio de búsqueda de llamadas (**Calling search space** ,CSS) es una **lista ordenada de particiones**.

Los css determinan las particiones que los dispositivos buscan al intentar completar una llamada:

- Teléfonos IP
- Softphones
- Gateways .

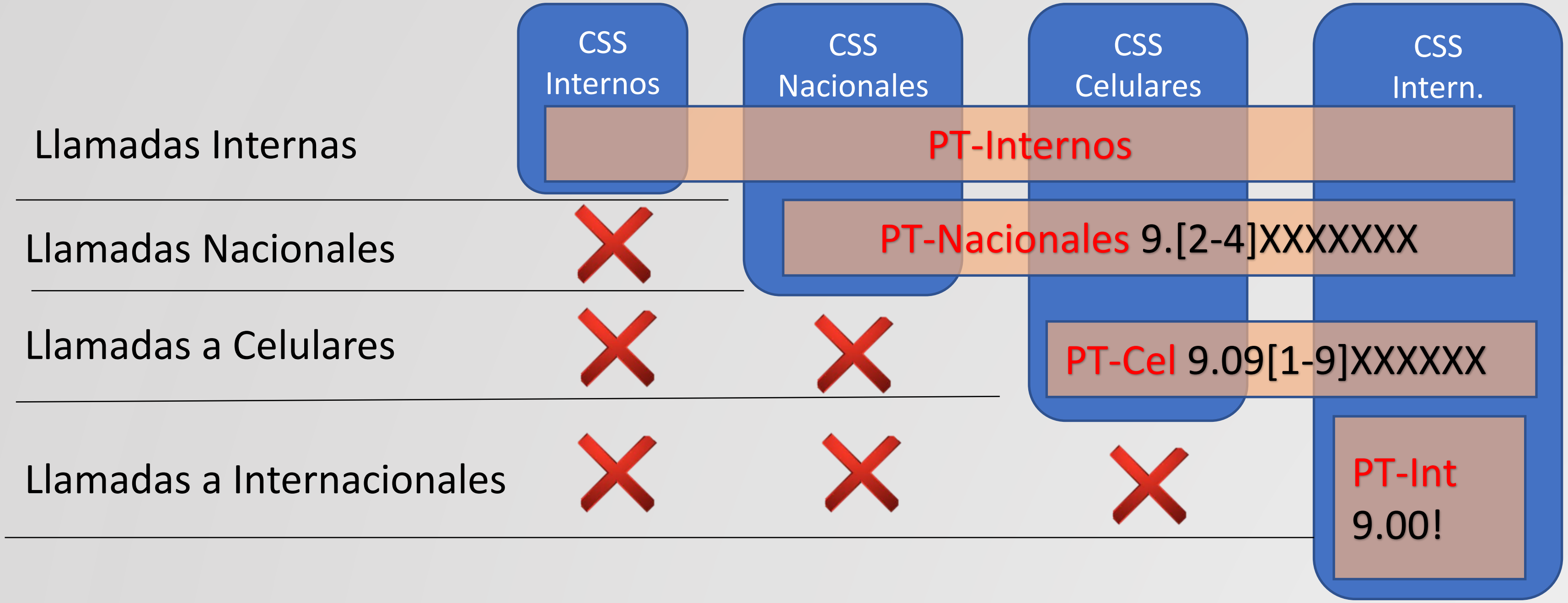
Un CSS ,con nombre “Supervisores” contiene 4 particiones : **Nacionales, Internacionales , Emergencia ,Celulares**

Un CSS, con nombre “Invitados” contiene 2 particiones: **Nacionales, Emergencia**

Si un IP-Phone o DN está configurado con el CSS "Invitado", solo busca en las particiones “Nacionales" y “Emergencia" cuando inicia la llamada.

Si un usuario que llama desde este número intenta marcar un número internacional, no se produce una coincidencia y la llamada no se puede enrutar.

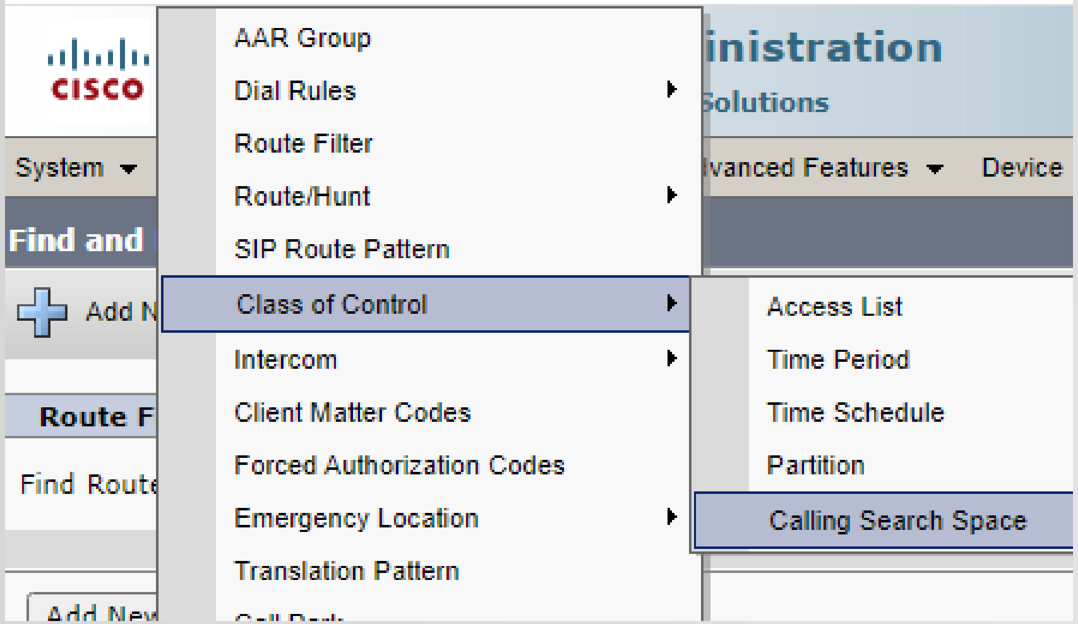
CUCM – CSS / Particiones

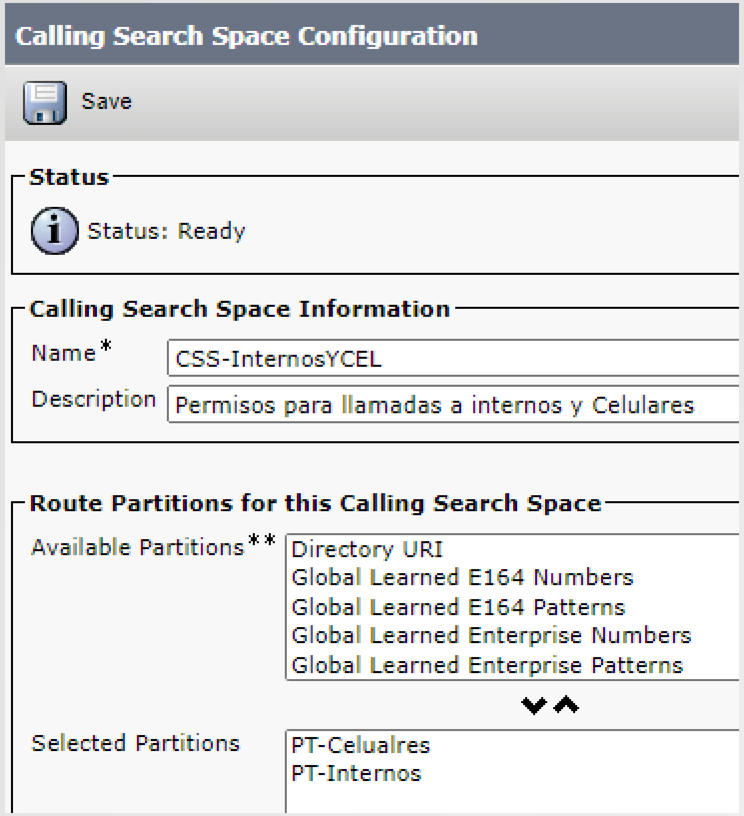


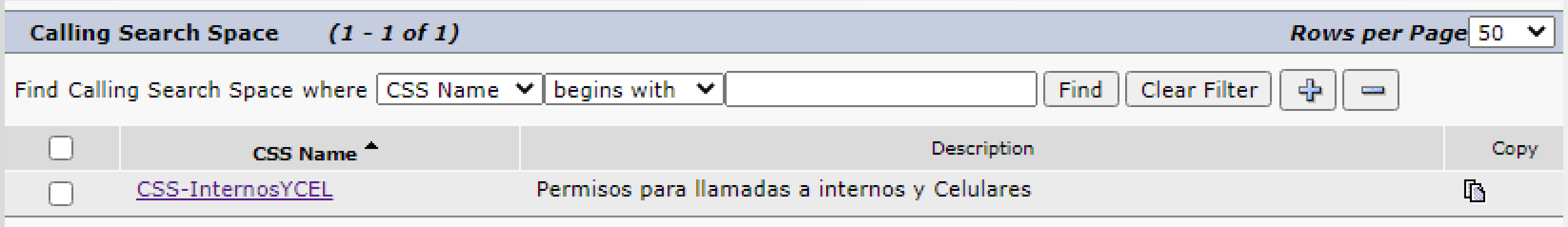
1. A donde puede llamar alguien
2. Ruteo Geográfico

Lista de Particiones

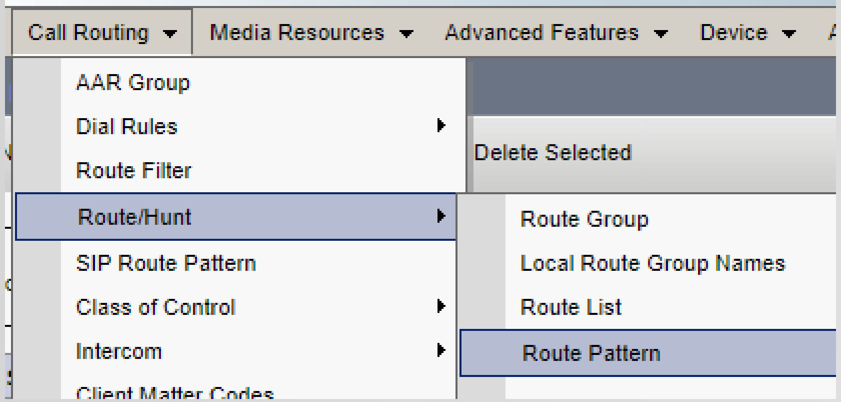
CUCM - CSS

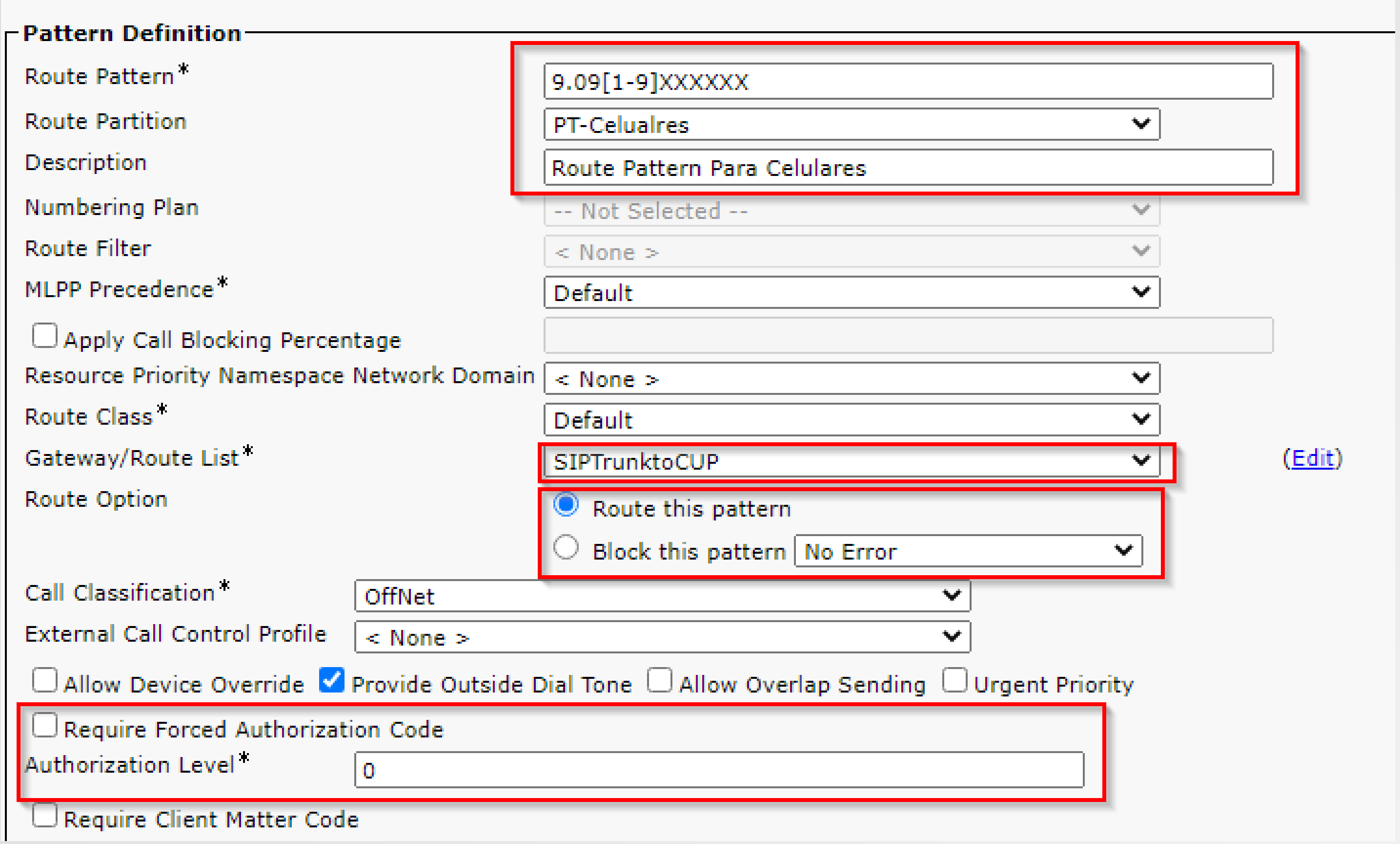

AAR Group
Dial Rules
Route Filter
Route/Hunt
SIP Route Pattern
Class of Control
Intercom
Client Matter Codes
Forced Authorization Codes
Emergency Location
Translation Pattern
Access List
Time Period
Time Schedule
Partition
Calling Search Space
System
Device

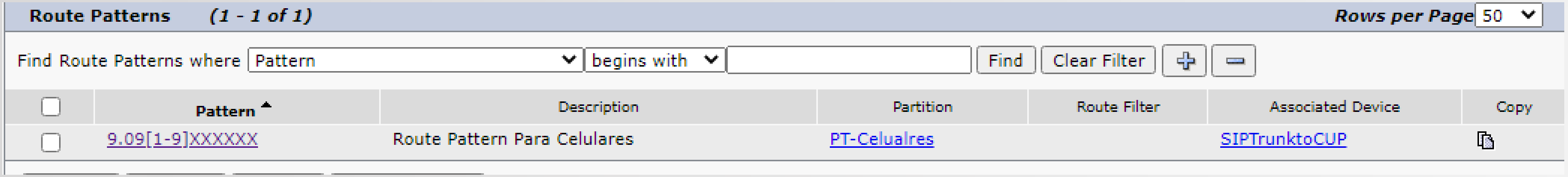

Calling Search Space Configuration
Save
Status
Status: Ready
Calling Search Space Information
Name*
CSS-InternosYCEL
Description
Permisos para llamadas a internos y Celulares
Route Partitions for this Calling Search Space
Available Partitions**
Directory URI
Global Learned E164 Numbers
Global Learned E164 Patterns
Global Learned Enterprise Numbers
Global Learned Enterprise Patterns
Selected Partitions
PT-Celualres
PT-Internos


Calling Search Space (1 - 1 of 1)
Rows per Page 50
Find Calling Search Space where CSS Name begins with
Find
Clear Filter
CSS Name
Description
Copy
CSS-InternosYCEL
Permisos para llamadas a internos y Celulares

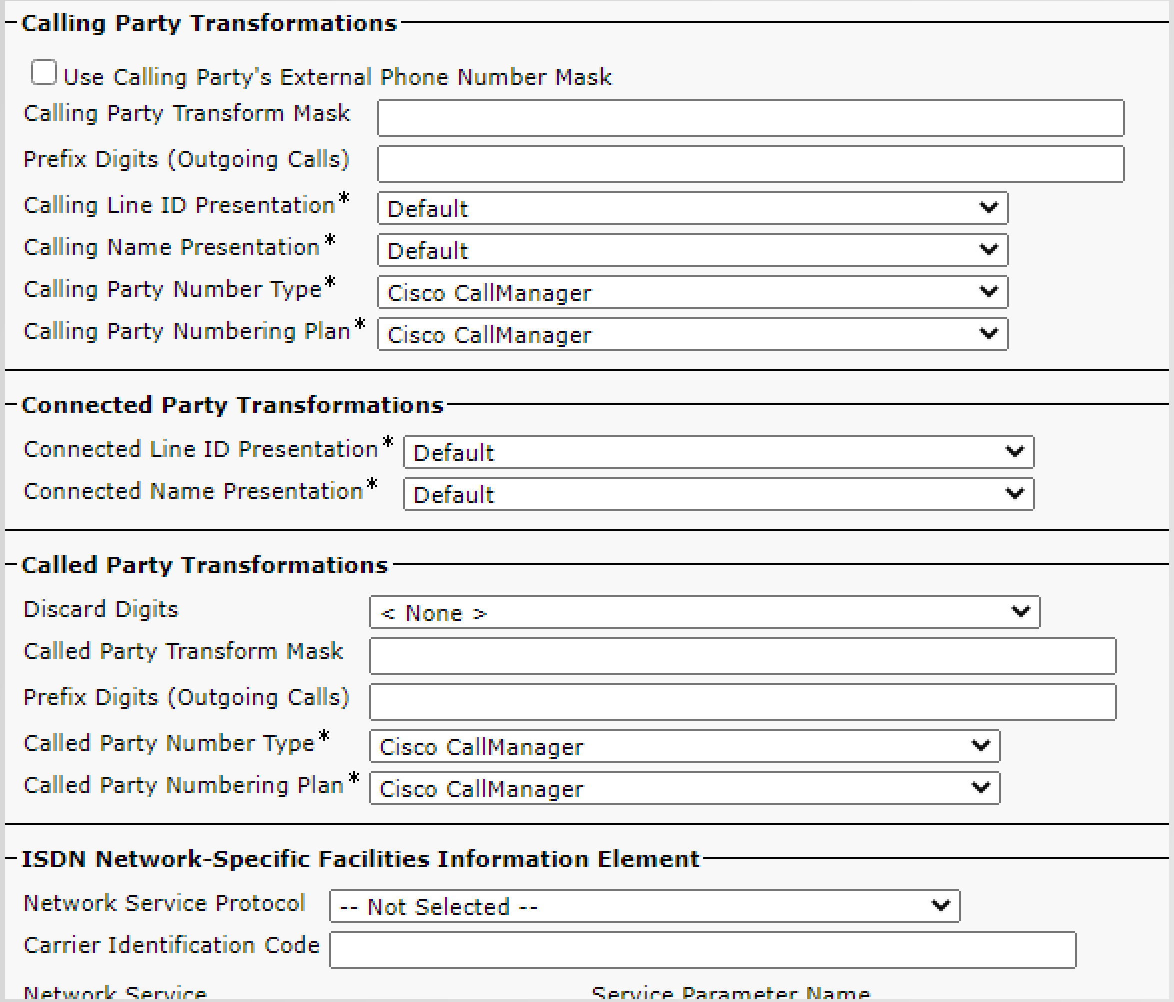
CUCM – Route Patterns

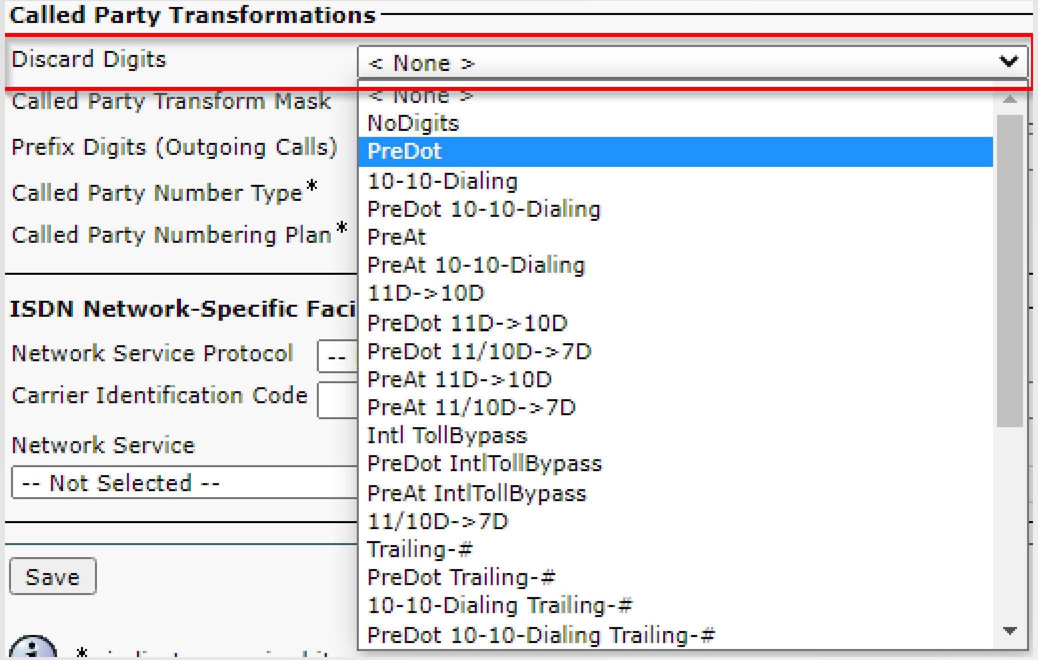






CUCM - Route Patterns





CUCM –NULL Partition

2600 PT-Interno

2601 no partition

Ninguno de los 2 teléfonos tienen configurado un CSS

¿Pueden llamarse?

CUCM – Los CSS cumplen 2 Reglas

Siempre se puede llamar a un DN que no este en una partición

Solo se puede llamar a un DN, si la partición de ese DN esta en el CSS

CUCM – Ejercicio CSS

1 Usuario disca 2000
2 Usuario disca 3000
3 Secretaria disca 3000
4 Secretaria disca 2000
5 Jefe disca 2000
6 Jefe disca 4000
7 Usuario disca
9.001-801-240-7845
8 Jefe disca
9.099-801-478

Path Selection

CUCM – Route Group

Un **route group** es una lista de devices (gateways, trunks).

Un route group puede ser configurado para manejar estas distribuciones

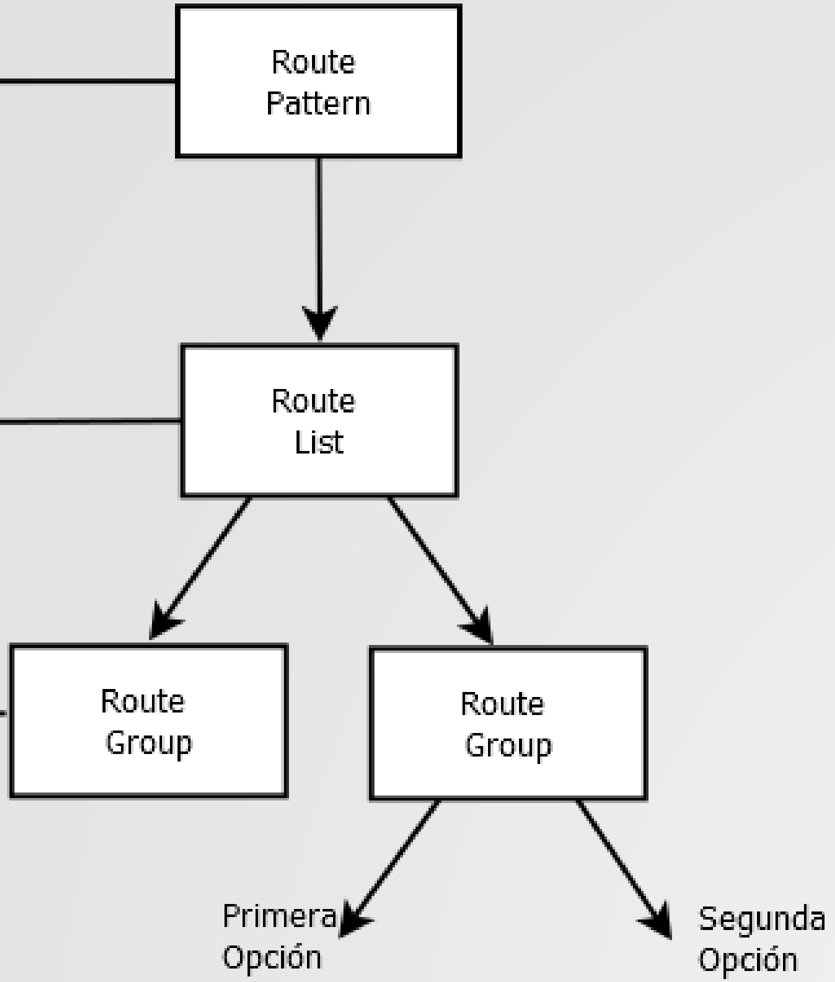
- **distribución circular (round-robin)**
  - es usada para distribuir recursos de carga
- **distribución arriba-abajo (top-down)**
  - para priorizar el uso de una gateway dentro de un route group

CUCM – Selection Path

Route Pattern
* Matchea el número discado para llamadas externas
* Realiza manipulación de Dígitos (Opcional)
* Apunta a una Route List para ruteo

Route List
* Primer nivel de Selección de Path
* Realiza Manipulación de Dígitos
* Apunta a un/s group priorizado para ruteo

Route Group
* Segundo nivel de selección de Path
* Apunta al dispositvo real.



CUCM – Path Selection

Para implementar una elección de camino (path selection) en el CUCM, la lógica del procesamiento de llamada debe ser construida desde abajo hacia arriba (bottom-up)

- Cuando se crea un route group, recién ahi se pueden cargar los dispositivos que va a formar parte del grupo.
- Si los devices no existen todavía, no hay nada que se pueda correlacionar al route group

Los siguientes pasos debe realizarse en el orden dado

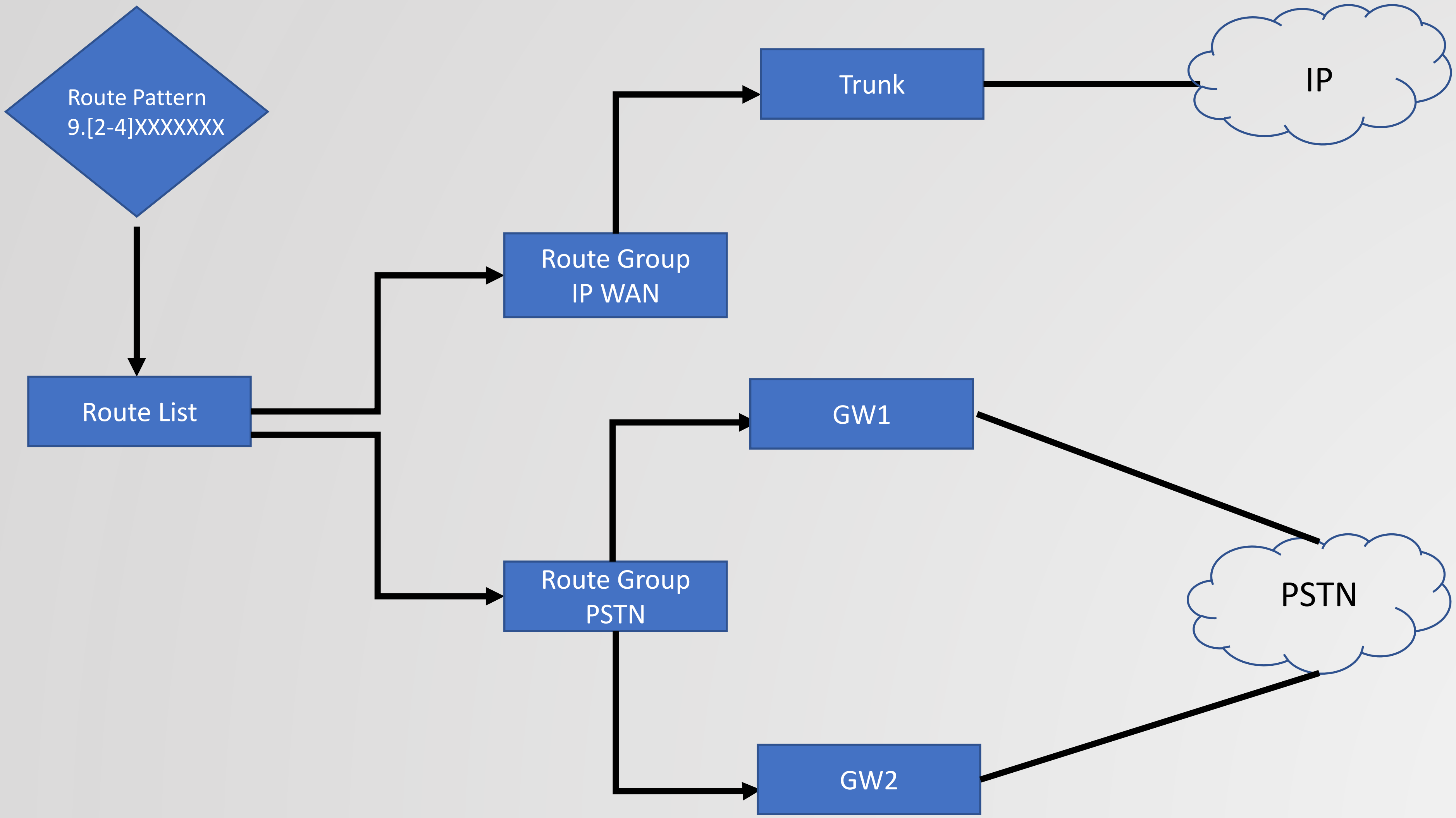
Paso 1 Agregar Dispositivos (Gateways y trunks).

Step 2 Crear route groups con devices disponibles

Step 3 Crear route list con route group disponibles.

Step 4 Crear route patterns apuntando a los route lists

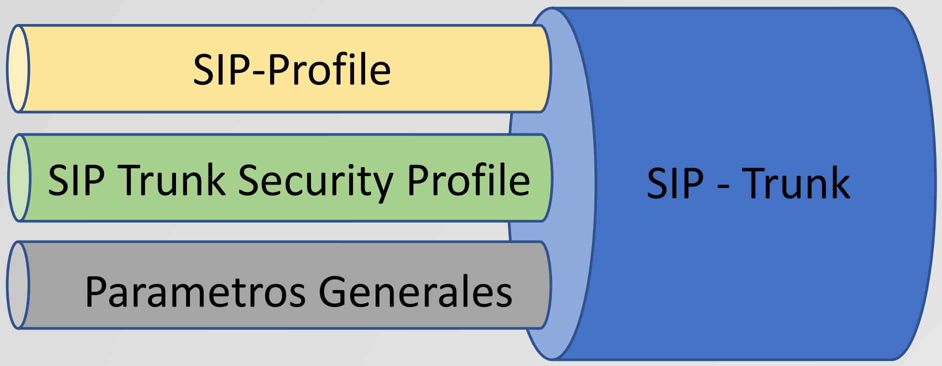
CUCM -


Route Pattern
9.[2-4]XXXXXXX
Route List
Route Group
IP WAN
Route Group
PSTN
Trunk
GW1
GW2
IP
PSTN

SIP TRUNK:

SIP Profile &
Sip Trunk Security Profile

CUCM – Sip Profile

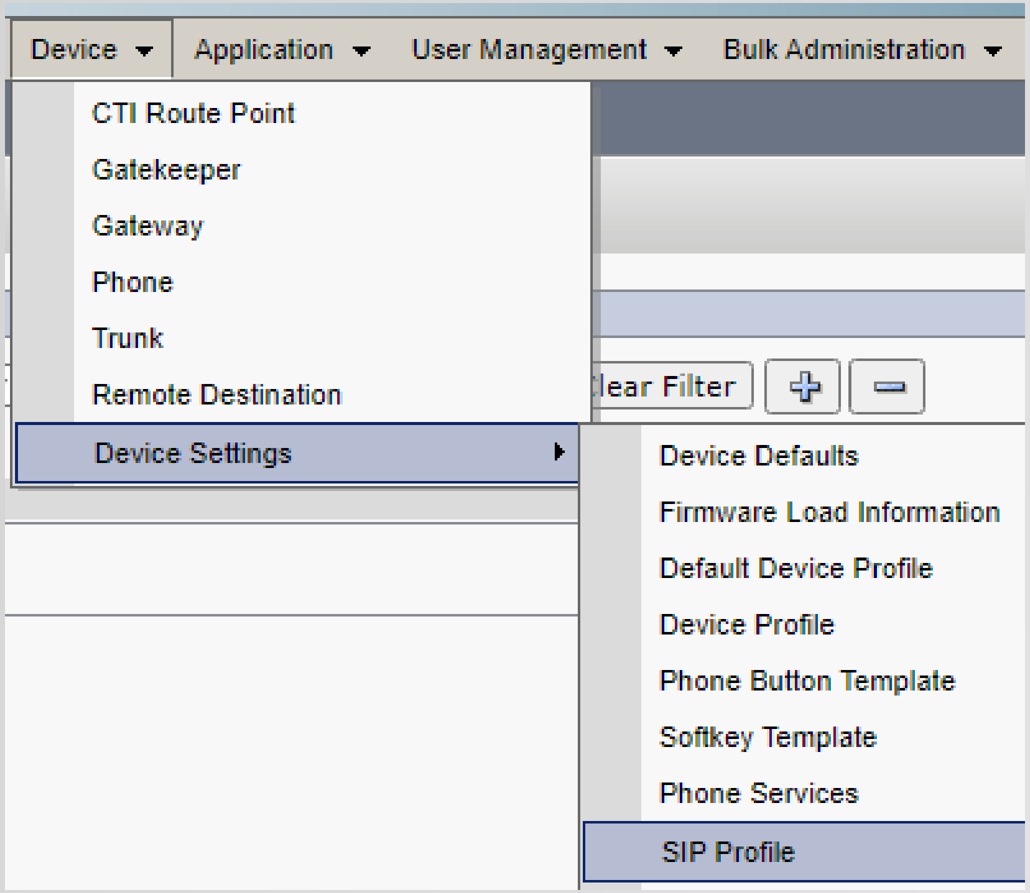


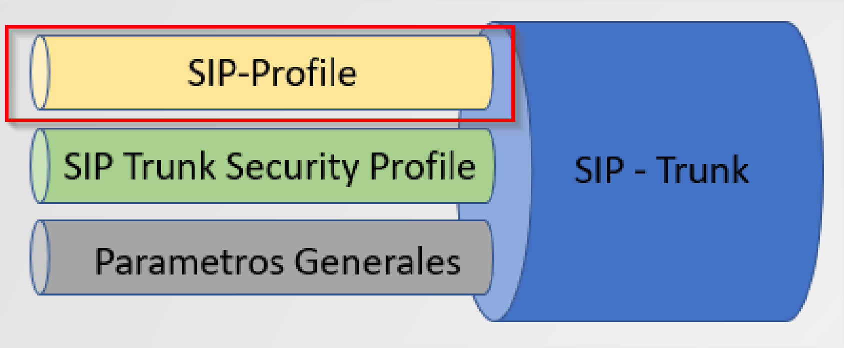
- Un perfil SIP (Sip-Profile) :

- Es un **template** que comprende configuraciones SIP comunes.
- Al implementar troncales SIP (Sip-Trunk) en la red , se puede aplicar configuraciones SIP comunes a grupos de dispositivos a través del SIP-Profile.
- Sin el perfil SIP, tendría que configurar los ajustes individualmente para cada troncal y dispositivo SIP en la red.

CUCM – SIP Profile

- Un perfil SIP (Sip-Profile) es un template que comprende configuraciones SIP comunes.





SIP Profile *(1 - 5 of 5)*

Find SIP Profile where Name begins with Find Clear Filter

Name	Description
Standard SIP Profile	Default SIP Profile
Standard SIP Profile For Cisco VCS	Default SIP Profile For Cisco Video Communication Server
Standard SIP Profile For TelePresence Conferencing	Default SIP Profile For Cisco TelePresence Conferencing
Standard SIP Profile For TelePresence Endpoint	Default SIP Profile For Cisco TelePresence Endpoint
Standard SIP Profile for Mobile Device	Default SIP Profile for Mobile Device

Status

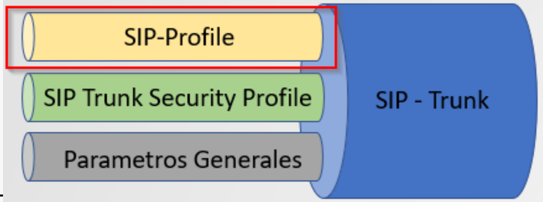
Status: Ready

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

SIP Profile Information

Name*	Standard SIP Profile
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

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	< None >
Accept Audio Codec Preferences in Received Offer*	Default

- [ ] Require SDP Inactive Exchange for Mid-Call Media Change
- [ ] Allow RR/RS bandwidth modifier (RFC 3556)

Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	(•) Common Port Range for Audio and Video ( ) Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [ ] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Normalization Script

Normalization Script: < None >

- [ ] Enable Trace

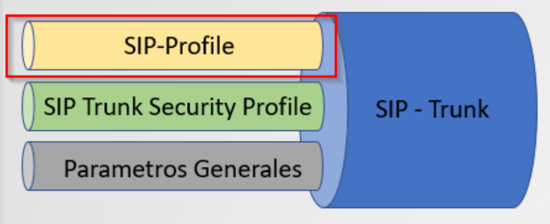
	Parameter Name
1	

External Presentation Information

- [ ] Anonymous External Presentation

External Presentation Number:

External Presentation Name:



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*	Never
Resource Priority Namespace List	< None >
SIP Rel1XX Options*	Disabled
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
- [ ] Enable External Presentation Name and Number
- [ ] Reject Anonymous Incoming Calls
- [ ] Reject Anonymous Outgoing Calls
- [ ] Send ILS Learned Destination Route String
- [ ] Connect Inbound Call before Playing Queuing Announcement

SIP OPTIONS Ping

- [ ] Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

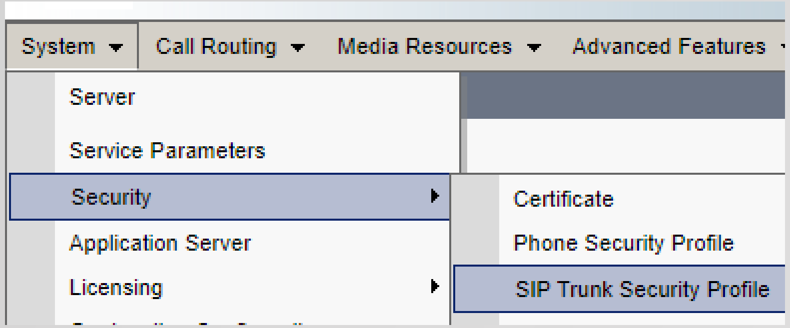
Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

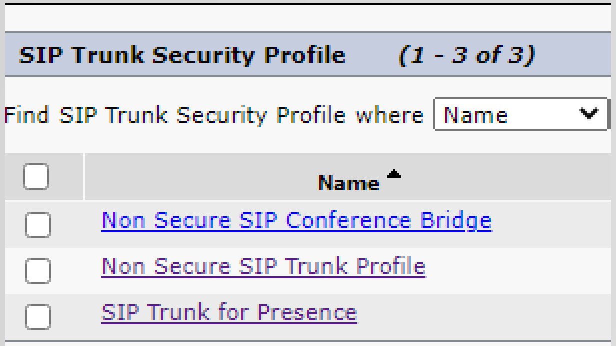
SDP Information

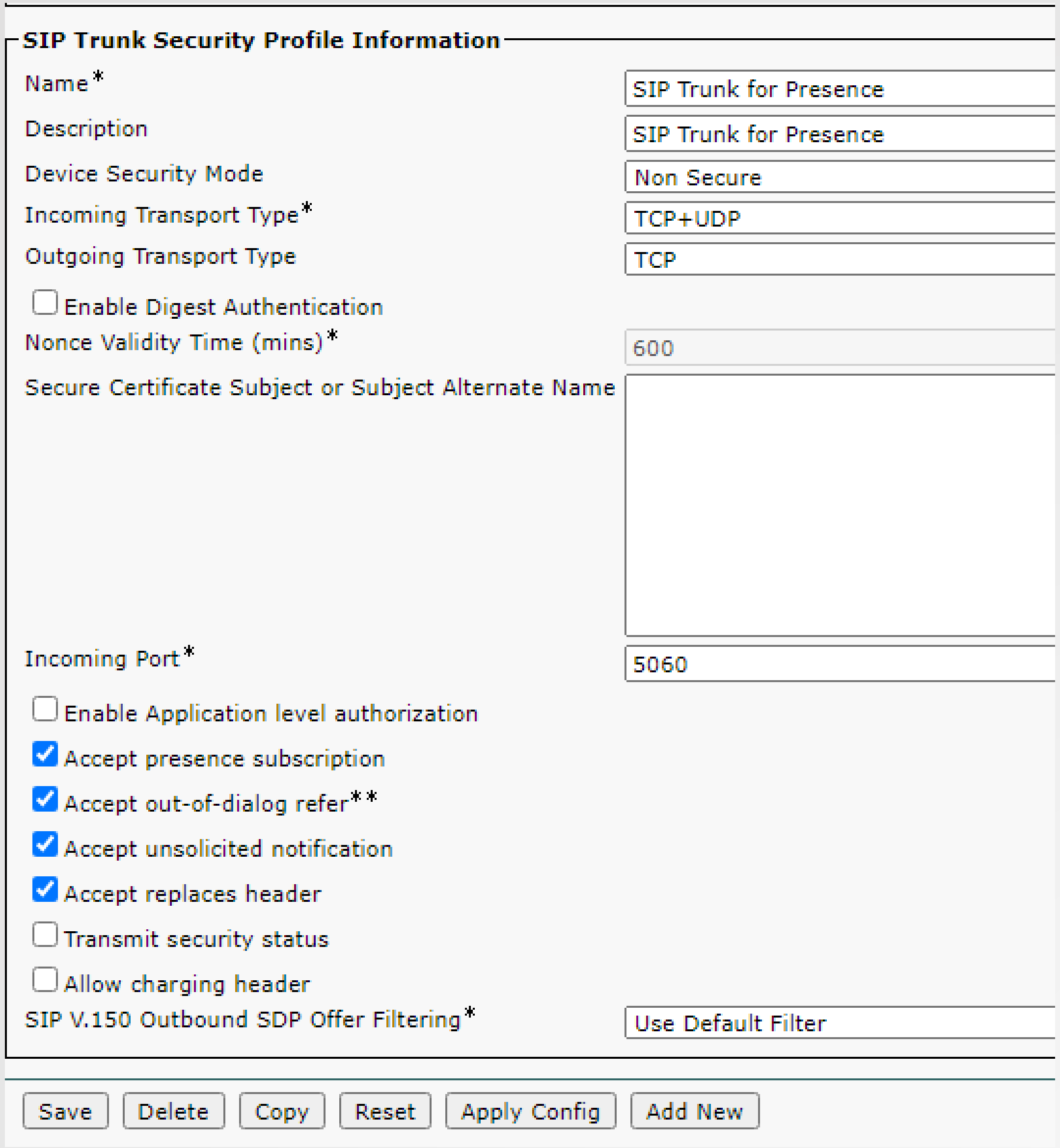
- [ ] Send send-receive SDP in mid-call INVITE
- [ ] Allow Presentation Sharing using BFCP
- [ ] Allow iX Application Media
- [ ] Allow multiple codecs in answer SDP

CUCM – SIP Security Profile

SIP Trunk Security Profile







CUCM proporciona un perfil de seguridad SIPtrunk no seguro y predefinido para el registro automático.

Device Pool

CUCM – Device Pool

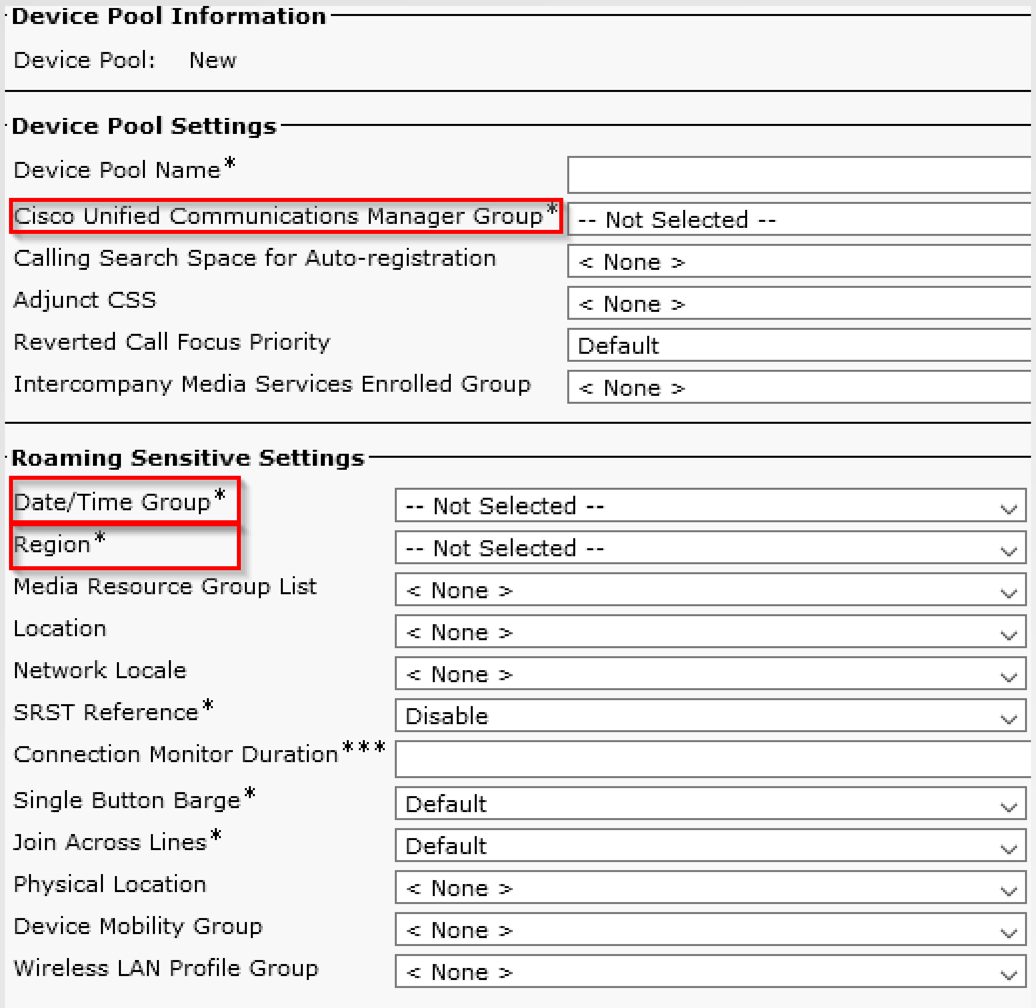
- Requerido
- Opcional

Propósito:

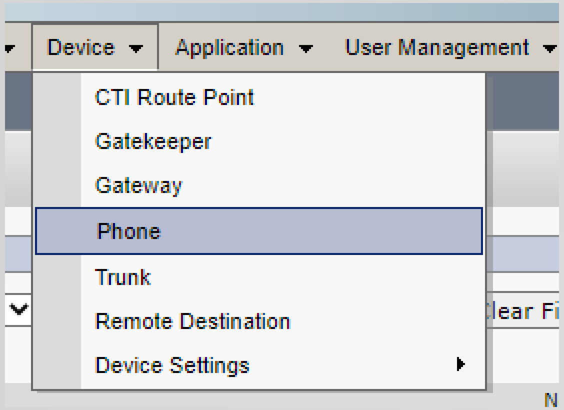
- Brindar una manera conveniente de definir un conjunto de características comunes que se pueden agregar a los dispositivos, en lugar de asignarlas como características individuales

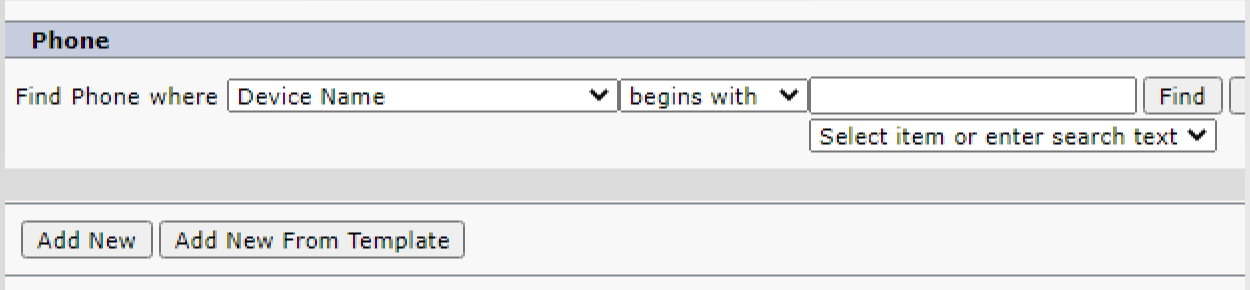
Previo a Configurar el Device Pool, se deben configurar:

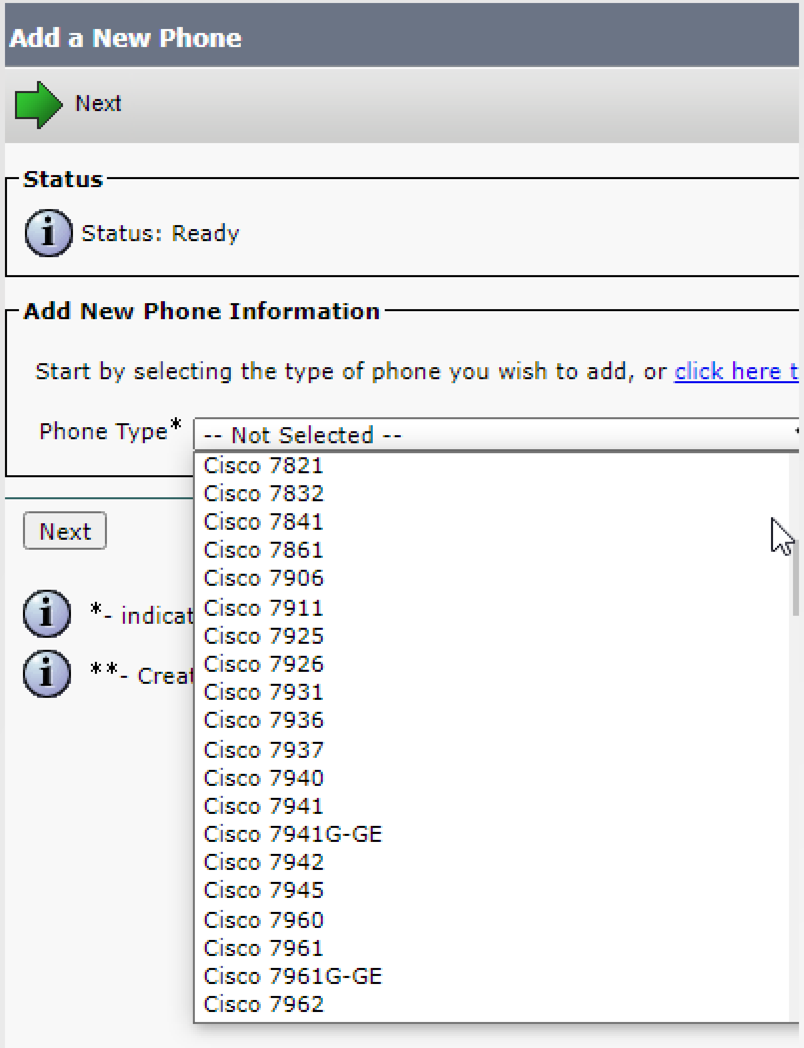
- Cisco Unified Communications Manager group.
  - Lista de servidores para Registración del Teléfono
- Date/time group.
- Region.
  - Se usa si se quiere frozar el uso de codecs entre diferentes regiones logicas . De lo contrario se usa el Defualt
- **SRST reference.**
- **Media resource group list.**
- **Calling search space for auto-registration.**



CUCM – IP Phone







CUCM – IP Phone

Phone Type

Product Type: **Cisco 7821**
Device Protocol: **SIP**

Device Information

☑ Device is trusted

MAC Address*	
Description	
☐ Require Activation Code for Onboarding	
Device Pool*	-- Not Selected --
Common Device Configuration	< None >
Phone Button Template*	-- Not Selected --
Softkey Template	< None >
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	< None >
User Hold MOH Audio Source	< None >
Network Hold MOH Audio Source	< None >
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >
Built In Bridge*	Default
Privacy*	Default
Device Mobility Mode*	Default
Owner	◉ User ○ Anonymous (Public/Shared Space)
Owner User ID*	
Mobility User ID	< None >
Phone Personalization*	Default
Services Provisioning*	Default

Protocol Specific Information

Packet Capture Mode*	None
Packet Capture Duration	0
BLF Presence Group*	Standard Presence group
SIP Dial Rules	< None >
MTP Preferred Originating Codec*	711ulaw
Device Security Profile*	-- Not Selected --
Rerouting Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile*	< None >
Digest User	< None >

☐ Media Termination Point Required
☐ Unattended Port
☐ Require DTMF Reception