



COMUNICACIONES 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)

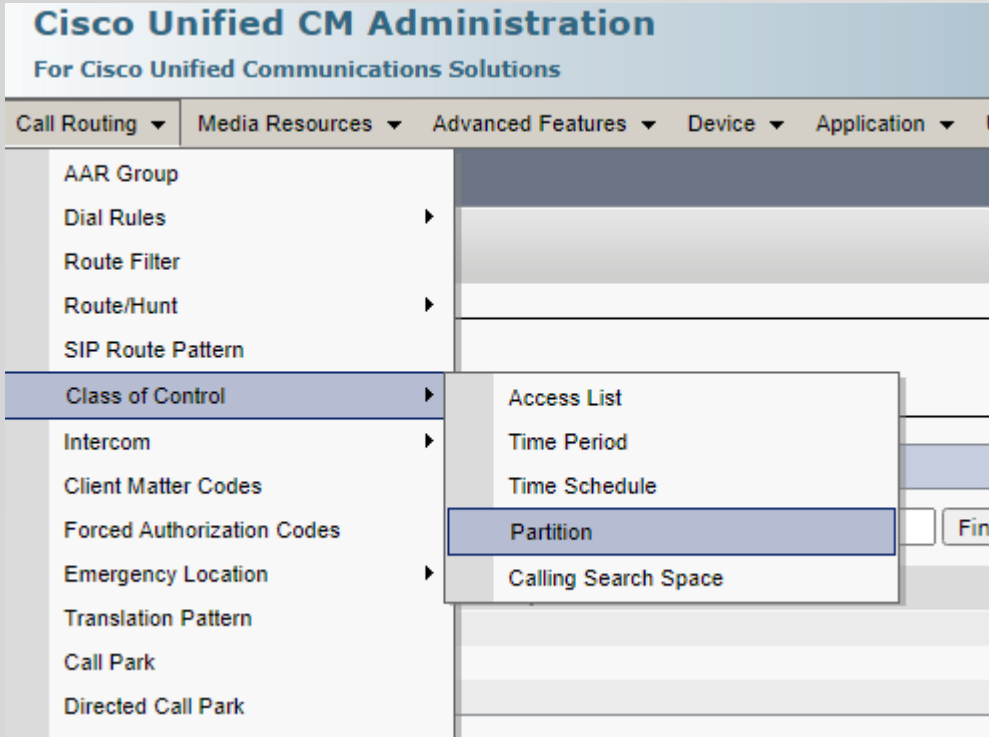
Para simplificar, los nombres de las particiones suelen reflejar sus características, como

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

¿Quién Puede llamarme?



CUCM -



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

<input type="checkbox"/>		
<input type="checkbox"/>	PT-Celualres	Particion para Celulares
<input type="checkbox"/>	PT-Internos	Particion para Telefonos Internos

CUCM - NO PARTITION / PARTITION

Phone Configuration

Save Delete Copy Reset Apply Config

Status

Status: Ready

Association

Modify Button Items

1 Line [1] - 2000 (no partition)

2 Line [2] - Add a new DN

3 Add a new SD

4 Add a new SD

5 Add a new SD

----- Add On Module(s) -----

6 Add a new SD

Phone Type

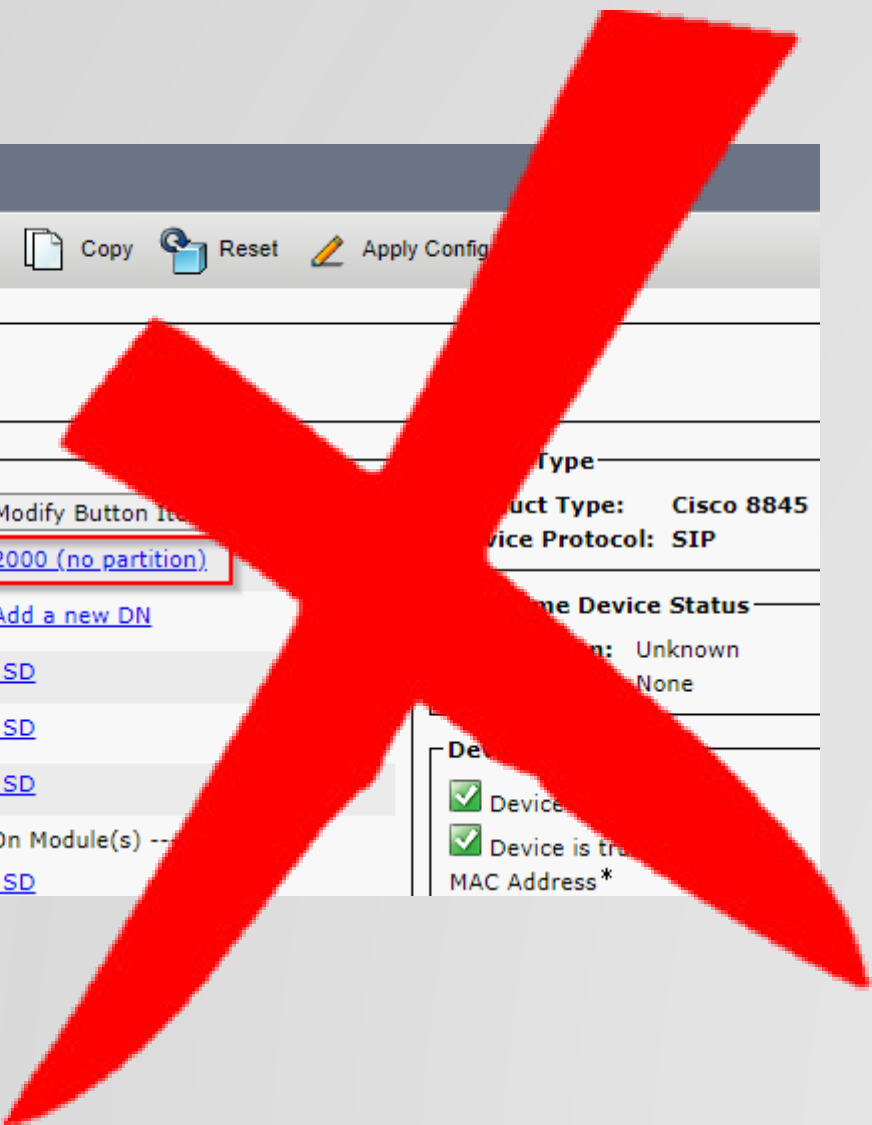
Product Type: Cisco 8845
Device Protocol: SIP

Real-time Device Status

Registration: Unknown
IPv4 Address: None

Device Information

☒ Device is Active
☒ Device is trusted
MAC Address*



Phone Configuration

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

Association

Modify Button Items

1 Line [1] - 2000 in PT-Internos

2 Line [2] - Add a new DN

3 Add a new SD

4 Add a new SD

5 Add a new SD

----- Add On Module(s) -----

6 Add a new SD

Phone Type

Product Type: Cisco 8845
Device Protocol: SIP

Real-time Device Status

Registration: Unknown
IPv4 Address: None

Device Information

☒ Device is Active
☒ Device is trusted
MAC Address*

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 .

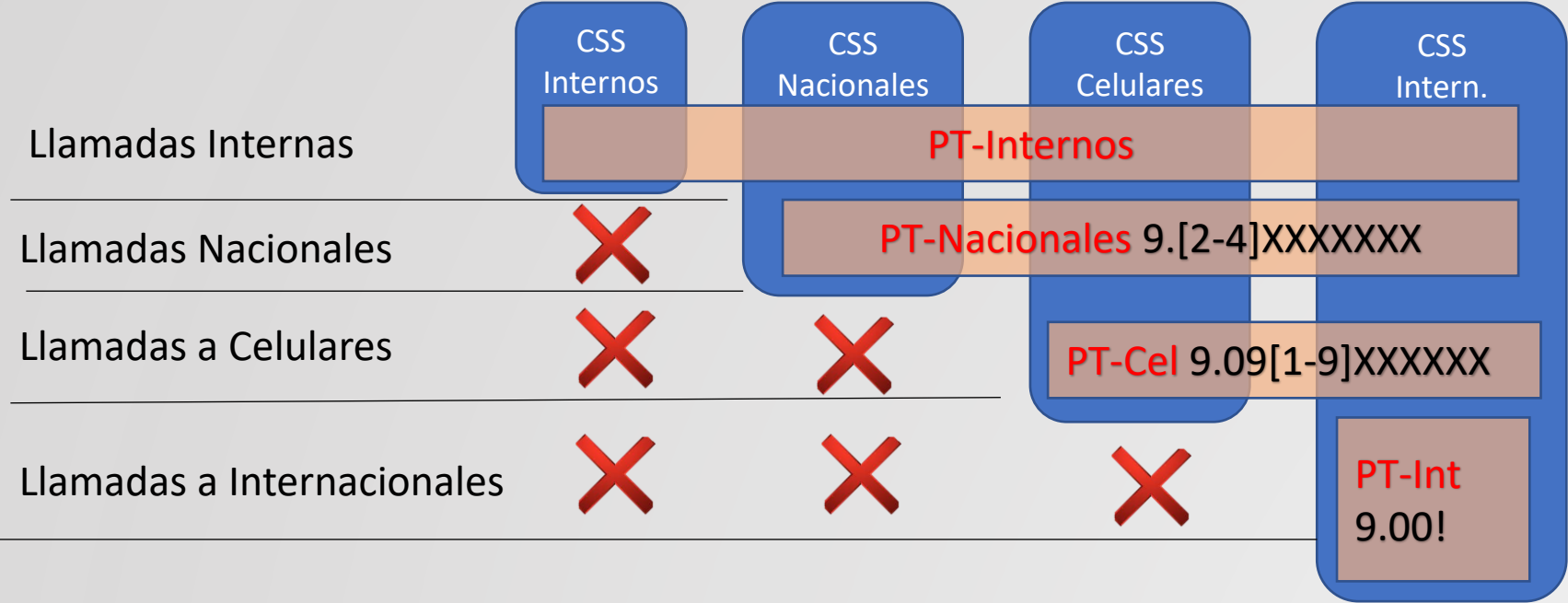
Un CSS ,con nombre “Supervisores” contiene 4 particiones : **Nacionales, Internacionales , Emergencia ,Celulares**

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.

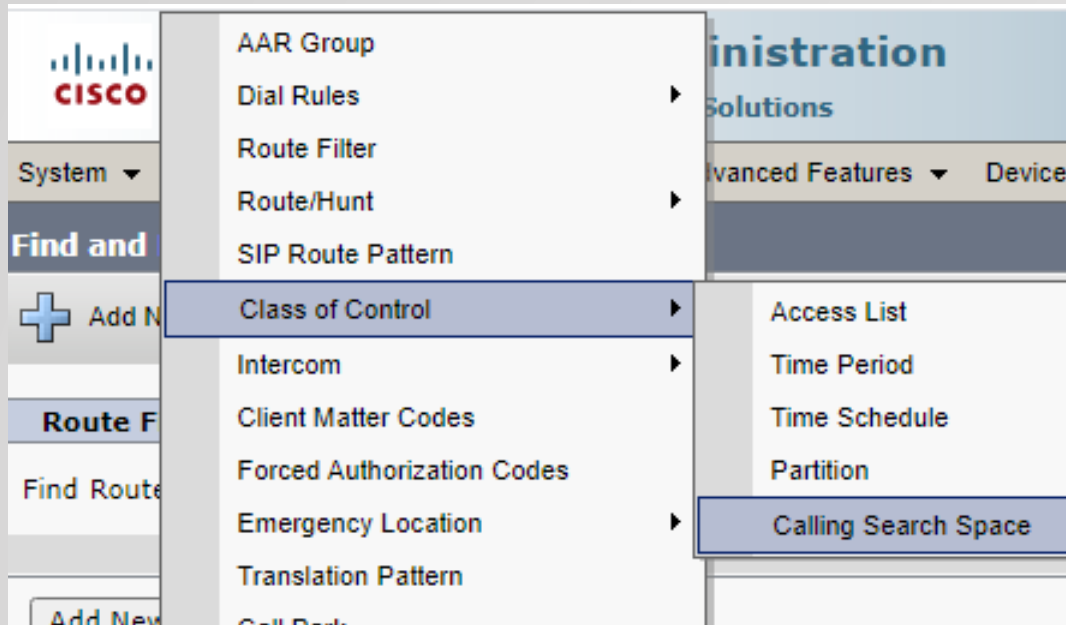
CUCM – CSS / Particiones



- 1. A donde puede llamar alguien
- 2. Ruteo Geográfico

Lista de Particiones

CUCM - CSS



Calling Search Space Configuration

Save

Status

Status: Ready

Calling Search Space Information

Name*

Description

Route Partitions for this Calling Search Space

Available Partitions**

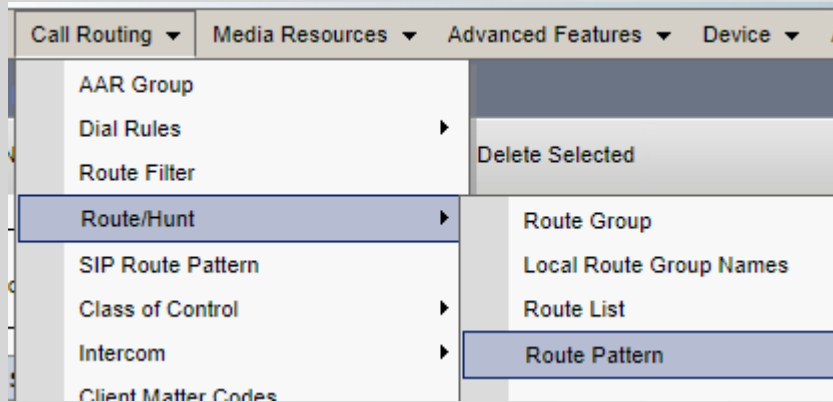
- Directory URI
- Global Learned E164 Numbers
- Global Learned E164 Patterns
- Global Learned Enterprise Numbers
- Global Learned Enterprise Patterns

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 <input type="text"/> <input type="button" value="Find"/> <input type="button" value="Clear Filter"/> <input type="button" value="+"/> <input type="button" value="-"/>			
<input type="checkbox"/>	CSS Name ^	Description	Copy
<input type="checkbox"/>	CSS-InternosYCEL	Permisos para llamadas a internos y Celulares	

CUCM – Route Patterns



Pattern Definition

Route Pattern* 9.09[1-9]XXXXXX

Route Partition PT-Celualres

Description Route Pattern Para Celulares

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* SIPTrunktoCUP

Route Option

☒ Route this pattern

☐ Block this pattern No Error

Call Classification* OffNet

External Call Control Profile < None >

☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

☐ Require Forced Authorization Code

Authorization Level* 0

☐ Require Client Matter Code

(Edit)

Route Patterns (1 - 1 of 1)						Rows per Page 50	
Find Route Patterns where <input type="text" value="Pattern"/> begins with <input type="text"/> <input type="button" value="Find"/> <input type="button" value="Clear Filter"/> <input type="button" value="+"/> <input type="button" value="-"/>							
<input type="checkbox"/>	Pattern ^	Description	Partition	Route Filter	Associated Device	Copy	
<input type="checkbox"/>	9.09[1-9]XXXXXX	Route Pattern Para Celulares	PT-Celualres		SIPTrunktoCUP		

CUCM - Route Patterns

Calling Party Transformations	
<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	<input type="text"/>
Prefix Digits (Outgoing Calls)	<input type="text"/>
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager
Connected Party Transformations	
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Called Party Transformations	
Discard Digits	< None >
Called Party Transform Mask	<input type="text"/>
Prefix Digits (Outgoing Calls)	<input type="text"/>
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager
ISDN Network-Specific Facilities Information Element	
Network Service Protocol	-- Not Selected --
Carrier Identification Code	<input type="text"/>
Network Service	Service Parameter Name

Called Party Transformations	
Discard Digits	< None >
Called Party Transform Mask	< None >
Prefix Digits (Outgoing Calls)	NoDigits
Called Party Number Type*	PreDot
Called Party Numbering Plan*	10-10-Dialing
	PreDot 10-10-Dialing
	PreAt
	PreAt 10-10-Dialing
	11D->10D
	PreDot 11D->10D
Network Service Protocol	PreDot 11/10D->7D
Carrier Identification Code	PreAt 11D->10D
	PreAt 11/10D->7D
Network Service	Intl TollBypass
-- Not Selected --	PreDot IntlTollBypass
	PreAt IntlTollBypass
	11/10D->7D
	Trailing-#
	PreDot Trailing-#
	10-10-Dialing Trailing-#
	PreDot 10-10-Dialing Trailing-#

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

USUARIO



- Linea 1 1000 PT_INTERNO
- Linea 2 2000 PT_INTERNO
- CSS:
 - PT_SECRETARIA
 - PT_INTERNO
 - PT_CELULARES
 - PT_NACIONALES

SECRETARIA



- Linea 1 2000 PT_SECRETARIA
- Linea 2 3000 PT_INTERNO
- CSS:
 - PT_JEFE
 - PT_INTERNO
 - PT_CELULARES

JEFE



- Linea 1 3000 PT_JEFE
- CSS:
 - PT_INTERNO
 - PT_SECRETARIA
 - PT_NACIONALES

Path Selection



CUCM – Route Group

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

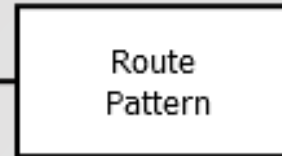
Un route group puede ser configurado para manejar estas distribuciones

- **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

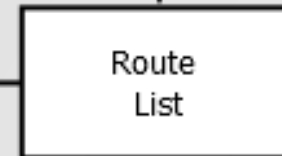
Route Pattern

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



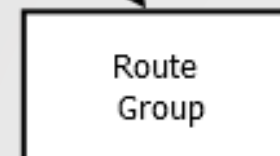
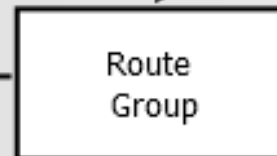
Route List

- * 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 dispositivo real.



Primera
Opción

Segunda
Opción

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 ahí 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

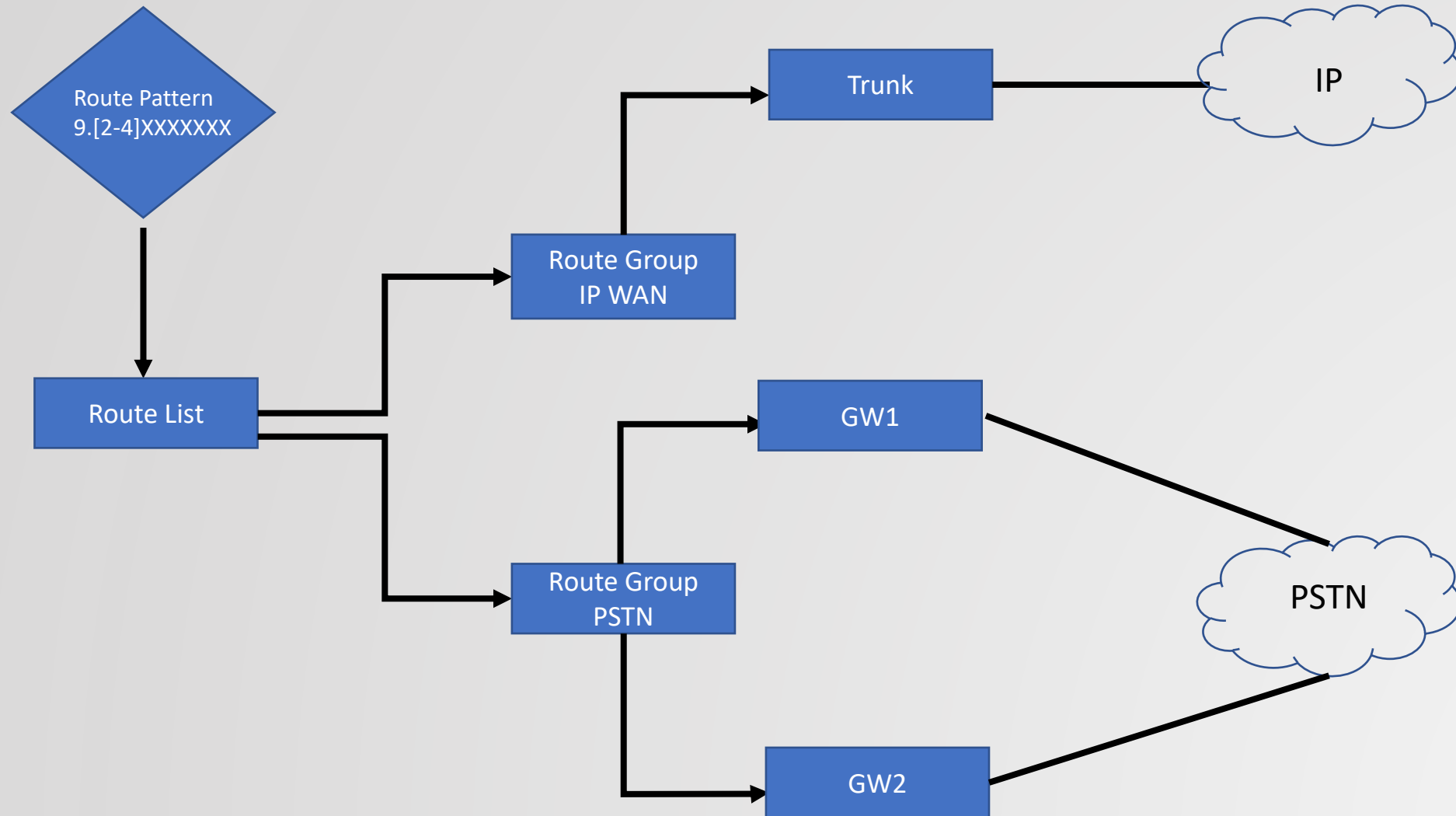
Paso 1 Agregar Dispositivos (Gateways y trunks).

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 -

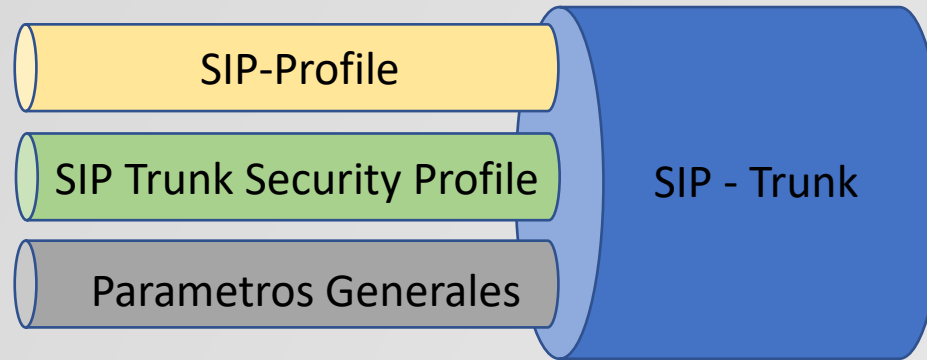





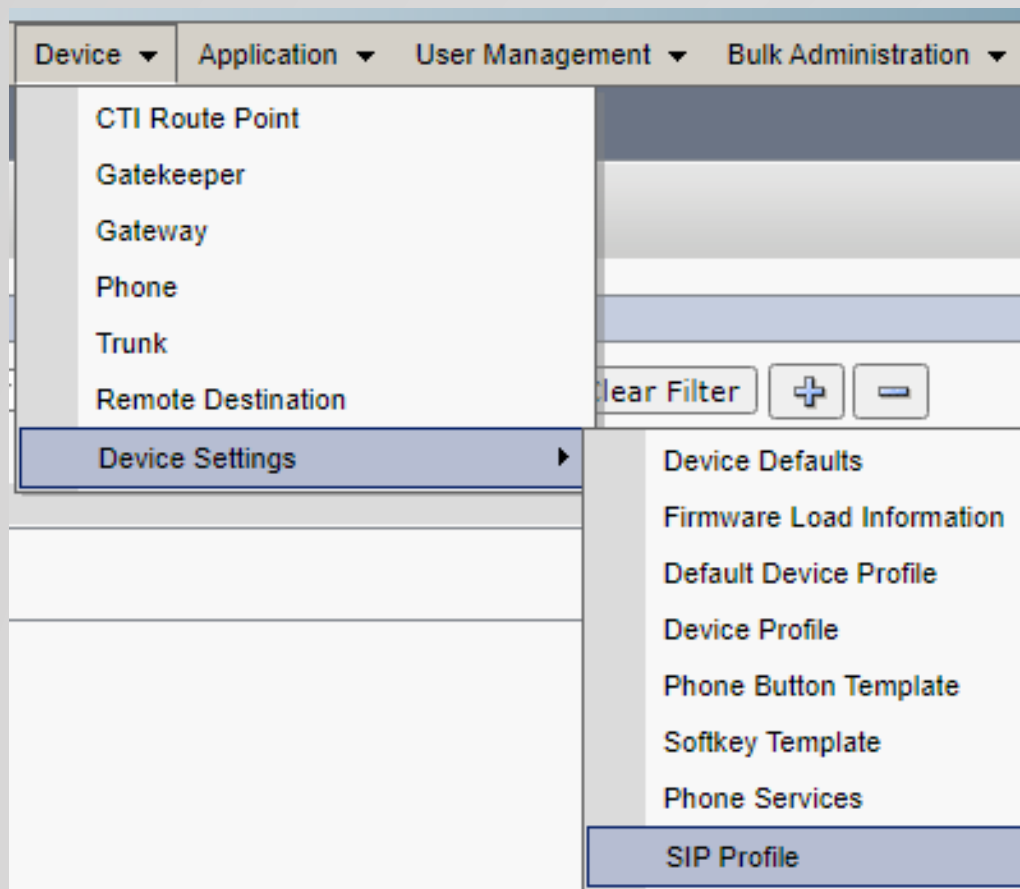
SIP TRUNK:

SIP Profile &
Sip Trunk Security Profile

CUCM – Sip Profile

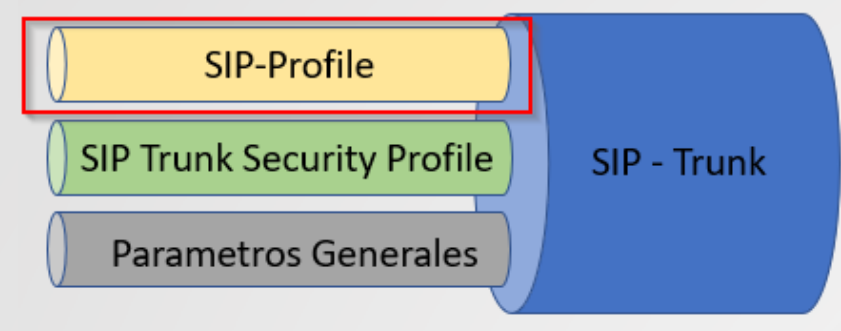


- Un perfil SIP (**Sip-Profile**) : 
- Es un **template** que comprende configuraciones SIP comunes.
- Al implementar troncales SIP (**Sip-Trunk**) en la red , se puede aplicar configuraciones SIP comunes a grupos de dispositivos a través del **SIP-Profile**.
- Sin el perfil SIP, tendría que configurar 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 <input type="text"/> begins with <input type="text"/> Find Clear Filter <input type="button" value="+"/> <input type="button" value="-"/>		
<input type="checkbox"/>	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**

SIP-Profile

SIP Trunk Security Profile

Parametros Generales

SIP - Trunk

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

☒ Conference Join Enabled

☐ RFC 2543 Hold

☒ 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

SIP-Profile

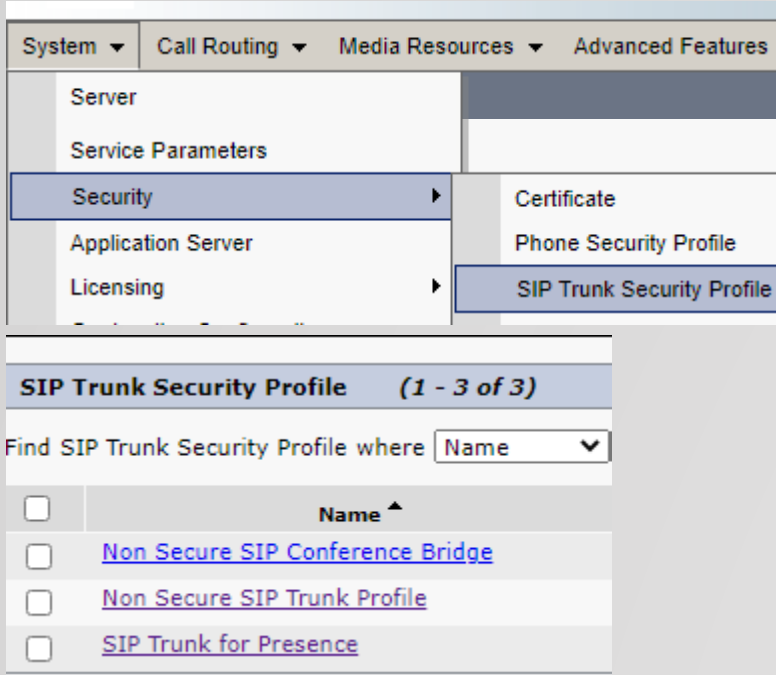
SIP Trunk Security Profile

Parametros Generales

SIP - Trunk

CUCM – SIP Security Profile

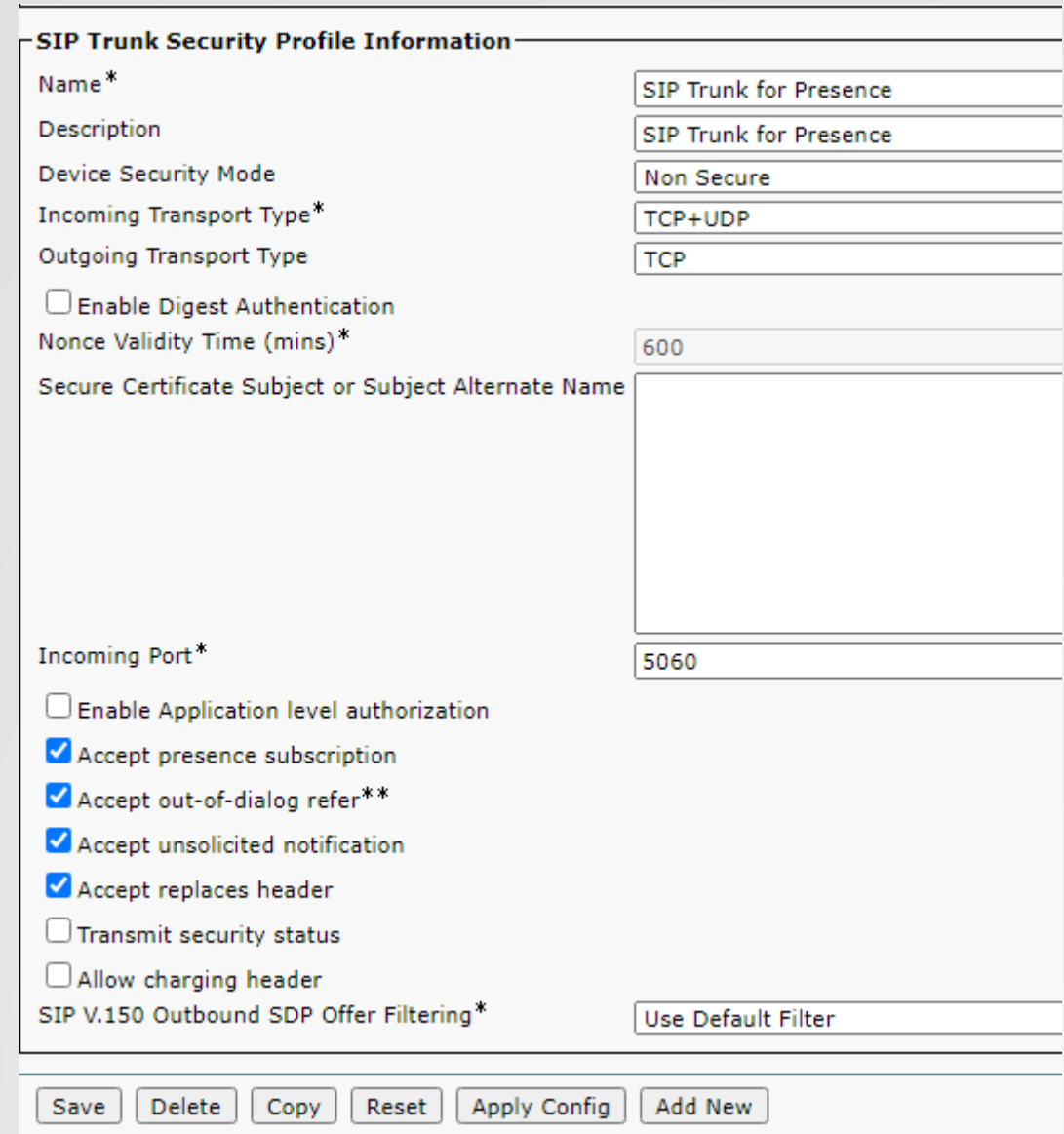
SIP Trunk Security Profile



The screenshot shows the CUCM navigation menu on the left with 'Security' selected, leading to 'SIP Trunk Security Profile'. Below this, a list of profiles is shown with 'SIP Trunk for Presence' selected.

SIP Trunk Security Profile (1 - 3 of 3)	
Find SIP Trunk Security Profile where <input type="text" value="Name"/>	
<input type="checkbox"/>	Name ^
<input type="checkbox"/>	Non Secure SIP Conference Bridge
<input type="checkbox"/>	Non Secure SIP Trunk Profile
<input checked="" type="checkbox"/>	SIP Trunk for Presence

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



The screenshot shows the configuration page for the 'SIP Trunk Security Profile Information'. The profile is named 'SIP Trunk for Presence' and is set to 'Non Secure' mode. It uses 'TCP+UDP' for incoming transport and 'TCP' for outgoing transport. The 'Enable Digest Authentication' checkbox is unchecked, and the 'Nonce Validity Time (mins)' is set to 600. The 'Secure Certificate Subject or Subject Alternate Name' field is empty. The 'Incoming Port' is set to 5060. The 'Enable Application level authorization' checkbox is unchecked, and the 'Accept presence subscription', 'Accept out-of-dialog refer**', 'Accept unsolicited notification', and 'Accept replaces header' checkboxes are checked. The 'Transmit security status' and 'Allow charging header' checkboxes are unchecked. The 'SIP V.150 Outbound SDP Offer Filtering' is set to 'Use Default Filter'.

SIP Trunk Security Profile Information	
Name*	SIP Trunk for Presence
Description	SIP Trunk for Presence
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	TCP
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application level authorization	
<input checked="" type="checkbox"/> Accept presence subscription	
<input checked="" type="checkbox"/> Accept out-of-dialog refer**	
<input checked="" type="checkbox"/> Accept unsolicited notification	
<input checked="" type="checkbox"/> Accept replaces header	
<input type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter

Save Delete Copy Reset Apply Config Add New

Device Pool



CUCM – Device Pool

- Requerido
- Opcional

Propósito:

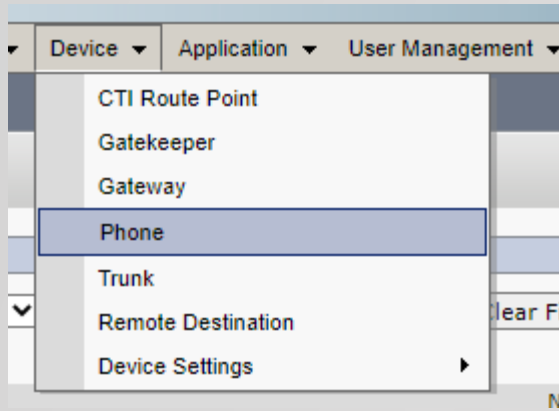
- Brindar una manera conveniente de definir un conjunto de características comunes que se pueden agregar a los dispositivos, en lugar de asignarlas como características individuales

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 forzar el uso de codecs entre diferentes regiones logicas . De lo contrario se usa el Default
- SRST reference.
- Media resource group list.
- Calling search space for auto-registration.

Device Pool Information	
Device Pool:	New
Device Pool Settings	
Device Pool Name*	
Cisco Unified Communications Manager Group*	-- Not Selected --
Calling Search Space for Auto-registration	< None >
Adjunct CSS	< None >
Reverted Call Focus Priority	Default
Intercompany Media Services Enrolled Group	< None >
Roaming Sensitive Settings	
Date/Time Group*	-- Not Selected --
Region*	-- Not Selected --
Media Resource Group List	< None >
Location	< None >
Network Locale	< None >
SRST Reference*	Disable
Connection Monitor Duration***	
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >
Wireless LAN Profile Group	< None >

CUCM – IP Phone



Phone

Find Phone where

Add a New Phone

Status

Status: Ready

Add New Phone Information

Start by selecting the type of phone you wish to add, or [click here to view the phone types](#)

Phone Type*

*- indicates a phone that is not supported in this version of CUCM

**- Create a new phone type

- Cisco 7821
- Cisco 7832
- Cisco 7841
- Cisco 7861
- Cisco 7906
- Cisco 7911
- Cisco 7925
- Cisco 7926
- Cisco 7931
- Cisco 7936
- Cisco 7937
- Cisco 7940
- Cisco 7941
- Cisco 7941G-GE
- Cisco 7942
- Cisco 7945
- Cisco 7960
- Cisco 7961
- Cisco 7961G-GE
- Cisco 7962

CUCM – IP Phone

Phone Type	
Product Type: Cisco 7821	
Device Protocol: SIP	
Device Information	
<input checked="" type="checkbox"/> Device is trusted	
MAC Address*	
Description	
<input type="checkbox"/> Require Activation Code for Onboarding	
Device Pool*	-- Not Selected -- View
Common Device Configuration	< None > View
Phone Button Template*	-- Not Selected --
Softkey Template	< None >
Common Phone Profile*	Standard Common Phone Profile View
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	<input checked="" type="radio"/> User <input type="radio"/> 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 >
<input type="checkbox"/> Media Termination Point Required	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> Require DTMF Reception	