



Advanced Dial Plan Design

BRKUCC-3000

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Meet the Engineer

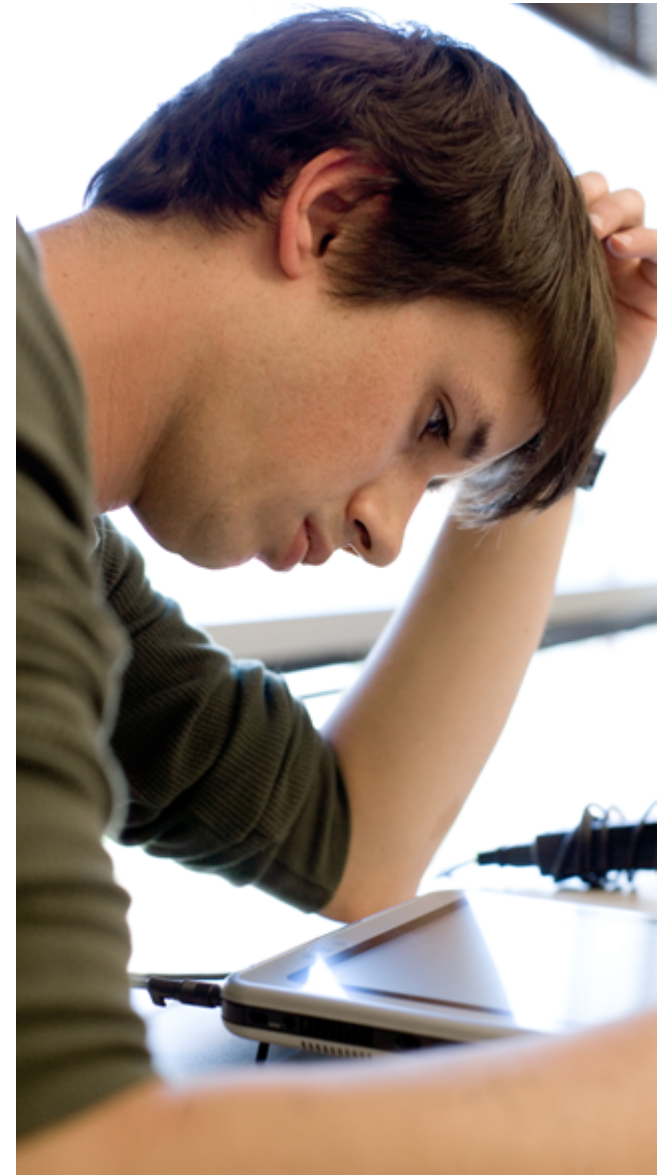
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Session Scope and Objectives

- To explore the various architectural challenges of planning an IP-based telephony network because it can do more than a traditional telephony system, because it breaks all the common boundaries (**few, if any, PBXs have hundreds of sites**)
- To explore the design and implementation possibilities of Cisco Unified Communications
 - Design based on Cisco Unified Communications Manager 4.X through 8.X
 - New functionality in 7.X and 8.X!!!!



Overall Agenda

- Planning Considerations



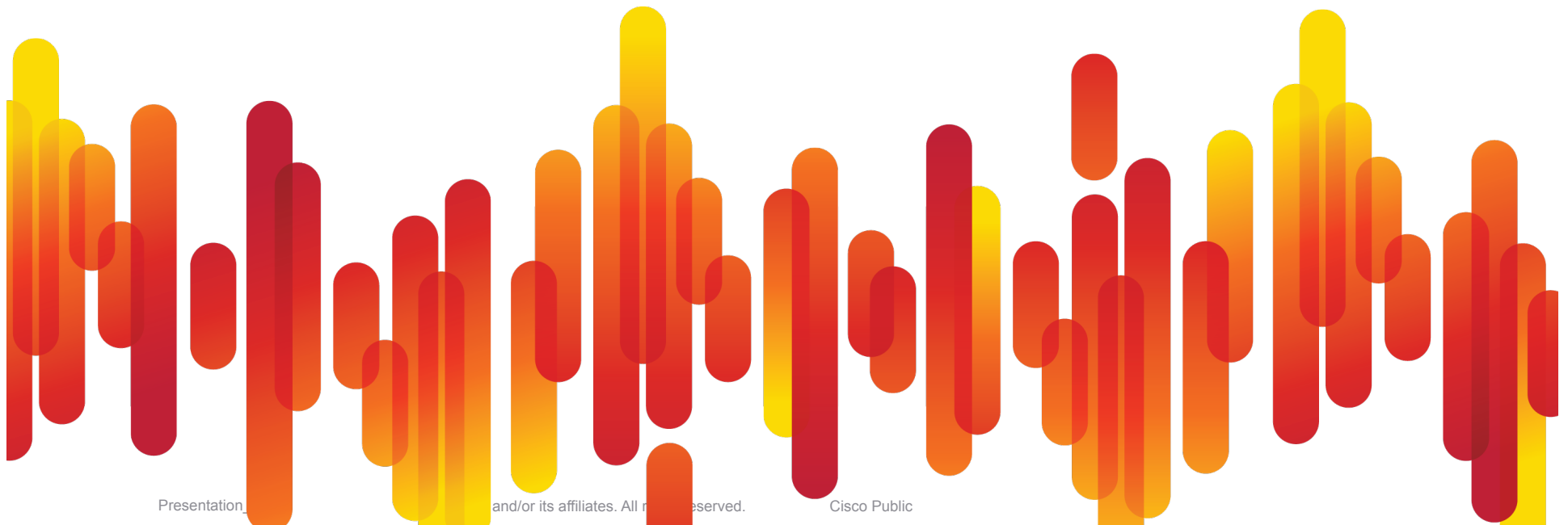
- Design Guidelines



- Conclusions



Planning Considerations



Dial Plan: Think Globally, Act Locally

- More than just a cute phrase: it actually applies to Unified Communications
- Even a **local** only company will make calls to, or receive calls from, international locations
 - Mobility users tend to travel: their mobile phone thus would be best equipped with global address books
 - You may have to click-to-dial international numbers from your CUPC (or other soft phone)
- Bottom line: we need, more than ever, to think of dial plans in global terms, and configure them in a locally significant way
- The next few slides offer food for thought, followed by the technical discussion

Dial Plan: Think Globally, Act Locally (Cont.)

- Manual dialing 0 0011 1 408 902 3574, **0 0011 1 418 723 1111**
- Manual dialing + 1 408 902 3574, **+1 418 723 1111**
- Manual dialing: 12345
- Click-to-dial + 1 408 902 3574 **+1 418 723 1111**
- From missed / received + 1 408 902 3574, **+1 418 723 1111**

- A call from Sydney may ring in as 011 61 2 8446 6000, whereas the received calls list shows +61 2 8446 6000

- Best to show calls coming from full E.164

- Manual dialing 0 00 1 408 902 3574, **0 00 1 418 723 1111**
- Dial from missed / received calls lists + 1 408 902 3574, **+1 418 723 1111**

- Manual dialing 0011 1 408 902 3574

Routing plan

- Called number
- Calling number

- Called/calling number: as per local requirements of PSTN

- On-net
- Off-net
- - Unified Mobility

Australian User 66000

French User 66000

Luc's Phone +1 408 902 3574 or 23574

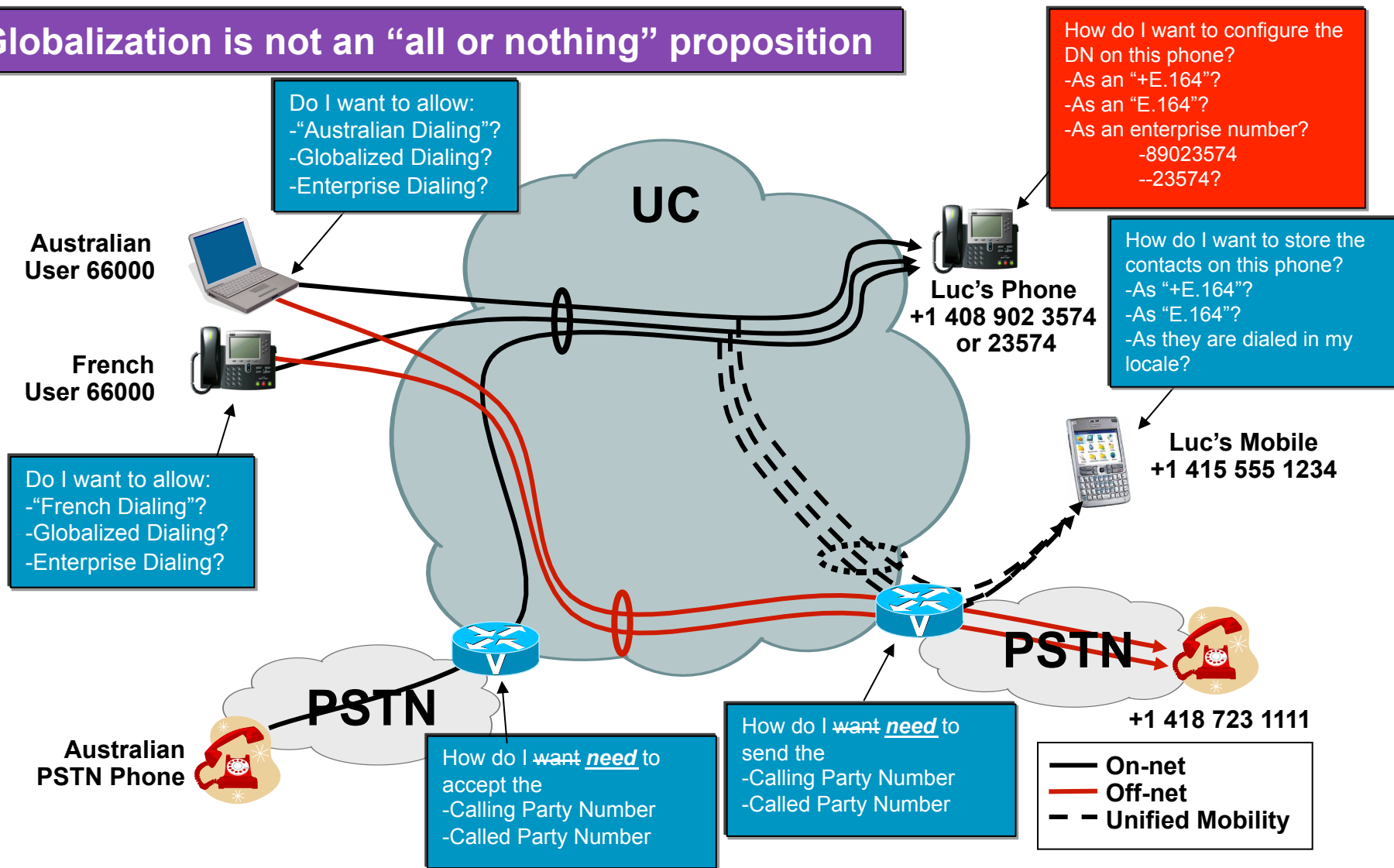
Luc's Mobile +1 415 555 1234

Australian PSTN Phone

+1 418 723 1111

A "Global" Dial Plan: not a monolith!

Globalization is not an "all or nothing" proposition



Dial Plan: Think Globally, Act Locally (Cont.)

- Dialing modes

We **must** accommodate local dialing habits: allow the Australians to dial the way they are used to, the French, the Canadians, etc... that means creating different telephony user interfaces for each dial-plan domain

We also should accommodate click-to-dial, directory dial, and other automated forms of dialing. Would it not be nice if these would work wherever you are roaming?

We should allow for edit-free dialing of missed and received calls—no matter what phone does it: French, Australian, Canadian, etc...

TUI is still site-specific if we allow for intra-site abbreviated dialing. Dialing 66000 in Paris must match Maurice, not Nigel in Sydney (whose local extension is 66000 also!)

- Routing plan

Route on called number, as always

Do we want to create and maintain routing tables on local dialing habits? No!

... instead, we want to create routing tables based on a universal, routable number

... that number can now be a full E.164 representation, including the + sign

... but applications may prevent you from using + on the DN itself

E.g.: Unity cannot accept the + (yet). Use vmail profiles to transit from +E.164 DNs to E.164 vmail box identifiers.

E.g.2: “cordon off” DNs used for Call Centre apps, or attendant console apps, etc... in their own partition (for the moment) until the apps support + natively

Dial Plan: Think Globally, Act Locally (Cont.)

- Decisions, Decisions, Decisions:

You do NOT have to adopt globalizations features as a monolith
Allowing globalized user input (e.g.: allowing a user to dial +14156134820 manually, from the missed/received calls lists, from contacts, click-to-call) is “risk free”.

Placing a +E.164 on a DN:

-**If you do**, be aware of the CTI and Messaging caveats (next two pages)

-**If you do not**, you *need* to manage the calling party numbers (e.g.: 89023574 calls a dual mode phone: what number do you want in the missed calls list? Hint: you need to hit dial from outside the enterprise...

-AND you need to manage how to match “89023574” when someone dials +14089023574

- Sending/receiving calls to/from an external network

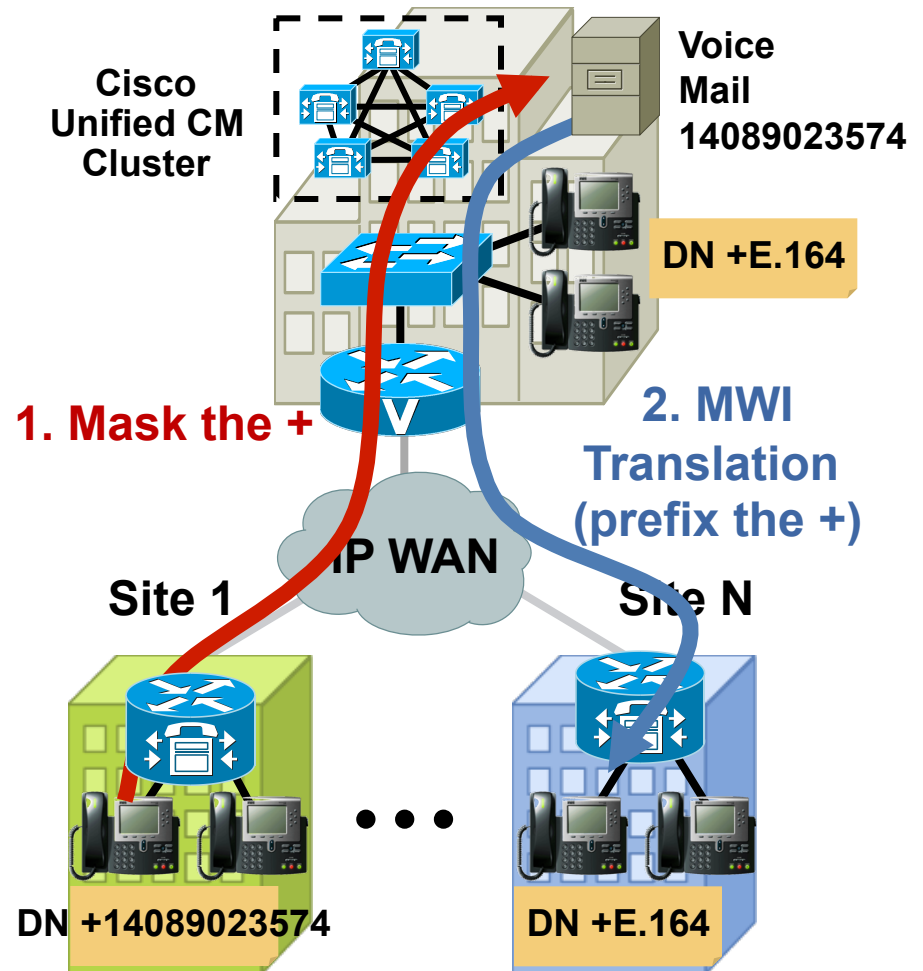
What form of the calling/called number will you be given by the external network when receiving calls?

What form of the calling/called number is expected/demanded by the external network?

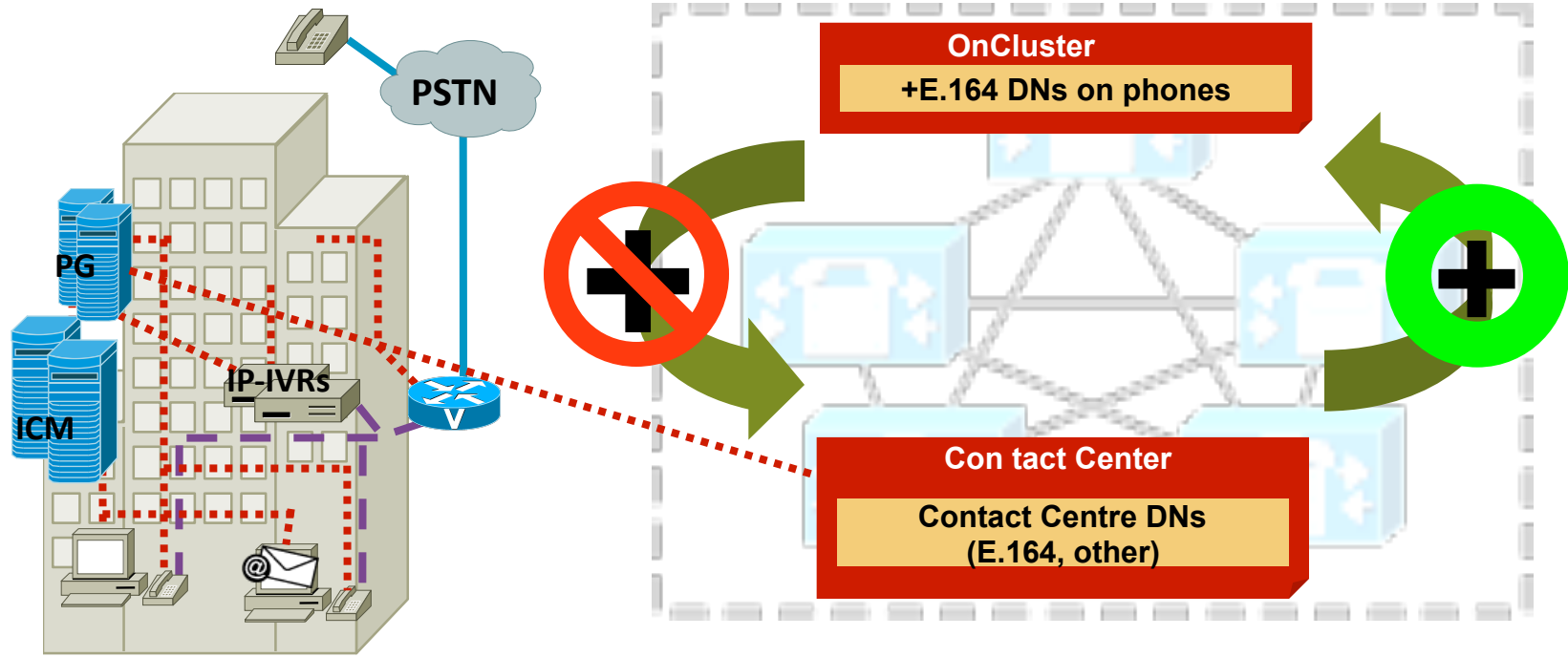
Dial Plan: Think Globally, Act Locally (Cont.)

Voice Mail Integration – When going from +E.164 to E.164

- When the DNs are +E.164, and the voice mail system does not support + (yet)
- Voice mail boxes need a unique DN
- Need to **mask off the + in the DNs** when accessing VM
- Message Waiting Indicator (MWI) messages from VM system need to be **prefixed with +** to match appropriate DN/partition



Dial Plan: Think Globally, Act Locally (Cont.)



- IP Voice
- TDM Voice
- ⋯ Call Control and CTI Data

For CTI-based apps not yet able to control +based DNs, use different partitions to separate the +DNs from the non + DNs. Use translation patterns to control calls between these groups of phones. Add + to the calling and called parties when calling from a CC phone to a +DN, and remove the + when calling *to* a CC phone.

Dial Plan: Think Globally, Act Locally (Cont.)

- Routing plan

For non-DID numbers, what to do?

Using “0” as a country code, you can have as many non-DID ranges as you want

This is the equivalent of using “RFC1918” addressing for your phones

Use “real” country codes as region codes

e.g.: +017085551000 is a non-DID number in US (1) , Chicago (708)...

Dial Plan: Think Globally, Act Locally (Cont.)

- Calling number

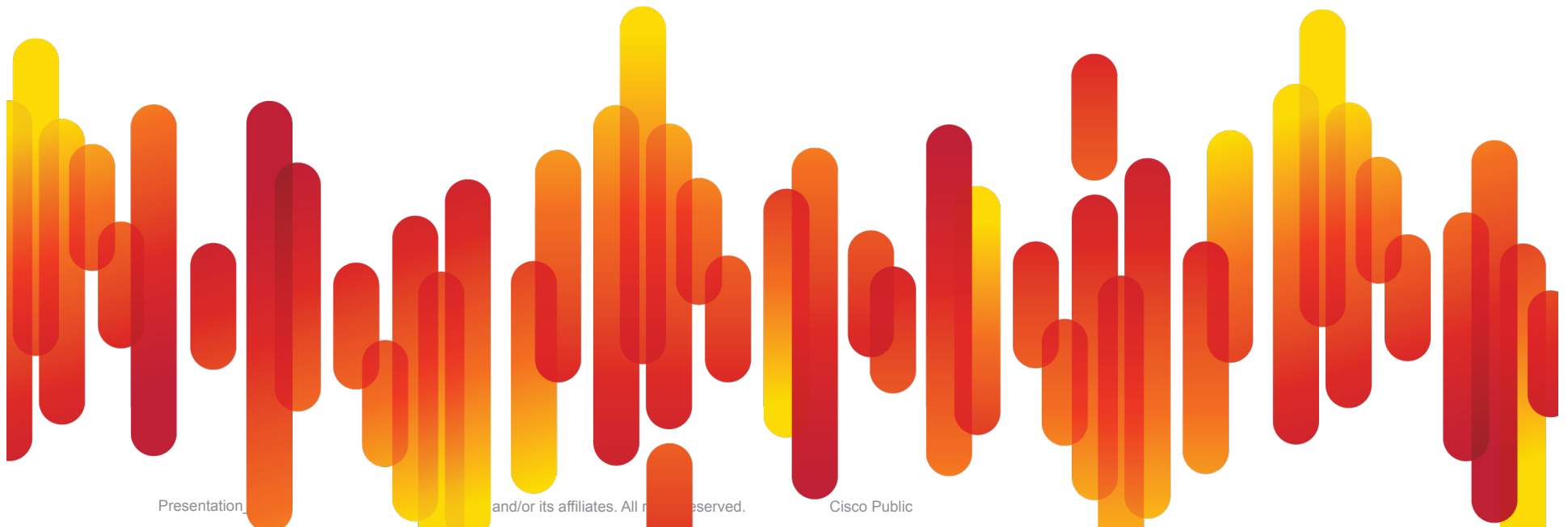
Calling number is best represented by a global form, which is used for call backs. E.g.: +1 408 902 3574 is stored in missed calls list if you get a call from me, as opposed to 0 0011 408 902 3574; that works on a French phone too!

form presented to user during ringing can be transformed to adapt to local habits: you **may** want 0 0011 1 408 902 3574 to be presented to an Aussie phone, whereas the French phone gets 00 1 408 902 3574

- Calling number

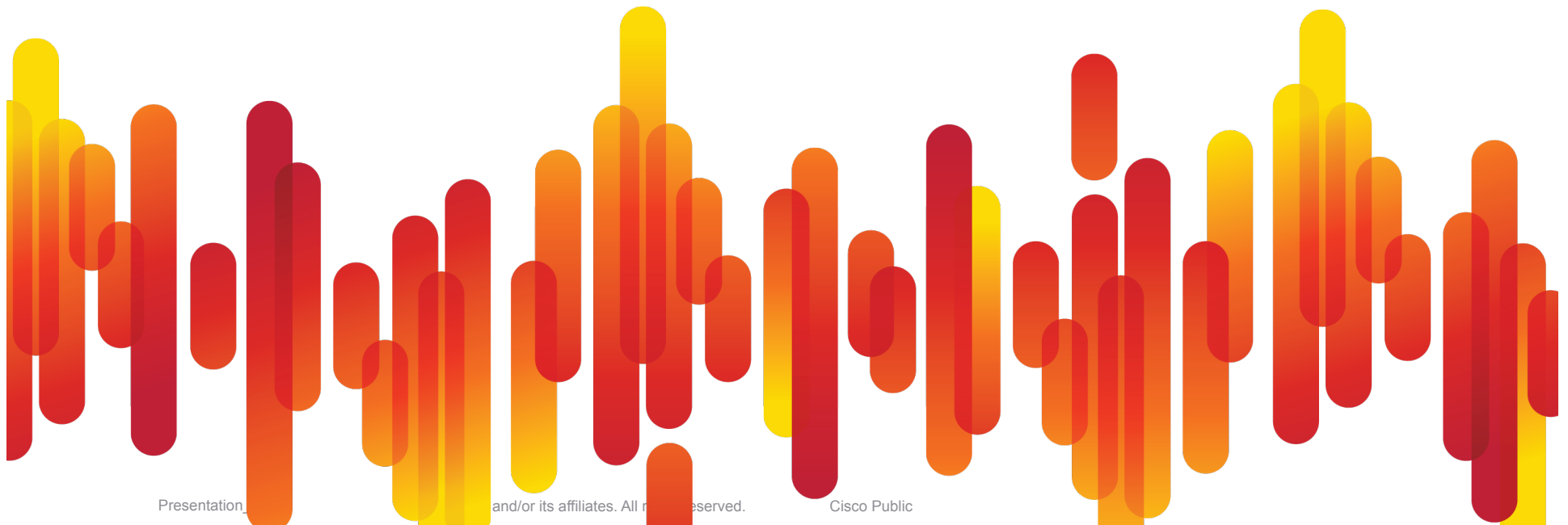
If a call is sent to a mobility user, and comes from an enterprise user, you may want to send the mobile phone a global representation of the calling number (i.e.: DID number of associated desk phone)

Design Guidelines



Design Guidelines Agenda

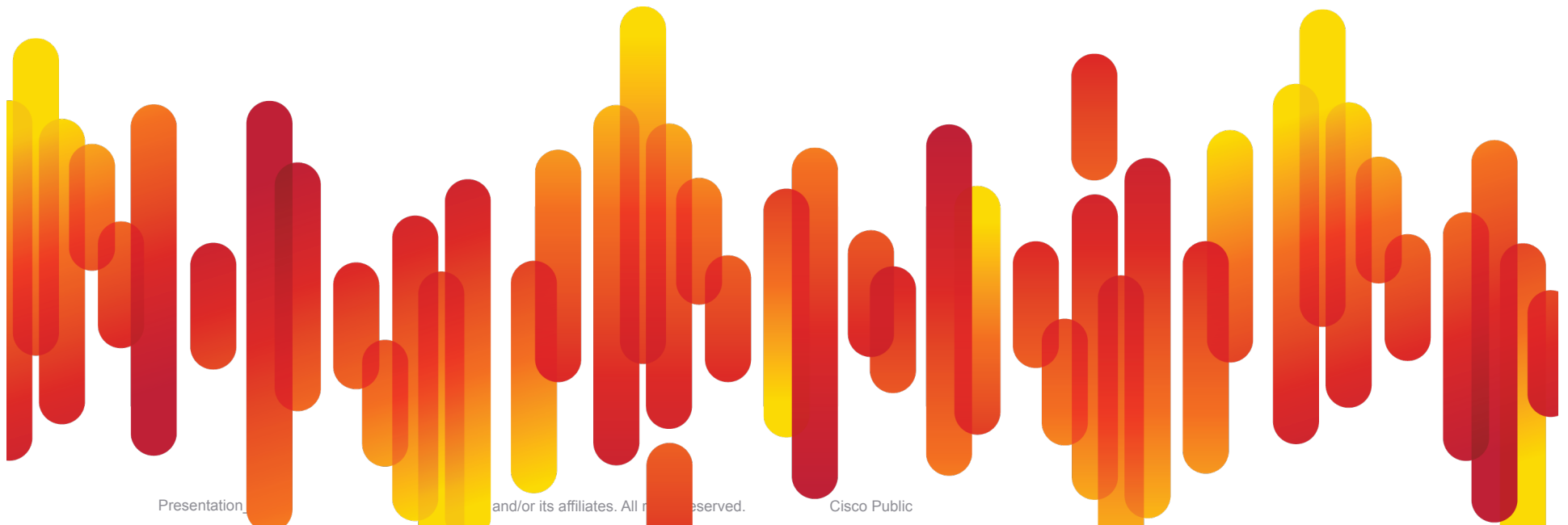
- 7.0 and 7.1 Updates
- 8.0 updates
- Multisite Deployments
- Mobility Considerations



Think Globally Act Locally, a.k.a.: 7.0 Update

- Local route group
- + sign support
- Calling/called number transformations
 - GW incoming call prefixing based on numbering plan
- Combined benefits

Local Route Group a Scalability Gem and an Enabler of Features



Local Route Group

What It Is: Concept

- Allow the site-specificity of call routing to be established by the calling device's location (as derived from device pool)
- Different endpoints in different sites would be associated with different local route groups: they can all call the same patterns, and the calls will be routed **differently, based on the caller's currently associated local route group**
- In practical terms, route patterns (i.e., patterns to **off-cluster destinations**) are no longer site-specific, and can be used for callers of different sites

Local Route Group

What It Is: Screen Shot

- Device pool is site-specific
- Local route group is associated with device pool
- Local route group is thus associated with all devices using a given device pool: e.g., phones, gateways

Device Pool Configuration - Mozilla Firefox

File Edit View History Bookmarks Tools Help

https://192.168.2.205:8443/ccmadmin/devicePoolEdit.do?key=93a9d36e-b22c-6869-

Customize Links Windows Marketplace

Google Search

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

System Call Routing Media Resources Voice Mail Device Application User Management Bulk

Device Pool Configuration

Save Delete Copy Reset Add New

Status
Status: Ready

Device Pool Information
Device Pool: sfo_device_pool (1 members**)

Device Pool Settings
Device Pool Name* sfo_device_pool
Cisco Unified Communications Manager Group* sfo_ucm_group
Calling Search Space for Auto-registration US_9011_911_dev_css
Reverted Call Focus Priority Default
Local Route Group sfo_local_route_group

Roaming Sensitive Settings
Date/Time Group* sfo_date_time
Region* sfo_region

Local Route Group

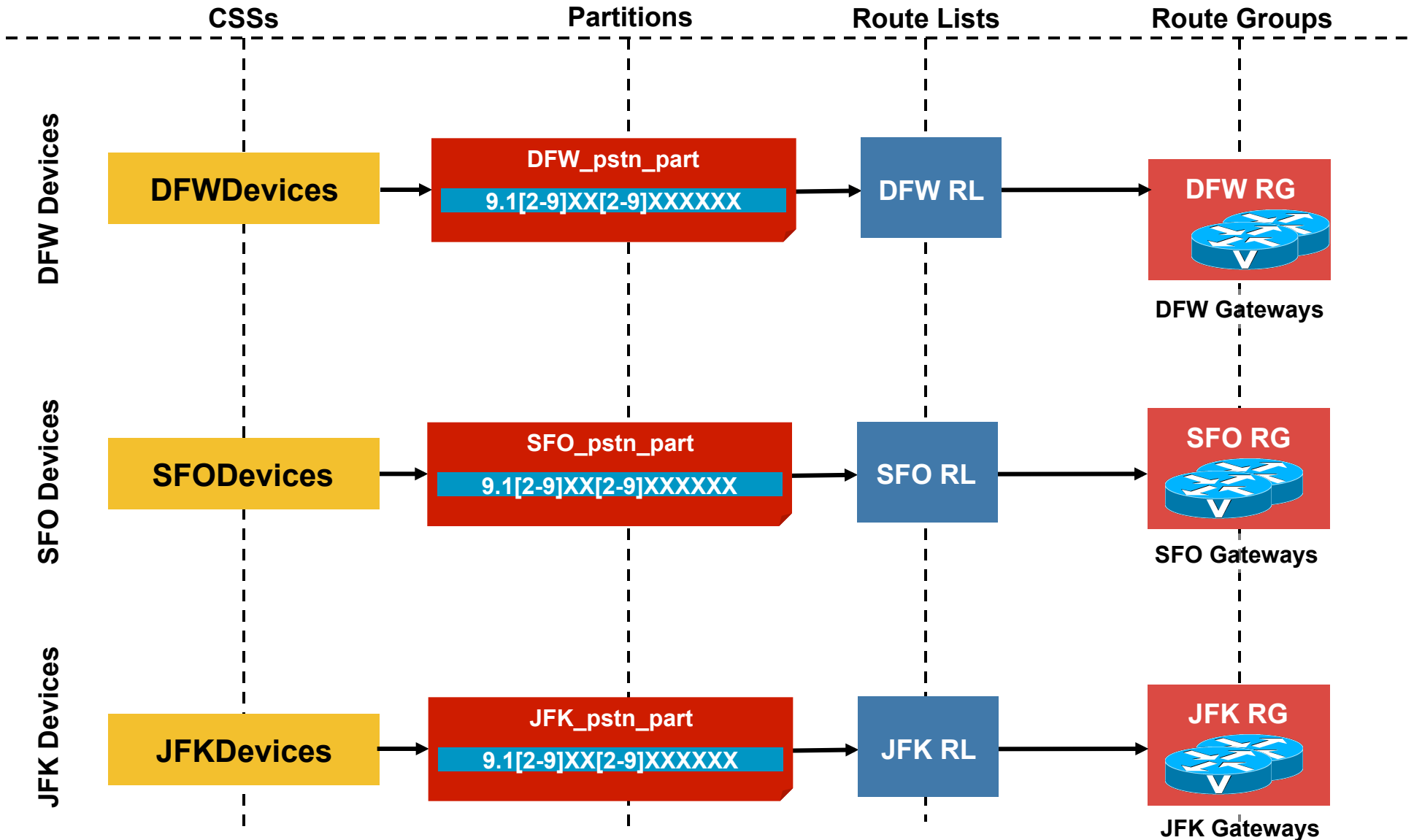
What It Is: Screen Shot

- Route lists can now refer to local route groups as well as regular route group
- Allows for simple local failover
- In this example, calls go to the centralised US GW (in site HQ), and fallback to the local route group

The screenshot shows the 'Route List Configuration' page. At the top, there are navigation buttons: Save, Delete, Copy, Reset, and Add New. The 'Status' section shows 'Status: Ready'. The 'Route List Information' section includes fields for Name* (US_LD_route_list), Description, and Cisco Unified Communications Manager Group* (Default). There is a checkbox for 'Enable this Route List' which is checked. The 'Route List Member Information' section has two list boxes: 'Selected Groups**' containing 'HQ_route_group' and 'Standard Local Route Group', and 'Removed Groups***' which is empty. An 'Add Route Group' button is next to the selected groups list. The 'Route List Details' section at the bottom lists the selected groups with their respective icons and names: 'HQ_route_group' and 'Standard Local Route Group'.

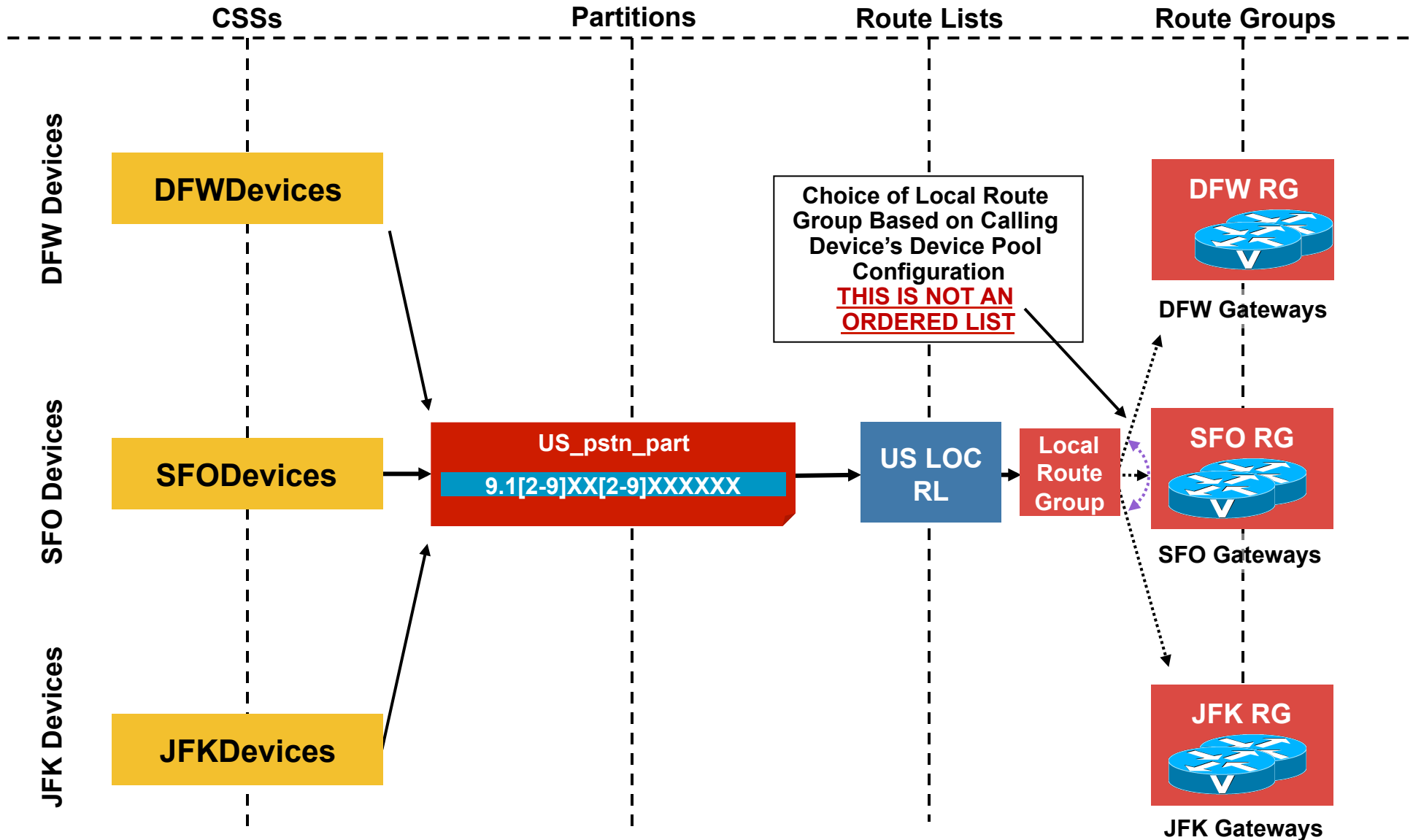
Local Route Group

Without It: GW Chosen by the Route Pattern



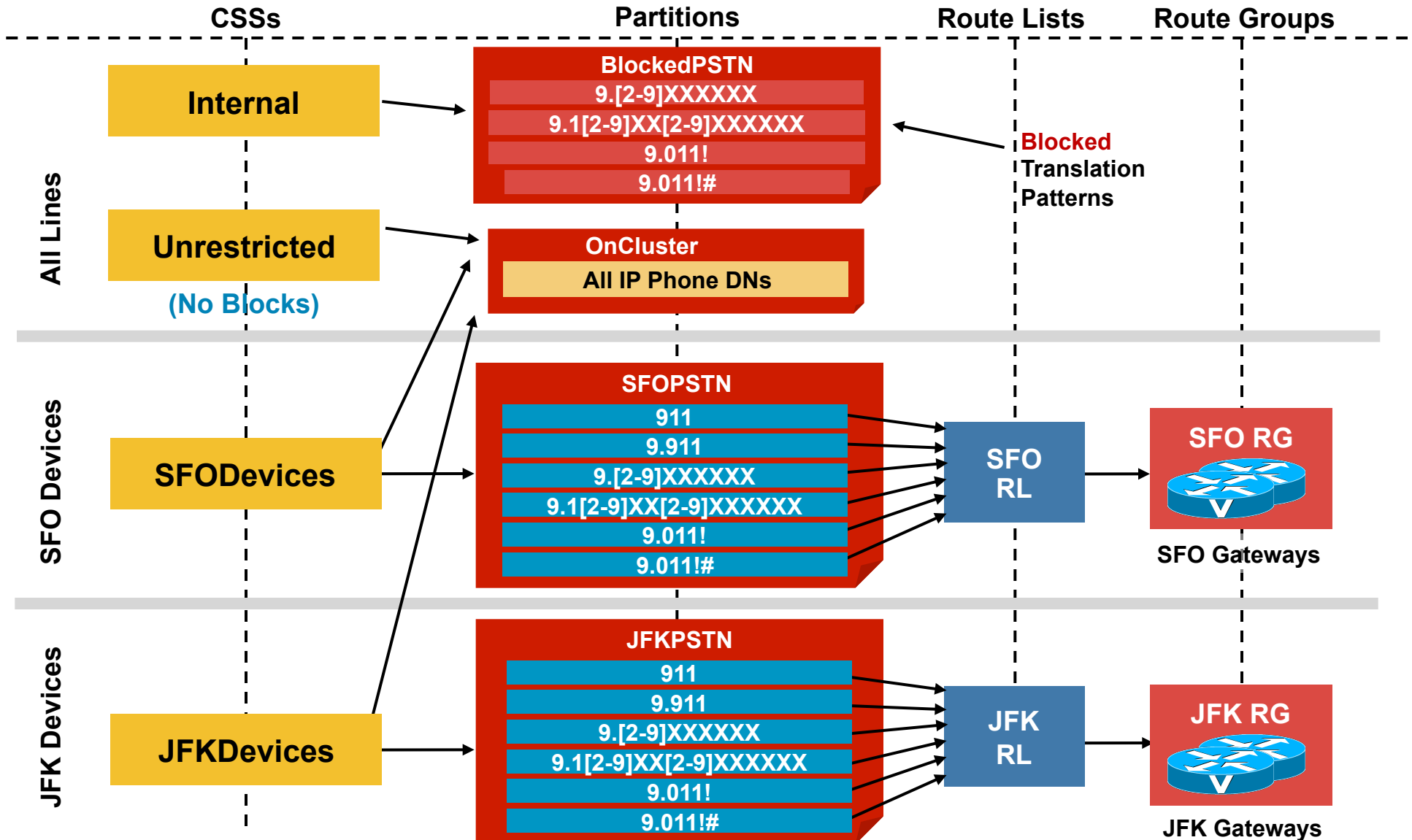
Local Route Group

With It: GW Chosen by Association to Calling Device



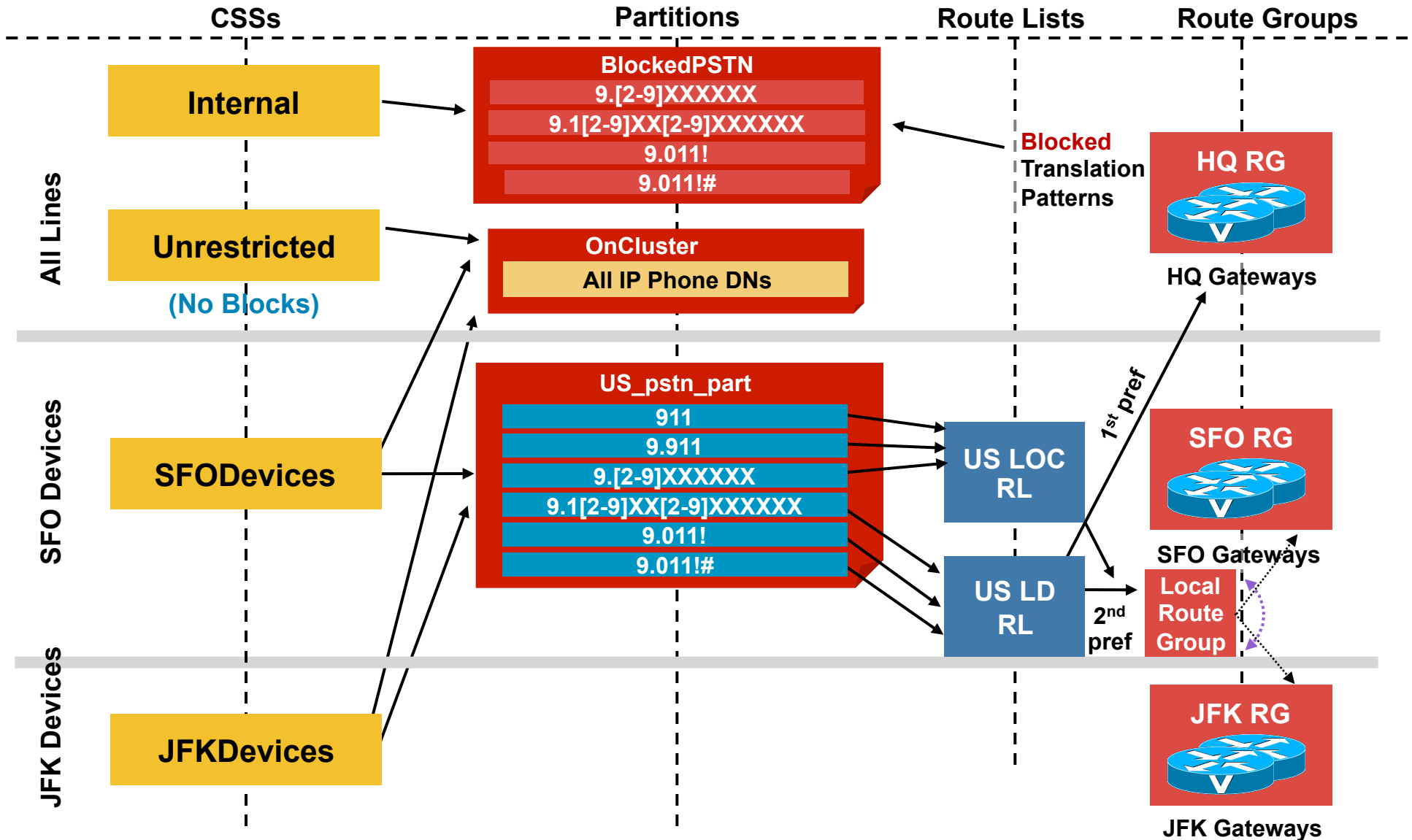
Local Route Group

With It—We Can Start from this, for Two Sites



Local Route Group

With It—and End Up with this, for Two Sites



Local Route Group

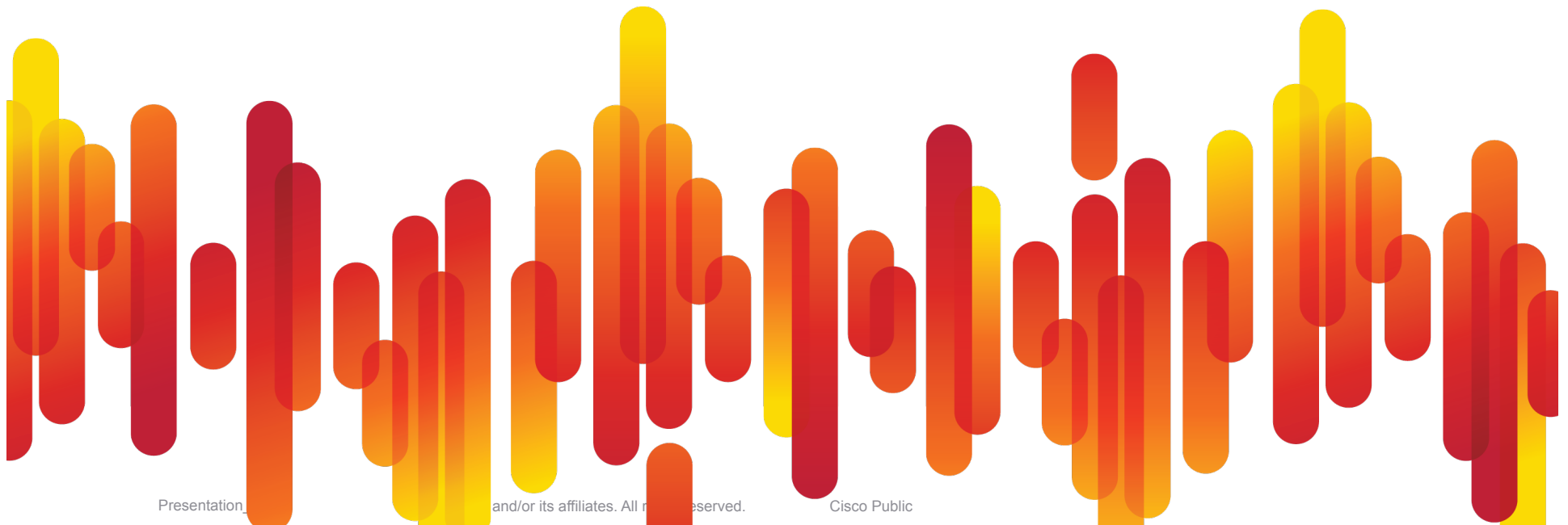
With It—Key Takeaways

- We go from route patterns that are site-specific to patterns that are type-specific
 - e.g., local, national, international
- We now group by dial-plan domains
 - e.g., US dialing habits of nine plus seven, nine plus ten, 91 plus ten, 9011 plus ???, 911, 9911). I could not add a French site to the preceding example without creating patterns for 112, 0112, 00[1-6]XXXXXXXX, 000!, 000!#
- We get site-specific failover for **free** on long-distance patterns
- We now have much fewer things to configure per site

QUIZ!!!

Can I, in the Preceding Example, Use a Single CSS for All Sites?

+ Sign Support Enabling Globalised Number Routing



+ Sign Support

What It Is: Concept

- E.164 support includes the use of + to **wildcard** international access codes AND to avoid overlap between globalized numbers and other ranges (e.g.: calls to India (+91XXXXXXXXXX) and NANP toll calls (912125551234))
- +33144522919 is the E.164 (global) representation of City Hall in the 19th arrondissement in Paris. It is accessed by different localised methods:
 - In Paris, send 0144522919 to an **intra-France** gateway
 - In Sydney, send 0001133144522919 to an international gateway in Australia
 - In San Francisco, send 01133144522919 to an international gateway in the US
 - From anywhere, by sending +33144522919, into a network that can digest it; e.g., most mobile GSM carriers, and now, our UC system 7.0**
- Supporting the + sign allows UCM-based systems to:
 - Route calls based on a directory's entry using the E164 notation
 - Either in a dual mode phone or click-to-dial from softclient
 - Store numbers in a non-site specific form in extension mobility profiles
 - Allows CallForwardAll destinations to use local route groups
 - Allows AAR destinations to be globalised, thereby simplifying AAR configuration
 - ... and many other things
- Most phones do not support the + sign for keypad entry (7921 and 7925 do), but support the + sign in display and missed/received calls menus
- Let's look at some screen shot examples

+ Sign Support

What It Is: Screen Shots

Route Patterns Now Support +

Route Pattern Configuration

Save

Status

Pattern Definition

Route Pattern* \+!

Route Partition pstn

Description cluster-wide pstn route, using loc. route group

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* Default

Space Network Domain < None >

pstn_rl (Edit)

Route this pattern

\ Is Used as an Escape Character:
\+ Means the Literal + Sign

\+! Matches Any Pattern Beginning with + (e.g., E.164)

Intent Is to Route All Calls to the Caller's Co-located PSTN GW

Speed Dial to +33144522919 Would Match this Pattern, and Be Sent to the Calling Phone's Local Route Group

This Points to the Local Route Group

+ Sign Support

What It Is: Screen Shots

The screenshot shows the 'Directory Number Configuration' page. At the top, there are buttons for 'Save', 'Delete', 'Reset', and 'Add New'. Below that is the 'Status' section, which shows 'Status: Ready'. The main section is 'Directory Number Information', which contains several fields: 'Directory Number*' with the value '\+33497232651', 'Route Partition' with a dropdown menu set to 'all_cluster_phones', 'Description', 'Alerting Name', and 'ASCII Alerting Name'. Below these fields is a section for 'Device from CTI' with a text box containing 'SEP001380FDCCEF'. At the bottom right of this section are two buttons: 'Edit Device' and 'Edit Line Appearance'. A speech bubble points to the 'Directory Number*' field, containing the text 'You Can Even Use the + Sign as Part of the DN of a Phone'.

You Can Even Use the + Sign as Part of the DN of a Phone

**E164 Can Be on the DN Directly, or in the External Phone Number Mask.
Note: “\” Shows on the Phone if Configured in DN and Phone Number Mask Is Left Blank**

Watch for CTI application support of + directly in the DN of phones!

+ Sign Support

What It Is: Screen Shots

If the Administrator Sets the Prefix to Default this Indicates Call Processing Will Use Prefix at the Next Level Setting (DevicePool/Service Parameter). Otherwise, the Value Configured Is Used as the Prefix Unless the Field Is Empty in Which Case There Is No Prefix Assigned

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is no prefix assigned.

Incoming Calling Party National Number Prefix

Incoming Calling Party International Number Prefix

Incoming Calling Party Unknown Number Prefix

Incoming Calling Party Subscriber Number Prefix

Clear Prefix Settings Default Prefix Settings

+1

+

Default

+1415

SIP tags all calls as "unknown"

Incoming Calls from a GW Can Now Have Their Calling-Party Number Globalised on a per-GW Basis. This Is an Example for San Francisco.

Update: We Can Now Strip and Prefix on Incoming Party Settings!

From the SRND:

The Notation Takes the Form PP:SS, Where PP Represents the Digits to Be Prefixed and SS Represents a Quantity of Digits to Be Stripped. The Digit Stripping Operation Is Performed First on the Incoming Calling Party Number, and Then the Prefix Digits Are Added to the Resulting String. For Example, if the Prefix Digits Field Is Configured as +33:1 and the Incoming Calling Party Number Is 01 58 40 58 58, the Resulting String Will Be +33 1 58 40 58 58

Calling Party Transformations

Globalise on Ingress – Incoming Calling Party Settings

New in 7.1: Incoming calling party settings now allow for using Calling Party Transformation Patterns to manipulate the calling party number when calls enter the system from gateways. One CgPTP CSS is available for each numbering type. Note: all calls are tagged with numbering type “Unknown” on SIP Gateways and trunks. This allows digit manipulation to be based on regular expressions, for more flexible matching.

Geo Location Configuration

Geo Location

Geo Location Filter

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

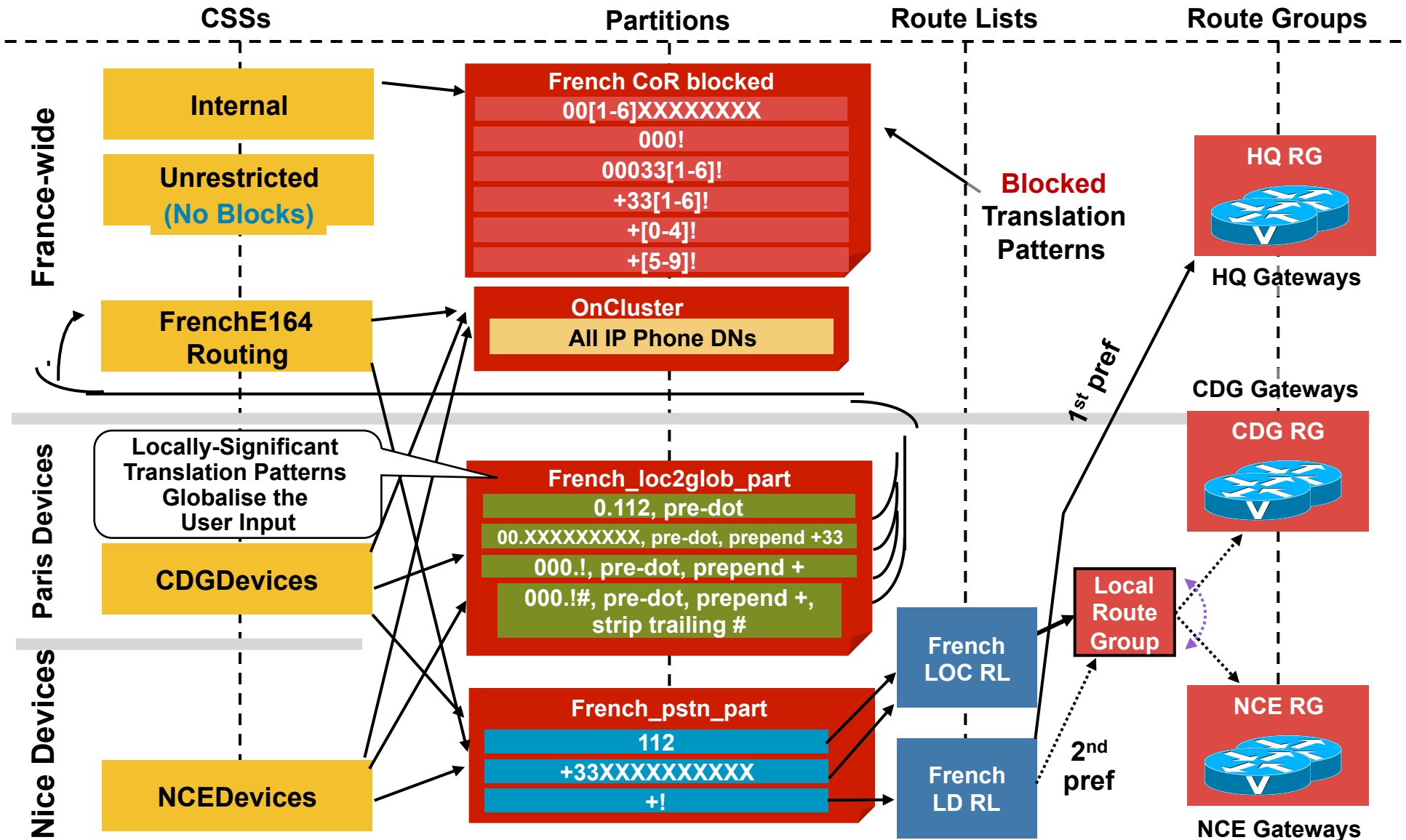
Number Type	Prefix	Strip Digits	Use Dev Pool CSS	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>

i *- indicates required item.

i **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

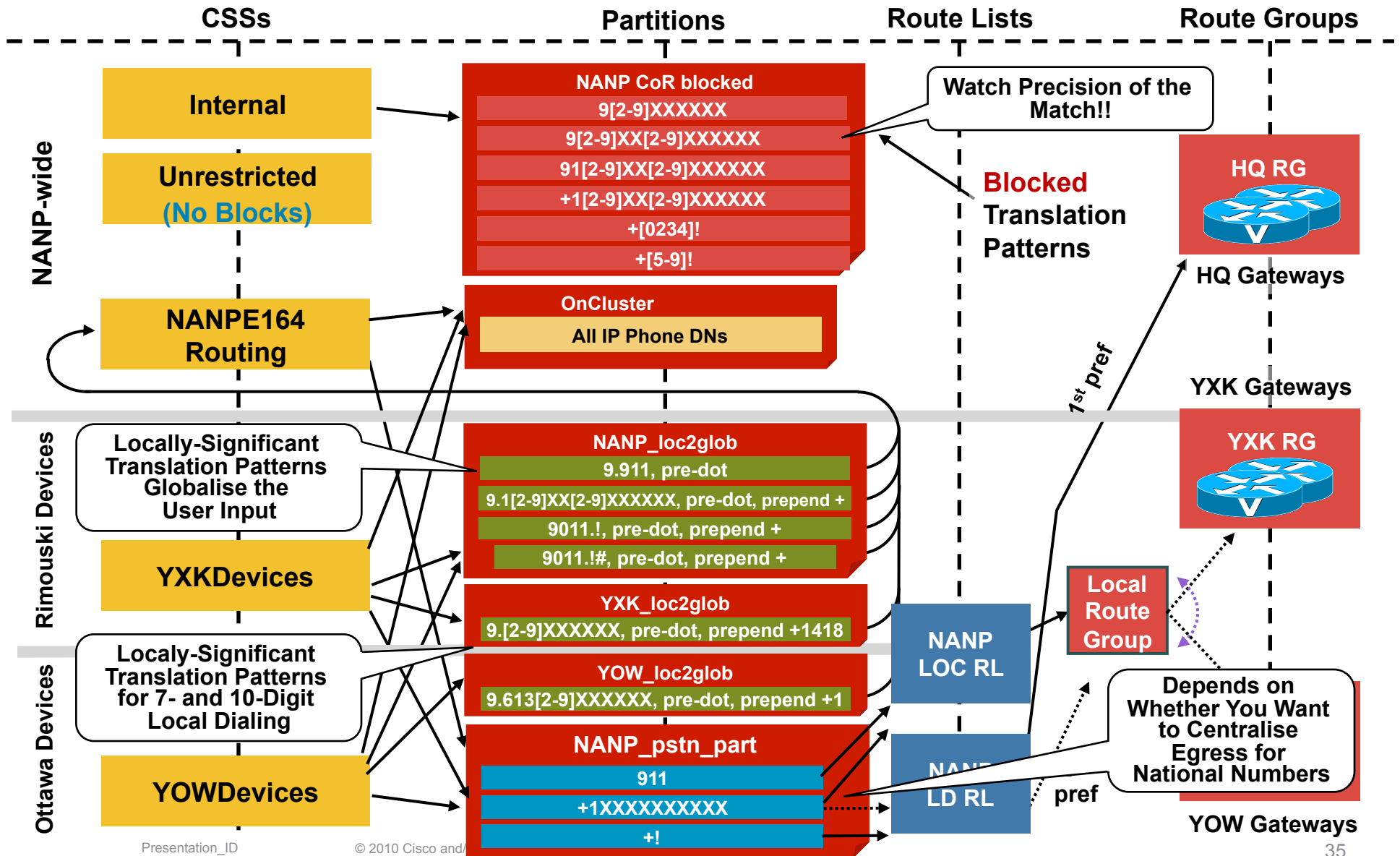
+ Sign Support

From the Phones: Allowing Globalised and Localised TUI



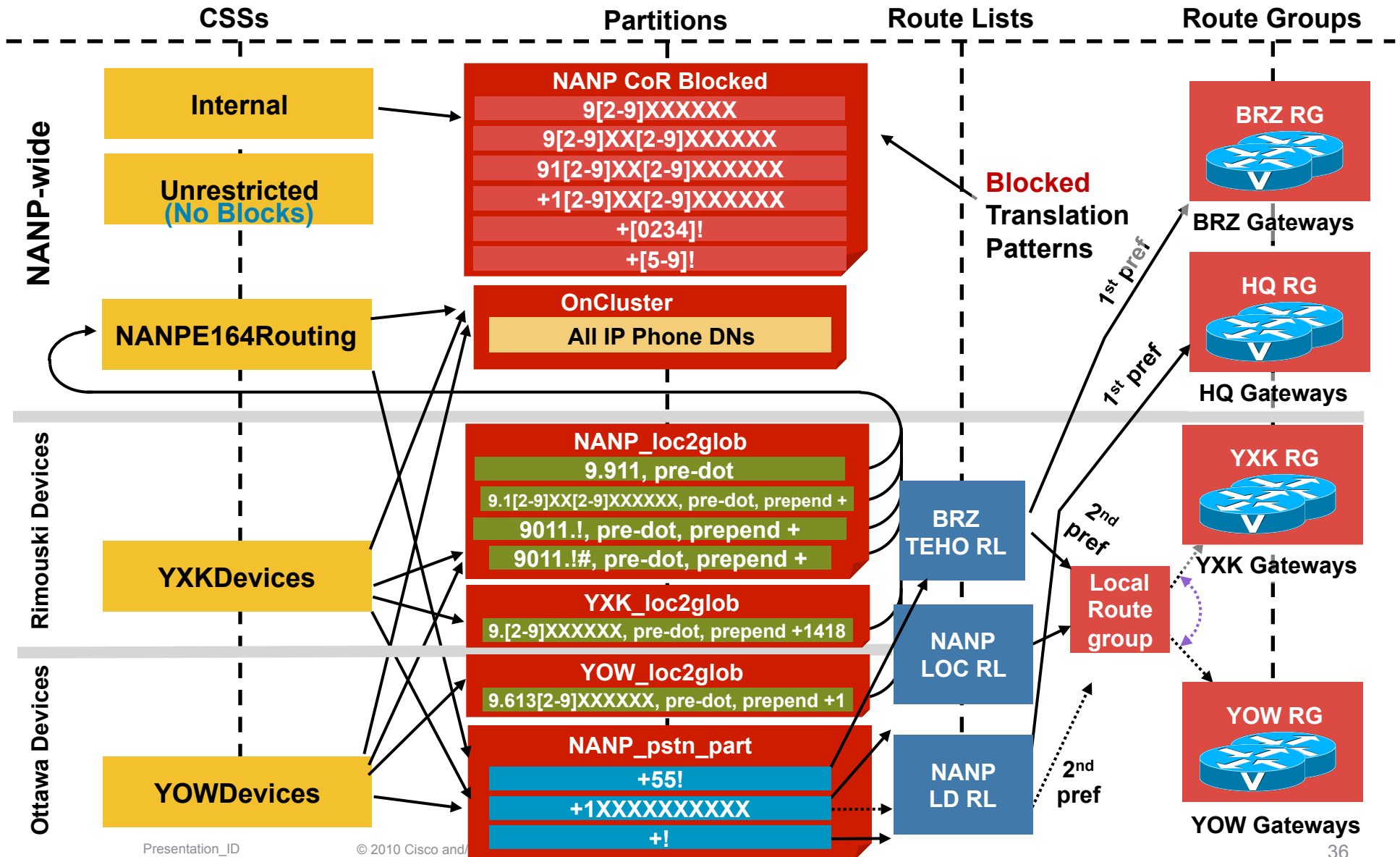
+ Sign Support

From the Phones: Localised TUI May Require Extra Effort



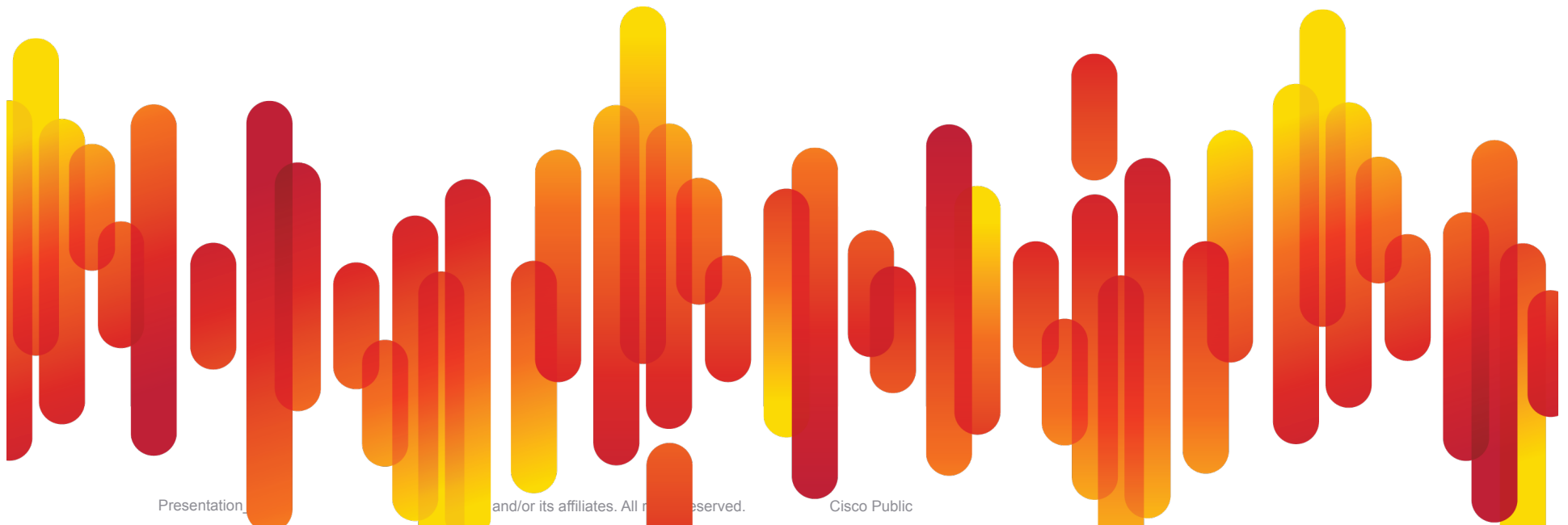
+ Sign Support

From the Phones: Tail End Hop Off Is Simple



Calling/Called Number Transformations:

Bridging Local and Global Forms



Calling/Called Number Transformations

What It Is: Concept

- Calls presented to a phone or a gateway typically require the calling and the called party numbers be adapted to the local preferences/requirements of:
 - The user receiving the call
 - The gateway through which the call is routed
 - The network to which the call is routed
- Calls received from an external network (e.g., the PSTN) typically present calls in a localised flavor. We can now adapt the received call based on:
 - The numbering plan presented by the network for a specific call
 - The called/calling number delivered into the UC system by the gateway
 - Combining the two elements above, we can globalise the number upon entry

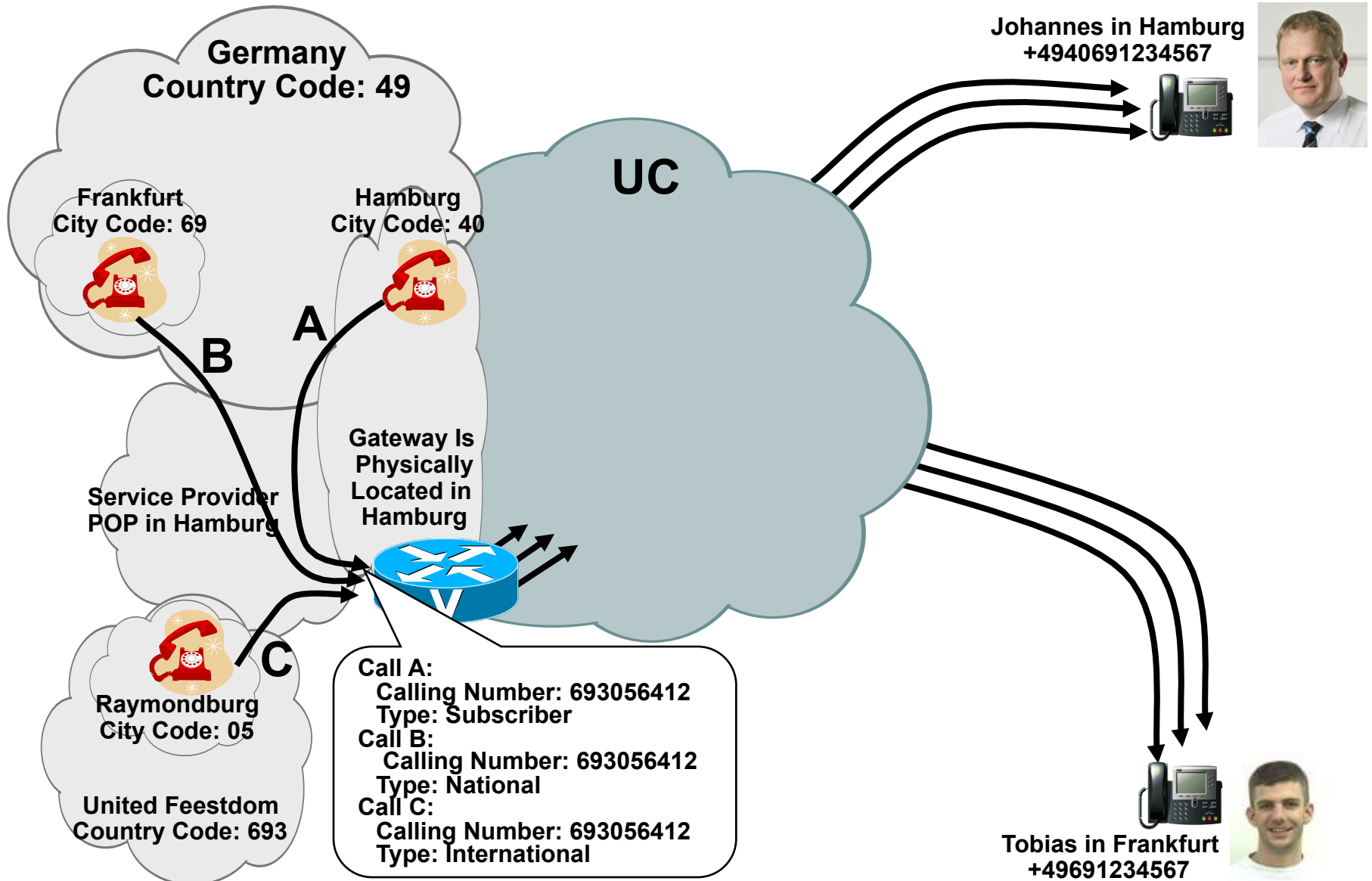
Calling/Called Number Transformations

What It Is: Concept (Cont.)

- The calling number may need to be:
 - Left in the global form; e.g., +1 408 902 3574. GSM networks may accept (or even require) this form
 - Changed to the locally-significant on-net abbreviated form; e.g., 23574 if the called party is colocated with me
 - Changed to an enterprise-significant form; e.g., 89023574 if I call someone in say, RTP's Cisco site, on-net
 - Changed to a nationally-significant form if I call a pizza shop in New York; e.g., 408 902 3574
 - Changed to a Brazilian-significant form if I call a shop in Rio: 0014089023574
- The called number may need to be adapted to enter another network with the correct numbering type and the correct numbering form
 - If I call +33144522919 using a US gateway, I may leave the number intact if the gateway **and** the carrier support the + sign
 - I may need to change the number to 011 33144522919 and set the numbering type to international
 - If I route the call through a French gateway, I may need to change the called number to 0144522919, and set the numbering type to national

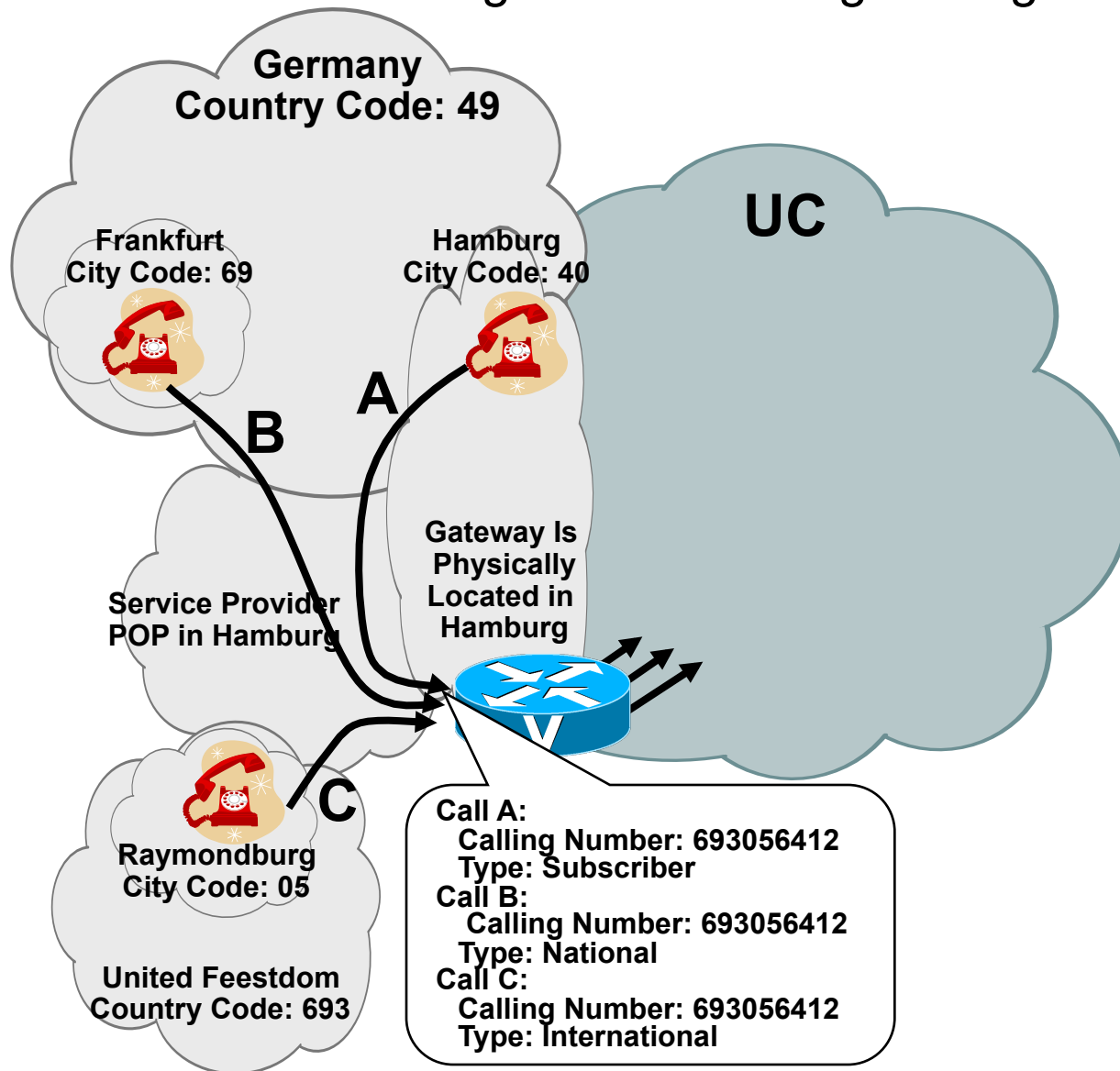
Calling-Party Transformations

Base Scenario



Calling-Party Transformations

Globalise on Ingress—Incoming Calling-Party Settings



- We need rules applied to the gateway to globalise the calling number on ingress
- There rules need to take into account:
 - The digits received
 - The number type
- Next screen looks at sample rules for German gateways

Calling-Party Transformations

Globalise on Ingress—Incoming Calling-Party Settings

For a Hamburg Gateway (Our Case):

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is no prefix assigned.

Incoming Calling Party National Number Prefix	+49
Incoming Calling Party International Number Prefix	+
Incoming Calling Party Unknown Number Prefix	Default
Incoming Calling Party Subscriber Number Prefix	+4940

For a Frankfurt Gateway (for the Sake of Argument):

Incoming Calling Party Settings

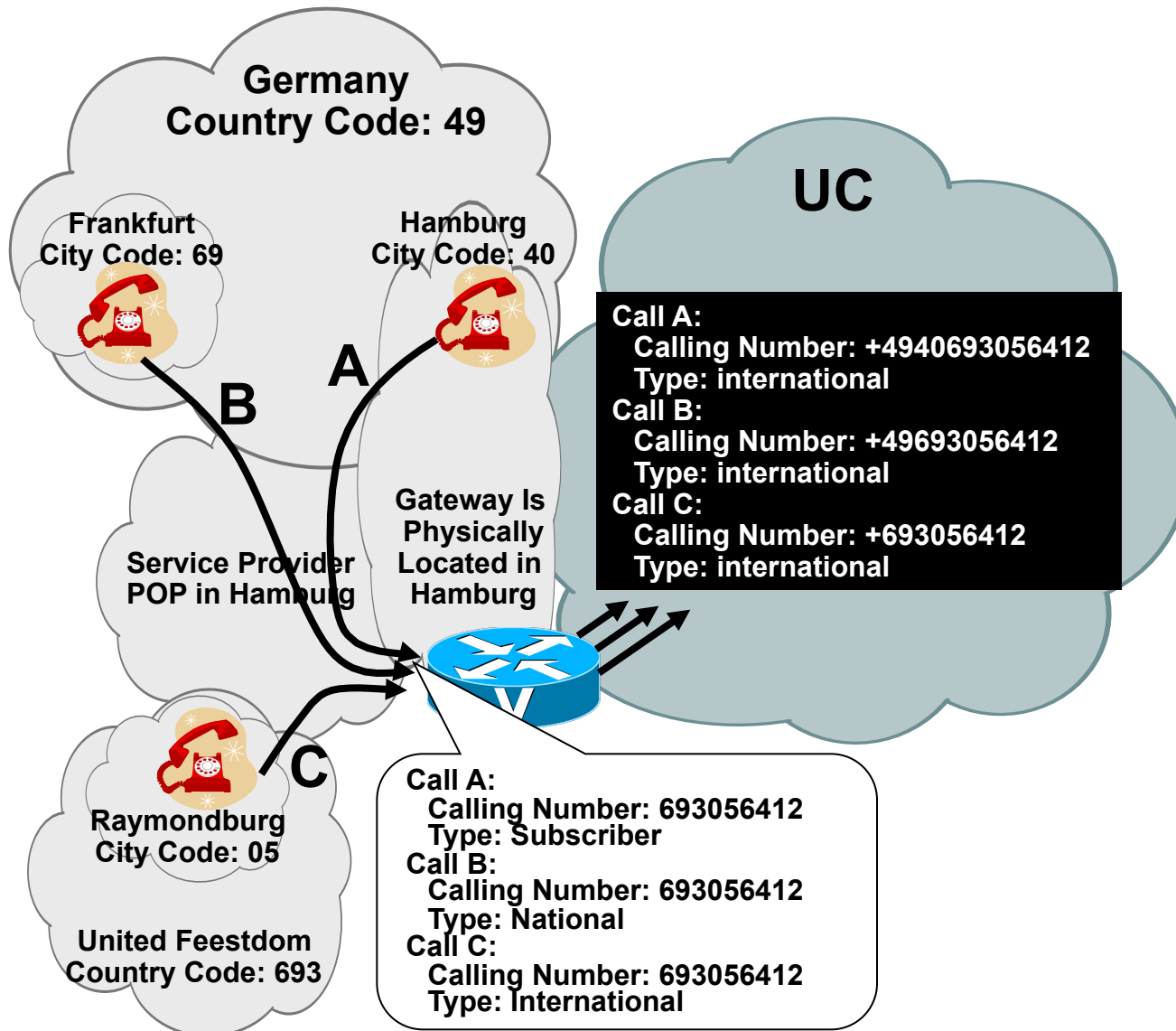
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is no prefix assigned.

Incoming Calling Party National Number Prefix	+49
Incoming Calling Party International Number Prefix	+
Incoming Calling Party Unknown Number Prefix	Default
Incoming Calling Party Subscriber Number Prefix	+4969

These Settings Can Be Applied at the Gateway, Device Pool, or Service Parameter Level, in Reverse Order of Precedence

Calling-Party Transformations

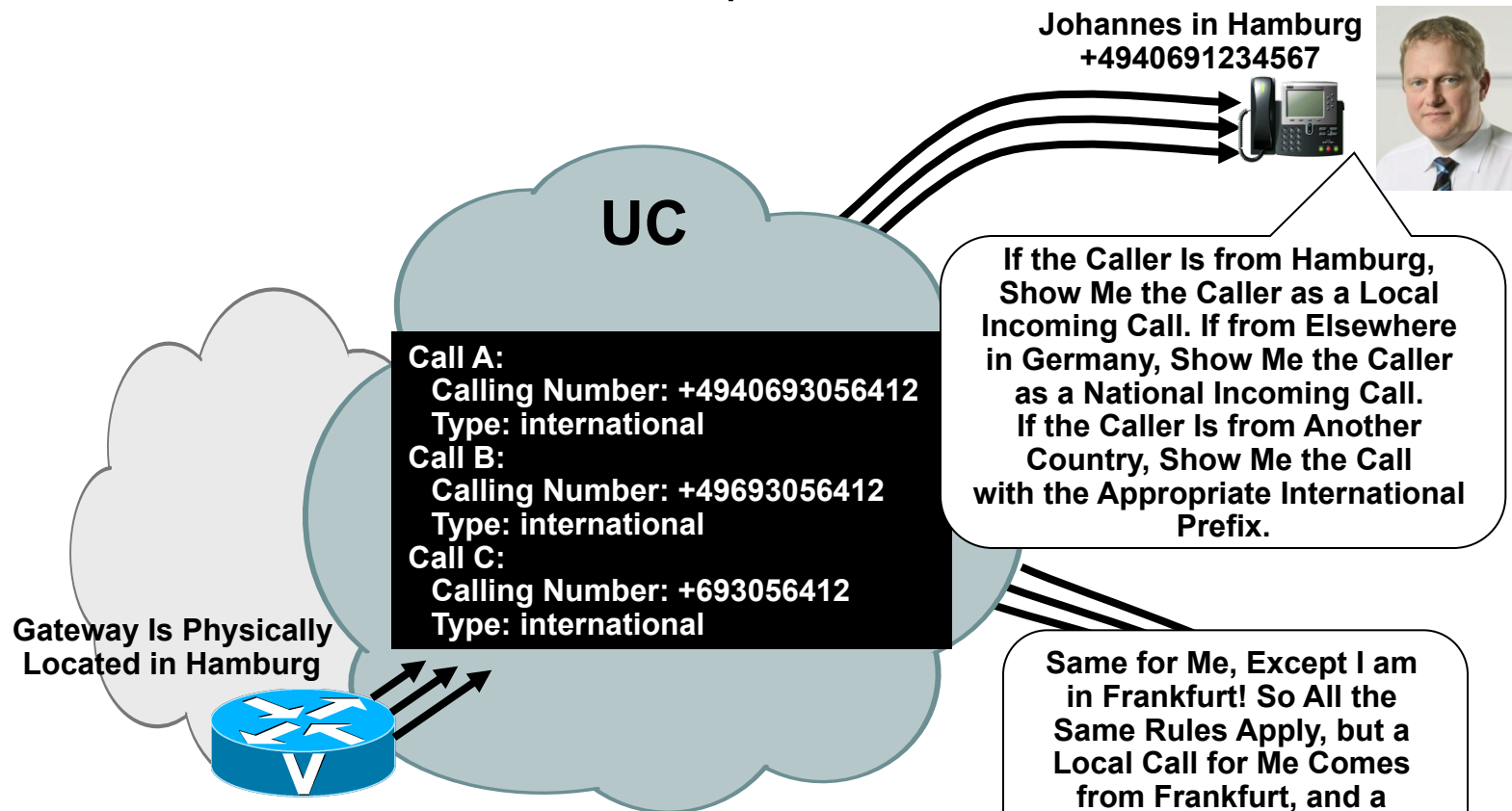
Globalise on Ingress



- We now have globalisation rules for the Hamburg gateway
- This allow us to process all calls on the presumption that the calling number is in a global format

Calling-Party Transformations

When Calls Are Presented to Endpoints



- We need rules applied to the destination endpoints (e.g., the phones)
- There rules must assume some common starting point: a global format for any call
- Next slides show sample config

Calling-Party Transformations

Calling-Party Transformation Patterns for Hamburg

Pattern Definition	
Pattern*	\+4940.!
Partition	hamburg
Description	localizing of incoming call, Hamburg to Hamburg
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority	

Calling Party Transformations	
<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Discard Digit Instructions	PreDot
Calling Party Transformation Mask	
Prefix Digits (Outgoing Calls)	0
Calling Line ID Presentation*	Default
Calling Party Number Type*	Subscriber
Calling Party Numbering Plan*	Cisco CallManager

This One Should Be Part of the Calling-Party Transformation Pattern CSS of Hamburg Devices Only

Calling-Party Transformations

Calling-Party Transformation Patterns for Frankfurt

Pattern Definition	
Pattern*	\+4969.!
Partition	frankfurt
Description	localizing of incoming call, Frankfurt Frankfurt
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority	

Calling Party Transformations	
<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Discard Digit Instructions	PreDot
Calling Party Transformation Mask	
Prefix Digits (Outgoing Calls)	0
Calling Line ID Presentation*	Default
Calling Party Number Type*	Subscriber
Calling Party Numbering Plan*	Cisco CallManager

This One Should Be Part of the Calling-Party Transformation Pattern CSS of Frankfurt Devices Only

Calling-Party Transformations

Calling-Party Transformation Patterns for German Sites

Pattern Definition	
Pattern*	\+49.!
Partition	Germany
Description	localizing of incoming call, Germany to Germany
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority	

Calling Party Transformations	
<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Discard Digit Instructions	PreDot
Calling Party Transformation Mask	
Prefix Digits (Outgoing Calls)	00
Calling Line ID Presentation*	Default
Calling Party Number Type*	National
Calling Party Numbering Plan*	Cisco CallManager

This One Should Be Part of the Calling-Party Transformation Pattern CSS of all German Sites

Calling-Party Transformations

Calling-Party Transformation Patterns for German Sites

Pattern Definition	
Pattern*	\+.
Partition	Germany
Description	localizing of incoming call, Intl to Germany
Numbering Plan	< None >
Route Filter	< None >
<input checked="" type="checkbox"/> Urgent Priority	

Calling Party Transformations	
<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Discard Digit Instructions	PreDot
Calling Party Transformation Mask	
Prefix Digits (Outgoing Calls)	000
Calling Line ID Presentation*	Default
Calling Party Number Type*	International
Calling Party Numbering Plan*	Cisco CallManager

This One Should Be Part of the Calling-Party Transformation Pattern CSS of all German Sites

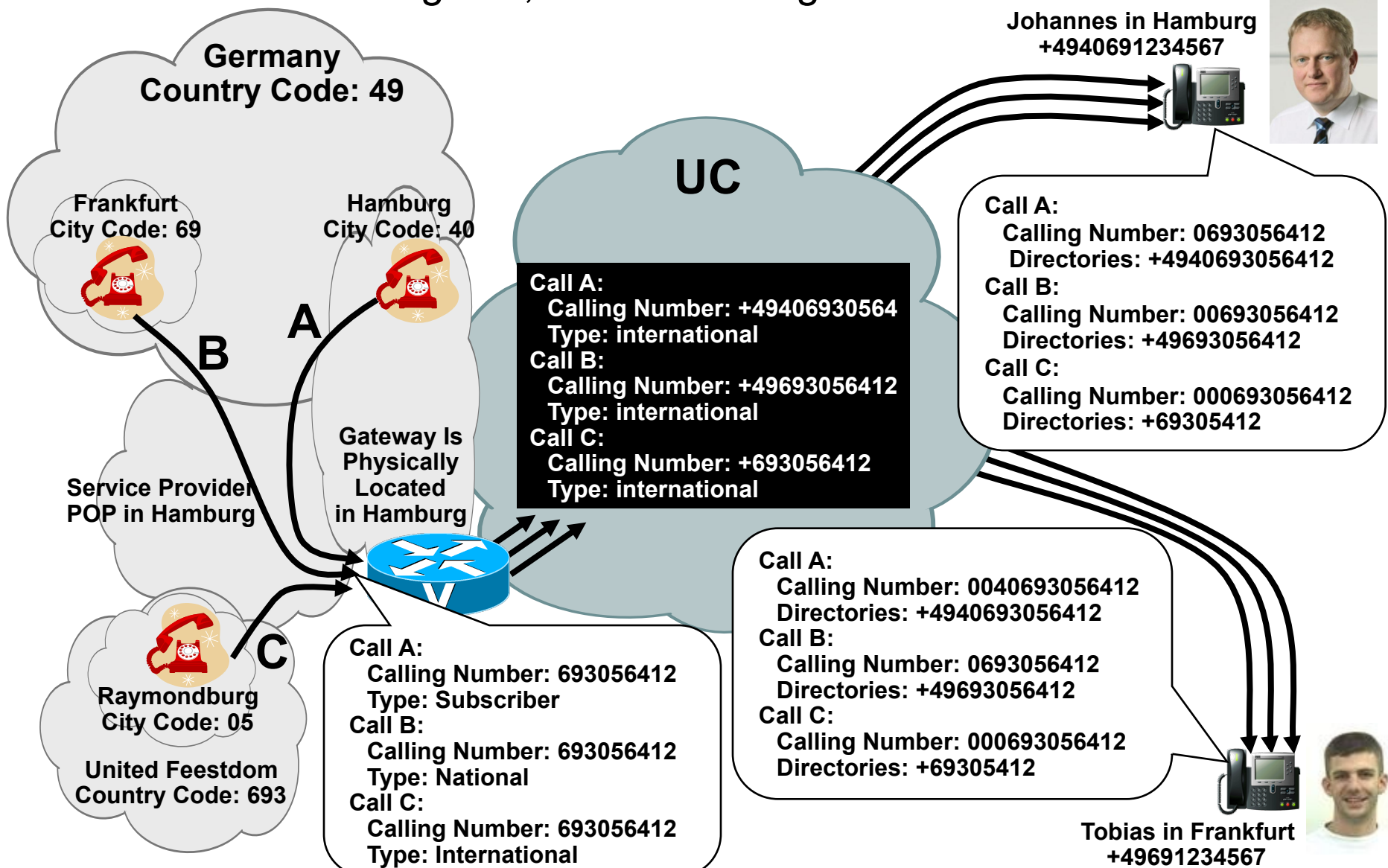
Calling-Party Transformations

Calling-Party Transformation Patterns—Note

- All but one of the preceding patterns match if the calling party number is from Hamburg
- The best match process will select the most precise pattern
- The nationally-significant patterns can be reused between sites
- Since all this is contained in a calling search space, the Germany-specific patterns can be used in Frankfurt **and** in Hamburg
- The transformation calling search spaces can be applied on the phone, or on the device pool, in order of precedence
- **Note:** these can be eliminated if the customer can accept to see E.164 numbers when the phone is ringing!

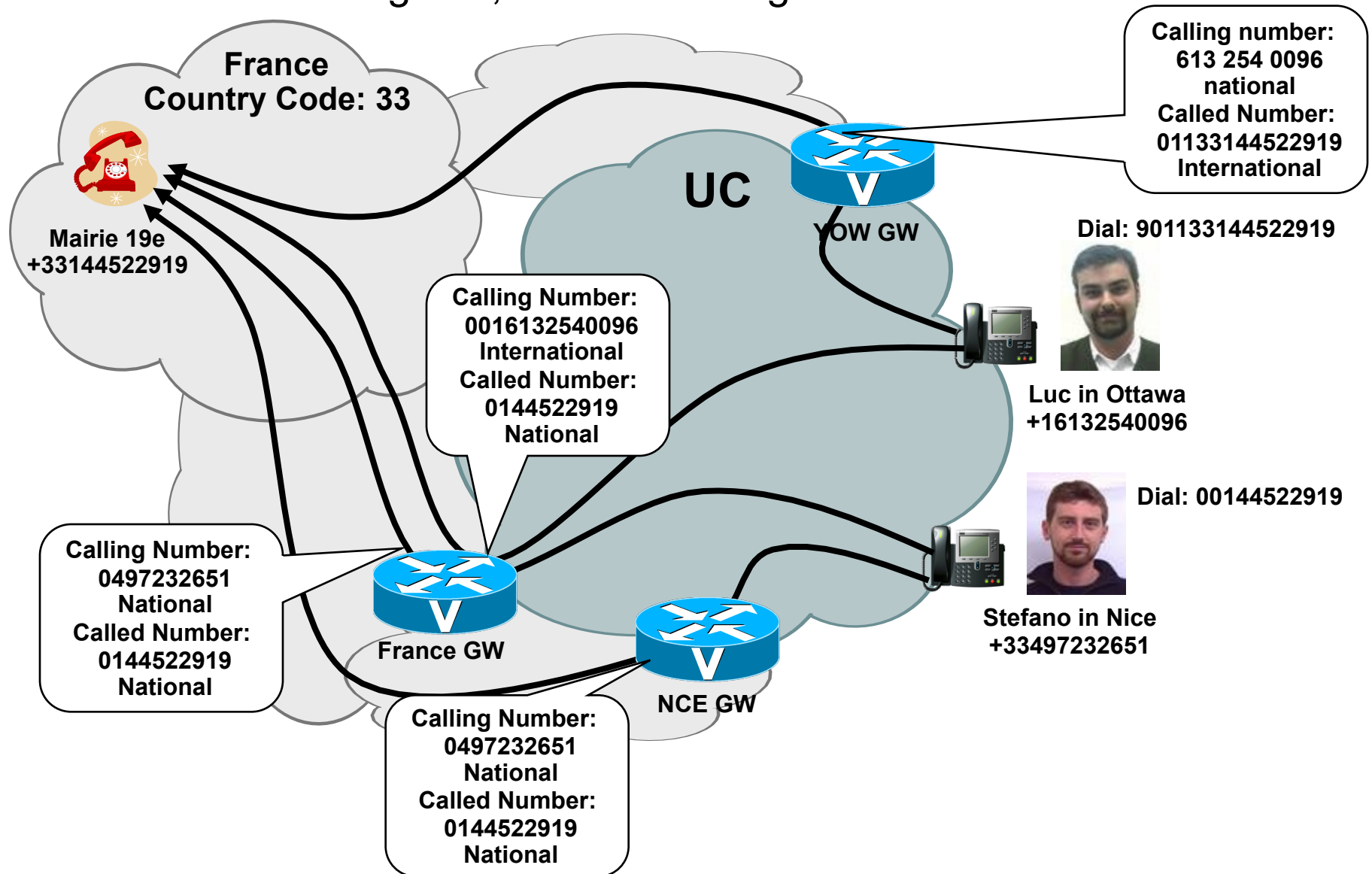
The Big Picture of Global Dial Plans

Globalise on Ingress, Localise on Egress



The Big Picture of Global Dial Plans

Globalise on Ingress, Localise on Egress



Called-Party Transformations

Localise on Egress: Screen Shots

Cdpts Are Applied Through a Device Pool to Calls Sent to Gateways

If Destination Number Is Any French PSTN Number in E.164 Format

Prepend the French National Routing Prefix

+33144522919
Would Be Transformed to 0144522919, Which the French PSTN Can Route

Keeps the Last Nine Digits

Sets the Resulting Number's Numbering Plan to National

The screenshot shows the 'Called Party Transformation Pattern Configuration' page. At the top, there is a 'Save' button and a 'Status' section showing 'Status: Ready'. The 'Pattern Definition' section includes a 'Pattern*' field with the value '+33.[1-6]!', a 'Transformation' dropdown set to 'cdg_called_party_xform_part', a 'Description' field with 'localization of french nat. numbers for cdg d.p.', and 'Numbering Plan' and 'Route Filter' dropdowns both set to '< None >'. Below this is the 'Called Party Transformations' section, which includes a 'PreDot' dropdown set to 'PreDot', a 'Party Transformation Mask' field, a 'Digits' field with the value '0', a 'Party Number Type*' dropdown set to 'National', and a 'Party Numbering Plan*' dropdown set to 'ISDN'.

Calling Party Transformation

Localise on Egress: Screen Shots

Calling Party Transformation Pattern Configuration

Save Delete Copy Add New

Status
Add successful

Pattern Definition

Pattern* \+33.!

Partition cdg_calling_party_xform_part

Description E.164 to national format, for French calling num.

Numbering

Route Filter

Urgent

Calling Party Transformations

Party's External P

Instructions

Transformation Mask

Outgoing Calls

Presentation* Default

Number Type* National

Numbering Plan* ISDN

Save Delete Copy Add New

Cgtps Are Applied to Calls Sent to Gateways and Phones, Through a Device Pool

If the Calling Number Is Any French PSTN Number In E.164 Format

Prepend the French National Routing Prefix

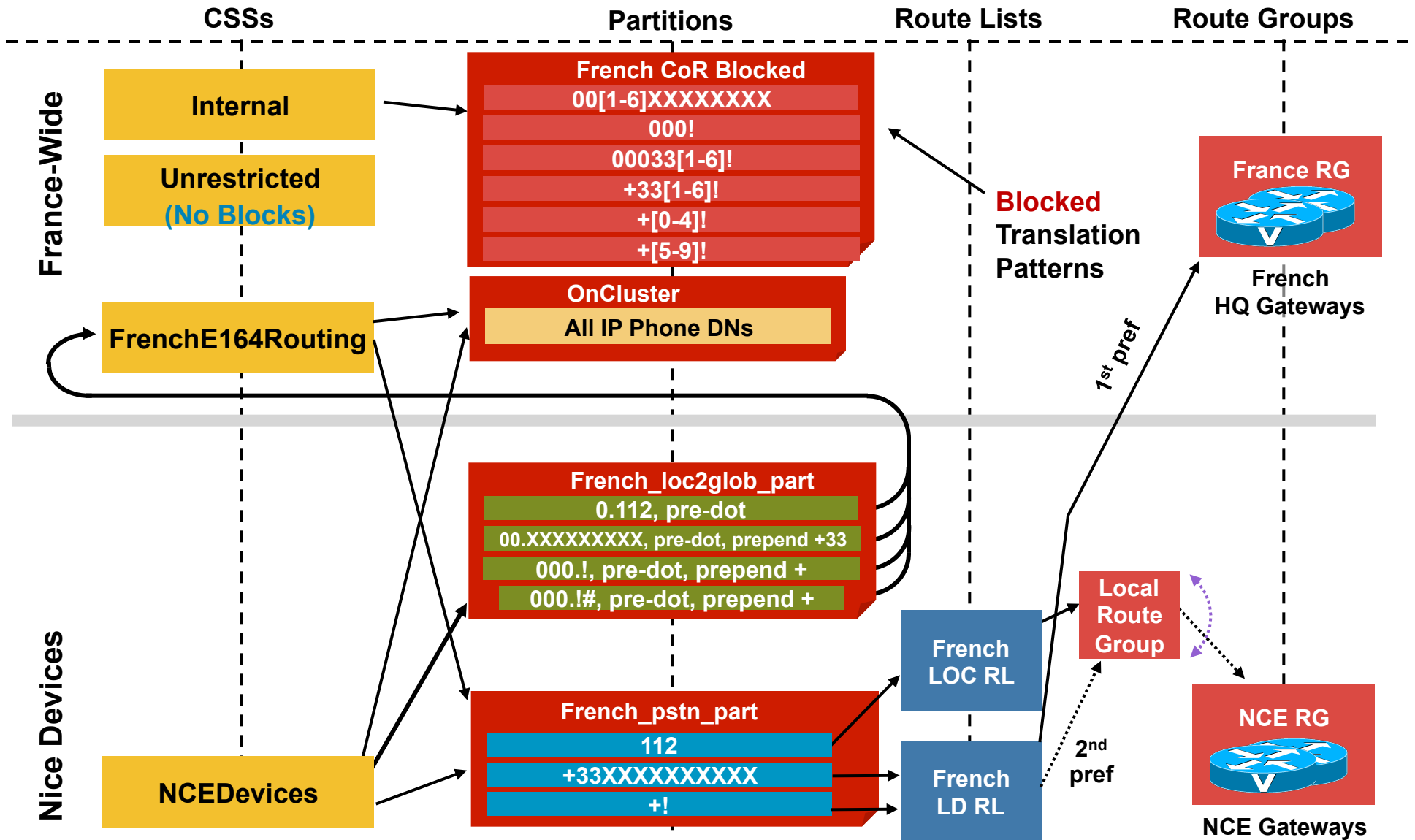
Keeps the Last Nine Digits

Sets the Resulting Number's Numbering Plan to National

If the Calling Party Is a French Number in E.164 Format, We Can Adapt It Here to Be Sent in the National Format: +33497232651 Becomes 0497232651

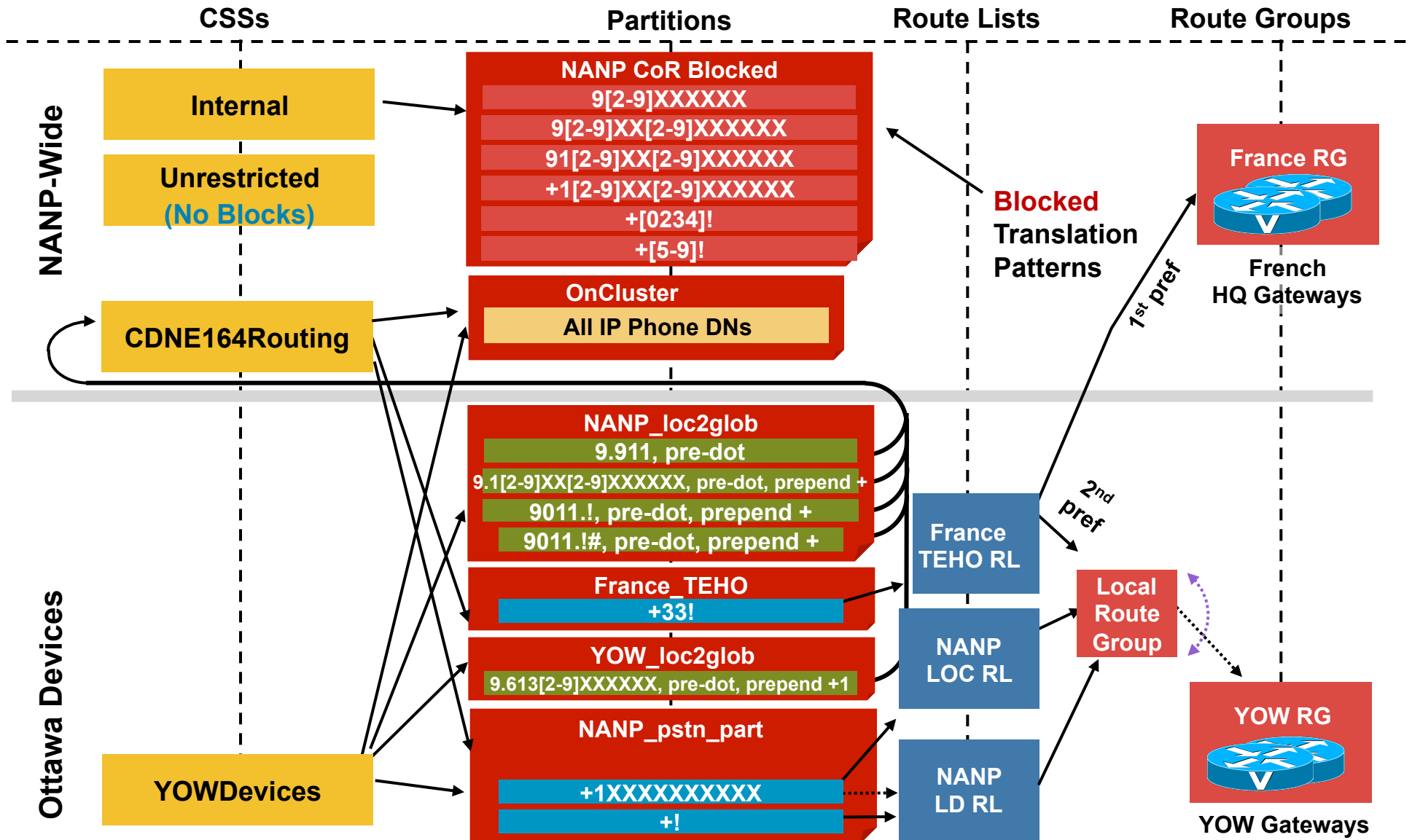
Dialed Pattern Translations

Stefano's Dial Plan: TUI and Routing



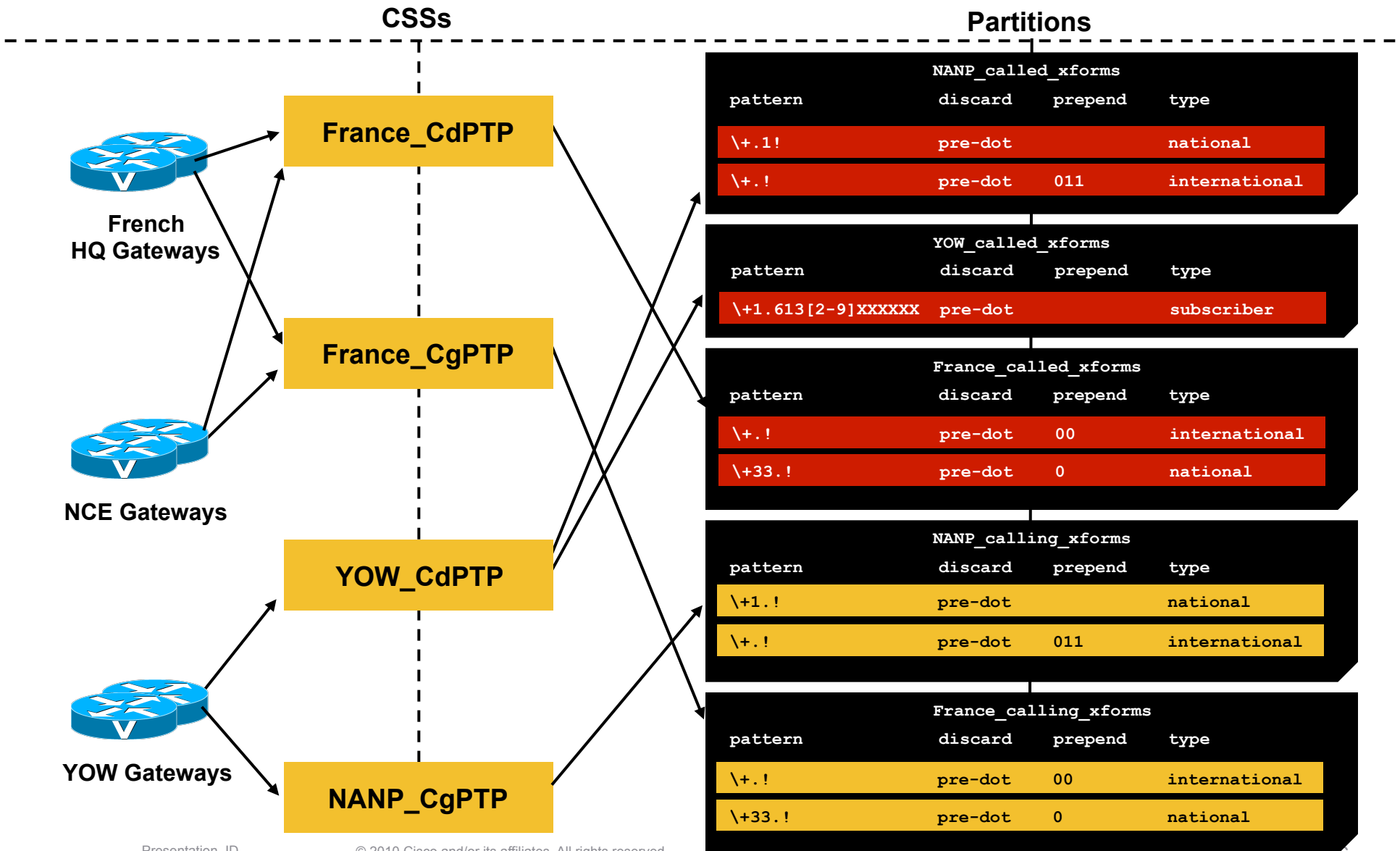
Dialed Pattern Translations

Luc's Dial Plan: TUI and Routing



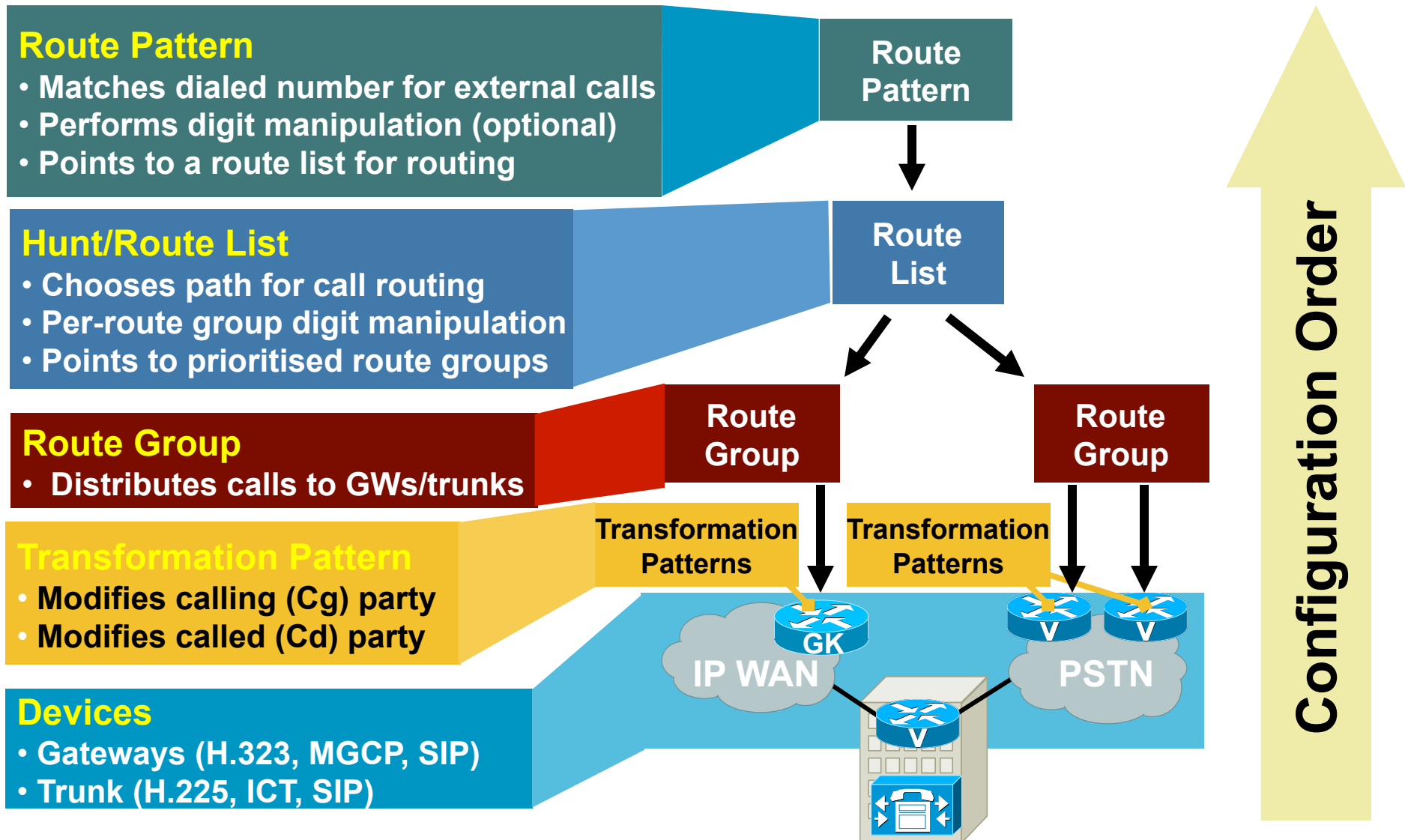
Calling/Called Number Transformations

Gateways: Calling and Called-Party Transforms



Number Transformations

Gateways: Calling and Called-Party Transforms



Combined Benefits

- Local CER failover
- CFUR routing
- AAR simplicity
- Mobility routing
- Speed dials—universal
- Missed/received calls list one-touch redial

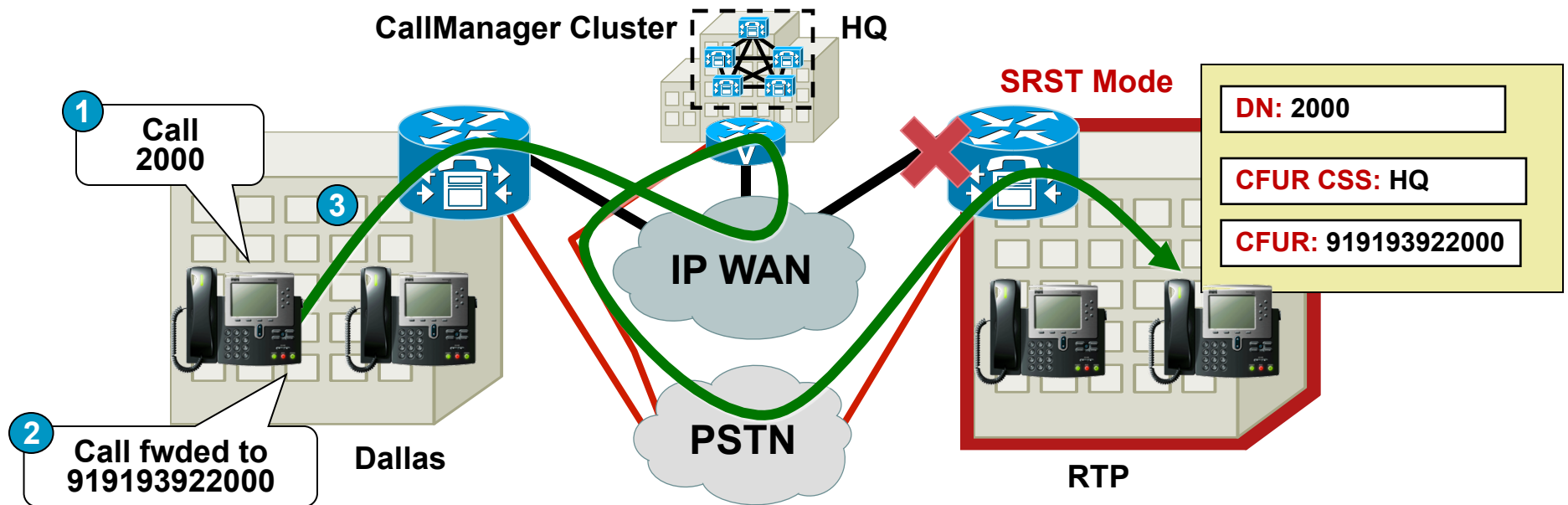
Combined Benefits

CER Local Failover

- When both CER servers in a CER group are down, pre-7.0 systems fall into a **one size fits all** default route
 - 911 CTI route point CFNA/CFB to 912 CTI route point
 - 912 CTI route point CFNA/CFB to 911, through a **single, cluster-wide** CSS
 - That CSS points to one gateway
- Now: place a 911 route pattern that route calls through the local route group in that CSS, and you have site-specific local failover for CER
- Bang! You are done!

Combined Benefits

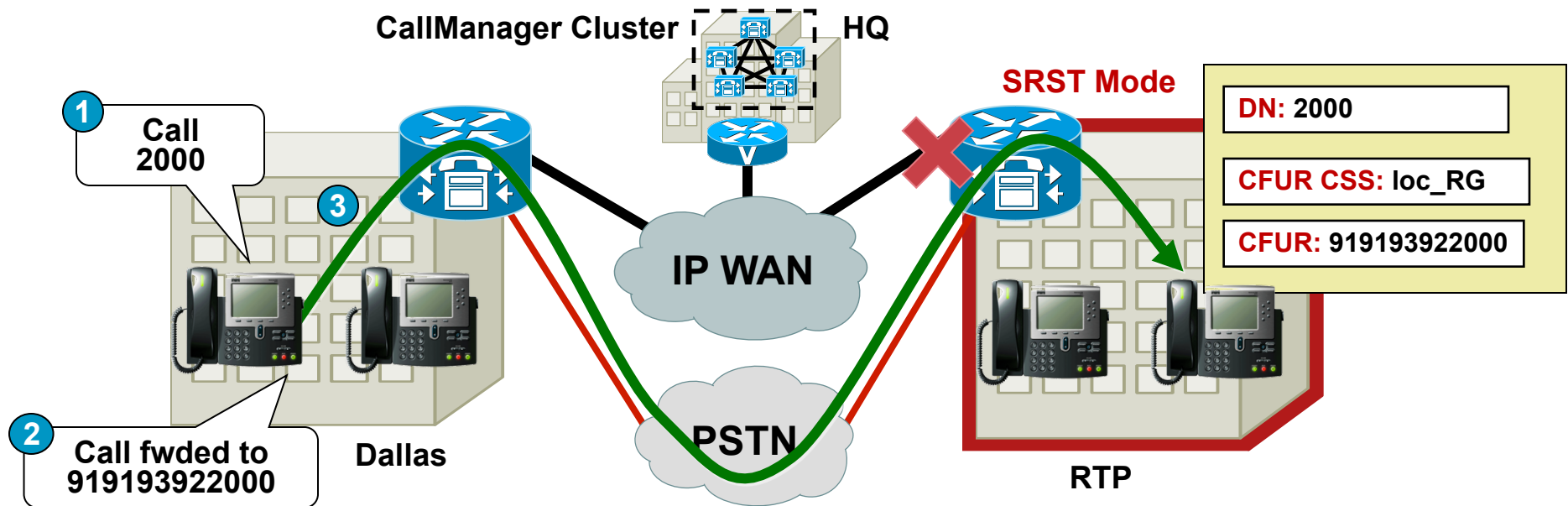
CFUR—Before



- Reroutes call to CFUR destination, which **must** match a pattern in CFUR CSS
- CFUR CSS points to a single, fixed egress gateway, which results in non-optimised call routing for all callers except those at HQ

Combined Benefits

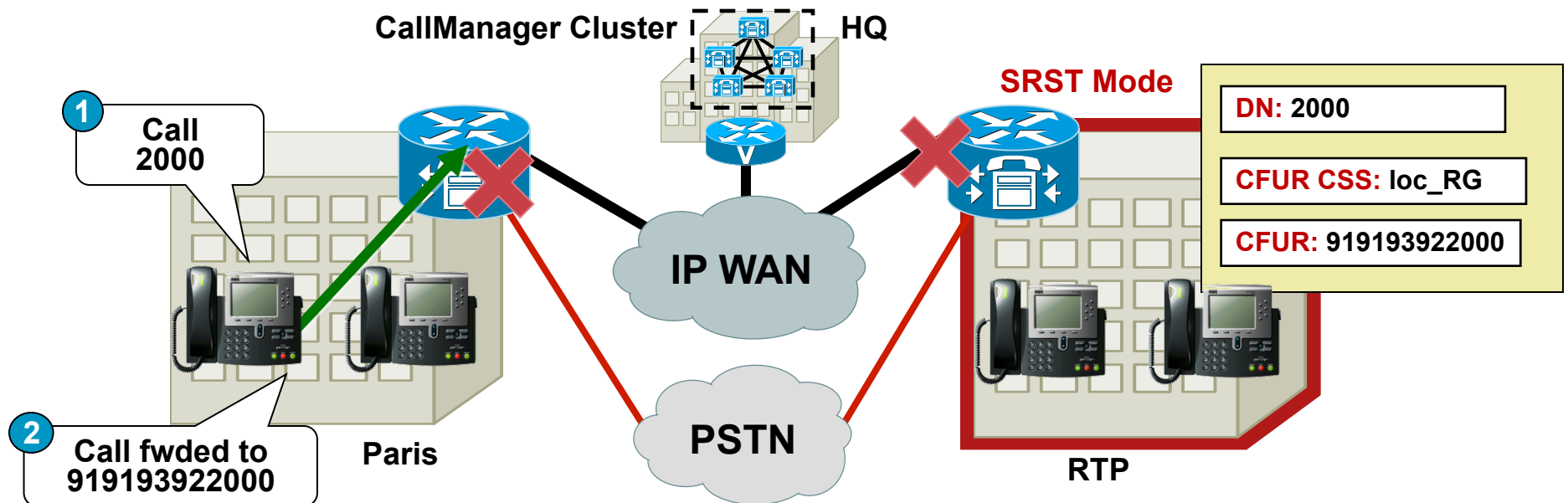
CFUR—Now



- Use a CFUR CSS that matches patterns pointing to the local route group of the caller. This creates optimum routing. This assumes that CFUR CSS **and** the local route group know how to deliver the call in a format that the PSTN connection can understand
- What if the caller is in Paris?

Combined Benefits

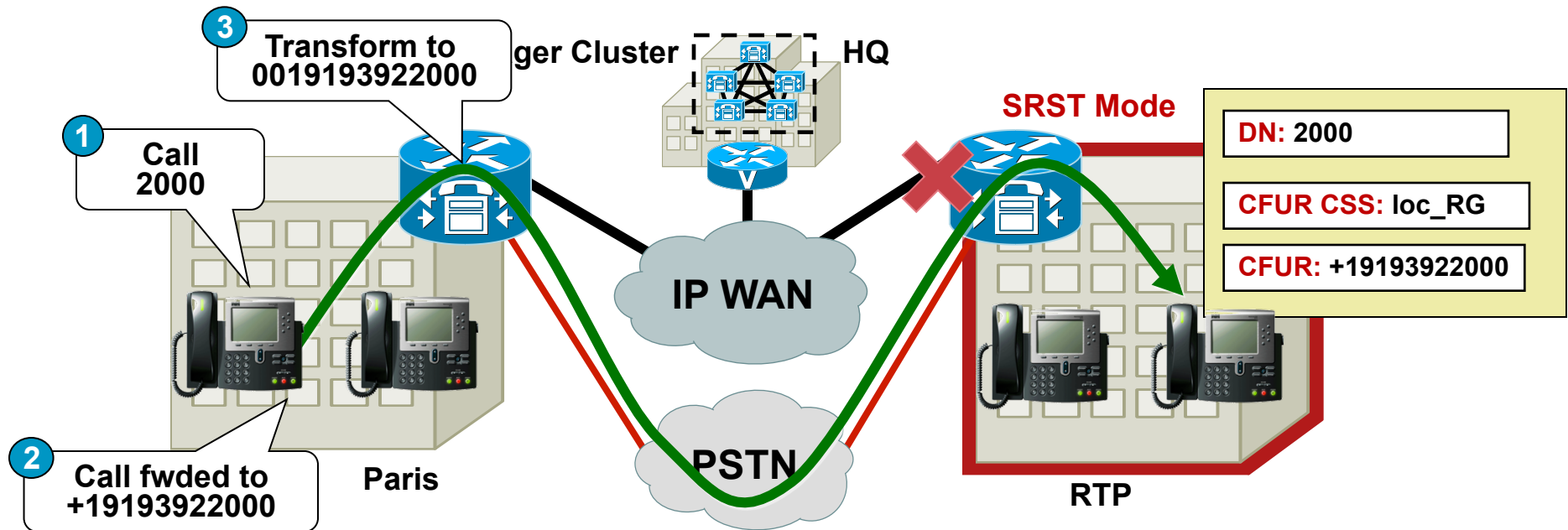
CFUR—Now



- Paris-originated calls fail, as the Paris local route group is not able to route calls made to 919193922000
- Solution: use E.164 notation in CFUR destination!

Combined Benefits

CFUR—Now



- Use full E.164 as CFUR destination
- Configure Paris device pool to feature:
 - Appropriate local route group
 - Substitution of +.!, strip pre-dot, prepend 00, number type: International, by using called-party transformation patterns
 - Transform the calling-party number, using calling-party transformation patterns to match carrier's expectation; e.g., if caller's DID is 0144522919, carrier may expect 33144522919, number type: international, OR 0144522919, number type national

Combined Benefits

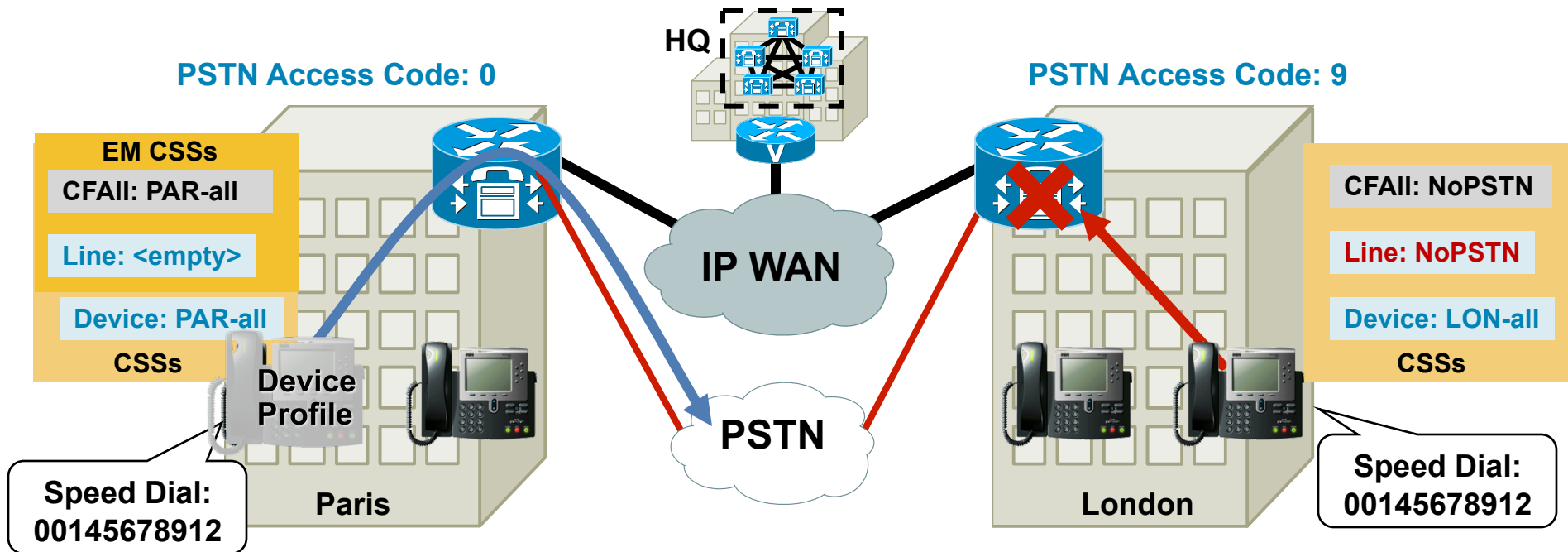
AAR

- In pre-7.0 systems, you need the AAR mask to determine the destination number, the AAR group to determine the appropriate off-net access codes, and the AAR CSS to route the call from a gateway colocated with the caller...
- In 7.0 system, make the AAR destination mask the E.164 destination of the phone, configure a single AAR group and put everyone in it, and make the AAR CSS point to the device pool's device mobility CSS for the site (and/or in 7.X use the LRG and use a single CSS for the entire cluster!!!!).
- Bang! You are done!

(Assuming you have configured all the rest of it 😊)

Combined Benefits

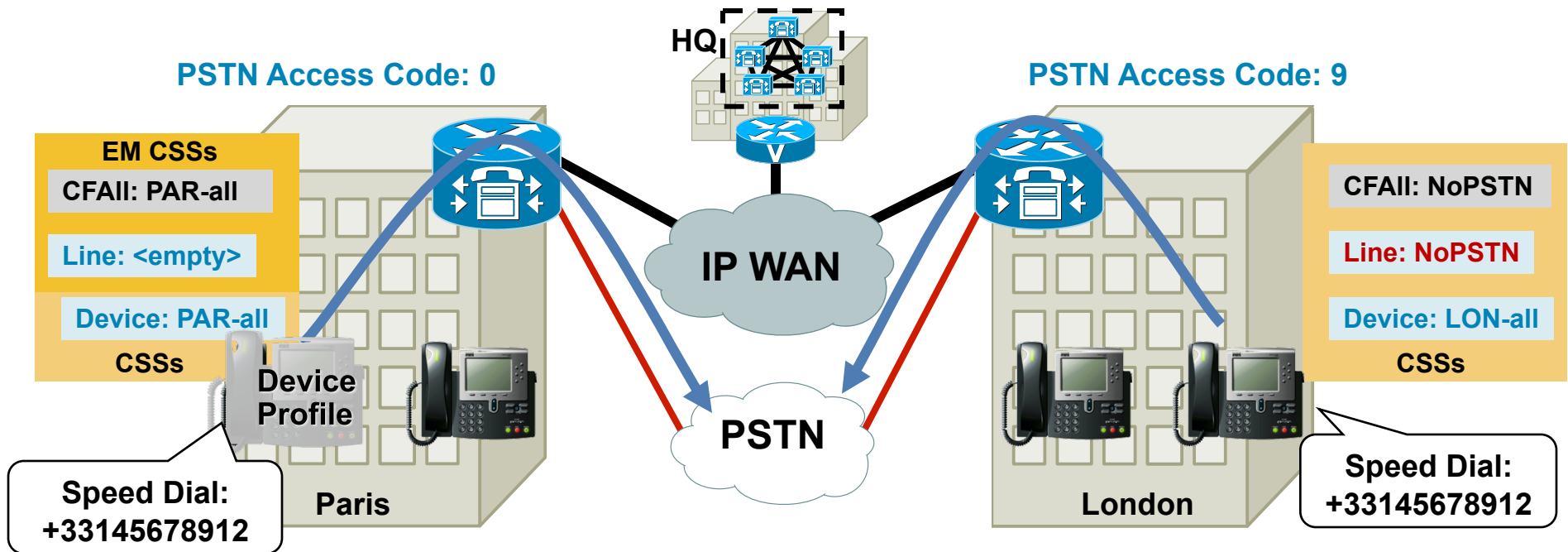
Before: Mobility and Speed Dials



- All PSTN calls are routed via local gateway
- User dialing habits and speed dials are not preserved across different dialing domains
- Even if call would route appropriately, it is placed using the local dialing habits of the home site, in a different dialing domain (e.g., French local number as opposed to international number from the UK).

Combined Benefits

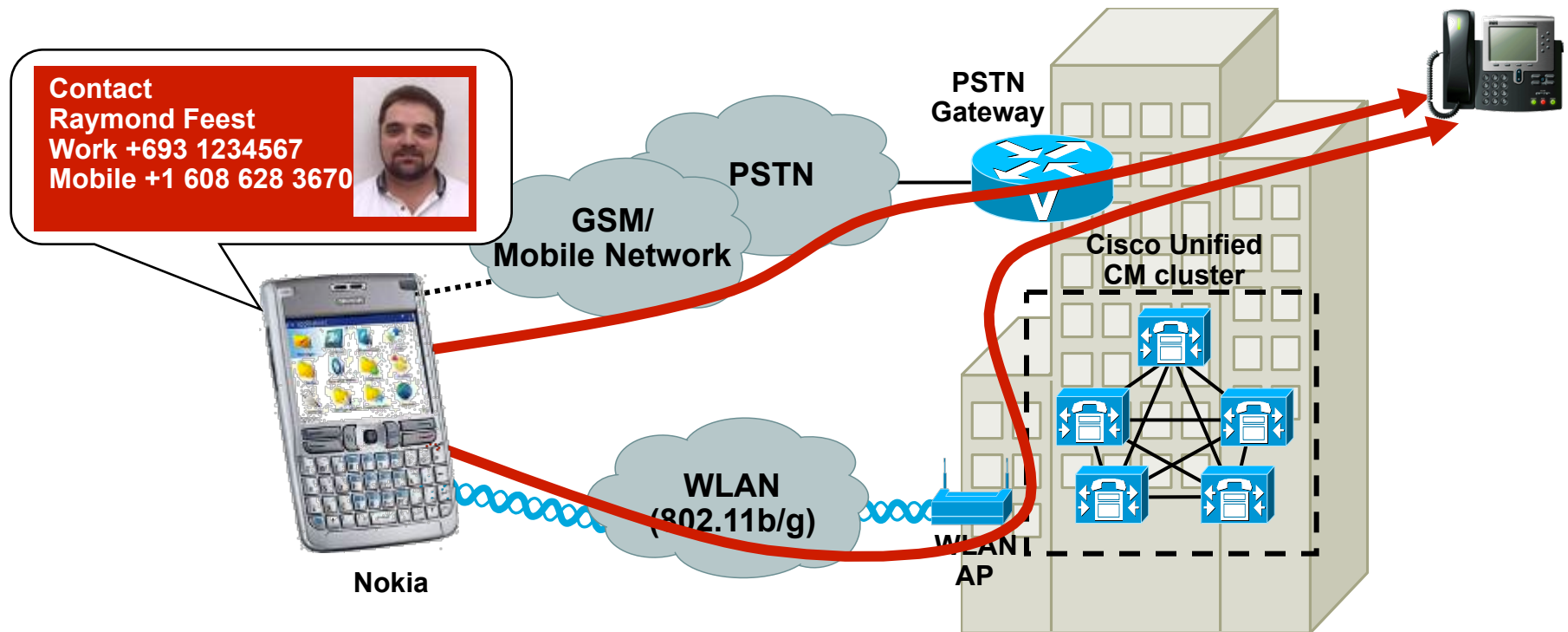
Now: Mobility and Speed Dials



- Put speed dials in using E.164 notation, as many mobile phone users know how to do today
- This pattern needs to be dialable from any dial plan you visit
- Local route group gets you local routing + sign support renders local dialing habits irrelevant, and calling/called party transformations adapt the called/calling numbers as they egress to the PSTN

+ Sign Support

Dial by Contact—GSM or IP Network Call Routing



Dual-Mode Phones Provide the Ability to Use Either PSTN/GSM or WLAN Connectivity for Making and Receiving Calls

- The GSM network can accept + signs
- The IP network now can accept + signs

Combined Benefits

Missed/Received Calls List One-Touch Redial

- The missed/received calls directories contain the globalised version of the incoming calls (e.g., in their E.164 form)
- Hit **dial** and the calls route, assuming that every phone's Device CSS can route calls in the **+** form
- Bang! You are done

Key Takeaways

The Key Takeaways of this Section Are:

- Localise the telephony user interface

Create translation patterns that accept local dialing habits, and expand the called number to a global form (E.164, or globally significant on-net form)

Create calling party transformation patterns that present incoming calls in the form expected by the local users

Are we ready to give that up and just use the + form?

- Globalise call routing within the UC system

Simplifies routing and enables features

- Localise egress to outside networks

By applying appropriate transformation patterns to the called and the calling numbers when offering the call to the gateway

Design Guidelines Agenda

- 7.0 and 7.1 Updates
- 8.0 updates
- Multisite Deployments
- Mobility Considerations

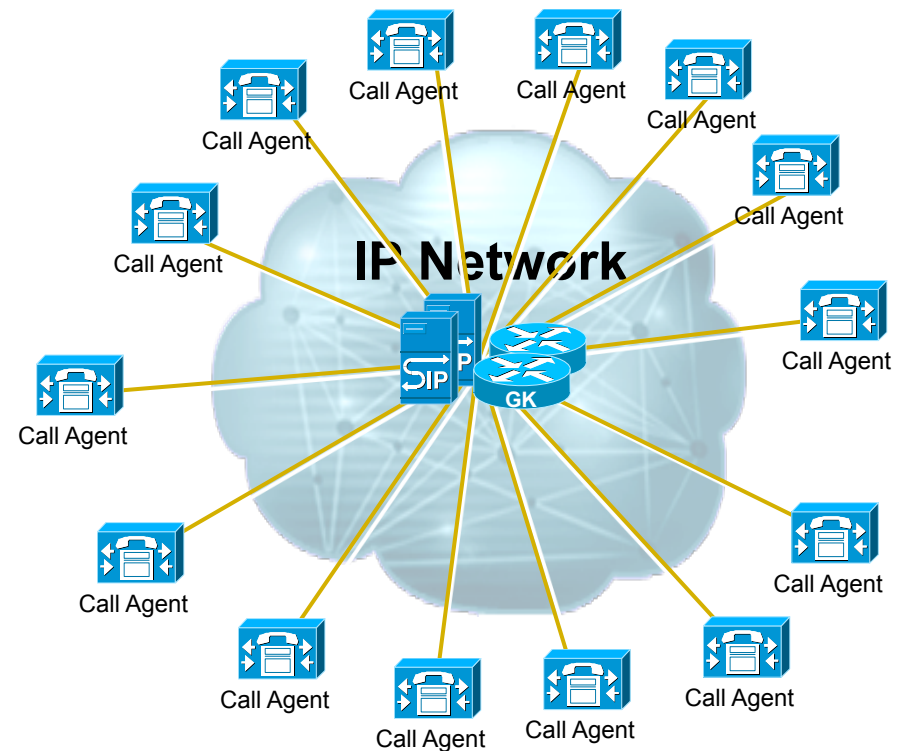
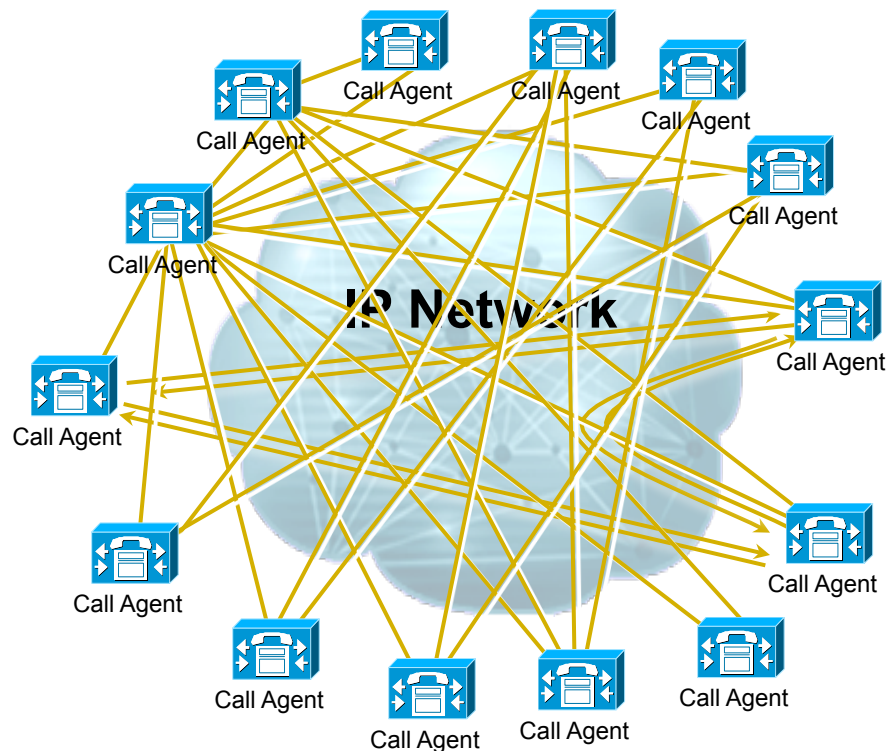
8.0 updates

- We shall focus on the Services Advertisement Framework's Call Control Discovery
- BRKUCC-2003 is a fantastic session covering SAF CCD in its entirety, not just the dial plan aspects of it.
- This section was “stolen” from Stefano's great work in BRKUCC-2003 “A New Approach to Call Routing and Dial Plans based on the Service Advertisement Framework”

Introduction

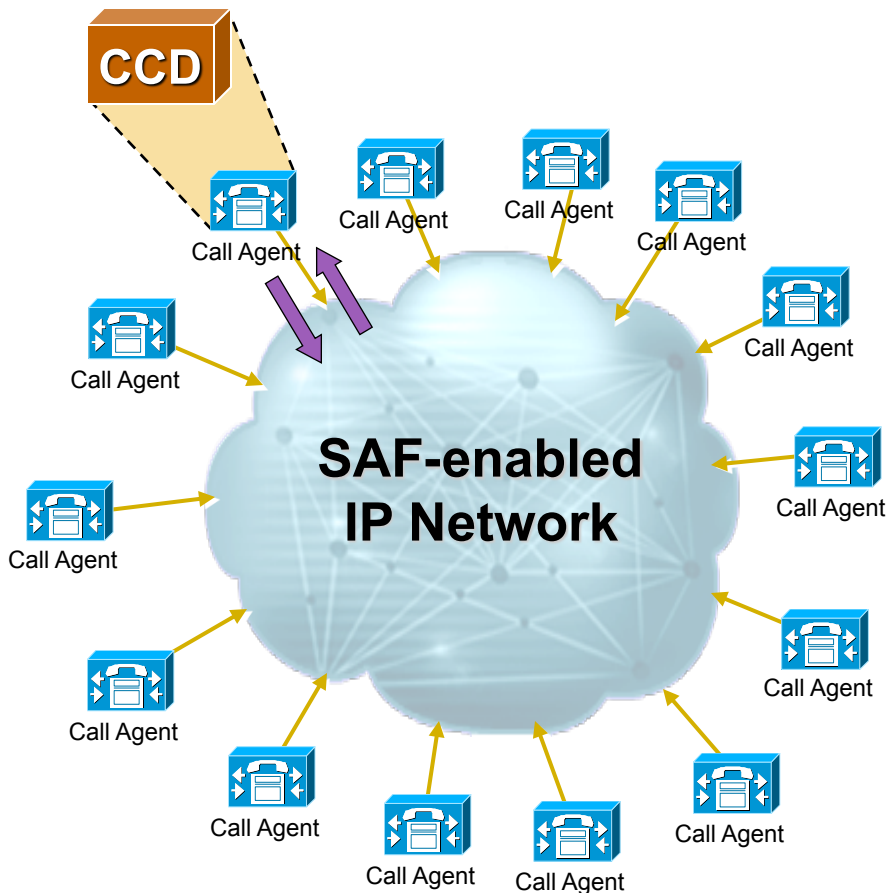
Limitations of Current Call Routing Approaches

- **Configuration complexity, Speed of deployment**
- **High operational cost, TCO**
- **Availability, Business Continuity**



Introduction

Call Control Discovery (CCD): a SAF Service



- Call agents 'discover' each other through the SAF network by:

Advertising their reachability information along with the DN ranges they own

Requesting to learn about other call agents in the network

- Call agents **dynamically** route calls to remote destinations based on received advertisements

Call Control Discovery (CCD)

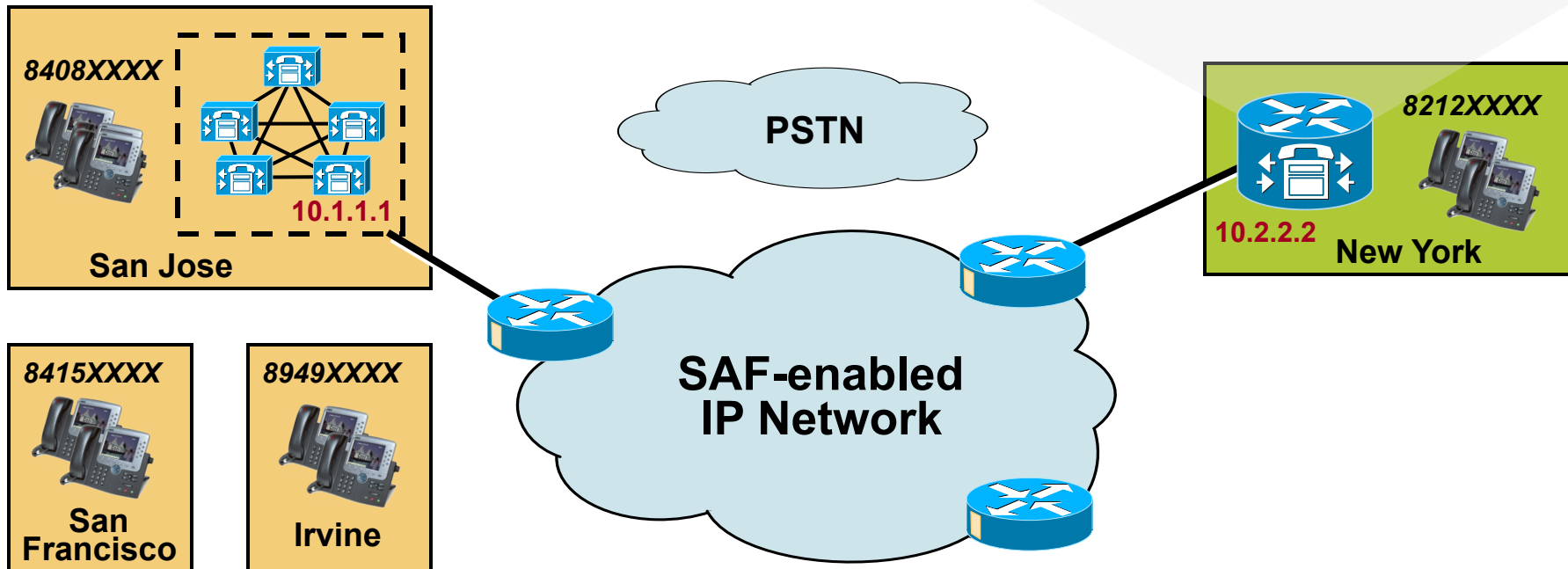
Advertising DN Ranges

Service Advertisement

IP address: 10.1.1.1
Protocol: SIP
DN Patterns:
 8408XXXX [4:+1408555],
 8415XXXX [4:+1415777],
 8949XXXX [4:+1949222]

New York CME Routing Table

DN Pattern	"to DID" rule	IP address	Protocol
8408XXXX	4:+1408555	10.1.1.1	SIP
8415XXXX	4:+1415777	10.1.1.1	SIP
8949XXXX	4:+1949222	10.1.1.1	SIP



Call Control Discovery (CCD)

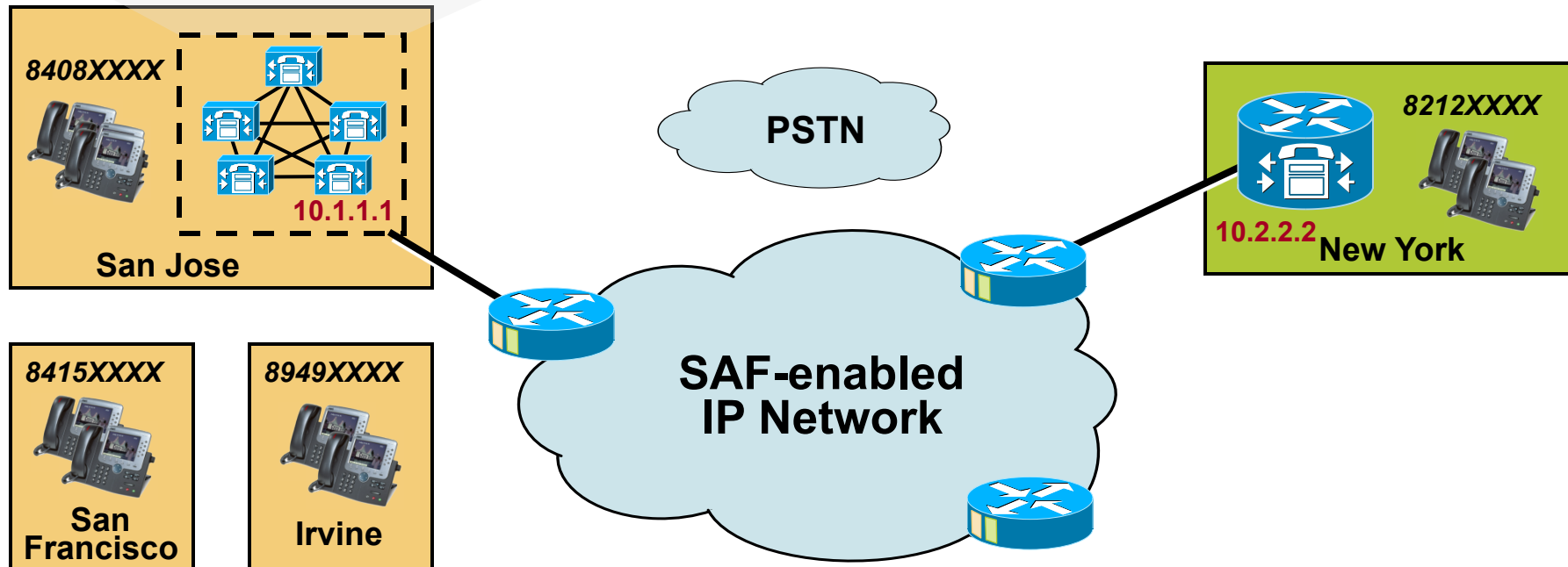
Learning DN Ranges

San Jose CUCM Routing Table

DN Pattern	"to DID" rule	IP address	Protocol
8212XXXX	4:+1212444	10.2.2.2	SIP

Service Advertisement

~~IP address: 10.2.2.2
Protocol: SIP
DN Patterns:
8212XXXX [4:+1212444]~~



Call Control Discovery (CCD)

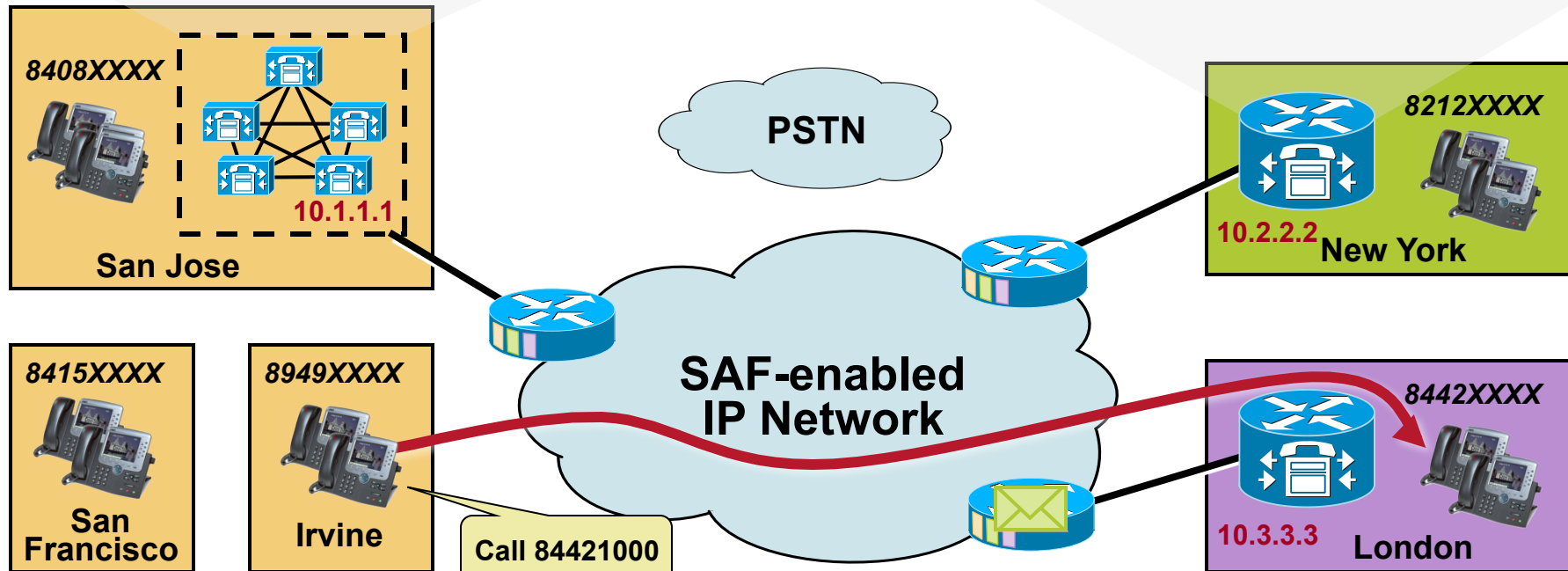
Dynamic Routing

San Jose CUCM Routing Table

DN Pattern	"to DID" rule	IP address	Protocol
8212XXXX	4:+1212444	10.2.2.2	SIP
8442XXXX	4:+442077111	10.3.3.3	H.323

New York CME Routing Table

DN Pattern	"to DID" rule	IP address	Protocol
8408XXXX	4:+1408555	10.1.1.1	SIP
8415XXXX	4:+1415777	10.1.1.1	SIP
8949XXXX	4:+1949222	10.1.1.1	SIP
8442XXXX	4:+442077111	10.3.3.3	H.323



Call Control Discovery (CCD)

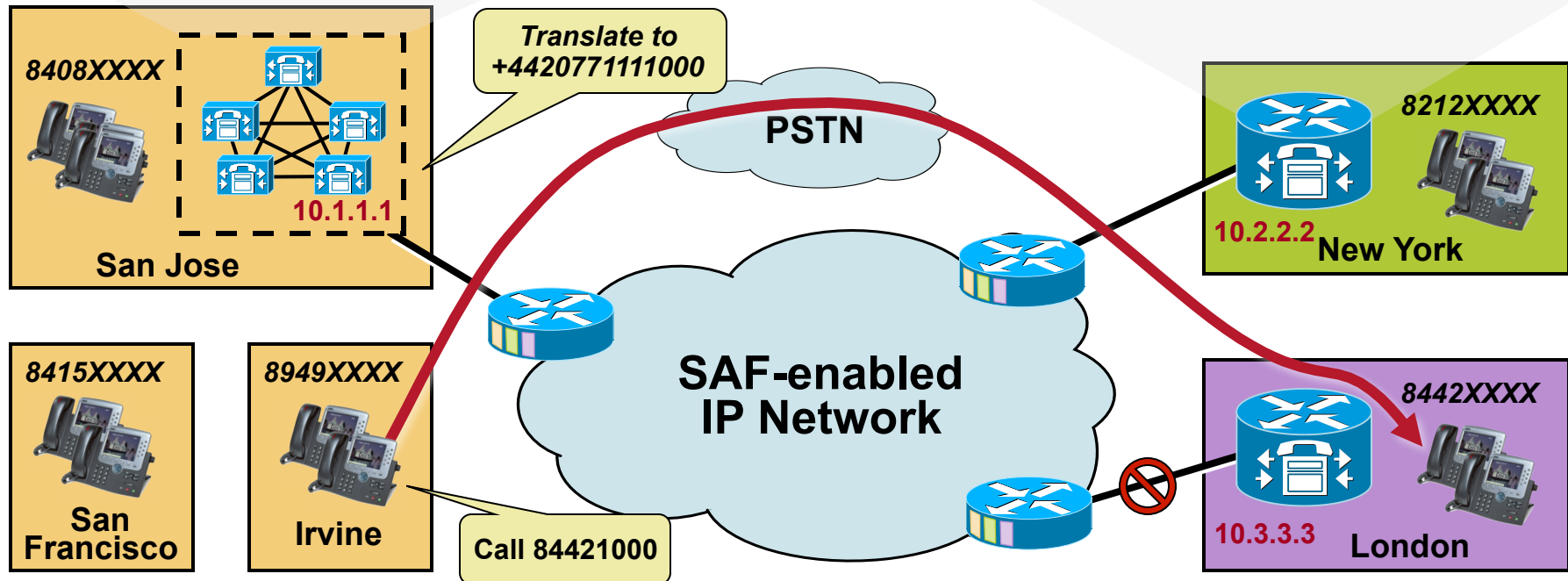
Automatic PSTN Failover

San Jose CUCM Routing Table

DN Pattern	"to DID" rule	IP address	Protocol
8212XXXX	4:+1212444	10.2.2.2	SIP
8442XXXX	4:+442077111	10.3.3.3	H.323

New York CME Routing Table

DN Pattern	"to DID" rule	IP address	Protocol
8408XXXX	4:+1408555	10.1.1.1	SIP
8415XXXX	4:+1415777	10.1.1.1	SIP
8949XXXX	4:+1949222	10.1.1.1	SIP
8442XXXX	4:+442077111	10.3.3.3	H.323



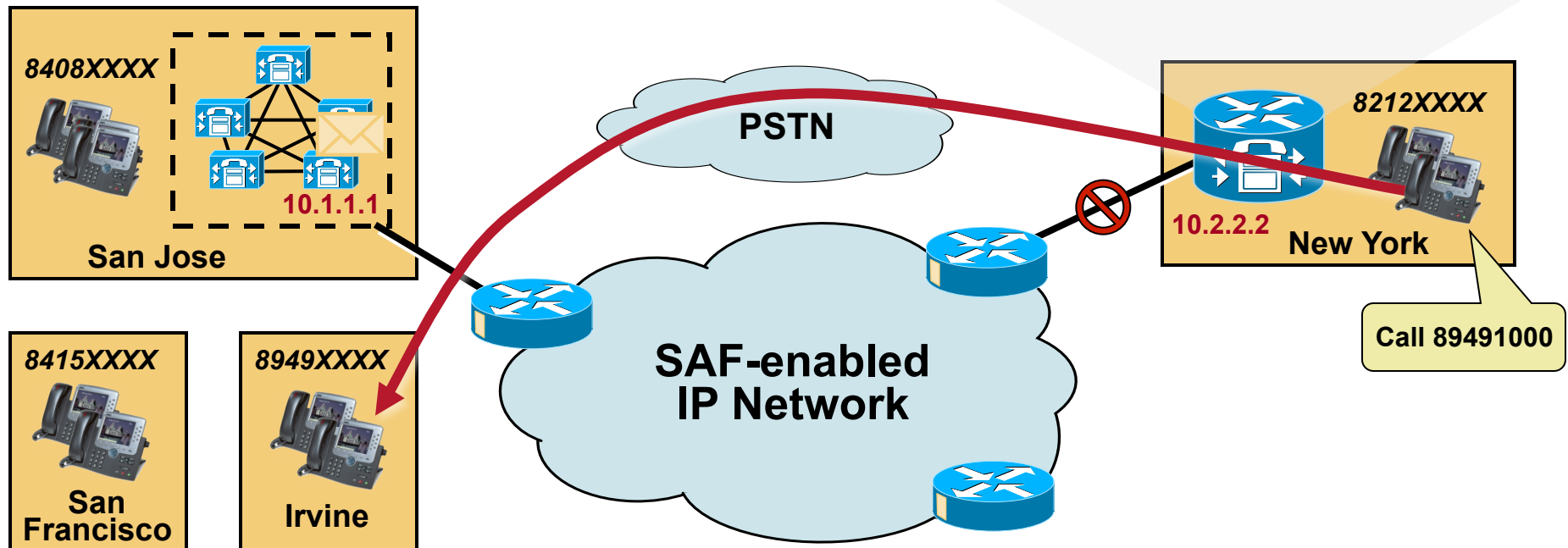
Call Control Discovery (CCD)

Automatic Rerouting for SRST

- SRST subscribes to CCD service but does not publish any patterns
- During WAN failures, SRST uses learned patterns to transparently re-route calls over the PSTN

New York SRST Routing Table

DN Pattern	"to DID" rule	IP address	Protocol
8408XXXX	4:+1408555	10.1.1.1	SIP
8415XXXX	4:+1415777	10.1.1.1	SIP
8949XXXX	4:+1949222	10.1.1.1	SIP



Call Control Discovery (CCD)

3rd Party IP PBX Integration

San Jose CUCM Routing Table

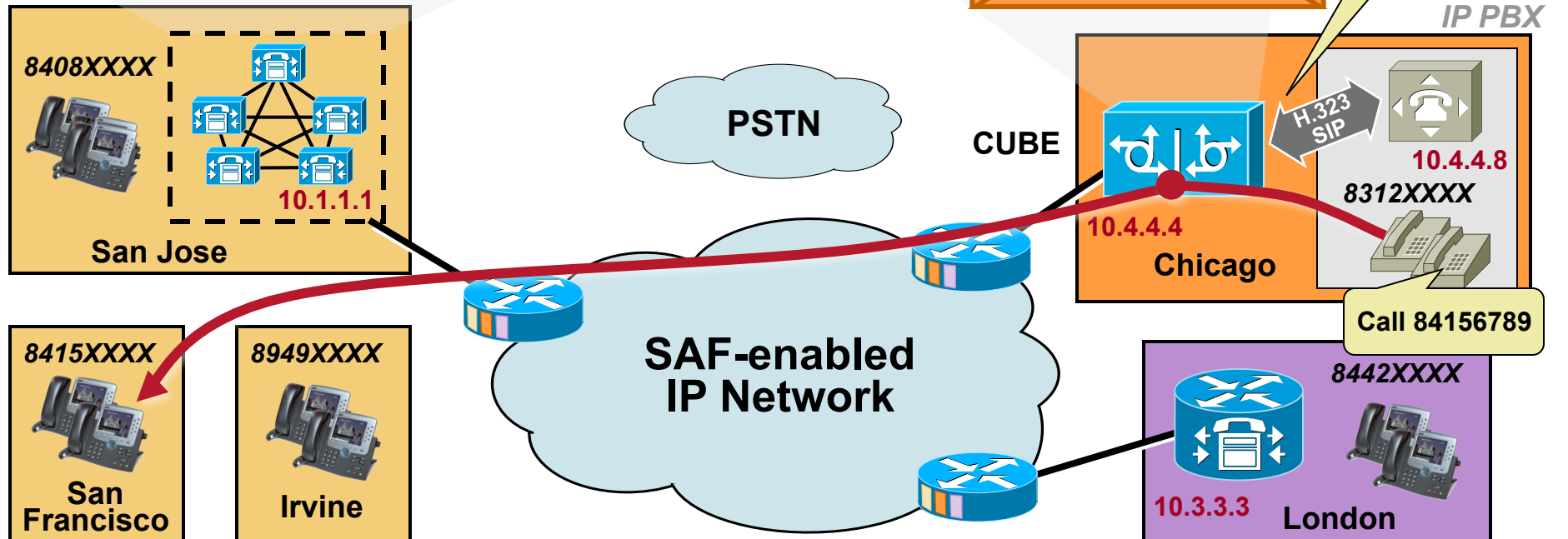
DN Pattern	"to DID" rule	IP address	Protocol
8442XXXX	4:+442077111	10.3.3.3	H.323
8312XXXX	4:+1312888	10.4.4.4	SIP

DN Pattern	"to DID" rule	IP address	Protocol
8408XXXX	4:+1408555	10.1.1.1	SIP
8415XXXX	4:+1415777	10.1.1.1	SIP
8949XXXX	4:+1949222	10.1.1.1	SIP
8442XXXX	4:+442077111	10.3.3.3	H.323

Chicago CUBE Routing Table

IP address: 10.4.4.4
 Protocol: SIP
 DN Patterns:
 8312XXXX [4:+1312888]

Static dial peer for destination 8312XXXX



Call Control Discovery (CCD)

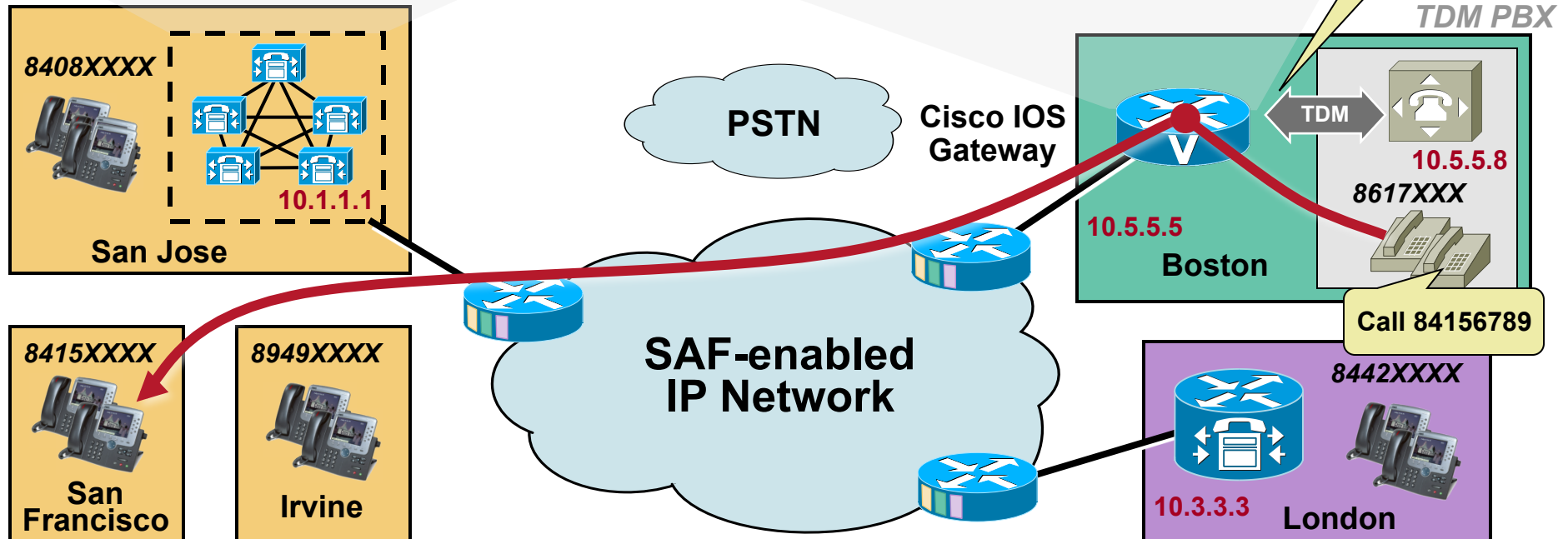
3rd Party TDM PBX Integration

San Jose CUCM Routing Table

DN Pattern	"to DID" rule	IP address	Protocol
8442XXXX	4:+442077111	10.3.3.3	H.323
8617XXXX	4:+1617999	10.5.5.5	SIP

Boston Gateway Routing Table

DN Pattern	"to DID" rule	IP address	Protocol
8408XXXX	4:+1408555	10.1.1.1	SIP
8415XXXX	4:+1415777	10.1.1.1	SIP
8949XXXX	4:+1949222	10.1.1.1	SIP
8442XXXX	4:+442077111	10.3.3.3	H.323



Call Control Discovery (CCD)

Cisco Unified CM details

- Always advertize numbers in a globalized form
e.g.: 89023574, +14089023574, or both.
- If need be, advertize both forms: this allows for the matching of calls to the DID or to the on-net forms of the number

In a remote cluster, if a user dials 89023574, or 0 00 1 408 902 3574 (which will be globalized to +14089023574), both calls will match into the CCD partition directly, and will route the call to IP as a first choice, and the PSTN as a second choice. This avoids having to configure translation patterns between the forms in each “listening” cluster

Call Control Discovery (CCD)

Cisco Unified CM Support Details

- Starting with release 8.0(1), ability to advertise and/or subscribe to the CCD service
- Learned DN patterns dynamically inserted in a specified partition
- Transparent PSTN failover when destination is unreachable
- Scalability:
 - Up to **2,000** advertised DN patterns per cluster
 - Up to **20,000** learned DN patterns per cluster
- DN patterns must be unique (*if duplicates, warning can be issued*)
- Ability to purge and block unwanted patterns (*e.g., from rogue or mis-configured call agents*)
- Extensive troubleshooting support through RTMT and traces

Call Control Discovery (CCD)

Unified CM Configuration – H.323 Trunk

Trunk Configuration Re

Save Delete Reset Apply Config Add New

Device Information

Product:	Inter-Cluster Trunk (Non-Gatekeeper Controlled)
Device Protocol:	Inter-Cluster Trunk
Device Name*	SAF_ICT
Description	
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	QSIG
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes

<input type="checkbox"/> SRTP Allowed - When this flag is checked, IPSec needs to be configured in the network to provide end to end security. Failure
<input type="checkbox"/> H.295 Pass Through Allowed
<input checked="" type="checkbox"/> Enable SAF
Use Trusted Relay Point* <input type="text" value="Default"/>
<input checked="" type="checkbox"/> PSTN Access

Call Control Discovery (CCD)

Unified CM Configuration – SIP Trunk

Trunk Configuration Related Links: [B](#)

Save Delete Reset Apply Config Add New

Status

Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	Call Control Discovery
Device Name*	SAFSIPICT
Description	

SIP Information

MTP Preferred Originating Codec*	711ulaw
Presence Group*	Standard Presence group
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	test1
SIP Profile*	Standard SIP Profile
DTMF Signaling Method*	No Preference

Call Control Discovery (CCD)

Unified CM Configuration – Hosted DN's

The screenshot displays two configuration panels in the Cisco Unified CM interface. The left panel, titled "Hosted DN Group Configuration", shows the configuration for a group named "HDNgrp1". The "PSTN Failover Strip Digits" field is set to "4" and "PSTN Failover Prepend Digits" is set to "+1972555". A red circle highlights these two fields, with a red arrow pointing to a yellow callout box. The right panel, titled "Hosted DN Pattern Configuration", shows the configuration for a pattern named "+1408555XXXX". The "PSTN Failover Strip Digits" field is set to "0" and "PSTN Failover Prepend Digits" is set to "888". A red circle highlights the "Use HostedDN as PSTN Failover" checkbox, which is checked, with a red arrow pointing to another yellow callout box. Both panels show a "Status" section with an "Update successful" message and a toolbar with "Save", "Delete", "Copy", and "Add New" buttons.

Hosted DN Group Configuration

Save Delete Copy Add New

Status
Update successful

Hosted DN Group Info

Name* HDNgrp1

Description

PSTN Failover Strip Digits 4

PSTN Failover Prepend Digits +1972555

Use HostedDN as PSTN Failover

Save Delete Copy Add New

Hosted DN Pattern Configuration

Save Delete Copy Add New

Status
Update successful

Hosted DN Patterns Info

Hosted Pattern* +1408555XXXX

Description

Hosted DN Group* HDNgrp1

PSTN Failover Strip Digits 0

PSTN Failover Prepend Digits 888

Use HostedDN as PSTN Failover

Save Delete Copy Add New

Applies the same "toDID" rules to all DN Patterns in this group

Used to advertise full E.164 ranges instead of internal numbers + "toDID" rules

Call Control Discovery (CCD)

Unified CM Configuration – Hosted DN's (2)

Find and List Hosted DN Patterns

+ Add New Select All Clear All Delete Selected

Status

3 records found

Hosted DN Pattern (1 - 3 of 3) Rows

Find Hosted DN Pattern where

<input type="checkbox"/>	Hosted Pattern ^	Description	Hosted DN Group
<input type="checkbox"/>	+9997XXX		HDNGrp2
<input type="checkbox"/>	7XXX		HDNgrp1
<input type="checkbox"/>	9727XXX		HDNgrp1

- Hosted DN patterns to be advertised are configured by the administrator
- Allows flexibility in designing on-net dial plan and choosing which DN ranges to advertise to other call agents

Call Control Discovery (CCD)

Unified CM Configuration – Advertising Service

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

CCD Advertising Service Configuration Related Links: [Find and List C](#)

Save Delete Copy Reset Add New

Status

Add successful

CCD Advertising Service Info

Name*

Description

SAF SIP Trunk ▾

SAF H323 Trunk ▾

HostedDN Group* ▾

Activated Feature

- Each HostedDN Group can be associated with only one CCD Advertising Service
- SAF Trunks can be re-used by different CCD Advertising Services and CCD Requesting Services
- The SAF trunks' Unified CM groups determine on which nodes this service runs and which IP addresses are advertised through SAF

Call Control Discovery (CCD)

Unified CM Configuration – Requesting Service

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

CCD Requesting Service Configuration

Save Delete Reset

CCD Requesting Service Info

Name*

Description

Route Partition

Learned Pattern Prefix

PSTN Prefix

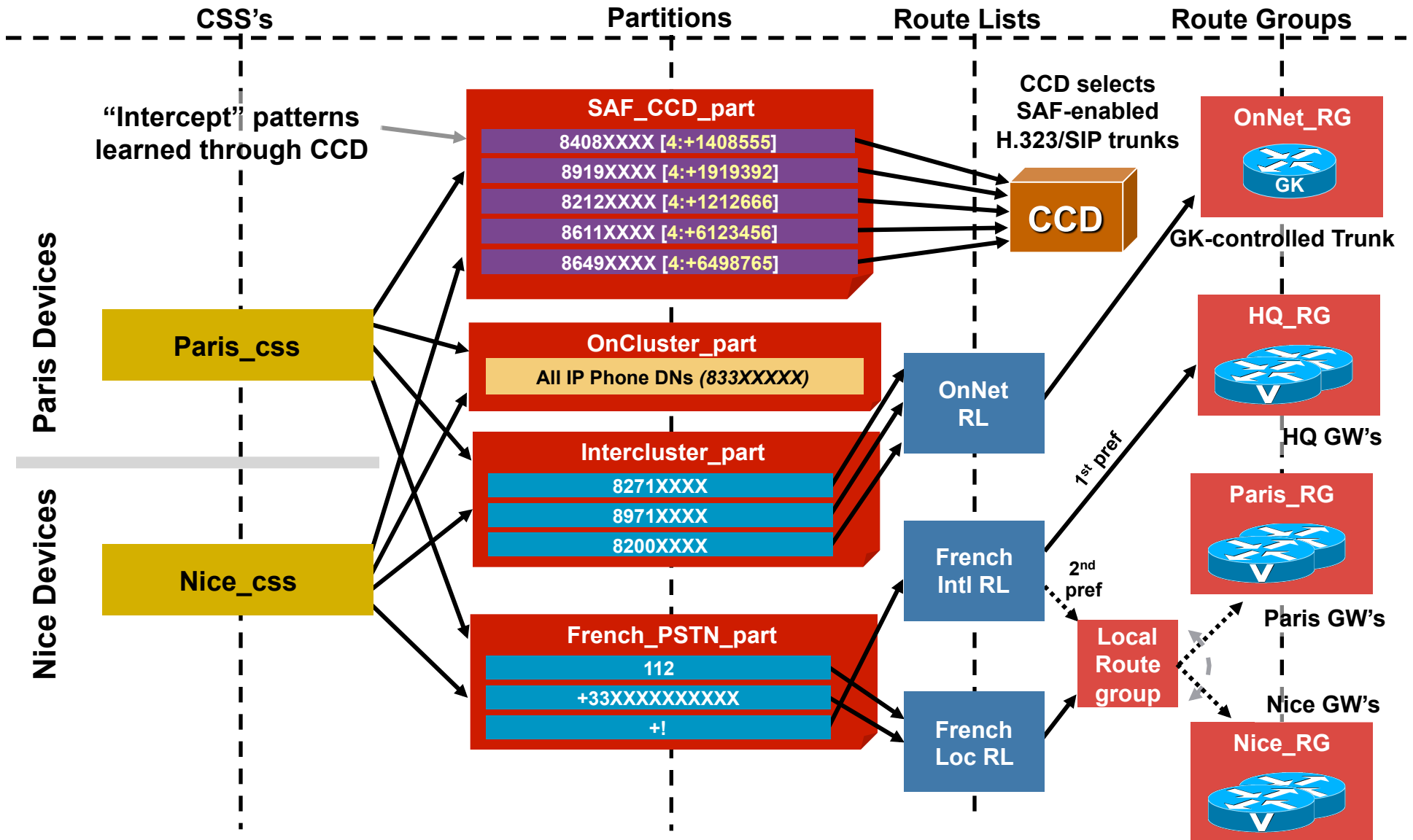
Available SAF Trunks

Selected SAF Trunks

Activated Feature

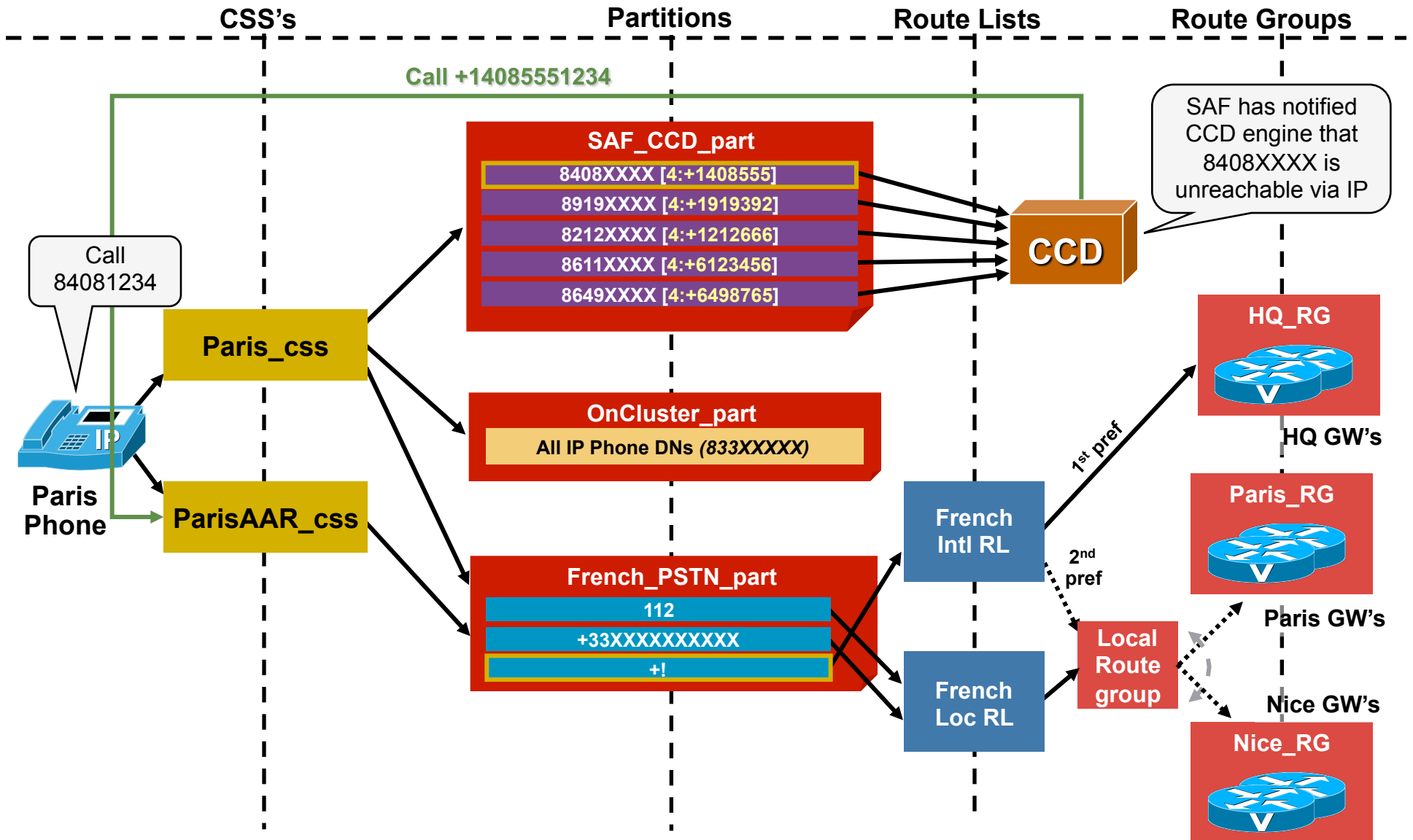
Call Control Discovery (CCD)

Integration with "Static Routing"



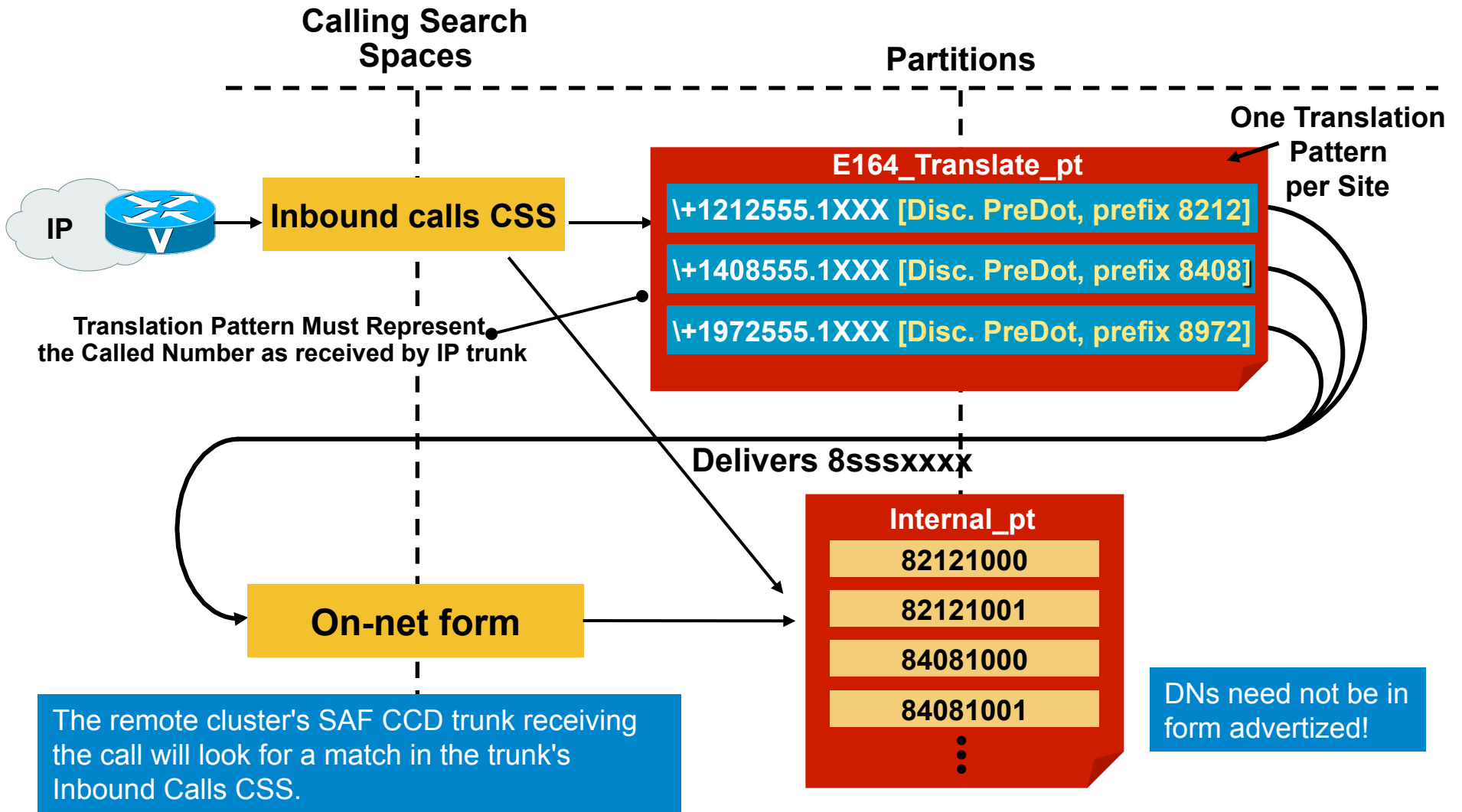
Call Control Discovery (CCD)

Integration with "Static Routing" – PSTN Failover



Call Control Discovery (CCD)

Inbound calls from CCD trunks



Call Control Discovery (CCD)

Monitoring and Troubleshooting for Unified CM

- RTMT is used to monitor learned routes and configured SAF Forwarders
- SAF/CCD tracing is included as part of the Unified CM SDI and SDL traces

The screenshot displays the Cisco Unified Communications Manager Real Time Monitoring Tool (RTMT) interface. The window title is "Cisco Unified Communications Manager Real Time Monitoring Tool (Currently Logged to: 10.194.121.30)". The menu bar includes File, System, CallManager, AnalysisManager, Edit, Window, Application, and Help. The main window is titled "Real Time Monitoring Tool For Cisco Unified Communications Solutions".

The interface is divided into several sections:

- System:** Contains a "Learned Pattern" tab.
- CallManager:** A navigation tree on the left with categories: Trunk Activity, SDL Queue, SIP Activity, Device (Device Summary, Device Search, Phone Summary), Service (Cisco TFTP, Heartbeat, Database Summary), CTI (CTI Manager, CTI Search), Report (Learned Pattern, SAF Forwarders), and Intercompany Media Services (Routing).
- Learned Pattern View:** A table showing learned patterns for the selected node "cucm-c1.cisco.com".

Pattern	TimeSta...	Status	Protocol	AgentId	IP Address	ToDID	CUC...
141XXX	2009/10/...	Reachable	SIP	C10.194...	10.194.121.14(5...		1
141XXX	2009/10/...	Reachable	H323(Q...	C10.194...	10.194.121.14(4...		1

At the bottom of the window, there are buttons for "Refresh", "Filter", "Clear Filter", "Find", and "Save". A status bar at the very bottom indicates "Report finishes downloading for node cucm-c1.cisco.com".

Design Guidelines Agenda

- 7.0 and 7.1 Updates
- 8.0 updates
- **Multisite Deployments**
- Mobility Considerations

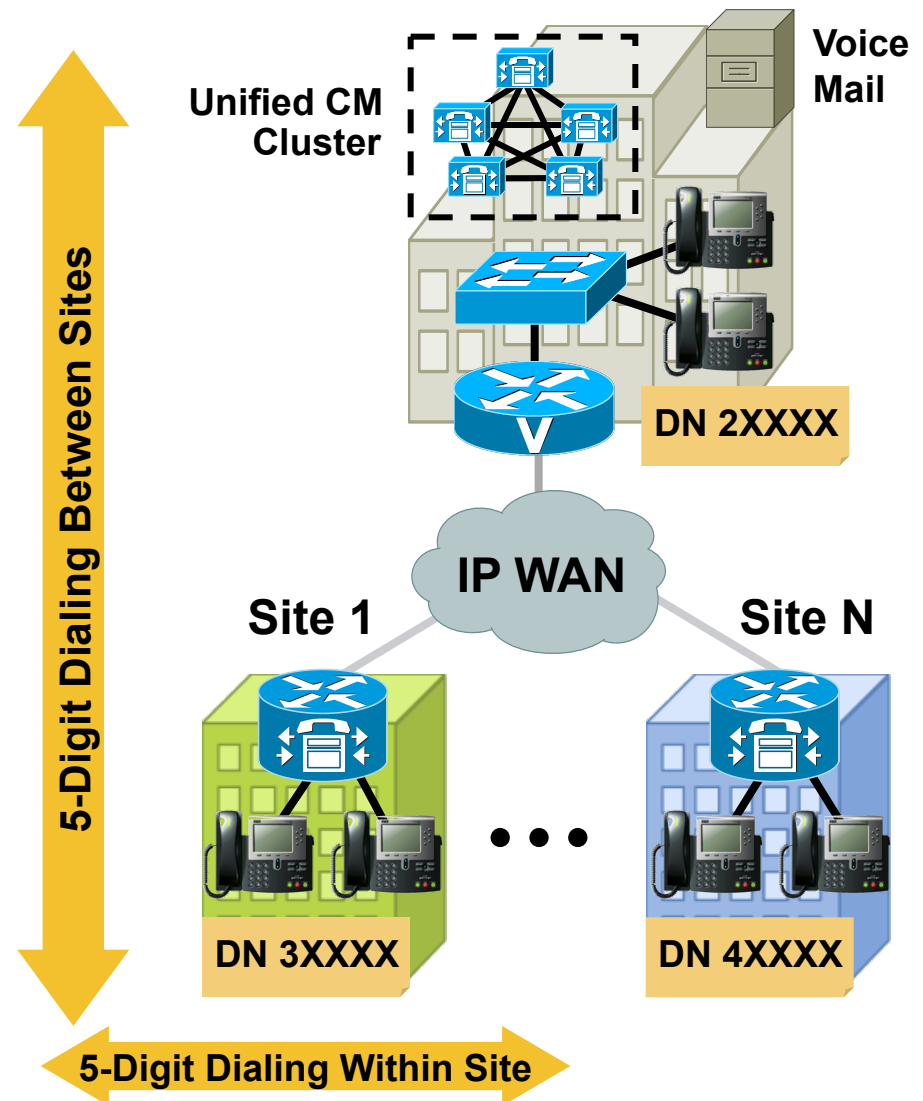
Design Best Practices Agenda

- 7.0 and 7.1 Updates
- 8.0 updates
- **Multisite Deployments**
 - Choosing a Dial-Plan Approach
 - Uniform On-Net Dialing → Moved to Appendix
 - Variable-Length On-Net Dialing with Partitioned Addressing → Moved to Appendix
 - Variable-Length On-Net Dialing with Flat Addressing
 - Tail End Hop Off (a.k.a. Toll Bypass)
- Mobility Considerations

Choosing a Dial Plan Approach

Uniform On-Net Dialing

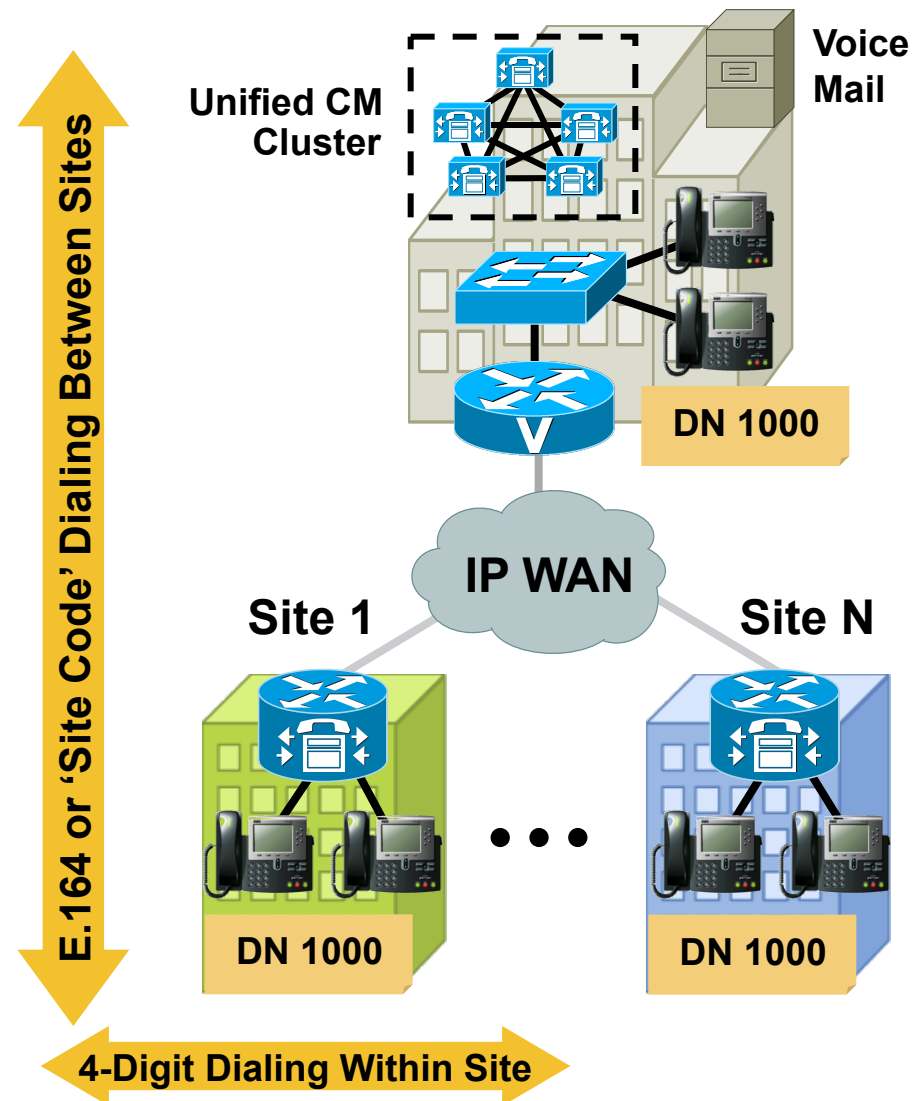
- Dialing within a site and across sites with same number of digits (e.g., five)
- Extensions are globally unique
- Easy to design and configure
- Limited scalability of the addressing method (**number of sites, number of extensions**)



Choosing a Dial Plan Approach

Variable-Length On-Net Dialing (VLOD)

- Abbreviated dialing within a site (four or five digits)
- Identical extensions (e.g., 1000) may appear at different sites
- Intersite calls use an **escape code** (e.g., 9 + full E. 164, or 8 + site code + extension)
- Easier scalability for large numbers of extensions and sites



Design Best Practices Agenda

- 7.0 and 7.1 Updates

- **Multi-Site Deployments**

 - Choosing a Dial Plan Approach

 - Uniform On-Net Dialing → Moved to Appendix

 - Variable-Length On-Net Dialing with Partitioned Addressing
→ Moved to Appendix

 - Variable-Length On-Net Dialing with Flat Addressing

 - Tail End Hop Off (a.k.a. Toll Bypass)

- **Mobility Considerations**

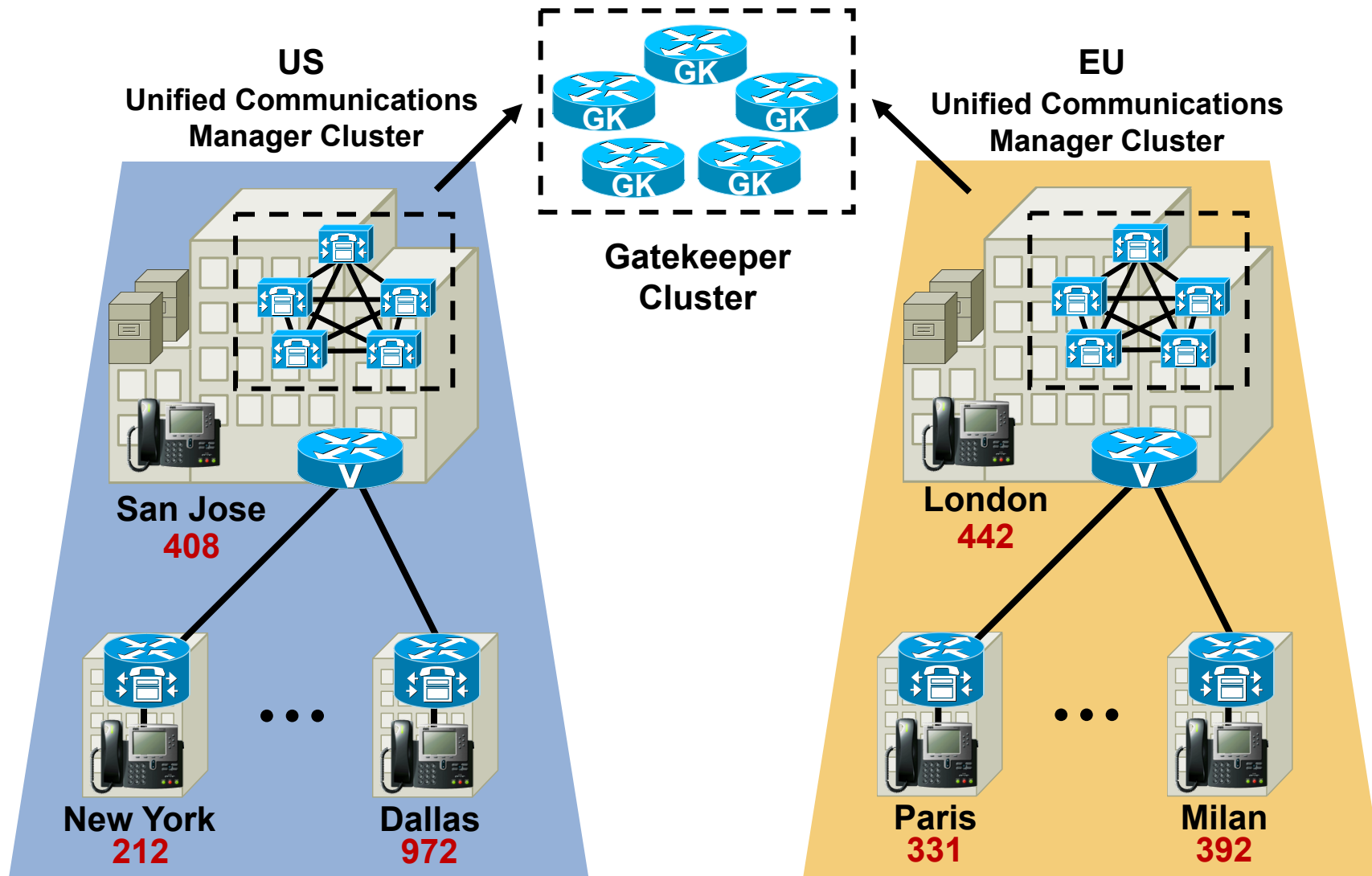
VLOD with Flat Addressing

Use This Model If...

- Branches interact often
- Users dial a **site code** for intersite calls
- Intersite calls go over IP WAN
- CTI applications are used across sites
- International deployment
- A global on-net dial plan is needed
- This approach is presumed by many upcoming features' design guidance. **If you can start with this approach, you will most likely be future-proofed**

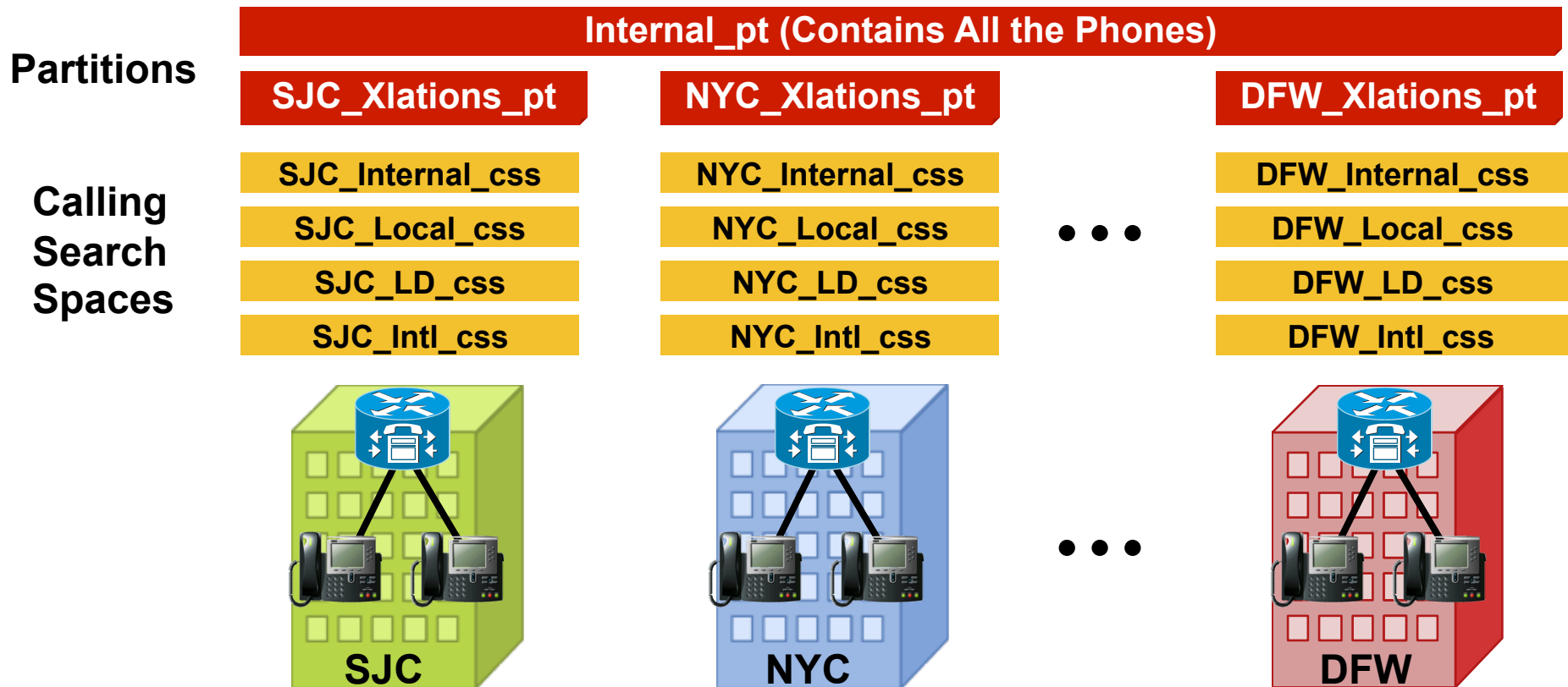
VLOD with Flat Addressing

Site Code Assignment



VLOD with Flat Addressing

Partitions and Calling Search Spaces



*Note: If Using the Line/Device CSS Approach or LRG, the Number of CSSs Can Be Reduced

VLOD with Flat Addressing Line Configuration

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Directory Number Configuration

[Configure Device \(SEP000785287409\)](#)
[Dependency Records](#)

Associated With

- ADP000785287409 (Line 2)
- SEP000785287409 (Line 2)

Directory Number: 82121000 (Internal)
Status: Ready
Note: Any update to this Directory Number automatically resets the associated devices

Update Remove from Device Reset Devices

Directory Number

Directory Number* 82121000
Partition Internal

Directory Number Settings

Voice Mail Profile <None>
(Choose <None> to use default)

Calling Search Space <None>

AAR Group <None>

User Hold Audio Source <None>

Line Settings for this Device

Display (External Caller ID) John Smith

Line Text Label 1000 *

External Phone Number Mask 2125551000

Message Waiting Policy Use System Policy

Ring Setting (Phone Idle) Use System Default

Ring Setting (Phone Active)** Use System Default



* Note: Line Text Label Is not Preserved in SRST Mode

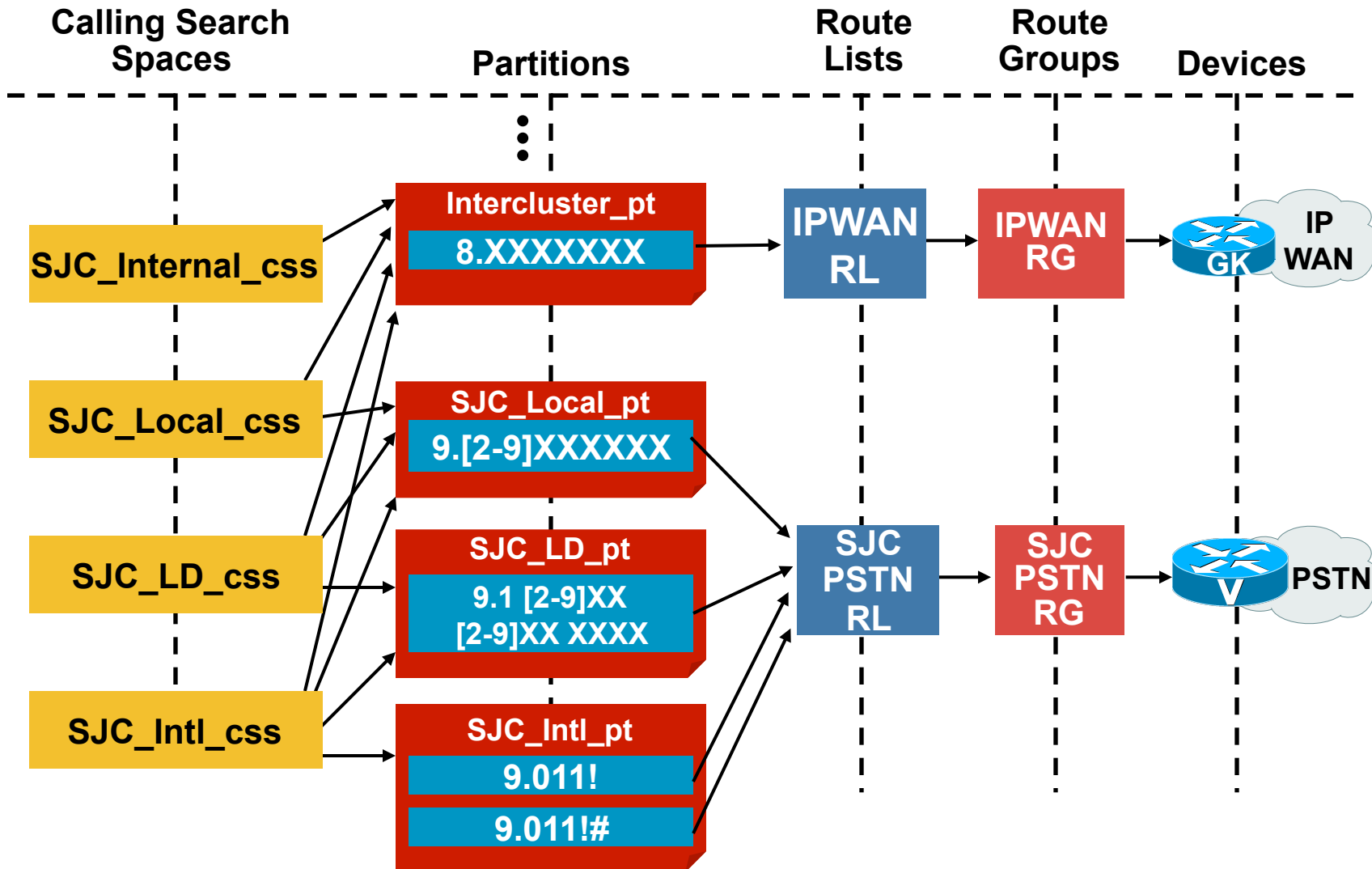
VLOD with Flat Addressing

Outgoing Inter-cluster WAN/PSTN Calls

- **Option one: eight digit only**
 - Simple, easy to maintain
 - No automatic PSTN failover (manual redial)
- **Option two: eight digit + E.164 with centralised PSTN failover**
 - A little more configuration and maintenance
 - Automatic PSTN failover using central gateway
 - (SJC in our example)
 - Possibility to place calls on-net even when dialed as PSTN
- **Option three: eight digit + E.164 with distributed PSTN failover**
 - A lot more configuration and maintenance (until 7.0!)
 - Automatic PSTN failover using local gateway

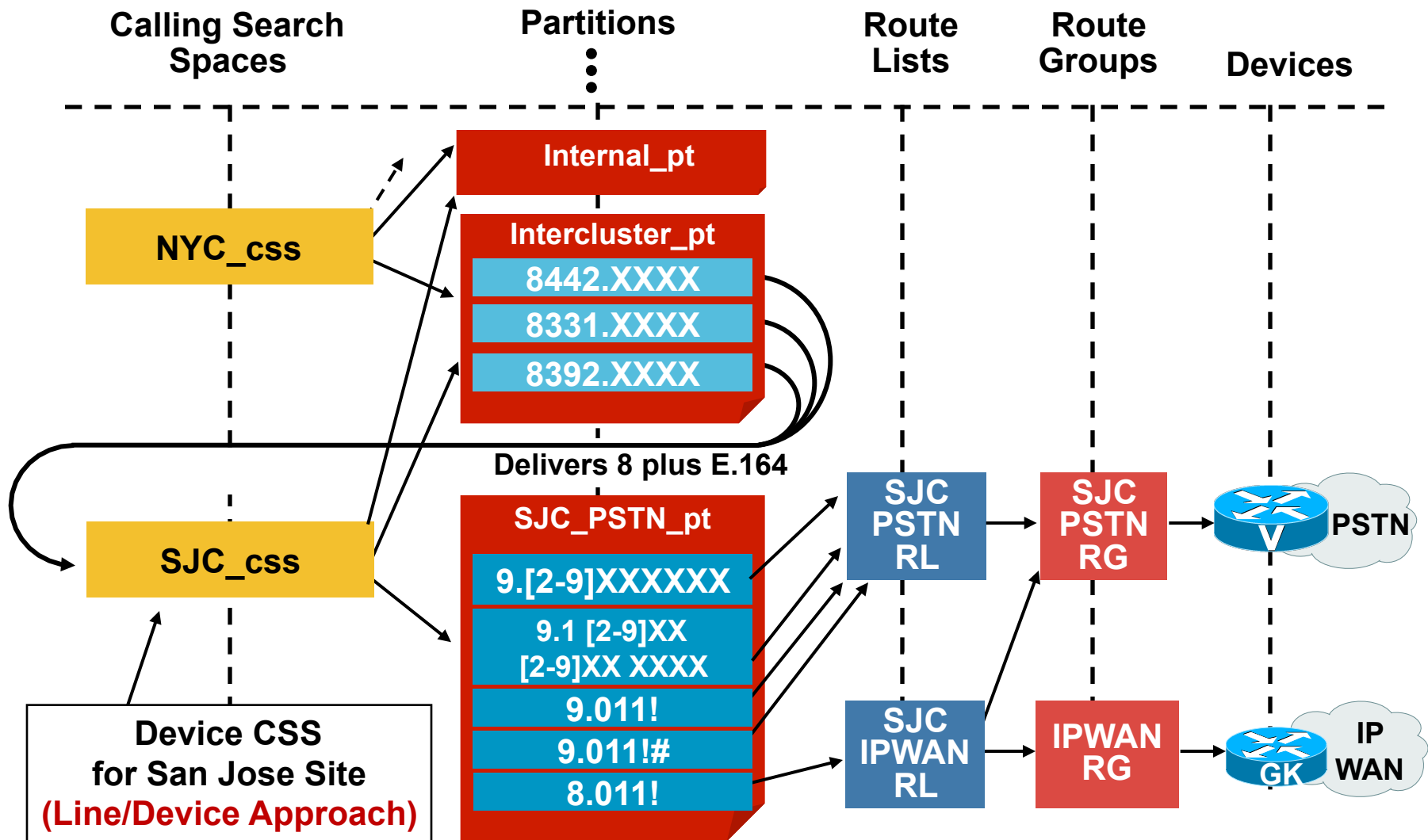
VLOD with Flat Addressing

Outgoing PSTN/IP WAN Calls: Option One



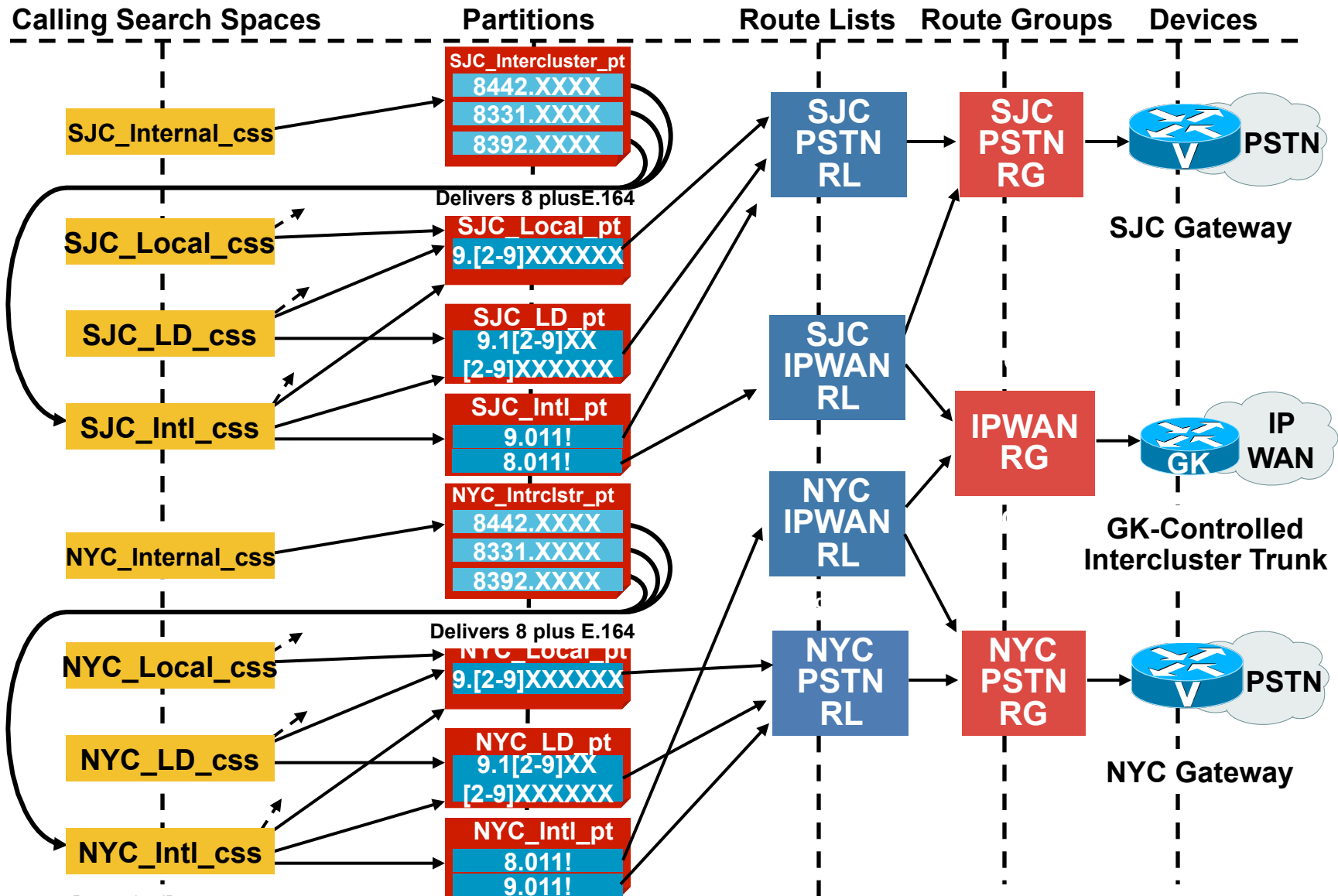
VLOD with Flat Addressing

Outgoing PSTN/IP WAN Calls: Option Two



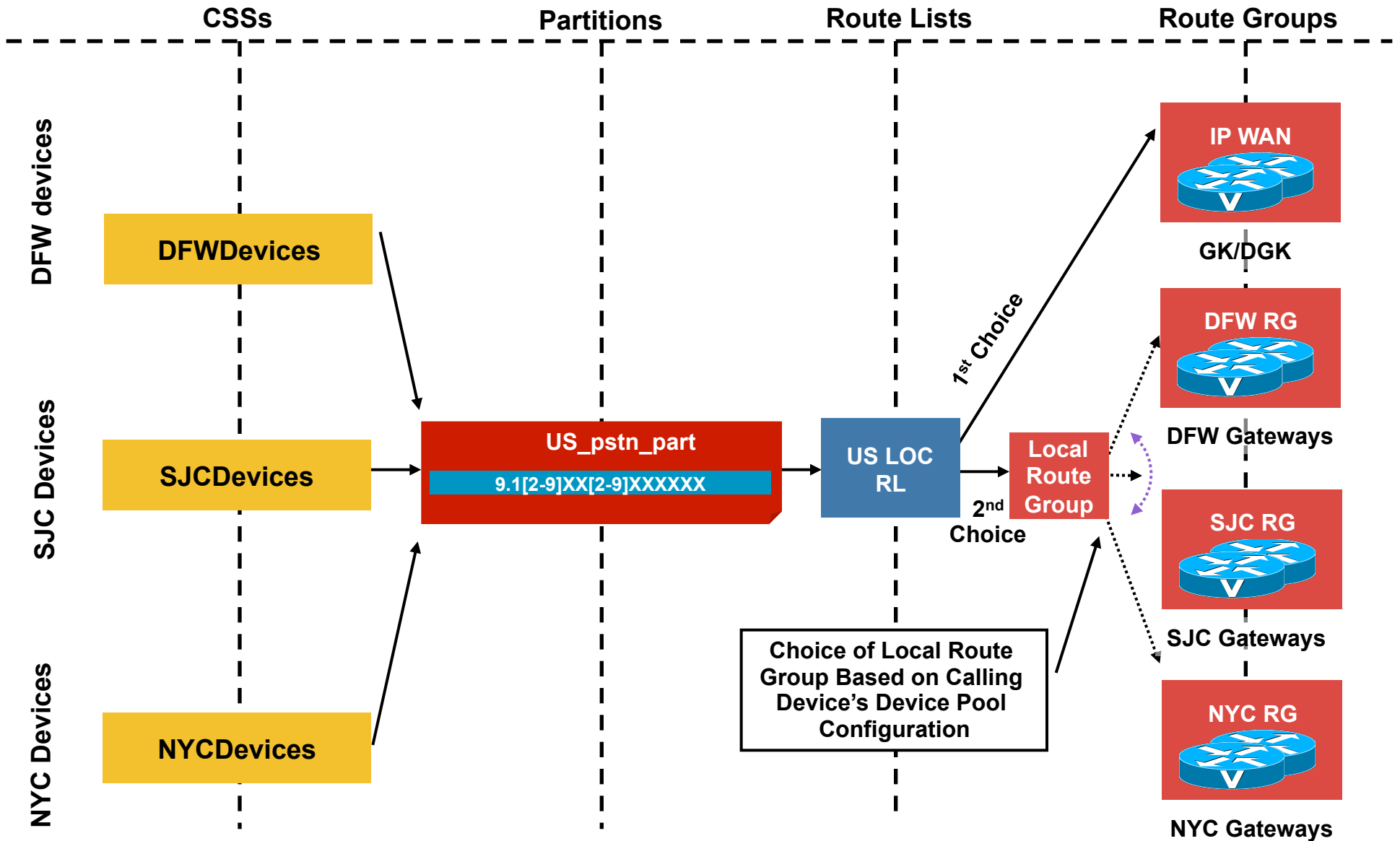
VLOD with Flat Addressing

Outgoing PSTN/IP WAN Calls: Option Three



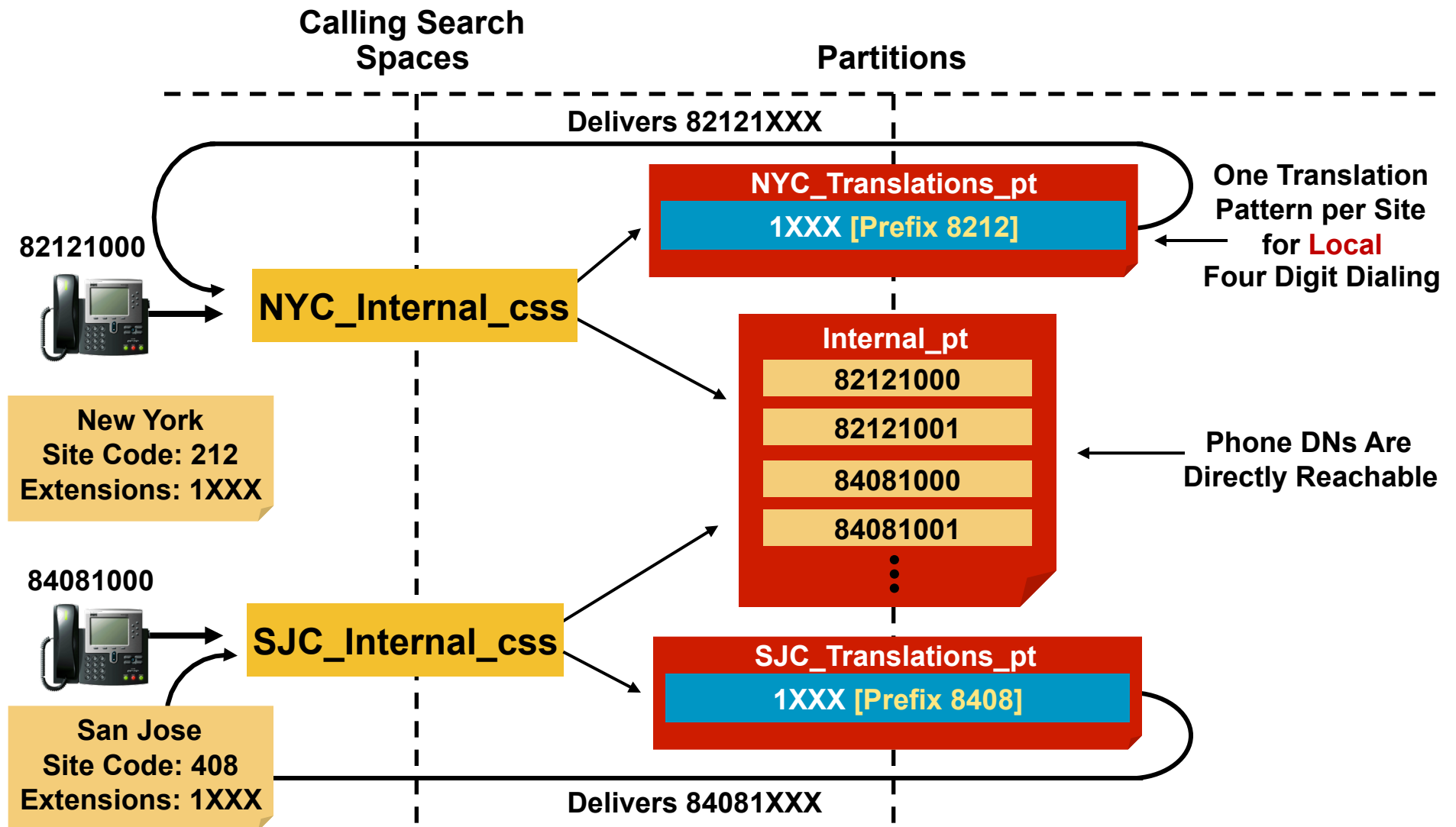
VLOD with Flat Addressing

Outgoing PSTN/IP WAN Calls: Option Three—LRG to the Rescue!



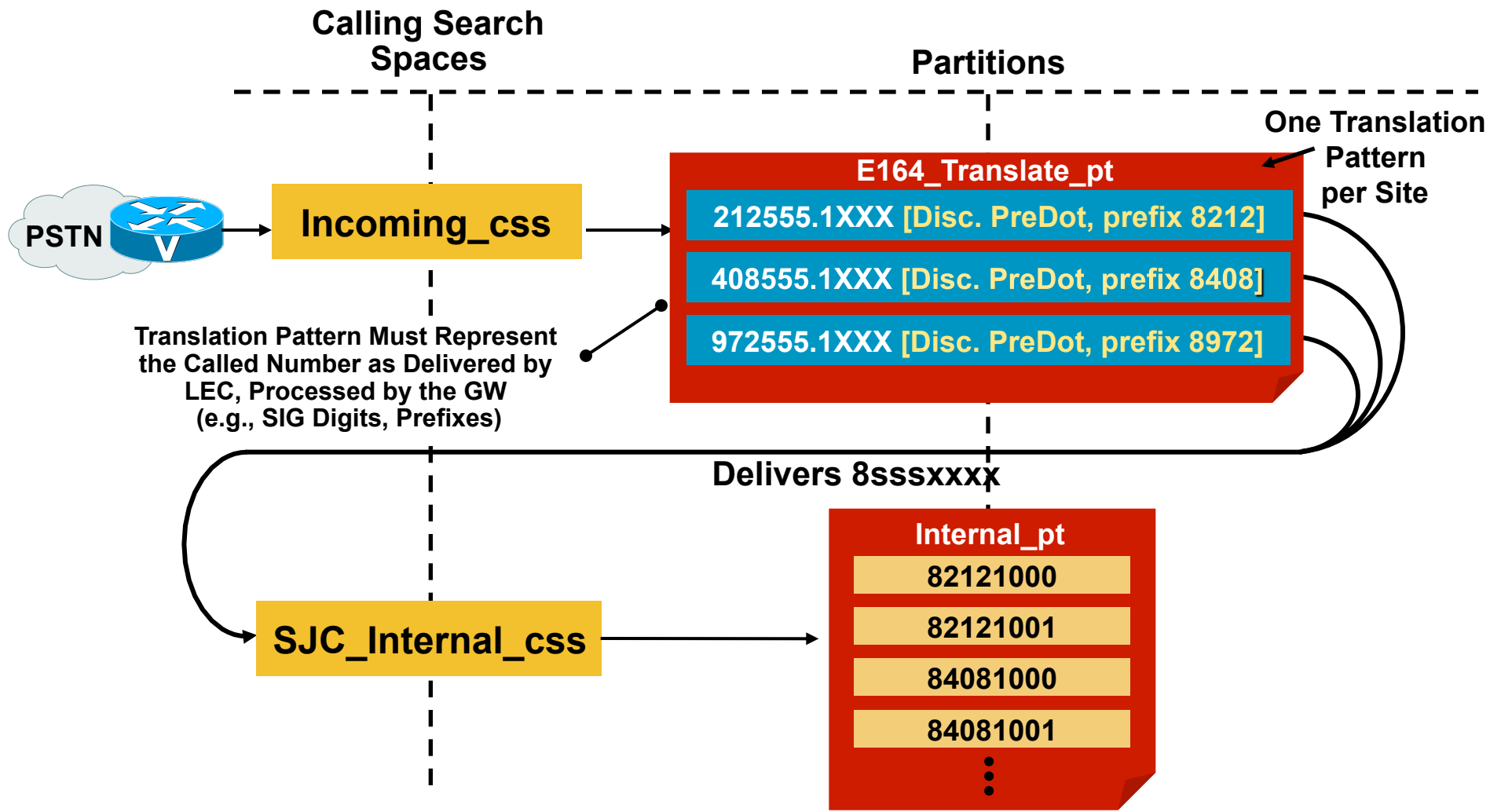
VLOD with Flat Addressing

Intra/Inter-site Calls Within a Cluster



VLOD with Flat Addressing

Incoming PSTN/IP WAN Calls (1/3 methods)



VLOD with Flat Addressing

Incoming PSTN/IP WAN Calls (2/3 methods)

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Gateway Configuration

[Back to Find/List Gateways](#)
[Dependency Records](#)

Assigned to Route Group:PSTN RG

Product : Cisco Catalyst 6000 T1 VoIP Gateway
Gateway : S0/DS1-0@SDA0001C96ACDDE
Device Protocol: Digital Access PRI
Registration: Registered with Cisco CallManager SJCCEM2
IP Address: 10.0.1.13

Status: Ready

Call Routing Information

Inbound Calls

Significant Digits*

Calling Search Space

AAR Calling Search Space

Prefix DN

Configure GW to Strip and Prefix Relevant Digits

VLOD with Flat Addressing

Incoming PSTN/IP WAN Calls (3/3 methods)

Incoming calling party settings now allow for using Calling Party Transformation Patterns to manipulate the calling party number when calls enter the system from gateways. One Calling Party Transformation Pattern CSS is available for each numbering type. Note: all calls are tagged with numbering type “Unknown” on SIP Gateways and trunks. This allows digit manipulation to be based on regular expressions, for more flexible matching.

Geo Location Configuration
Geo Location
Geo Location Filter

Incoming Calling Party Settings
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Use Dev Pool CSS	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input checked="" type="checkbox"/>	<input type="text" value="< None >"/>

i *- indicates required item.
i **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

VLOD with Flat Addressing

Gatekeeper Configuration

```
gatekeeper
zone local US cisco.com 10.9.11.1
zone local EU cisco.com 10.20.1.1
no zone subnet US default enable
no zone subnet EU default enable
zone subnet US 10.9.11.2/32 enable
zone subnet US 10.9.11.3/32 enable
zone subnet EU 10.20.1.2/32 enable
zone subnet EU 10.20.1.3/32 enable
zone prefix US 14085551...
zone prefix US 12125551...
zone prefix US 19725551...
zone prefix EU 442077881...
zone prefix EU 33144551...
zone prefix EU 390266771...
gw-type-prefix 1#* default-technology
bandwidth interzone zone US 256
bandwidth interzone zone EU 256
arq reject-unknown-prefix
no shutdown
```

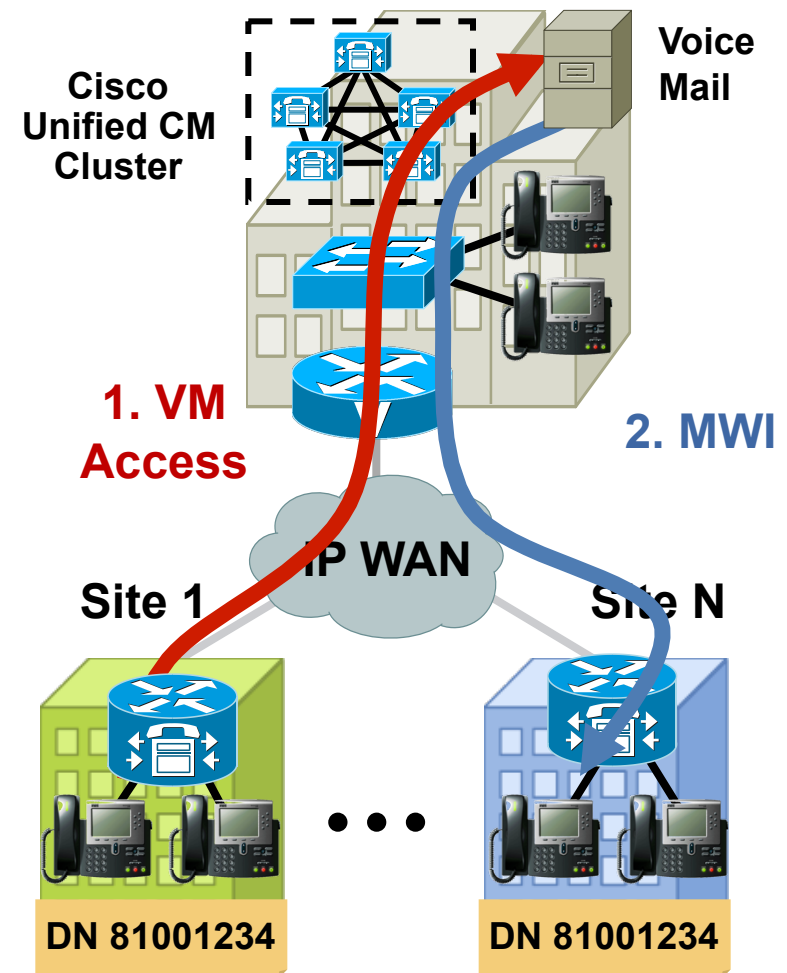
**! Replace E.164s with 8-Digit
! Numbers for Option 1**

!
zone prefix US 84081...
zone prefix US 82121...
zone prefix US 89721...
zone prefix EU 84421...
zone prefix EU 83311...
zone prefix EU 83921...
!

VLOD with Flat Addressing

Voice Mail Integration

- Each eight digit extension is unique → it can be used to identify a voicemail box
- No need to use masks in voicemail profile
- No translations necessary for MWI

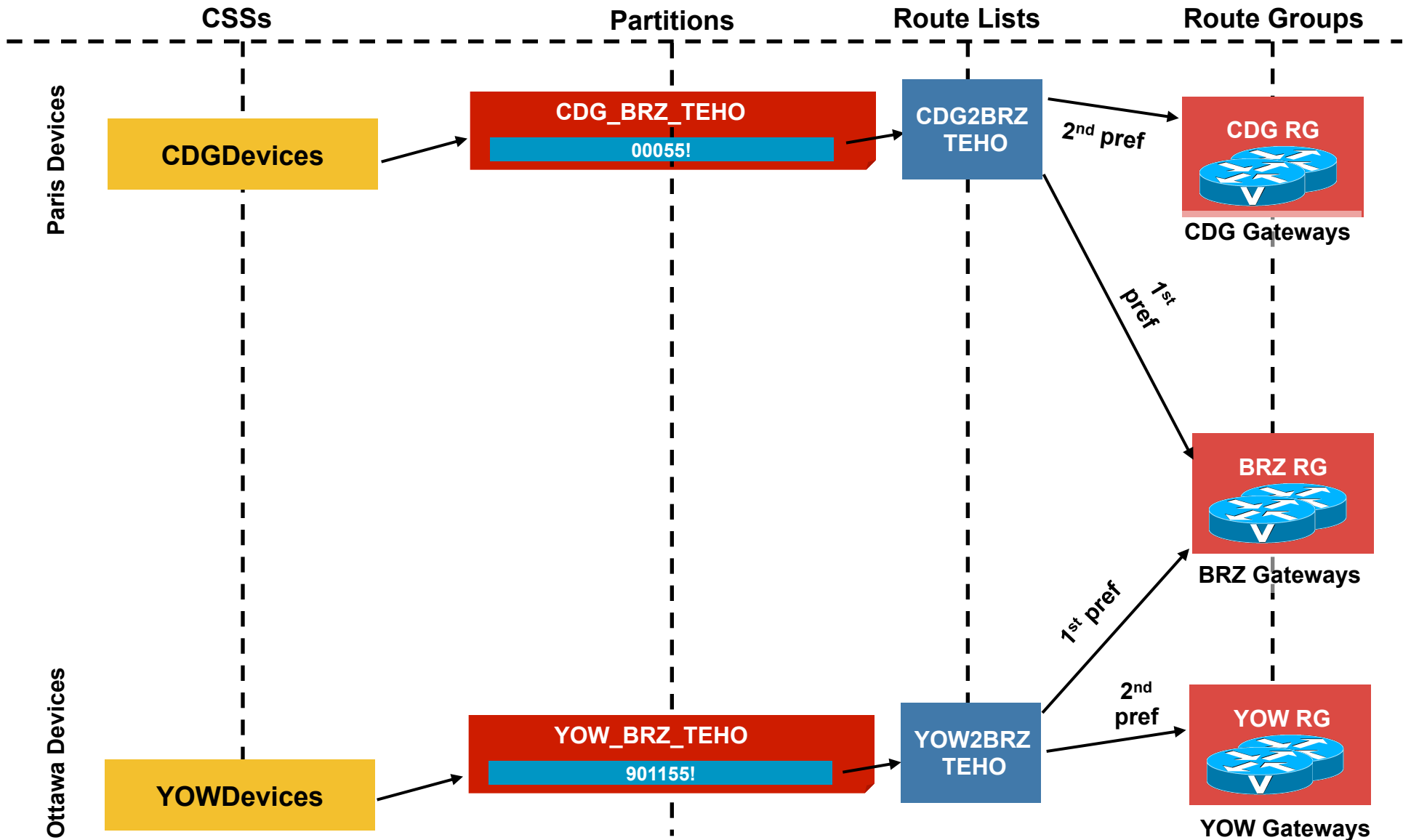


Design Best Practices Agenda

- 7.0 and 7.1 Updates
- **Multi-Site Deployments**
 - Choosing a Dial Plan Approach
 - Uniform On-Net Dialing → Moved to Appendix
 - Variable-Length On-Net Dialing with Partitioned Addressing → Moved to Appendix
 - Variable-Length On-Net Dialing with Flat Addressing
 - Tail End Hop Off (a.k.a. Toll Bypass) (Some Parts in Appendix)**
- Mobility Considerations

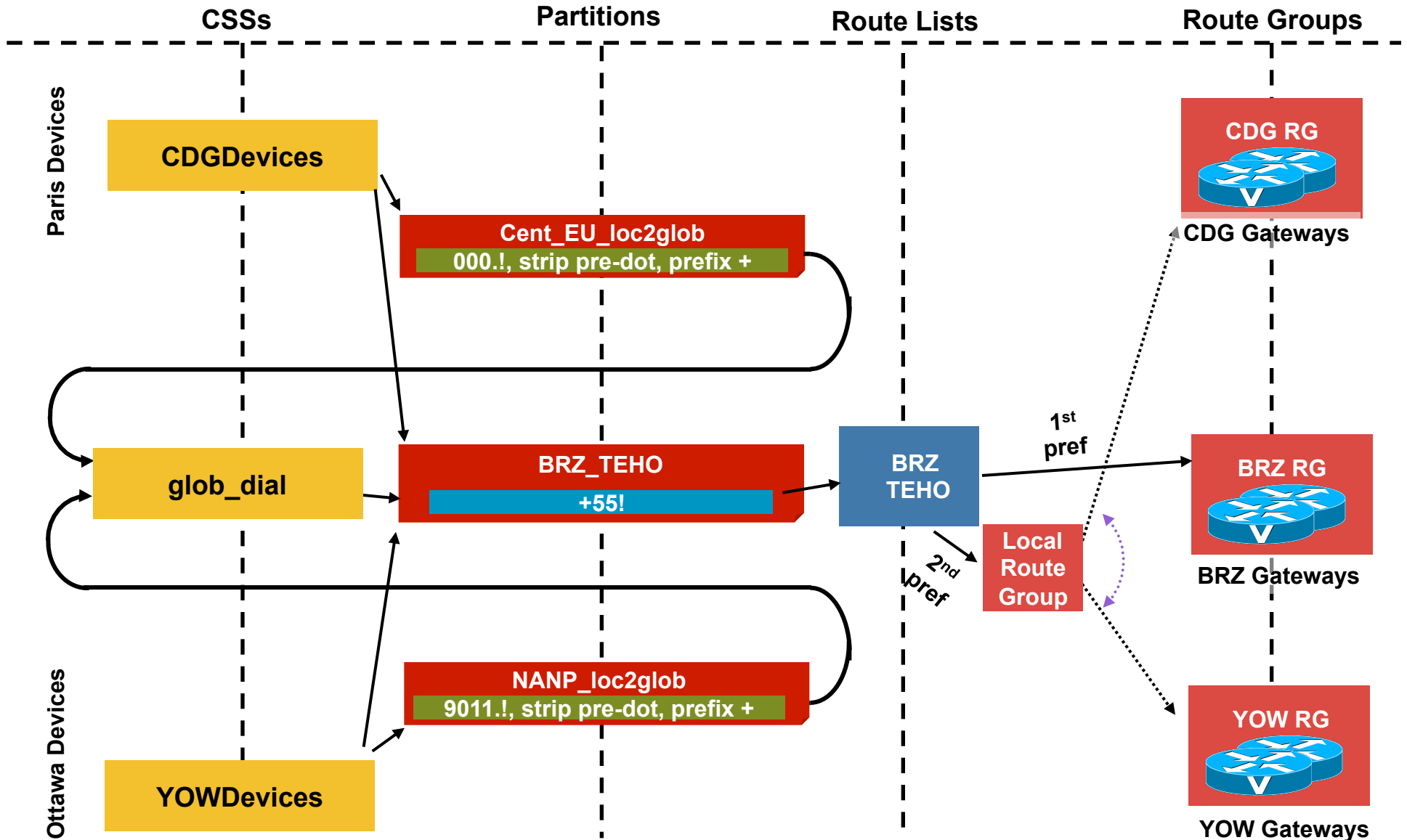
Tail-End Hop-Off (TEHO)

TEHO Without Local Route Group



Tail-End Hop-Off (TEHO)

TEHO with Local Route Group



Design Best Practices Agenda

- 7.0 and 7.1 Updates
- Multi-Site Deployments
- **Mobility Considerations**

Extension Mobility → Moved to Appendix

Device Mobility

Cisco Unified Mobility

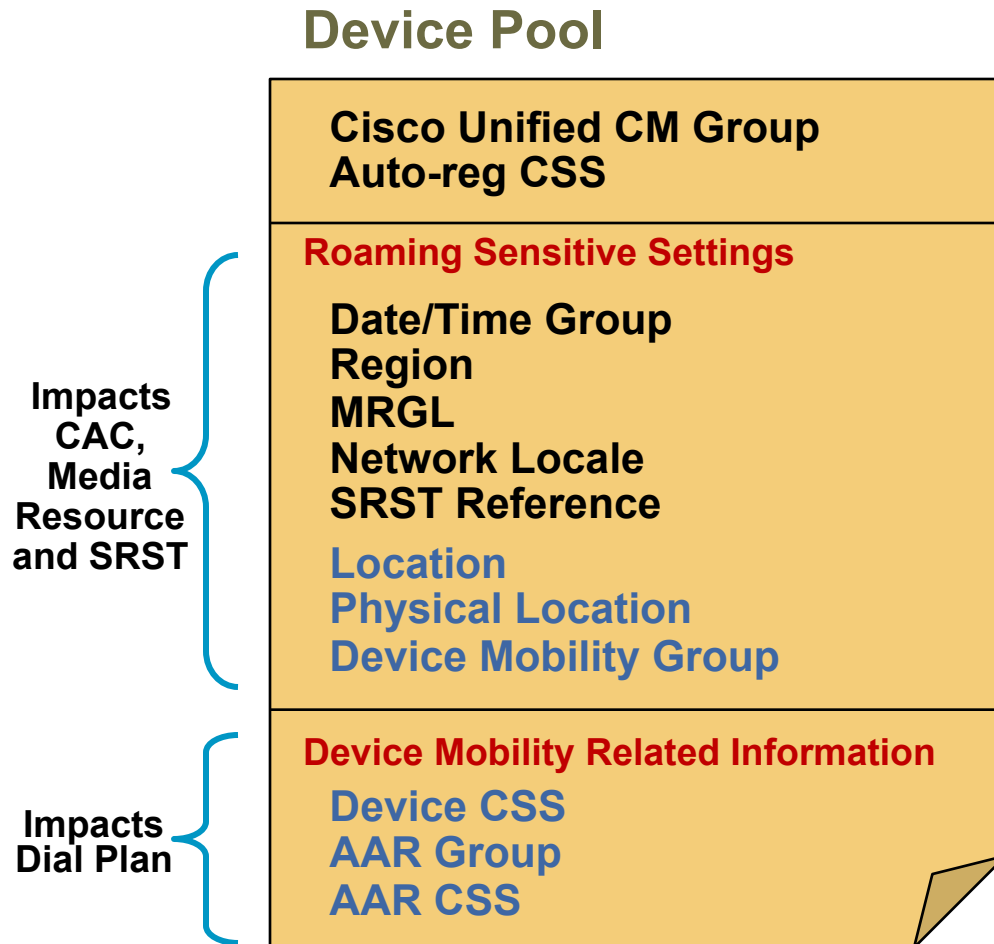
Device Mobility Considerations

High-Level Behaviour—Cisco Unified Communications Manager Versions 4.2 and 6.X, 7.X Only!

- Determines that the device has moved to new location based on the device's IP subnet
- Dynamically associates **roaming** device pool to devices that move to a different site
- Message displayed on phone screen for a few seconds when it registers with Cisco Unified Communications Manager:
 - Device in home location
 - Device in roaming location

Device Mobility

Device Pool Changes



Common Profile (new)

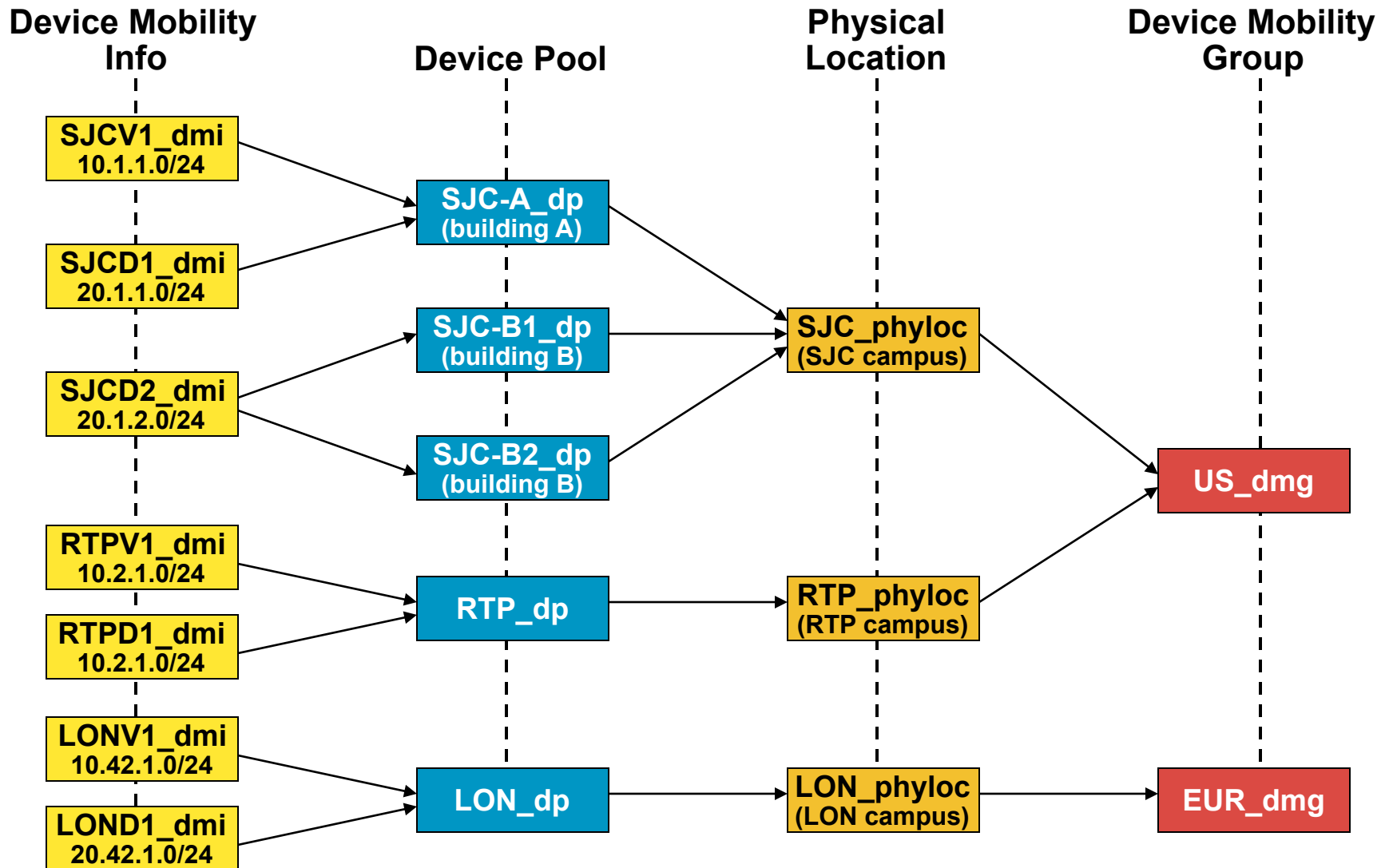
- Softkey Template
- Network Hold MoH Audio Source
- User Hold MoH Audio Source
- MLPP Indication
- MLPP Preemption
- MLPP Domain



Device

Device Mobility

New Concepts



Device Mobility Considerations

The Big Idea Is to Track Phones Based on Subnets

Voice Subnet: 10.1.1.0/24
Data Subnet: 20.1.1.0/24
Data Subnet: 20.1.2.0/24



Voice Subnet: 10.2.1.0/24
Data Subnet: 20.2.1.0/24



Voice Subnet: 10.42.1.0/24
Data Subnet: 20.42.1.0/24



Note: When Roaming from SJC to LHR, We Are Crossing DMGs Dial Plan-Related Information Does not Change.

Cisco CallManager Group	CM GroupLHR
Roaming Device Pool	LHR_DP
Location	LHR_Location
Region	LHR_Region
Network Locale	UK
AAR Group	SJC_AAR
AAR Calling Search Space	SJC_CSS
Device Calling Search Space	SJC_CSS
Media Resource Group List	LHR_MRGL
SRST	LHR_SRST

Roaming Sensitive Settings Change When Roaming Between Physical Locations. DMG not a Factor

Device Mobility Related Information Changes Only When Roaming Within the same DMG

Device Mobility Considerations

RTP Mobile User at Home Location



Phone: SEP00059BF19AA5 (Phone1- SEP00059BF19AA5)
 Registration: Registered with Cisco CallManager CLUSTER3-1
 IP Address: 10.1.110.1

Status: Ready

Copy Update Delete Reset Phone

Phone Configuration (Model = Cisco 7960)

Device Information

MAC Address* 00059BF19AA

Description Phone1- SEP00059BF19AA5

Owner User ID (Select User ID)

Device Pool* RTP_DP (View Details)

Common Profile <None> (View Details)

Calling Search Space RTP_CSS

AAR Calling Search Space RTP_CSS

Media Resource Group List <None>

User Hold Audio Source <None>

Network Hold Audio Source <None>

Location RTP_Location

AAR Group RTP_AAR

User Locale English United States

Network Locale United States

Device Security Mode Use System Default

Signal Packet Capture Mode None

Packet Capture Duration 60

Built In Bridge Default

Privacy Default

Device Mobility Mode Default (View Current Settings)

Cisco CallManager Group	CM Group 1
Roaming Device Pool	(None Selected)
Location	RTP_Location
Region	RTP-Region1
Network Locale	United States
AAR Group	RTP_AAR
AAR Calling Search Space	RTP_CSS
Device Calling Search Space	RTP_CSS
Media Resource Group List	(None Selected)
SRST	RTP_SRST

Device Mobility Considerations

RTP Mobile User at **SJC Roaming** Location

Phone: SEP00059BF19AA5 (Phone1- SEP00059BF19AA5)
Registration: Registered with Cisco CallManager CLUSTER3-1
IP Address: 10.1.120.2

Status: Ready

Copy Update Delete Reset Phone

Phone Configuration (Model = Cisco 7960)

Device Information

MAC Address* 00059BF19AA

Description Phone1- SEP00059BF19AA5

Owner User ID (Select User ID)

Device Pool* RTP_DP (View Details)

Common Profile < None > (View Details)

Calling Search Space RTP_CSS

AAR Calling Search Space RTP_CSS

Media Resource Group List < None >

User Hold Audio Source < None >

Network Hold Audio Source < None >

Location RTP_Location

AAR Group RTP_AAR

User Locale English United States

Network Locale United States

Device Security Mode Use System Default

Signal Packet Capture Mode None

Packet Capture Duration 60

Built In Bridge Default

Privacy Default

Device Mobility Mode Default (View Current Settings)

Cisco CallManager Group	CM Group 1
Roaming Device Pool	SJC_DP
Location	SJC_Location
Region	SJC-Region2
Network Locale	United States
AAR Group	SJC_AAR
AAR Calling Search Space	SJC_CSS
Device Calling Search Space	SJC_CSS
Media Resource Group List	(None Selected)
SRST	SJC_SRST



Design Best Practices Agenda

- 7.0 and 7.1 Updates
- Multi-Site Deployments
- **Mobility Considerations**

Extension mobility

Device Mobility

Cisco Unified Mobility

Cisco Unified Mobility

Configuration and Call Routing Concept

1 RD Profile per Mobility User

Remote Destination Profile

IP Phone



Line Level Configuration

DN: 408-555-1234
(Partition/Calling Search Space)

Shared Line

DN: 408-555-1234
(Partition/Calling Search Space)

Line Level Configuration

RD Profile Level Configuration:

- Device pool
- Calling search space
- Rerouting calling search space
- User/network hold audio source

Device Level Configuration:

- Device pool
- Common device configuration
- Calling search space
- Media resource group list
- User/network hold audio source

Call Routing and MoH Behaviour for Remote Destination Devices

Call Routing and MoH Behaviour for IP Phone

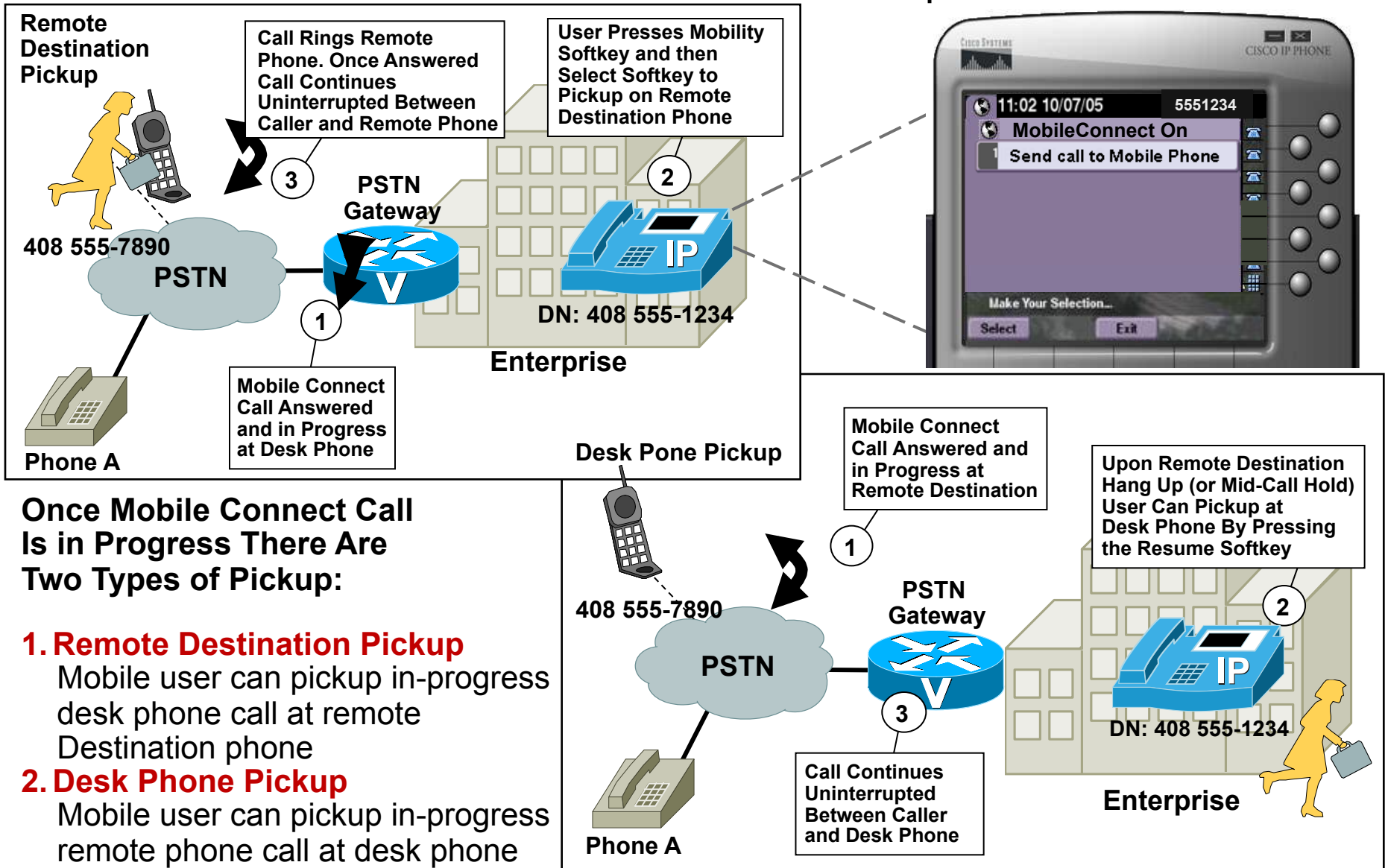


Configuration per Remote Destination Basis:

- Timers (answer too soon/late and delay before ringing)
- Allowed/blocked access lists

Cisco Unified Mobility

Remote Destination and Desk Phone Pickup



Once Mobile Connect Call Is in Progress There Are Two Types of Pickup:

1. Remote Destination Pickup

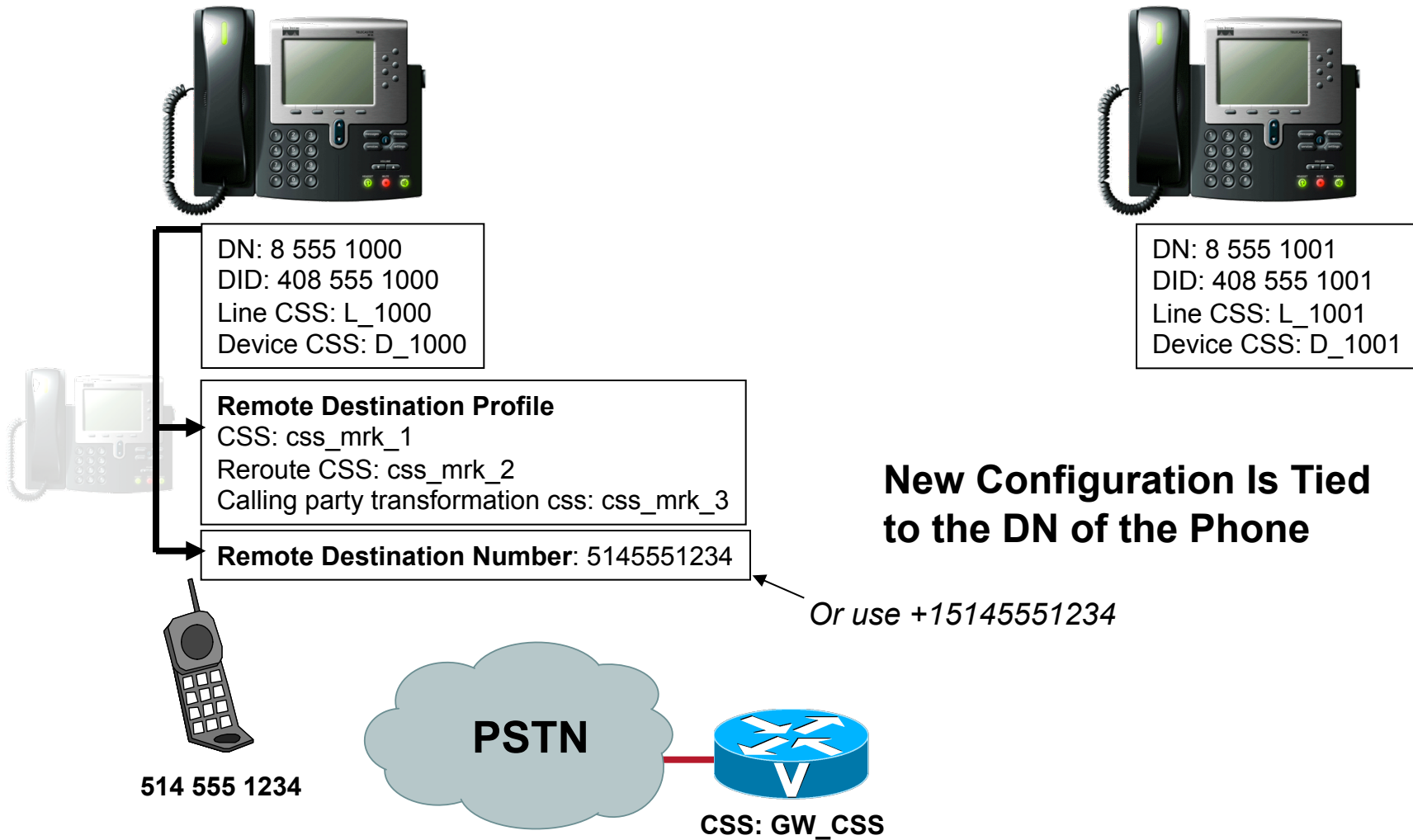
Mobile user can pickup in-progress desk phone call at remote Destination phone

2. Desk Phone Pickup

Mobile user can pickup in-progress remote phone call at desk phone

Unified Mobility: Dial Plan Implications

New Configuration



New Configuration Is Tied to the DN of the Phone

Unified Mobility: Dial Plan Implications

RDP and Remote Destination Number Associated to DN

Directory Number Information

Directory Number*

Route Partition

Description

Alerting Name

ASCII Alerting Name

Allow Control of Device from CTI

Associated Devices

Dissociate Devices

Directory Number Settings

Voice Mail Profile (Choose <None> to use system default)

Calling Search Space

Presence Group*

User Hold MOH Audio Source

Network Hold MOH Audio Source

Associated Remote Destinations

Name	Destination Number
john_doe_cell	5145551234

Unified Mobility: Dial Plan Implications

RDP and Remote Destination Number Associated to DN

Remote Destination Profile Configuration

Save Delete Copy Add New

Status
Status: Ready

Association Information

- 1 Line [1] - 85551000 in mrk_1
- 2 Line [2] - Add a new DN

Remote Destination Profile Information

Name* rdp_john_doe

Description

User ID* john_doe

Device Pool* Default

Calling Search Space css_mrk_1

User Hold Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

Privacy* TypeStatus.STATUS_OFF

Rerouting Calling Search Space css_mrk_2

Calling Party Transformation CSS css_mrk_3

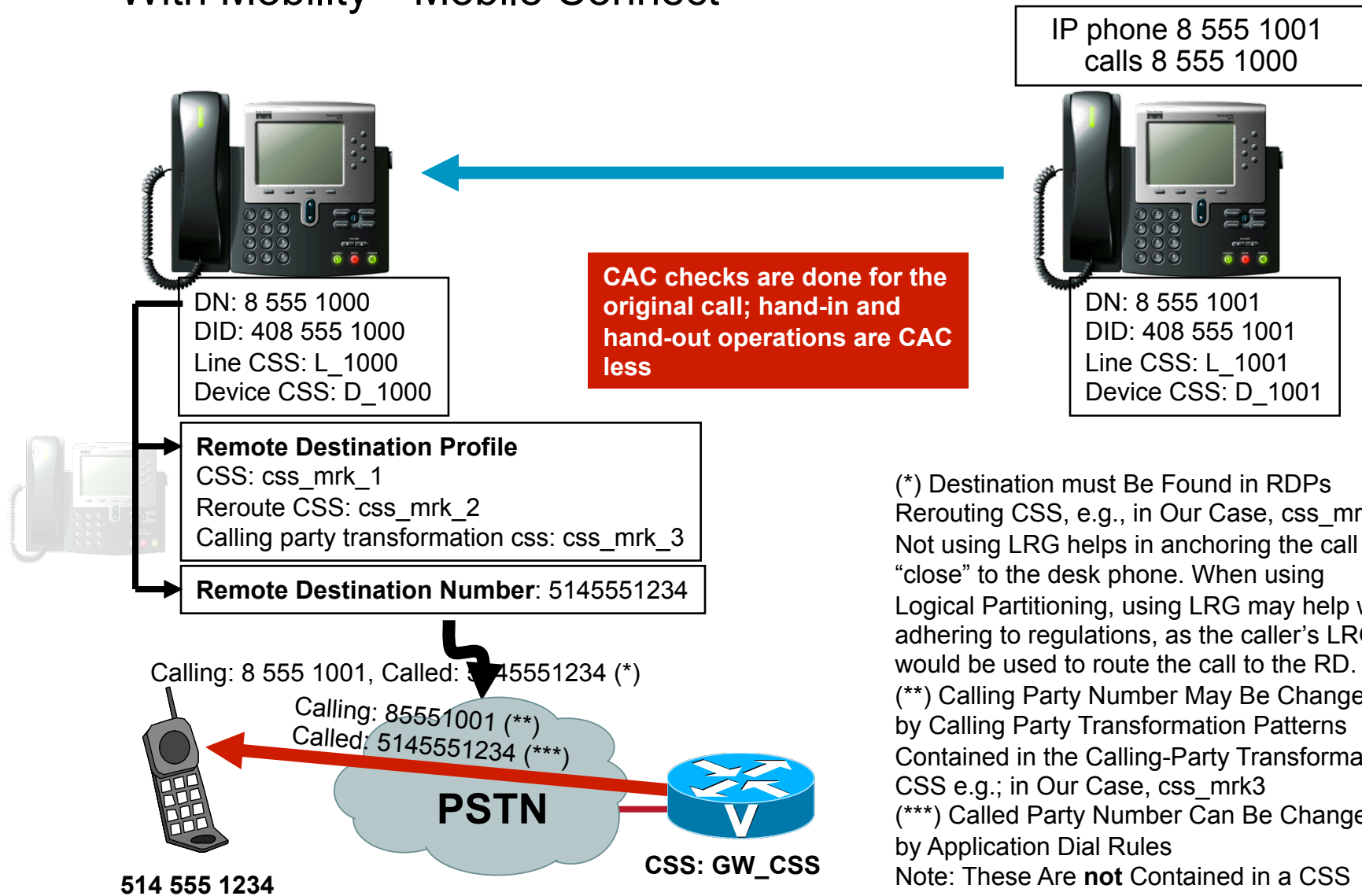
Ignore Presentation Indicators (internal calls only)

Associated Remote Destinations

Name	Destination Number
john_doe_cell	5145551234
Add a New Remote Destination	

Unified Mobility: Dial Plan Implications

With Mobility—Mobile Connect



(*) Destination must Be Found in RDPs Rerouting CSS, e.g., in Our Case, css_mrk2. Not using LRG helps in anchoring the call “close” to the desk phone. When using Logical Partitioning, using LRG may help with adhering to regulations, as the caller’s LRG would be used to route the call to the RD.

(**) Calling Party Number May Be Changed by Calling Party Transformation Patterns Contained in the Calling-Party Transformation CSS e.g.; in Our Case, css_mrk3




(***) Called Party Number Can Be Changed by Application Dial Rules


Note: These Are **not** Contained in a CSS OR: by CdPTPs at the GW level.

Unified Mobility: Dial-Plan Implications

3. With Mobility—Transformation Patterns

Calling Party Transformation Pattern Configuration

Save  Delete  Copy  Add New

Status
 Add successful

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

Urgent Priority

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transformation Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Save Delete Copy Add New

Unified Mobility: Dial-Plan Implications

3. With Mobility—Application Dial Rules

Application Dial Rule Configuration

Save Delete Add New

Status
 Status: Ready

Application Dial Rule Information

Name*

Description

Number Begins With

Number of Digits*

Total Digits to be Removed*

Prefix With Pattern

Application Dial Rule Priority

Name	Number Begins With	Number of Digits	Total Digits to be Remo
NPA415_NXX555	514555	10	0
CC1NPA514NXX555	1514555	11	1

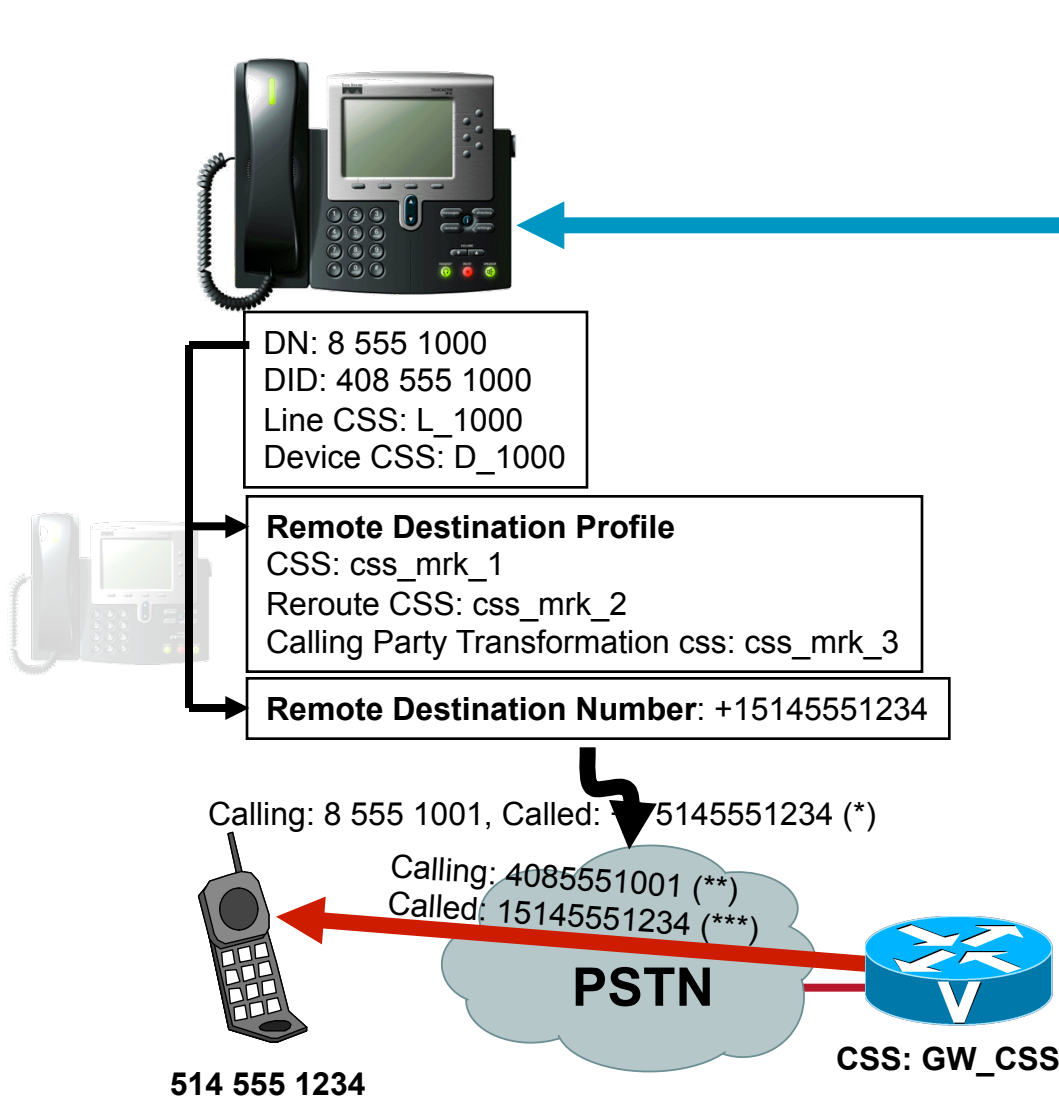
Save Delete Add New

*- indicates required item.

With 7.X, use **Called Party Transformation Patterns**, and you may want to require the number be entered in +E.164 format, and use Called Party Transformation Patterns to adapt the number to PSTN carrier requirements.

Unified Mobility: Dial Plan Implications

Mobility—Mobile Connect Enhanced



(*) Destination must Be Found in RDPs Rerouting CSS, e.g., in Our Case, css_mrk2. Not using LRG helps in anchoring the call “close” to the desk phone. When using Logical Partitioning, using LRG may help with adhering to regulations, as the caller’s LRG would be used to route the call to the RD.

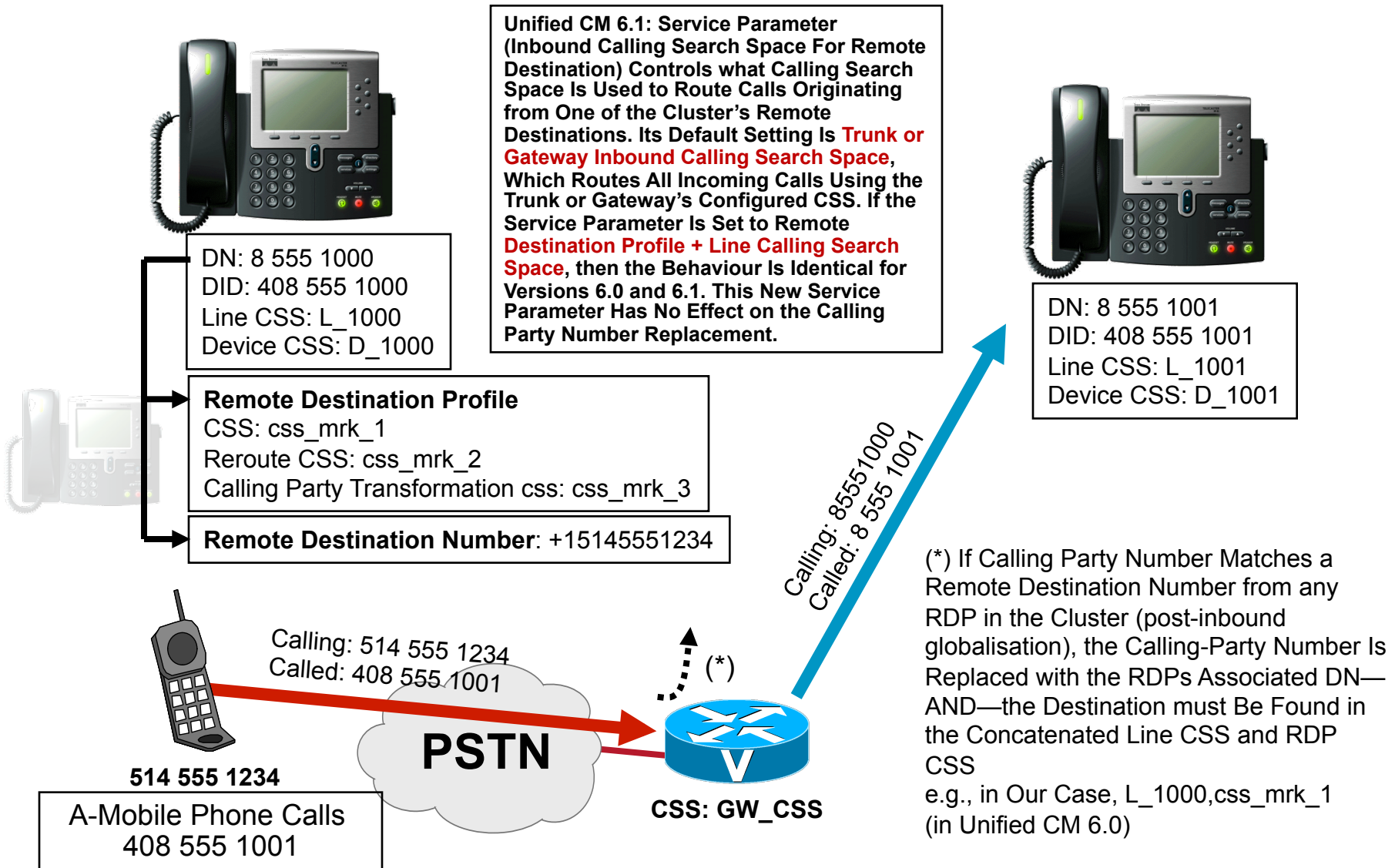
(**) Calling Party Number May Be Changed by Calling Party Transformation Patterns Contained in the Calling-Party Transformation CSS e.g.; in Our Case, css_mrk3

(***) Called Party Number Can Be Changed by Application Dial Rules

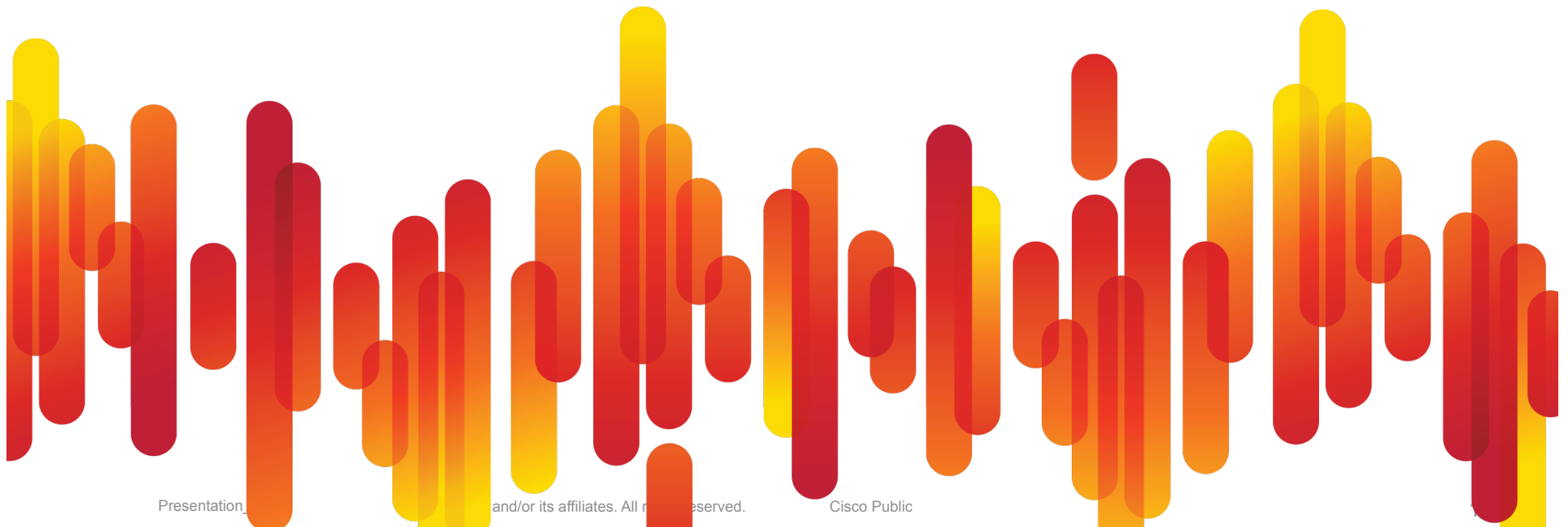
Note: These Are **not** Contained in a CSS OR: by CdPTPs at the GW level.

Unified Mobility: Dial Plan Implications

Mobility—Inbound Calls



Conclusions



Conclusions

General Recommendations

- **Keep It Simple!**

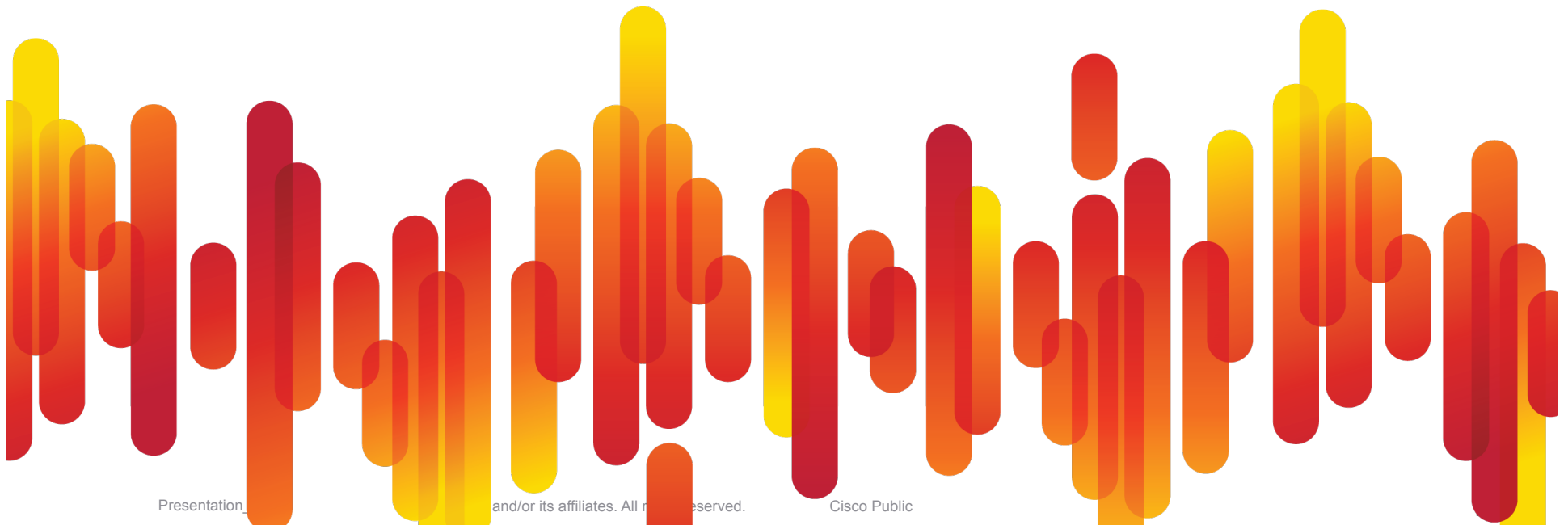
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Q and A



Meet the Engineer

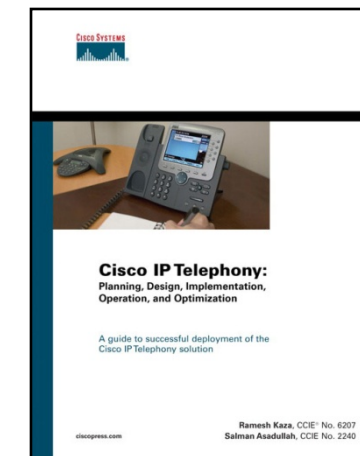
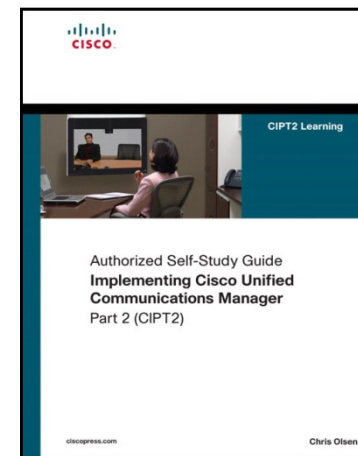
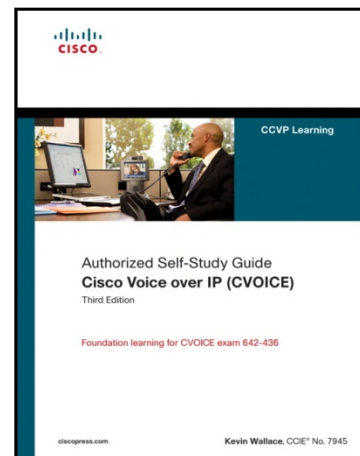
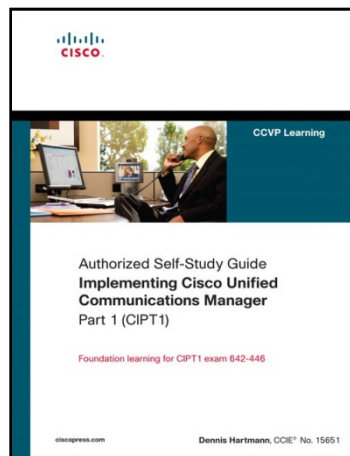
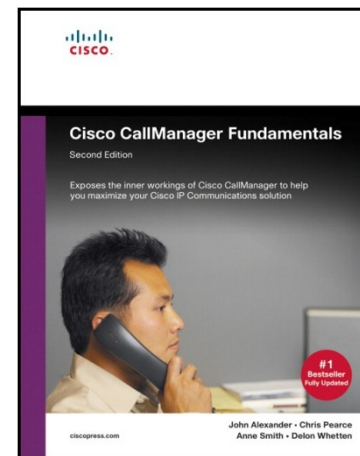
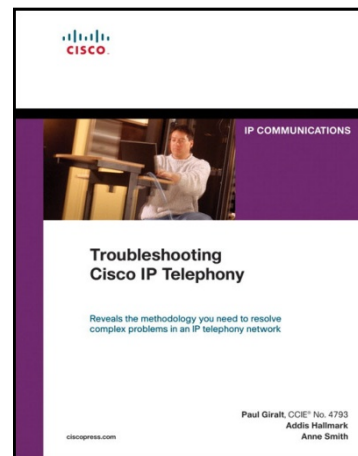
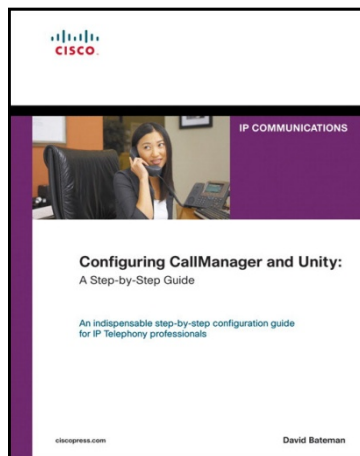
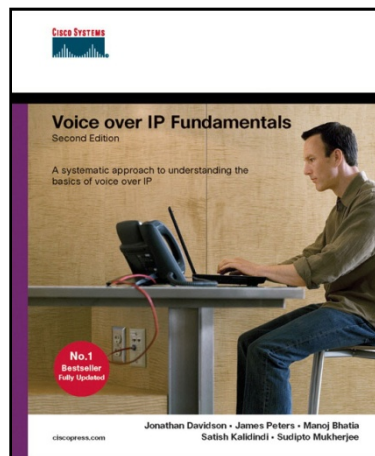
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Designed to provide a "big picture" perspective as well as "in-depth" technology discussions, these face-to-face meetings will provide fascinating dialogue and a wealth of valuable insights and ideas.

Visit the Meeting Centre reception desk located in the Meeting Centre in World of Solutions

Recommended Reading

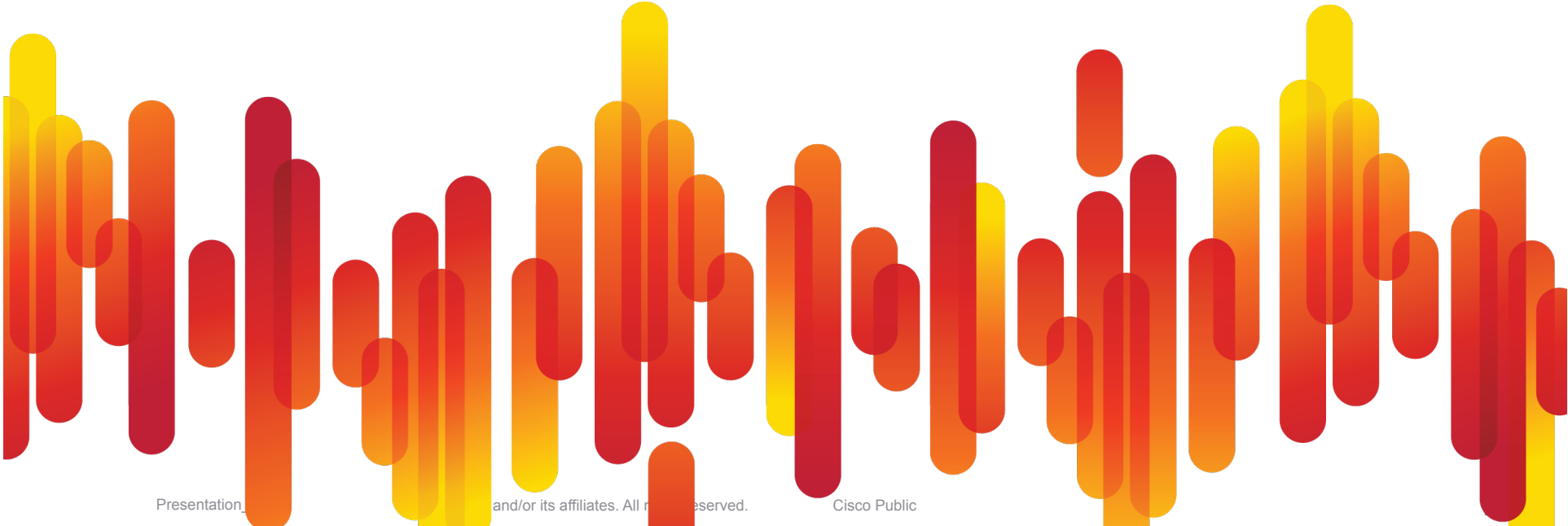
BRKUCC-3000



Source: Cisco Press



Appendix



Planning Considerations

The Fundamentals

A Few Things We All Like in a Good Dial Plan:

- Not reprinting business cards (i.e., not changing numbers because we change phone systems)
- Having abbreviated dialing within a site (e.g., five-digit dialing)
- Having a simple, direct correspondence between someone's DID number (i.e., business card) and their internal extension
- Keeping it simple, where even the new guy can use the phone system (i.e., dial **nine** or **zero** for an outside line, or five digits to reach colleagues)

Note: This Presentation Uses Some examples based on North-American Dialing Habits: Season to Taste...

Planning Considerations

The Fundamentals (Cont.)

A Few Things We All Like in a Good Dial Plan:

- Keeping it simple, where even the new system administrator can maintain the phone system (an area-code split would not destroy the plan)
- Future proofing, such that when the new office opens, we do not have to redo it all
- Have a good user experience (e.g., not having to wait for interdigit timeout when calling the guy in the next cube over)

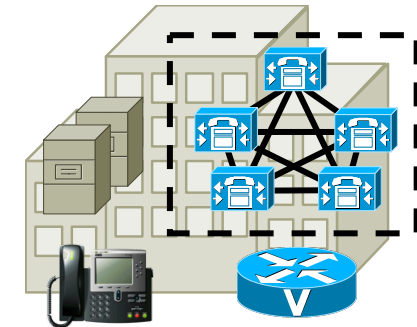
Remember: the Best Tool to Start with Is thi



Planning Considerations

Uniform Dial Plans Are Simple

- Q: Could this system use a **uniform** three-digit dial plan?
- A: No! Chicago and Dallas' DID ranges overlap in the last three digits
- Q: Ok, how about four-digit uniform dial plan?
- A: No! overlaps again!
- Because each time you call extensions 9110 through 9119 in Chicago, you get the police department (by calling 911)
- **And:** because the system cannot off-hand tell the difference between calling Al Capone at 9141, and calling long distance to a Toronto number (e.g. 9 1 416 555 1234) you will have to wait for interdigit timeout, even when calling from Anchorage!



Anchorage
907 507 18XX



New York
212 555 75XX



Chicago
708 552 91XX



Birmingham
205 937 54XX

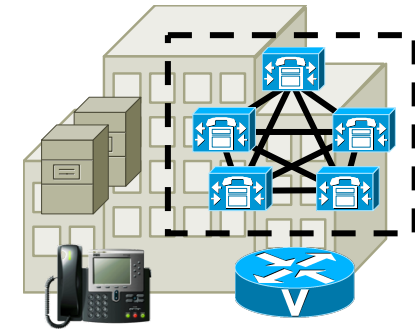


Dallas
972 553 11XX

Planning Considerations

Uniform Dial Plans Are Simple (Cont.)

- Q: Fine! How about a five-digit uniform dial plan?
- A: Currently, yes! No overlap in the current ranges of DID numbers assigned
- Q: Great! How about that new office we want to get in Hawaii? Room for it in our dial plan?
- A: Sure. Well, maybe: it cannot use a DID range where the third digit of the office code is nine or zero, and cannot overlap with 575XX, 291XX, 754XX, 311XX, or 718XX...



Anchorage
907 507 18XX



New York
212 555 75XX



Chicago
708 552 91XX



Birmingham
205 937 54XX



Dallas
972 553 11XX

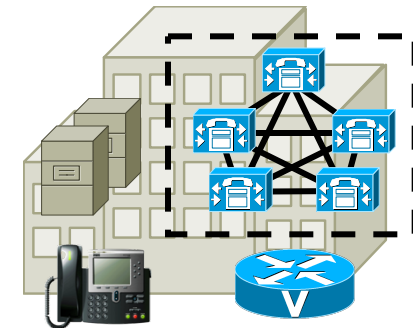


Hawaii
808 ??? ?????

Planning Considerations

Uniform Dial Plans Are Simple (Cont.)

- Q: If all I could get from Hawaii's telco is a DID range of 808 557 54XX, could I not dial six digits to reach a Hawaii phone, and five digits anywhere else? That way, I avoid the overlap between Hawaii and Birmingham
- A: No! Because calls to New York (e.g., 57540) will sometimes overlap with calls to Hawaii's phones e.g., 575403), forcing the interdigit timeout to occur before the call is routed (and a few other reasons: can you spot them?)
- Q: What do I do now? Go to six digits?
- A: No: Anchorage's second NXX digit is 0. Overlaps with the operator code...
- Q: Seven digits?
- A: No: Birmingham starts with a nine!



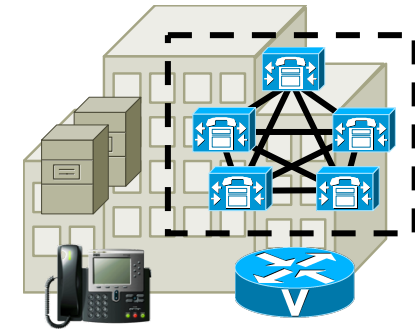
Anchorage
907 507 18XX



Planning Considerations

Uniform Dial Plans Are Simple (or So We Hoped)

- Q: Eight digits?
- A: Ok for now: but you'll never open an office in Raleigh (area code 919)
- Q: Nine digits? Oops. Forget about it!
That zero again (Four cases, no less)
- Q: Ten digits?
- A: Great idea! The North American dial plan will make sure that it never overlaps. You can even give up the outside access code. It is not really abbreviated dialing anymore though...



Anchorage
907 507 18XX



New York
212 555 75XX



Chicago
708 552 91XX



Birmingham
205 937 54XX



Dallas
972 553 11XX



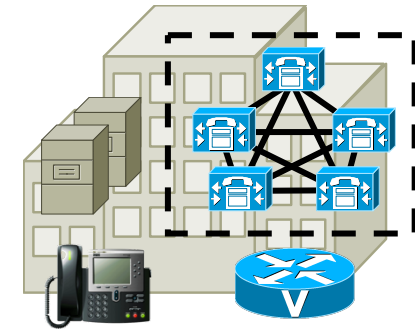
Hawaii
808 557 54XX

Planning Considerations

How About an On-Net, Intersite Access Code?

- Q: What about zero for operator, nine for outside line, and eight for intersite calls?
- A: Great idea
- Q: How many digits for intrasite calls, though?
- A: Not three (4XX and 1XX overlap)
Not four either (911!)
Five would work!

...no it would not... B'ham and Hawaii overlap still if you try to reach them from elsewhere...



Anchorage
907 507 18XX



New York
212 555 75XX



Chicago
708 552 91XX



Birmingham
205 937 54XX



Dallas
972 553 11XX



Hawaii
808 557 54XX

Planning Considerations

How About an On-Net, Intersite Access Code?

- Q: Ok: now I have it:

0 = operator

8 + 5 digits: intersite on-net

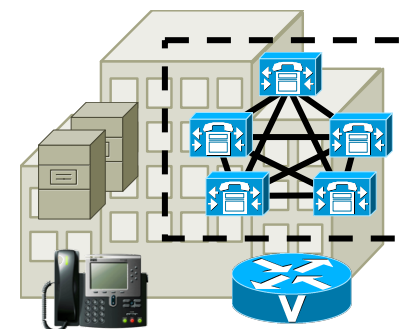
9 + 7 digits, 9 + 10 digits , 9 + 1 + 10 digits,
9 + 011... all off-net patterns

And then any five digits that begin with one
through seven are for an on-net, intrasite call

Am I good to go?

- A: Yes

- ...for now (except Hawaii and B'ham overlap ☹)



Anchorage
907 507 18XX



New York
212 555 75XX



Chicago
708 552 91XX



Birmingham
205 937 54XX



Dallas
972 553 11XX



Hawaii
808 557 54XX

Planning Considerations

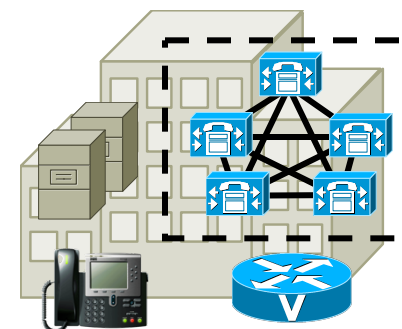
What If I Have Many, Many More Sites? More Users?

- Q: I have 250 branches, with over 90 with 100+ users, and a dozen with more than 1000 users, and a headquarter with 12000 users. Can I still use eight + five digits for on-net, intersite calls?
- A: No!

You essentially have the following to play with:

1XXXX, 2XXXX, 3XXXX, 4XXXX, 5XXXX, 6XXXX, 7XXXX

250 phone companies' DID ranges, the need for more than a whole five-digit range for a single site, and dividing the rest into 250 unequal parts. Future planning, area code splits, new office codes, etc...

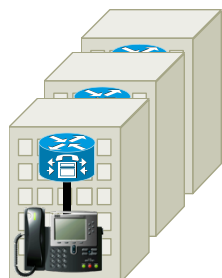


San Jose

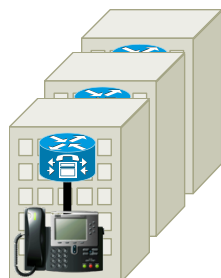
408 526 XXXX

408 853 XXXX

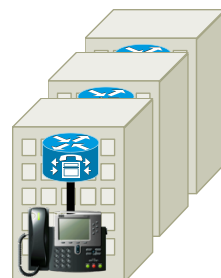
Site Codes 123 and 124



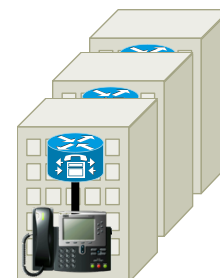
Baltimore
240 555 XXXX
Site Code 012



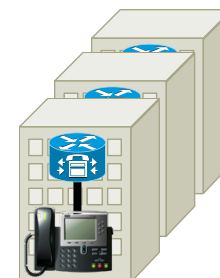
Oakland
510 555 51XX
Site Code 345



New Orleans
504 555 5XXX
Site Code 256



Philadelphia
267 555 1XXX
Site Code 390

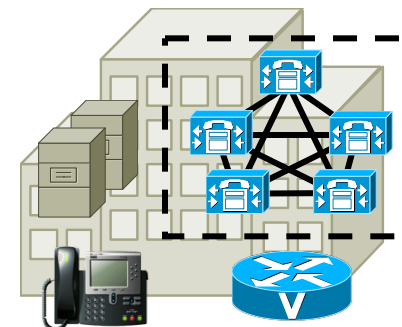


Hawaii
808 557 54XX
Site Code 822

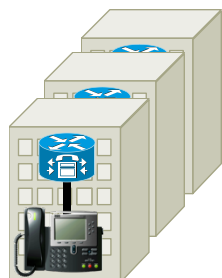
Planning Considerations

What if I Have Many, Many More Sites? More Users? (Cont.)

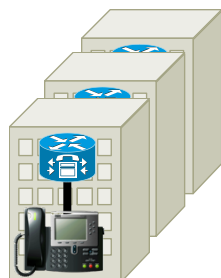
- Q: What to do?
- A: Site codes are a good idea
 - 0 = operator
 - 9 = outside line, all combinations
 - 8 + site code (three digits would work up to 1000 sites), followed by a four digit extension
 - [1-7]XXX: on-net, intrasite dialing
- Q: But I have a site with more than 10000 users?
- A: Would you be OK with using two site codes for that site? And having that site use five digit on-net? Using E.164?



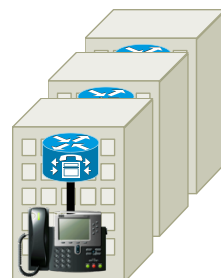
San Jose
408 526 XXXX
408 853 XXXX
Site Codes 123 and 124



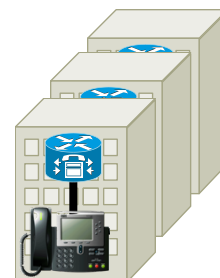
Baltimore
240 555 XXXX
Site Code 012



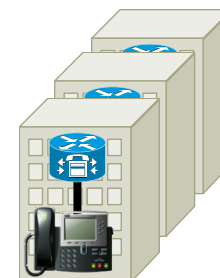
Oakland
510 555 51XX
Site Code 345



New Orleans
504 555 5XXX
Site Code 256



Philadelphia
267 555 1XXX
Site Code 390



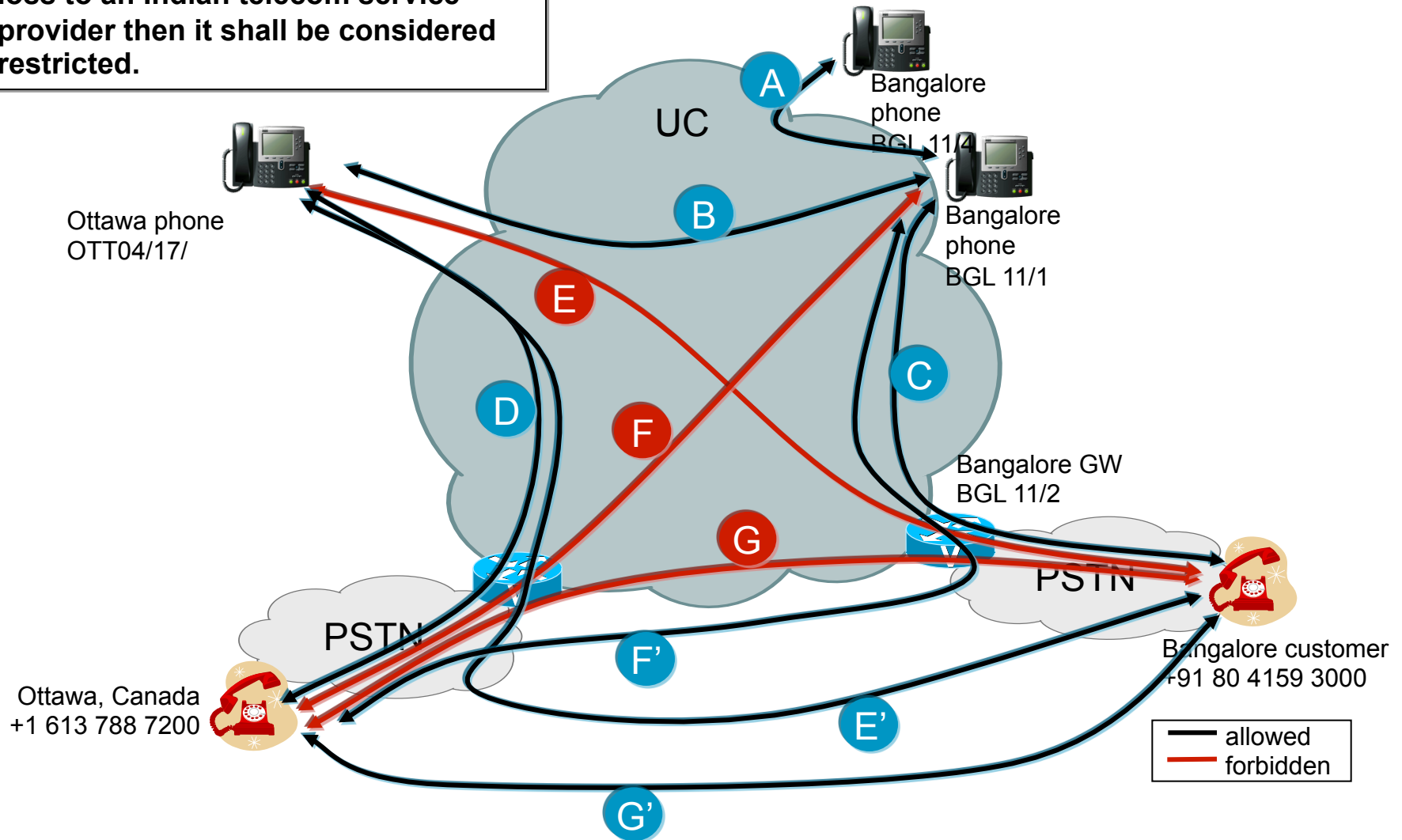
Hawaii
808 557 54XX
Site Code 822

Logical Partitioning (7.1 update)

- To satisfy regulatory requirements in markets where toll bypass is not allowed
- E.g.: In India, if a call leg results in revenue loss to an Indian telecom service provider then it shall be considered restricted.

Logical Partitioning ... to control the initiation of calls

In India, if a call leg results in revenue loss to an Indian telecom service provider then it shall be considered restricted.



Logical Partitioning ... to control the initiation of calls

- We can control the initiation of calls with traditional dial plan tools like CSSes and Partitions
- We can provision multiple line phones, where one line is used for calls within the enterprise (Closed User Group, or CUG), and another line is used for calls to/from the PSTN

QUIZ!!!

Why would I need more than CSSes and partitions?

Logical Partitioning ... to control mid-call features

3- Ottawa phone conferences-in Ottawa customer

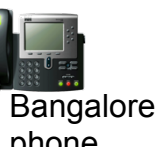
Ottawa phone
OTT04/17/

But this allows a user in India to do toll-bypass to a customer in Canada!
This is forbidden by regulations in India!!!

Ottawa, Canada
+1 613 788 7200



1- BGL phone calls another BGL phone



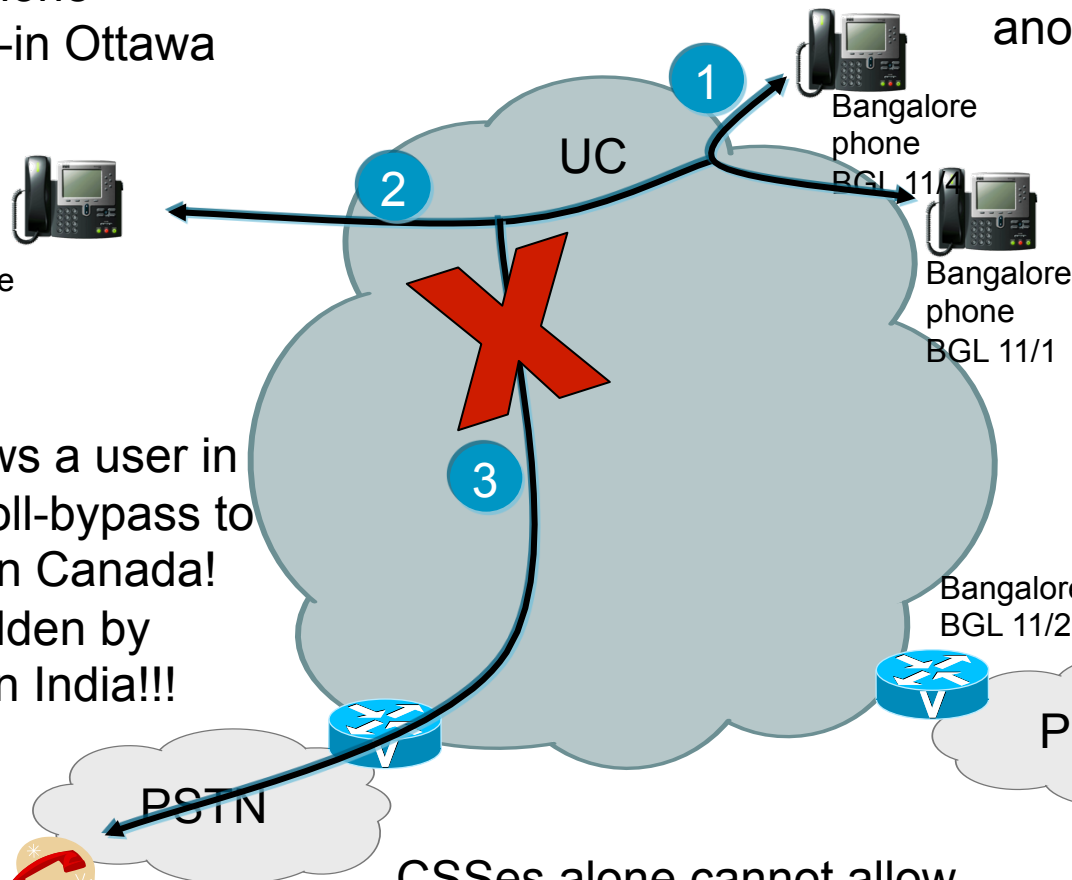
2- BGL phone conferences-in Ottawa phone

Bangalore GW
BGL 11/2

PSTN



Bangalore customer
+91 80 4159 3000



CSSes alone cannot allow the Canadian phone to call Ottawa customers AND prevent call leg 3!!!

Logical Partitioning ... to control call legs at any time

- CSSes and Partitions fail to control mid-call features
- Logical Partitioning controls the initiation of call legs at any time, based on CUCM-defined *policies* based on endpoint type and Geolocation
- as in: **NO** call, ad-hoc or meet-me conference, transfer, parked call retrieval, call pickup is allowed if the new call leg would break a policy based on geolocation

Logical Partitioning

... required ingredients: device types

border	Gateway (H.323, SIP) Inter-cluster trunk (ICT), gk-controlled or not H.225 trunk SIP trunk MGCP port (E1, T1, PRI, BRI, FXO)
interior	Phones (SCCP, SIP, third party) CTI route points VG224 analog phones MGCP port (FXS) Cisco Unity Voice Mail (SCCP)

These types are fixed, and not editable.

Logical Partitioning

... required ingredients: geo locations

Save Delete Copy Add New

Geo Location Configuration

Name*	BGL11/1/
Description	Bangalore building 11, floor 1
Country using the two-letter abbreviation	in
State, Region, or Province (A1)	karnataka
County or Parish (A2)	Varthur Hobli
City or Township (A3)	bangalore
Borough or City District (A4)	Cessna Business Park
Neighborhood (A5)	Kadubeesanahalli Village
Street (A6)	Sarjapur Marathalli Outer Ring
Leading Street Direction, such as N or W (PRD)	
Trailing Street Suffix, such as SW (POD)	
Address Suffix, such as Avenue, Platz (STS)	road
Numeric house number (HNO)	
House Number Suffix, such as A, 1/2 (HNS)	
Landmark (LMK)	SEZ Unit
Additional Location Information, such as Room Number (LOC)	E3-8
Floor (FLR)	1
Name of Business or Resident (NAM)	cisco
Zip or Postal Code (PC)	560 087

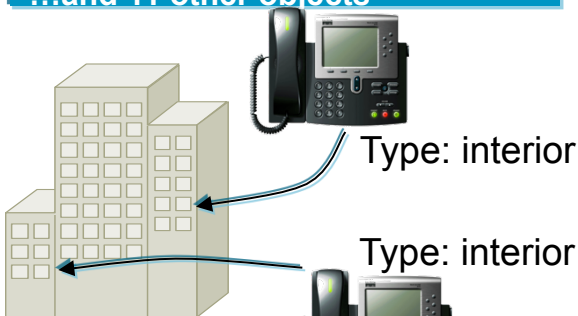
- A geo location is a list of up to 17 location objects conforming to RFC-4119
- In CUCM release 7.1, this is entirely a manual process
- Geo locations are associated with devices like GWs, trunks and phones.
- Geo Locations are configured either at the device, the device pool, or the enterprise parameters levels, in order of precedence

Logical Partitioning

... required ingredients: geo locations

Name: BGL11/4/
Description: Bangalore 11, floor 4
Country: IN
State (A1): Karnataka
City (A3): Bangalore
Street (A6): Sarjapur Marathalli...
Address suffix (STS): Road

Floor (FLR): 4
...and 11 other objects



Name: BGL11/1/
Description: Bangalore 11, floor 1
Country: IN
State (A1): Karnataka
City (A3): Bangalore
Street (A6): Sarjapur Marathalli...
Address suffix (STS): Road

Floor (FLR): 1
...and 11 other objects

- These two phones are practically at the same address, except for the floor
- Strictly speaking, their geo locations are **not** the same ...but
- For our policies, we should **treat** them the same
- In other words, we do not want one policy per floor!!!
- We will **filter** out the location objects we need not consider in the policies we want to apply to these phones

Logical Partitioning

... required ingredients: geo locationfilters

Geo Location Filter Configuration

Name*

Description

Match geolocations using the following criteria:

- Country using the two-letter abbreviation
- State, Region, or Province (A1)
- County or Parish (A2)
- City or Township (A3)
- Borough or City District (A4)
- Neighborhood (A5)
- Street (A6)
- Leading Street Direction, such as N or W (PRD)
- Trailing Street Suffix, such as SW (POD)
- Address Suffix, such as Avenue, Platz (STS)
- Numeric house number (HNO)
- House Number Suffix, such as A, 1/2 (HNS)
- Landmark (LMK)
- Additional Location Information, such as Room Number (LOC)
- Floor (FLR)
- Name of Business or Resident (NAM)
- Zip or Postal Code (PC)

- A geo location filter selects which of the 17 location objects will be carried forth for use in a policy
- When combined with the actual geo locations, it allows for policies to be based on the higher order objects
- This example “keeps” only 3 of the 17 location objects
- Geo locations thus filtered will be considered only on their country, state and city.
- Two different endpoints on different floors of the same building will yield the same *filtered* geolocation
- Geo location filters are associated to the device⁽¹⁾, the device pool, or the enterprise parameters, in order of precedence

*(1) No filter config on device for phones;
On device pool or enterprise parameters only.*

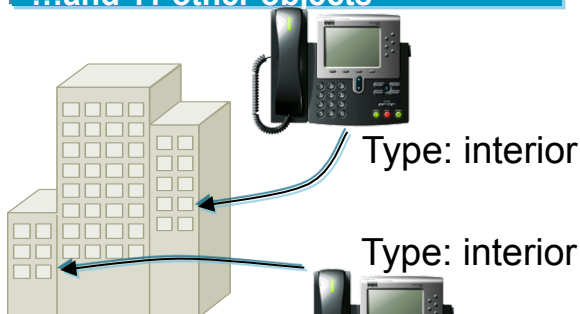
Logical Partitioning

... required ingredients: geo location filters

Geo location

Name: BGL11/4/
Description: Bangalore 11, floor 4
Country: IN
State (A1): Karnataka
City (A3): Bangalore
Street (A6): Sarjapur Marathalli...
Address suffix (STS): Road

Floor (FLR): 4
...and 11 other objects



Geo location

Name: BGL11/1/
Description: Bangalore 11, floor 1
Country: IN
State (A1): Karnataka
City (A3): Bangalore
Street (A6): Sarjapur Marathalli...
Address suffix (STS): Road

Floor (FLR): 1
...and 11 other objects

Geo location filter

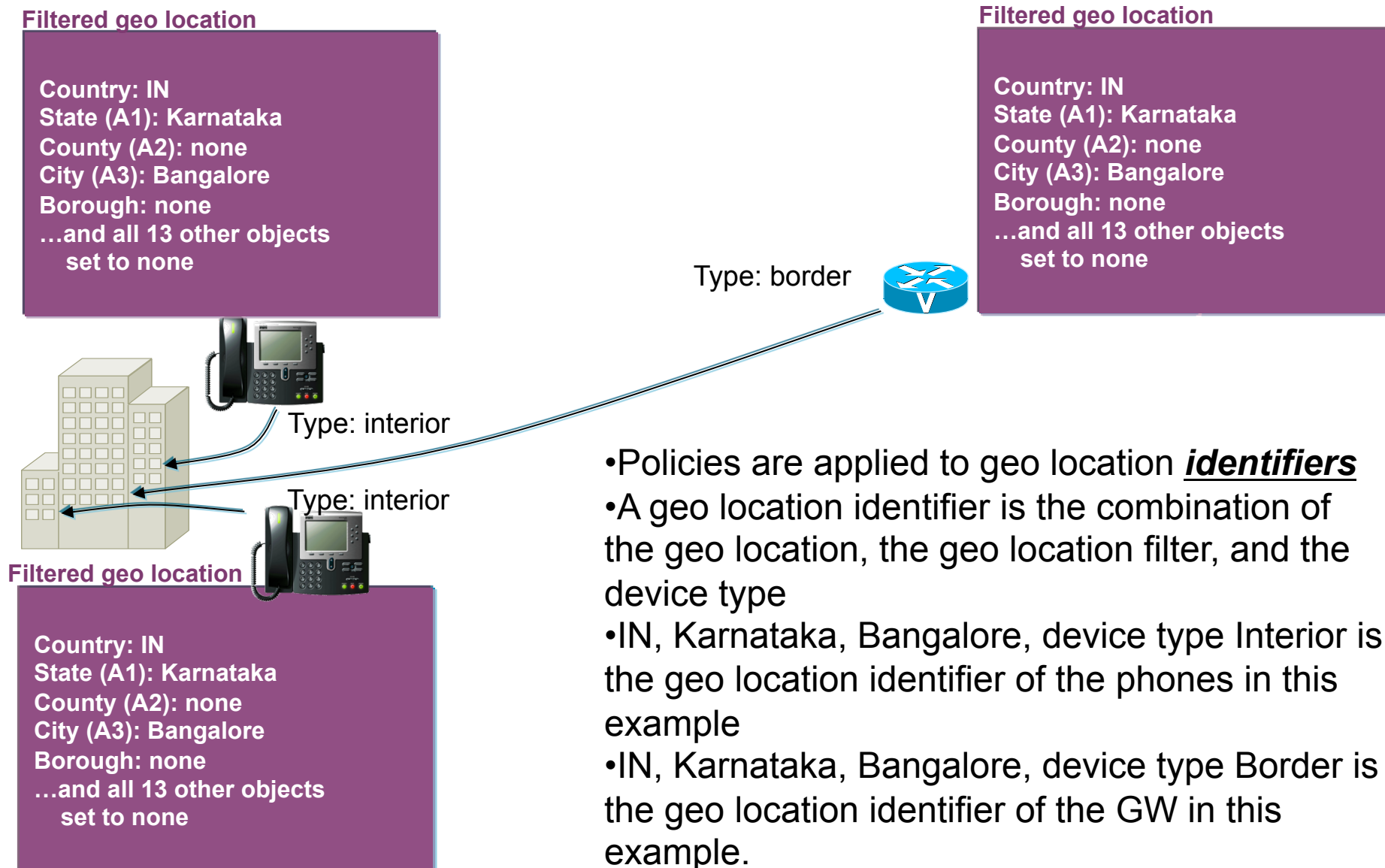
Name: Bangalore
Description: applied to any Ban...
✓ Country
✓ State (A1)
■ County (A2)
✓ City (A3)
■ Borough (A4)
...and none of the 12 other objects

Filtered geo location To be considered by policy

Country: IN
State (A1): Karnataka
County (A2): none
City (A3): Bangalore
Borough: none
...and all 13 other objects
set to none

Logical Partitioning

... required ingredients: geo location policies



Logical Partitioning

... required ingredients: geo location policies

Logical Partitioning Policy Configuration

Name* **BGL devices**

Description used for all call legs with bgl devices

Country in

A1 karnataka

A2 < None >

A3 bangalore

A4 < None >

A5 < None >

A6 < None >

PRD < None >

POD < None >

STS < None >

HNO < None >

HNS < None >

LMK < None >

LOC < None >

FLR < None >

NAM < None >

PC < None >

Configured Policies

Device Type	Geo Location Policy	Other Device Type	action
Interior	BGL devices	Border	Allow
Border	BGL devices	Border	Allow
Border	BGL devices	Interior	Allow

NOTE: Geo Location Policies that are not displayed use the Default Policy; To remove policies from the above list, set the respective policy to Use Default Policy

Configure Relationship to other Geo Location Policies

Device Type	Geo Location Policy	Other Device Type	action
Border	BGL devices BlankGeoLocPolicy	Border	Use Default Policy

This covers:
 BGL devices/Interior to BGL devices/border: allow
 BGL devices/border to BGL devices/border: allow
 BGL devices/border to BGL devices/interior: allow

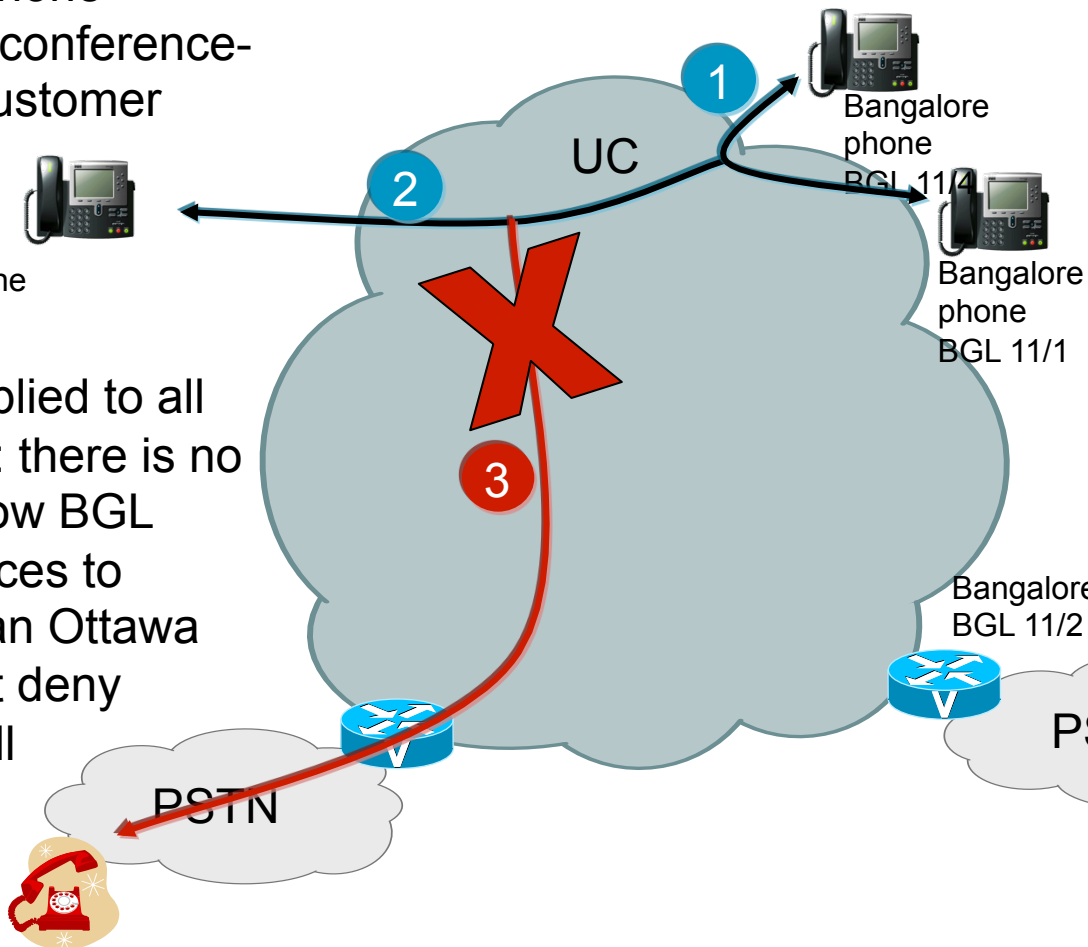
Quiz: why must I allow border to border in BGL?

Logical Partitioning ... policy in action

3- Ottawa phone attempts to conference-in Ottawa customer

Policy is applied to all participants: there is no policy to allow BGL interior devices to connect to an Ottawa GW: Implicit deny prevents call

Ottawa, Canada
+1 613 788 7200



1- BGL phone calls another BGL phone
All devices interior:
policy mechanism not invoked

2- BGL phone conferences-in Ottawa phone
All devices interior:
policy mechanism not invoked

Bangalore customer
+91 80 4159 3000

Logical Partitioning details – when a call is denied

- For the feature scenarios which are restricted due to logical partitioning configurations, a feature based “error” message will be displayed to an end user and a tone will be played as needed.

Transfer: - “External Transfer Restricted”

Pickup: - "PickUp is Unavailable"

Adhoc Conference: - " Conference is Unavailable"

Meet-Me Conference: - " MeetMe is Unavailable"

Park/DCallPark: - "Cannot Retrieve Parked Call".

Mobility Cell Pickup: - "Cannot Send Call to Mobile“

- on analog phone: no display: re-order tone is heard

Logical Partitioning details – feature interaction

- When LP enabled, LP config trumps the BlockOffNetToOffNetTransfer service parameter for deciding on the interconnection of the specific trunks, gateways or phones.
- With call pickup: when multiple calls are ringing on a phone: pickup attempt will start at longest ringing, check policy, if failed move on to next longest ringing, check policy, etc... until a policy allow is found, or until no other call is available to check for policy

Logical Partitioning details – feature interaction

- Shared lines

calls to a shared line: each phone is check individually for policy with calling device: if policy denies, phone does not ring. Denied phone is effectively not part of shared line for that call

Calls from a shared line: phone making call is checked for policy. If call succeeds, other phones are given call instance details only if their policy with destination device allows. A1 and A2 share a line; A1 calls PSTN GW1 (allowed). A2 not allowed to call PSTN GW1. A2 does not see call, and cannot retrieve the call from hold.

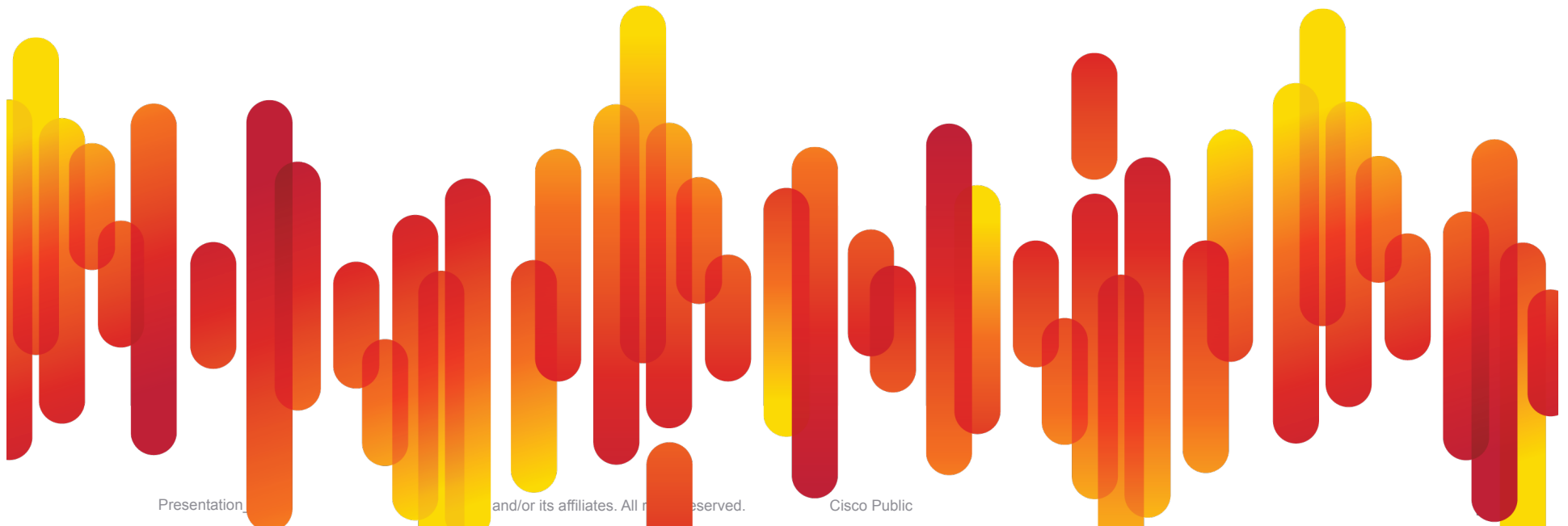
Logical Partitioning

Geolocations, filters, policies

- Gateways and trunks are associated with geolocations and geolocation filters at the device, device pool or enterprise parameter levels, in order of precedence
- Phones located in the home location (i.e.: not roaming as per device mobility) get their Geolocation from the device configuration; when roaming, from the Device Pool configuration
- Roaming or not, phones get their Geolocation filters from the device pool configuration

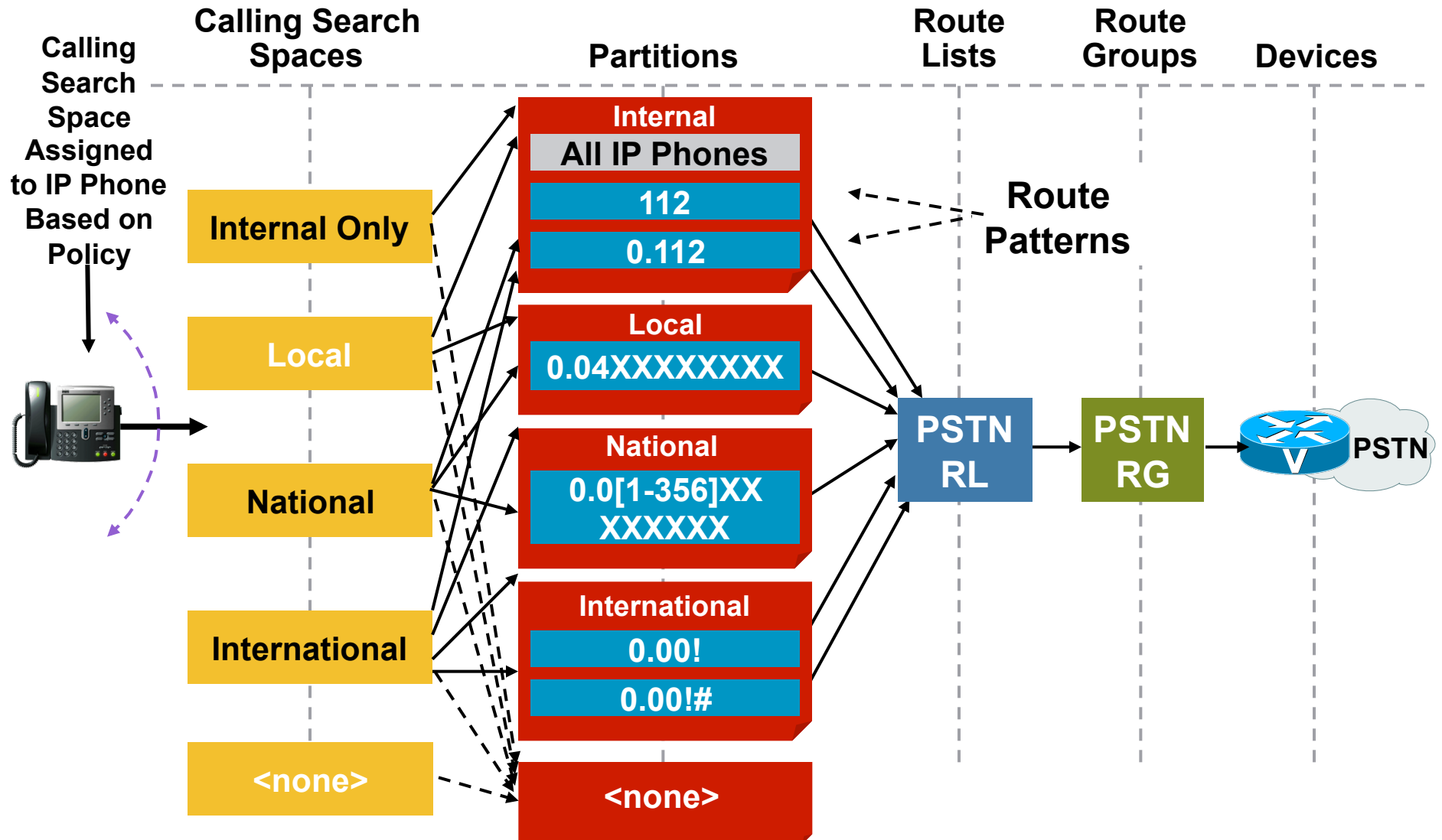
Appendix

Building Classes of Service



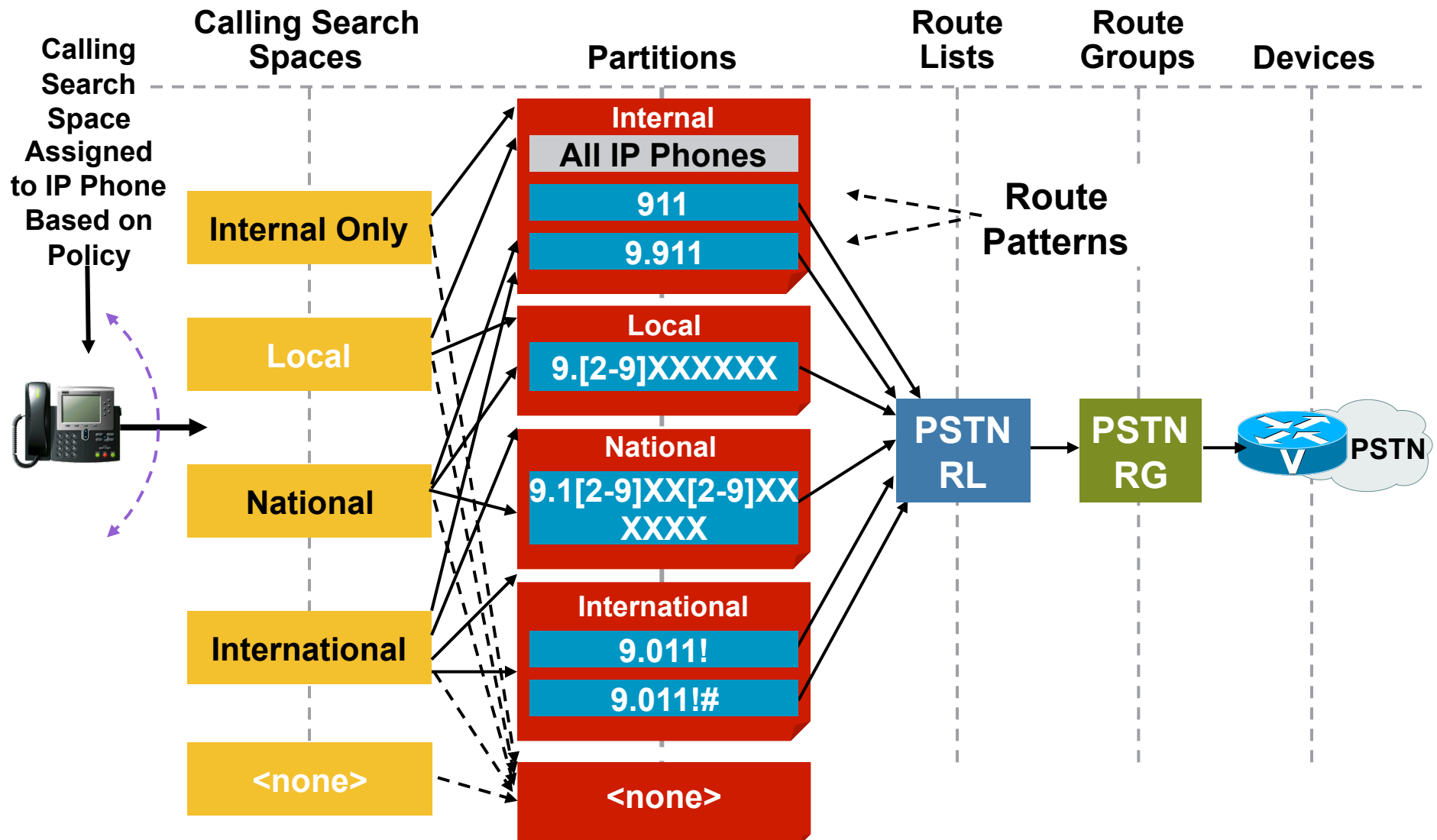
Traditional CSS Approach

Example of Composite View—France



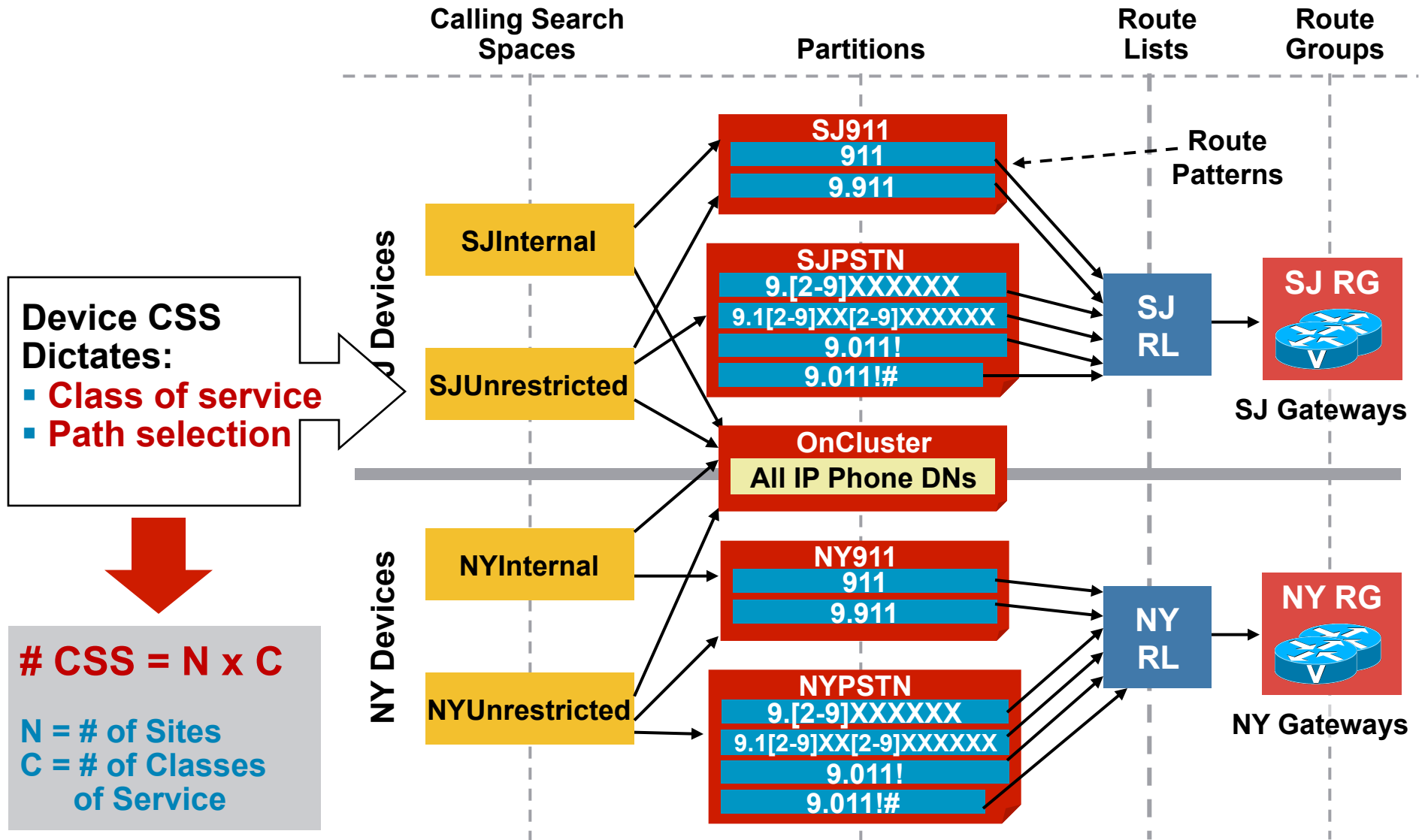
Traditional CSS Approach

Example of Composite View—North America



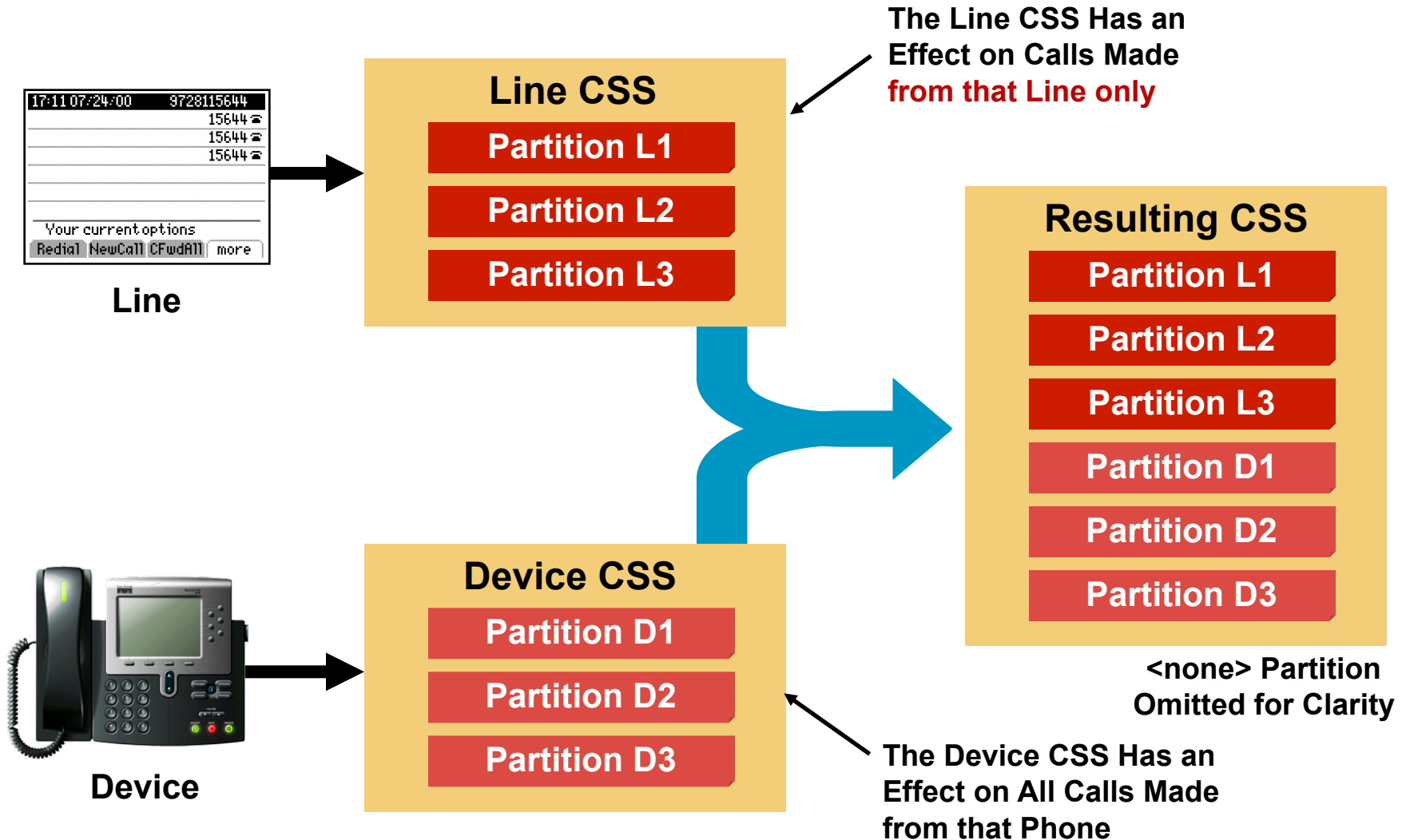
Traditional CSS Approach

Scalability for Centralised Deployments



The Line/Device CSS Approach

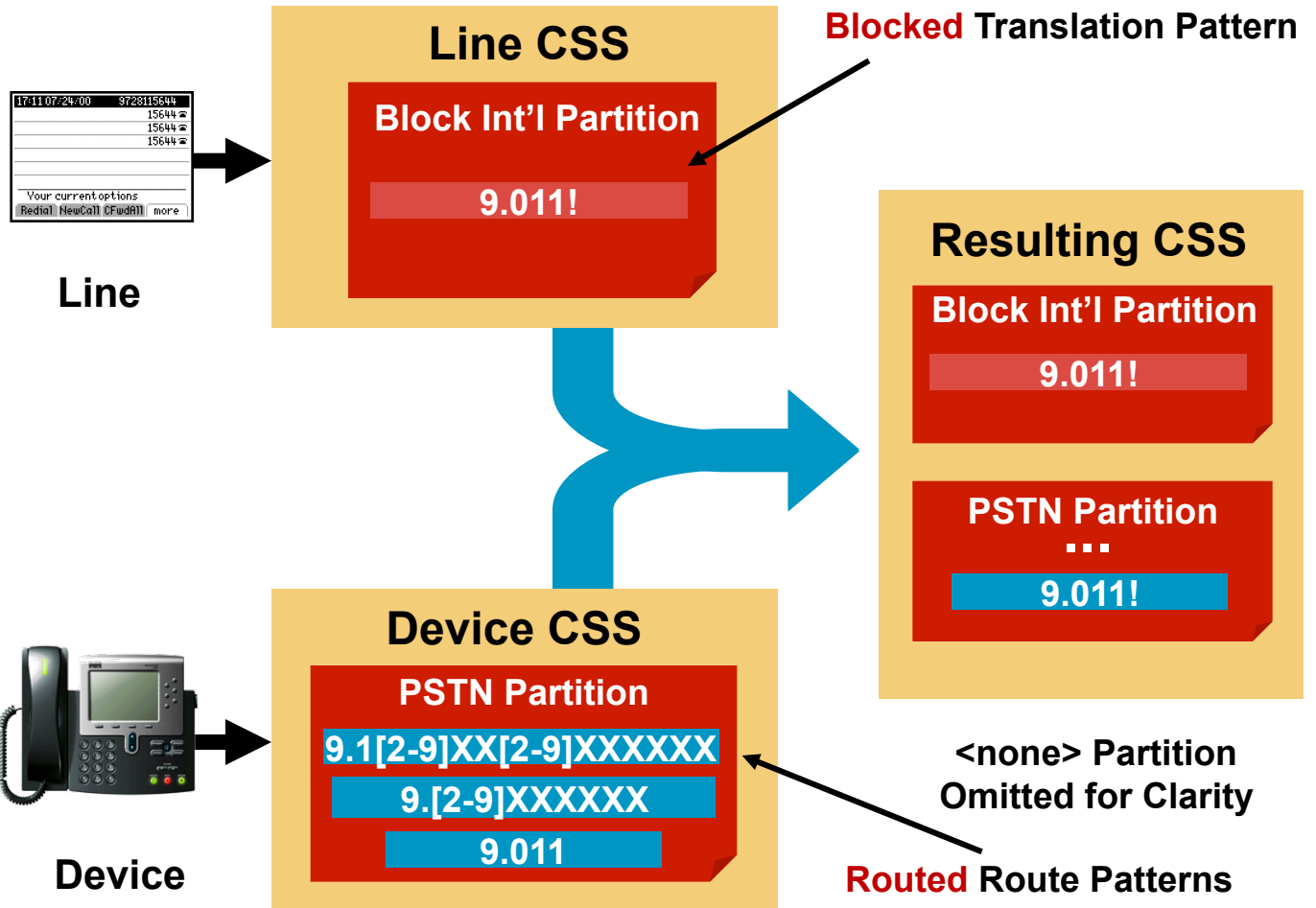
Line CSS vs. Device CSS



The Line/Device CSS Approach

Key Idea

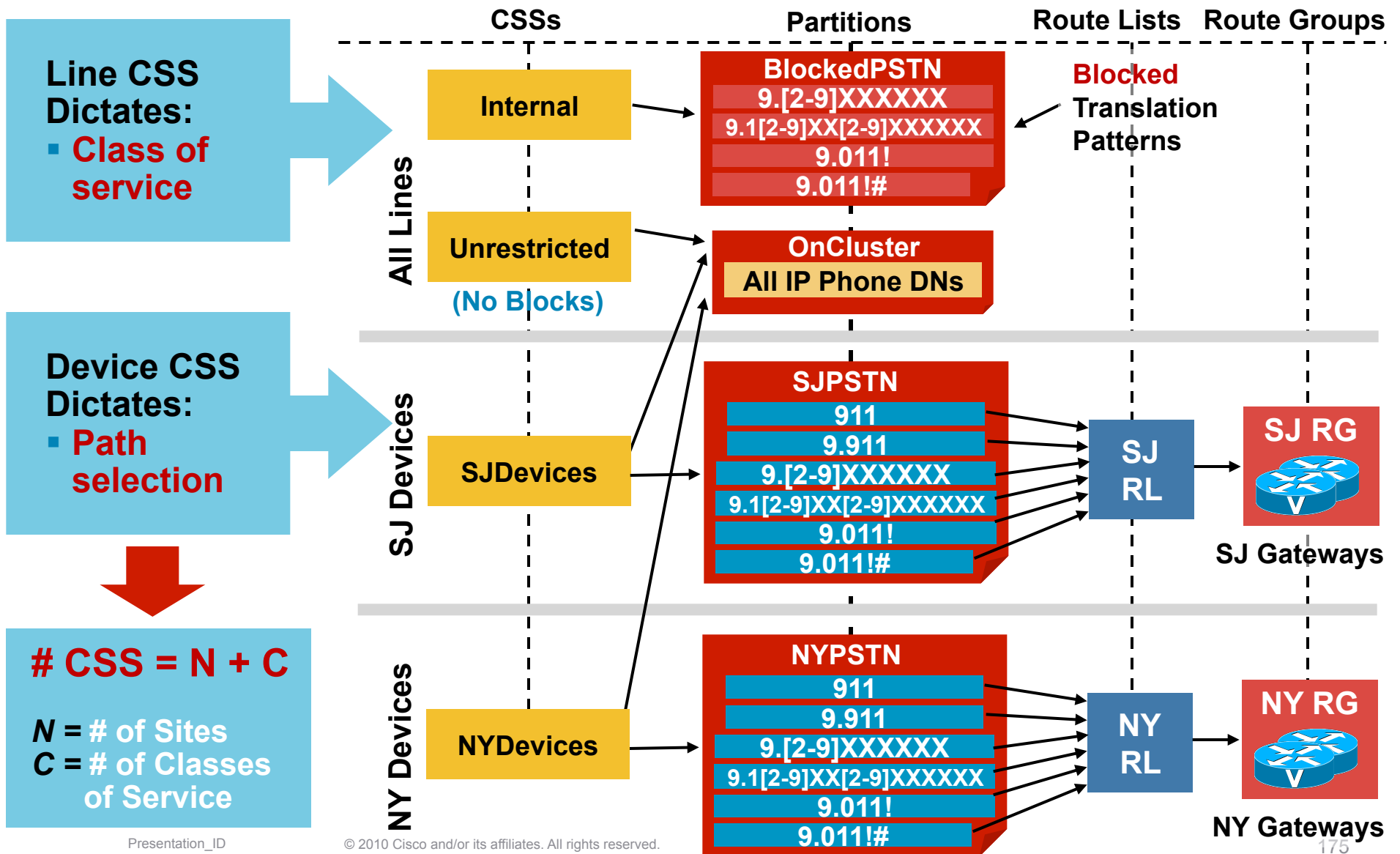
Line CSS
 Selectively Blocks
 Undesired Routes
 (According to
 Class of Service)



Device CSS
 Allows Access
 to All
 External Routes

The Line/Device CSS Approach

Scalability for Centralised Deployments



The Line/Device CSS Approach

CallForward Caveats

- Forwarded calls use the CallFwdxxx CSSs only; these values are not concatenated with line or device CSS
- If forwarded calls must have unrestricted privileges, set the CallFwdxxx CSSs to the site-specific device CSS
- If forwarded calls must be restricted to internal numbers only, set the CallFwdxxx CSSs to a single, global CSS with only internal partitions
- In Cisco Unified Communications Manager version 4.X: If forwarded calls must have some intermediate restriction (e.g., no international calls), this approach may lose efficiency, as additional site-specific CSSs will be needed



- In Cisco Unified Communications Manager 5.X and 6.X, a new CSS [Secondary Calling Search Space for CallForwardAll] has been added, allowing for CFA to have all the classes of service afforded by the line/device approach

The Line/Device CSS Approach

CallForward Caveats (Cont.)



Calling Search Space Activation policy (6.X only)

Use system default

the CFA CSS activation policy cluster-wide service parameter determines which forward all calling search space will be used

With configured CSS

The configures CFAAll and secondary CSS for CFAAll are used

With activating device/line CSS

The forward all calling search space and secondary calling search space for forward all automatically gets populated with the directory number calling search space and device calling search space for the activating device

- When a device is roaming in the same device mobility group, Cisco Unified Communications Manager uses the device mobility CSS to reach the local gateway. If a user sets call forward all at the phone, the CFA CSS is set to none, and the CFA CSS activation policy is set to With activating device/line CSS, then:

The device CSS and line CSS get used as the CFA CSS when the device is in its home location

If the device is roaming within the same device mobility group, the device mobility CSS

from the roaming device pool and the line CSS get used as the CFA CSS

If the device is roaming within a different device mobility group, the device CSS and

line CSS get used as the CFA CSS

The Line/Device CSS Approach

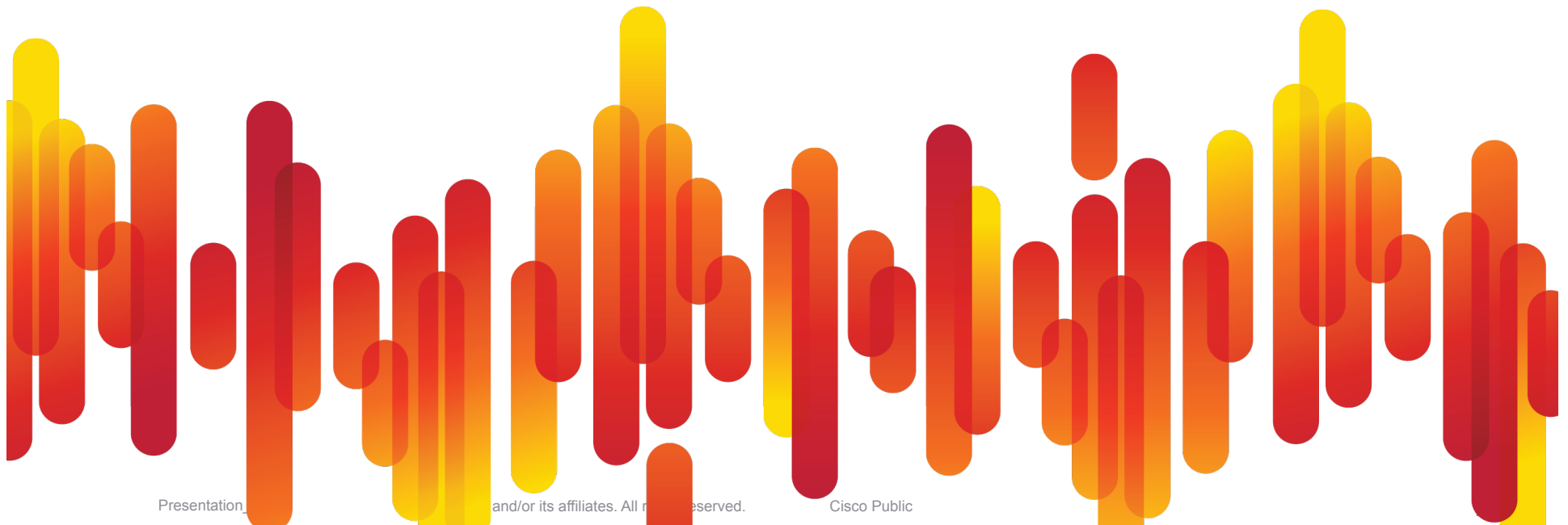
Other Caveats

- Blocking translation patterns configured within the Line CSS must be **at least as specific** as the route patterns configured within the device CSS
 - (Watch for the @ wildcard, as its patterns are very specific)
- AAR uses a different CSS for rerouted calls; in most cases, this CSS can be the same as the unrestricted site-specific Device CSS
- Priority order between line and device is reversed for CTI route points and CTI ports; therefore, the line/service CSS approach **cannot be directly applied to CTI devices**, such as Softphone (not Unified Personal Communicator)

In this case, it is viable only if blocked patterns are more specific than the routed ones (i.e., not relying on order of the partitions)

Appendix

Uniform Addressing



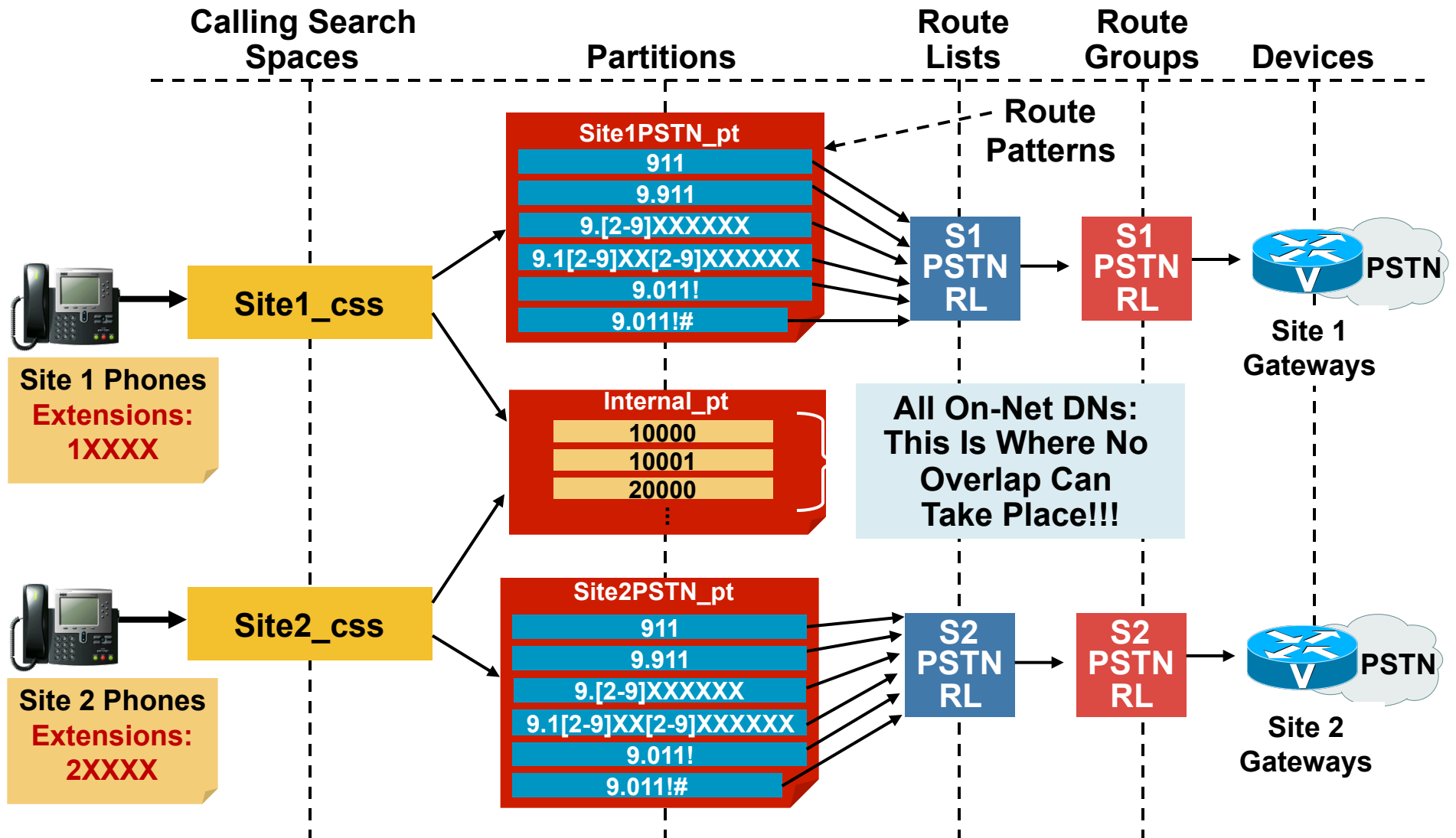
Uniform On-Net Dialing

Use this Model If...

- DID ranges do not overlap (based on chosen quantity of digits for internal calls)
- Number of sites is small
- Number of sites is not expected to grow significantly in the future
- DID ranges are deemed to be predictable (can anyone make that assumption??? One area code split, and you may be back to the drawing board!!!)

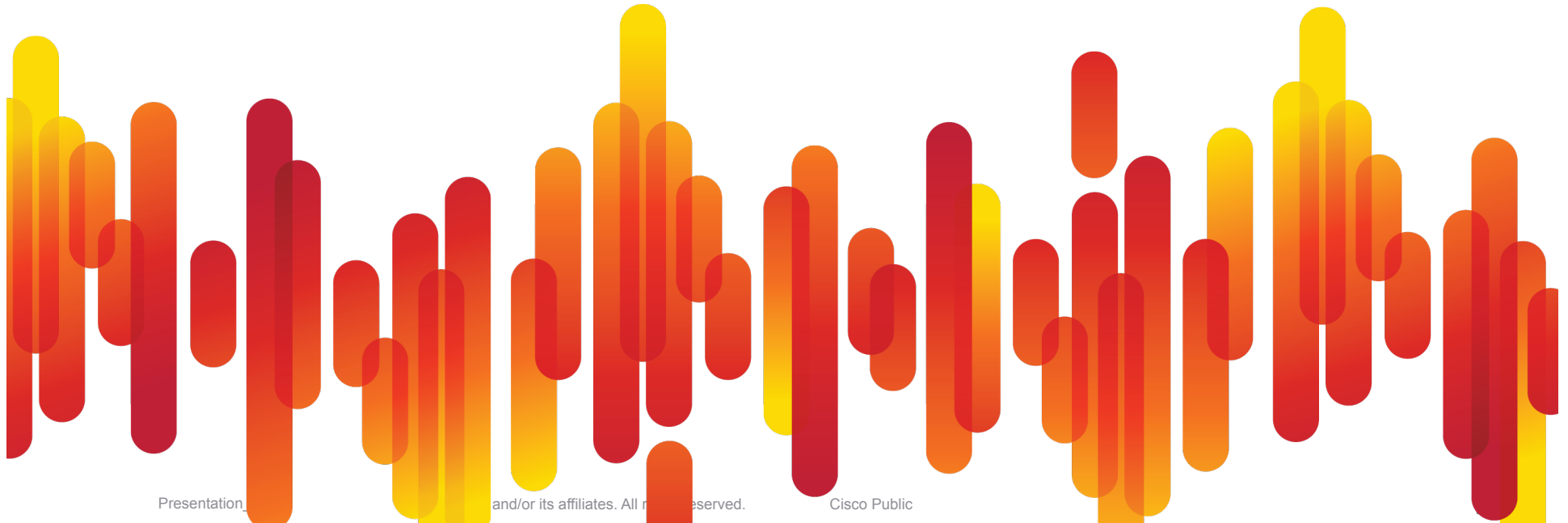
Uniform On-Net Dialing

Composite View



Appendix

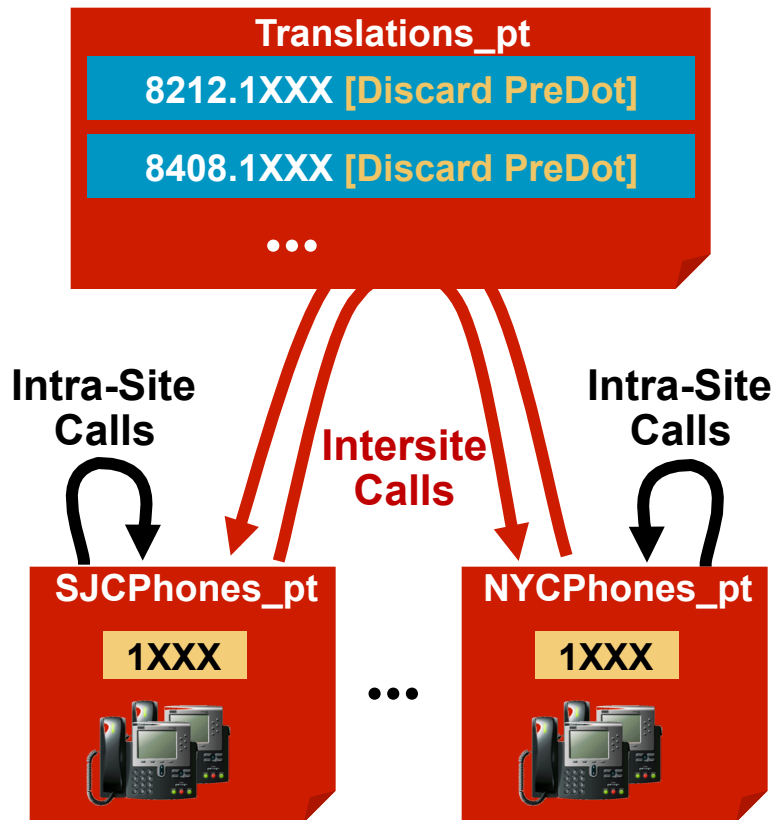
Variable-Length On-Net Dialing with Partitioned Addressing



Choosing a Dial Plan Approach

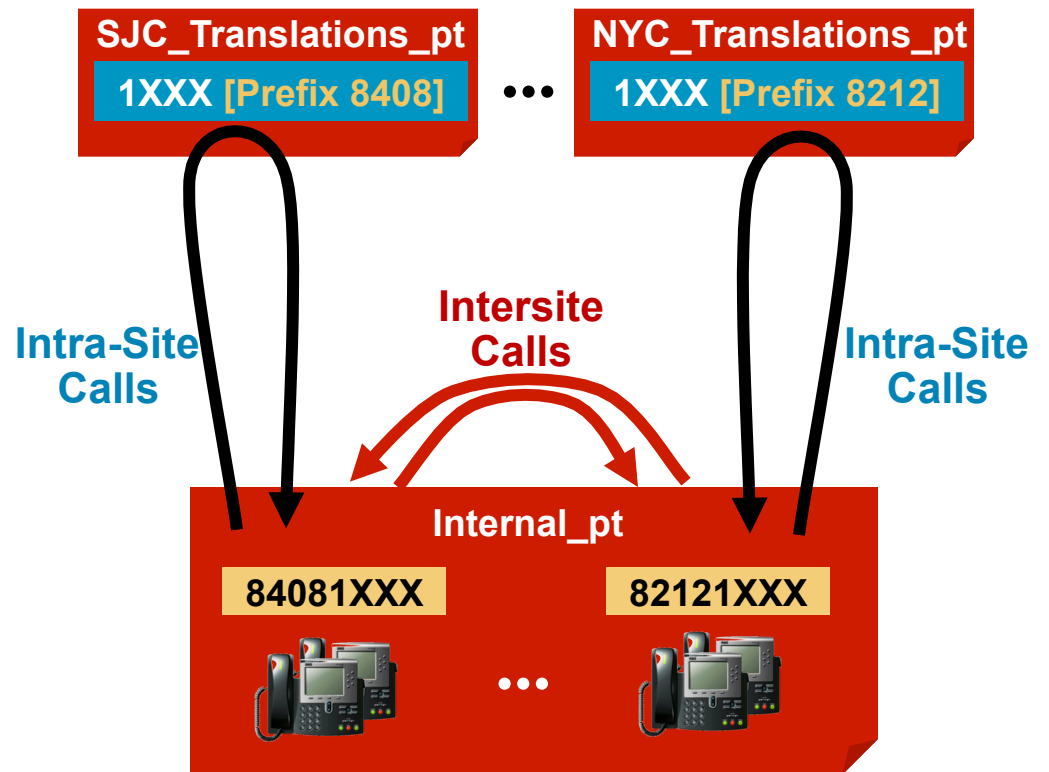
Addressing Methods for VLOD

Partitioned Addressing



- Phone DNs in different partitions
- Global Xlations for intersite calls

Flat Addressing



- Phone DNs in same global partition
- Per-site translations for intrasite calls

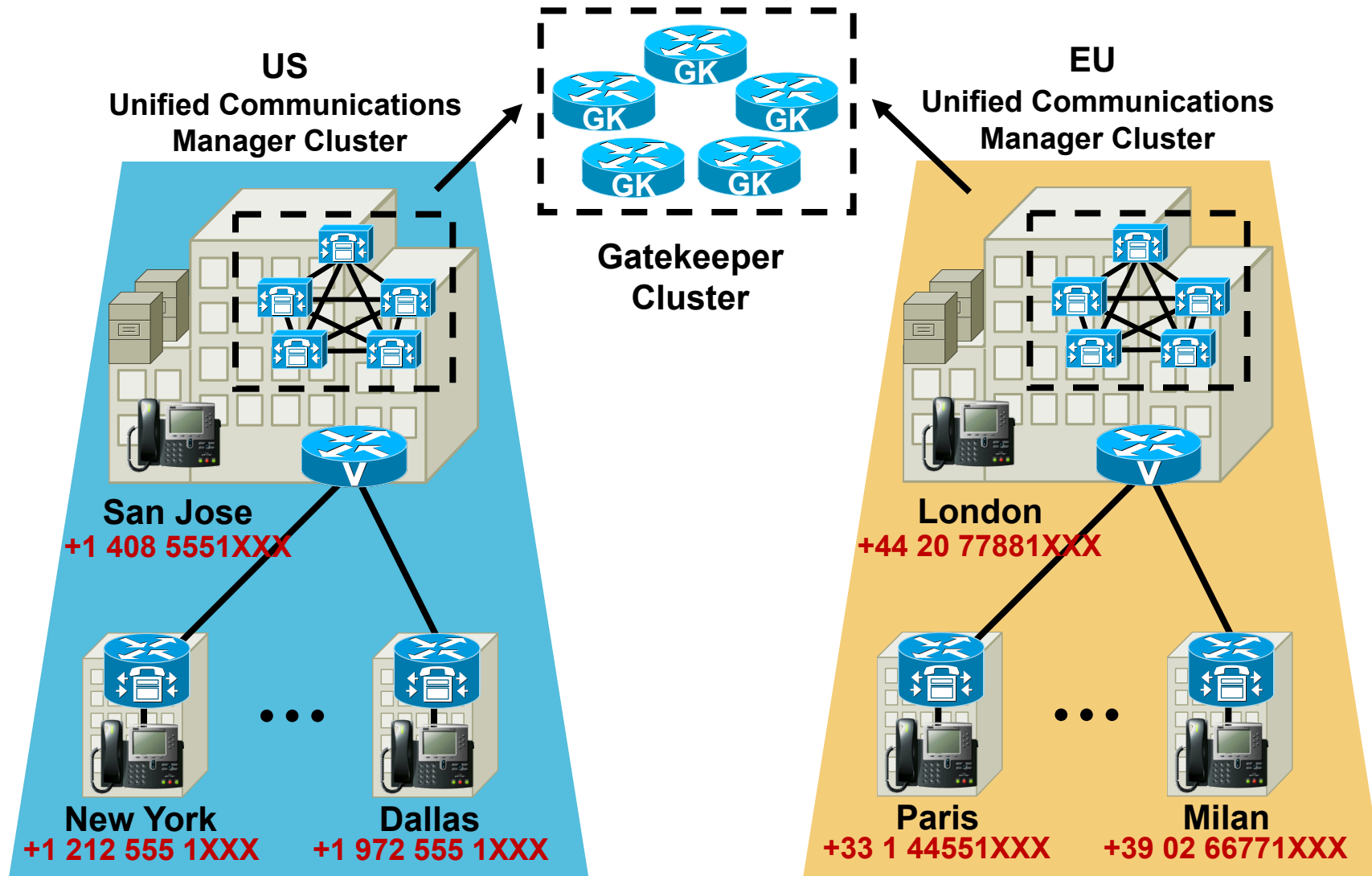
VLOD with Partitioned Addressing

Use this Model If...

- A global on-net numbering plan using site codes is not desired (**or possible**)
- Policy restrictions must be applied to on-net intersite calls (**that is, some or all users are not allowed to dial other sites on-net**)
- Intersite calls are always routed over the PSTN
- CTI applications are not used across sites
- You have to because the system was built this way from the start...

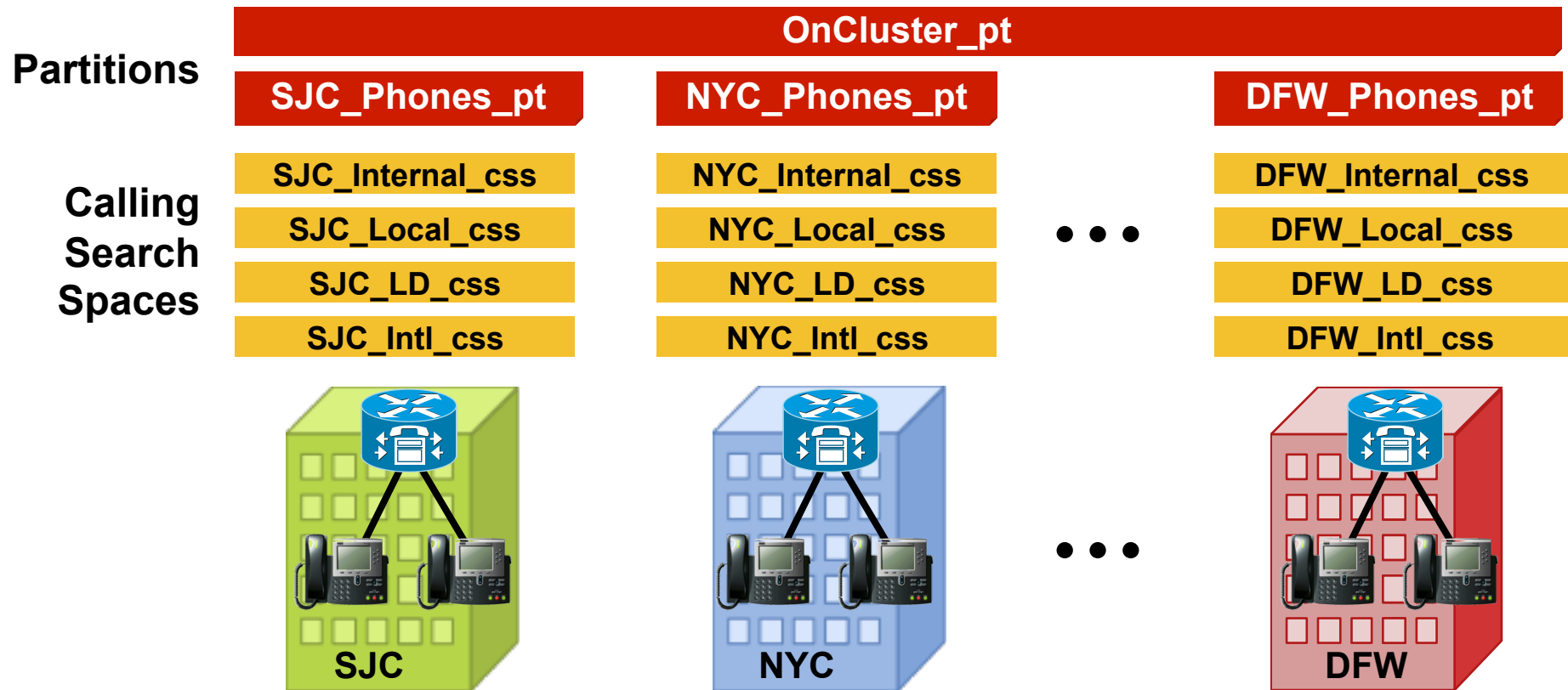
VLOD with Partitioned Addressing

Hypothetical Customer Example



VLOD with Partitioned Addressing

Partitions and Calling Search Spaces



*Note: If Using the Line/Device CSS Approach, the Number of CSSs Can Be Reduced

VLOD with Partitioned Addressing

Line Configuration

System Route Plan Service Feature Device User Application Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

Directory Number Configuration [Configure Device \(SEP000D294DFC13\)](#)
[Dependency Records](#)

Associated With
SEP000D294DFC13
791E (Line 1)

Directory Number: 1000 (NYCPhones_pt)
Status: Update completed
Note: Any update to this Directory Number automatically resets the associated devices

Update Remove from Device Reset Devices

Directory Number

Directory Number* 1000
Partition NYCPhones_pt

Directory Number Settings

Voice Mail Profile <None>
(Choose <None> to use default)

Calling Search Space <None>

AAR Group <None>

User Hold Audio Source <None>

Line Settings for this Device

Display (Internal Caller ID) John Smith

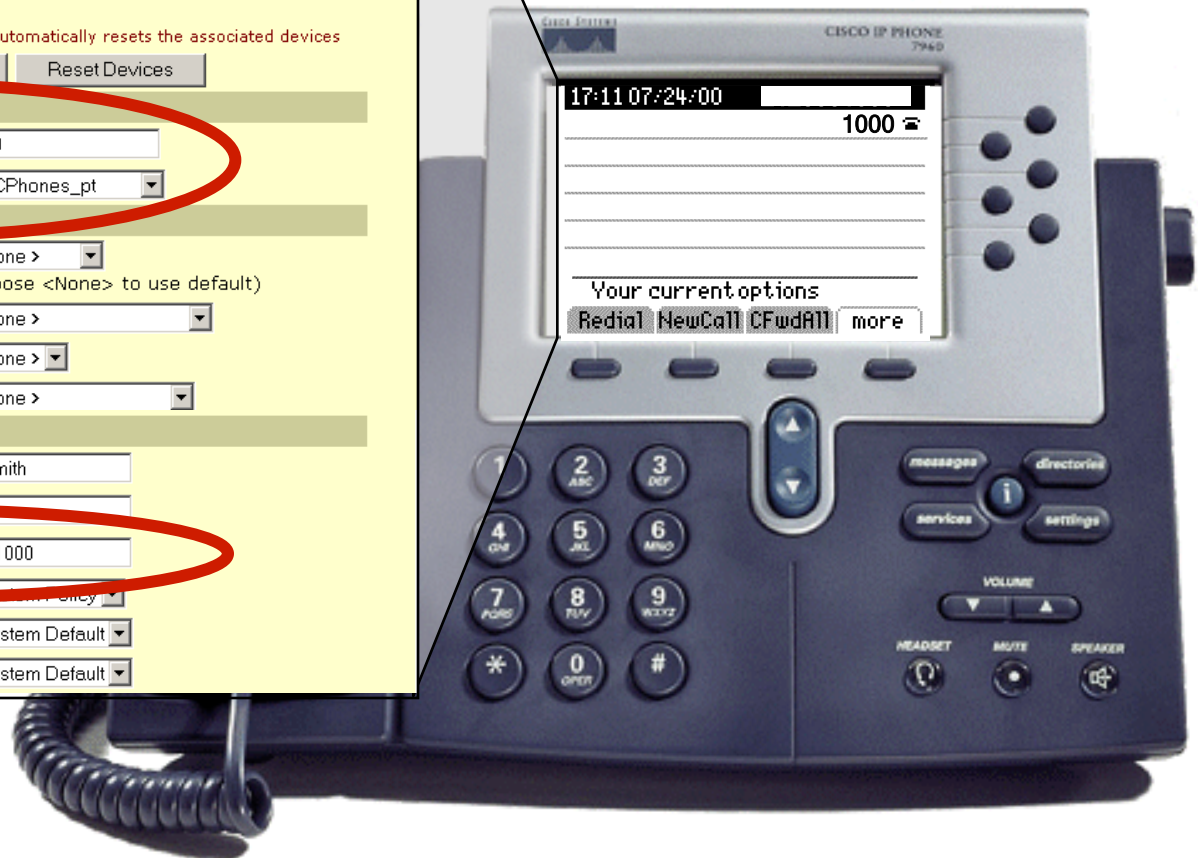
Line Text 1

External Phone Number Mask 2125551000

Message Waiting Tone Policy Use System Policy

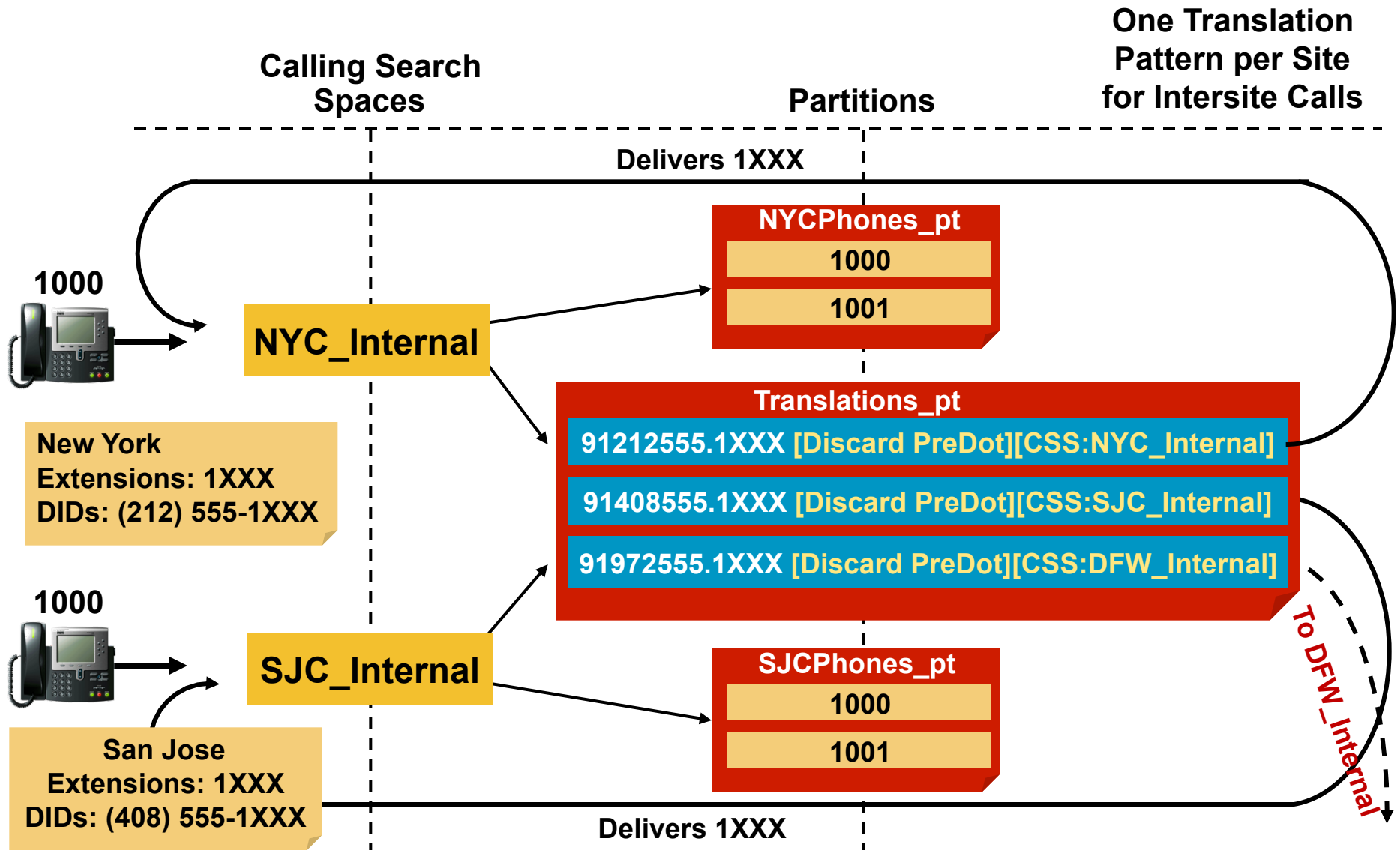
Ring Setting (Phone Idle) Use System Default

Ring Setting (Phone Active)** Use System Default



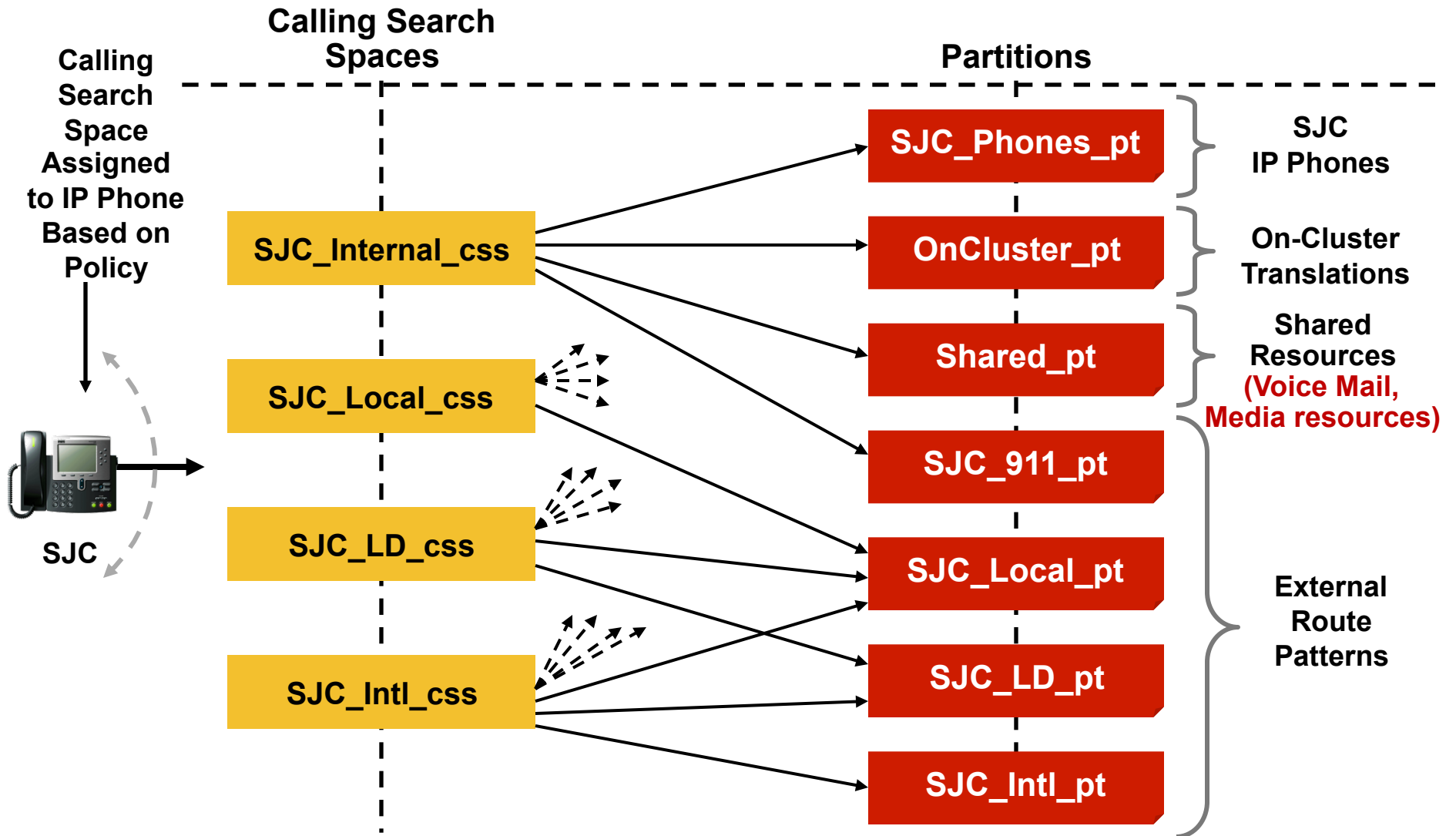
VLOD with Partitioned Addressing

Intersite Calls Within a Cluster



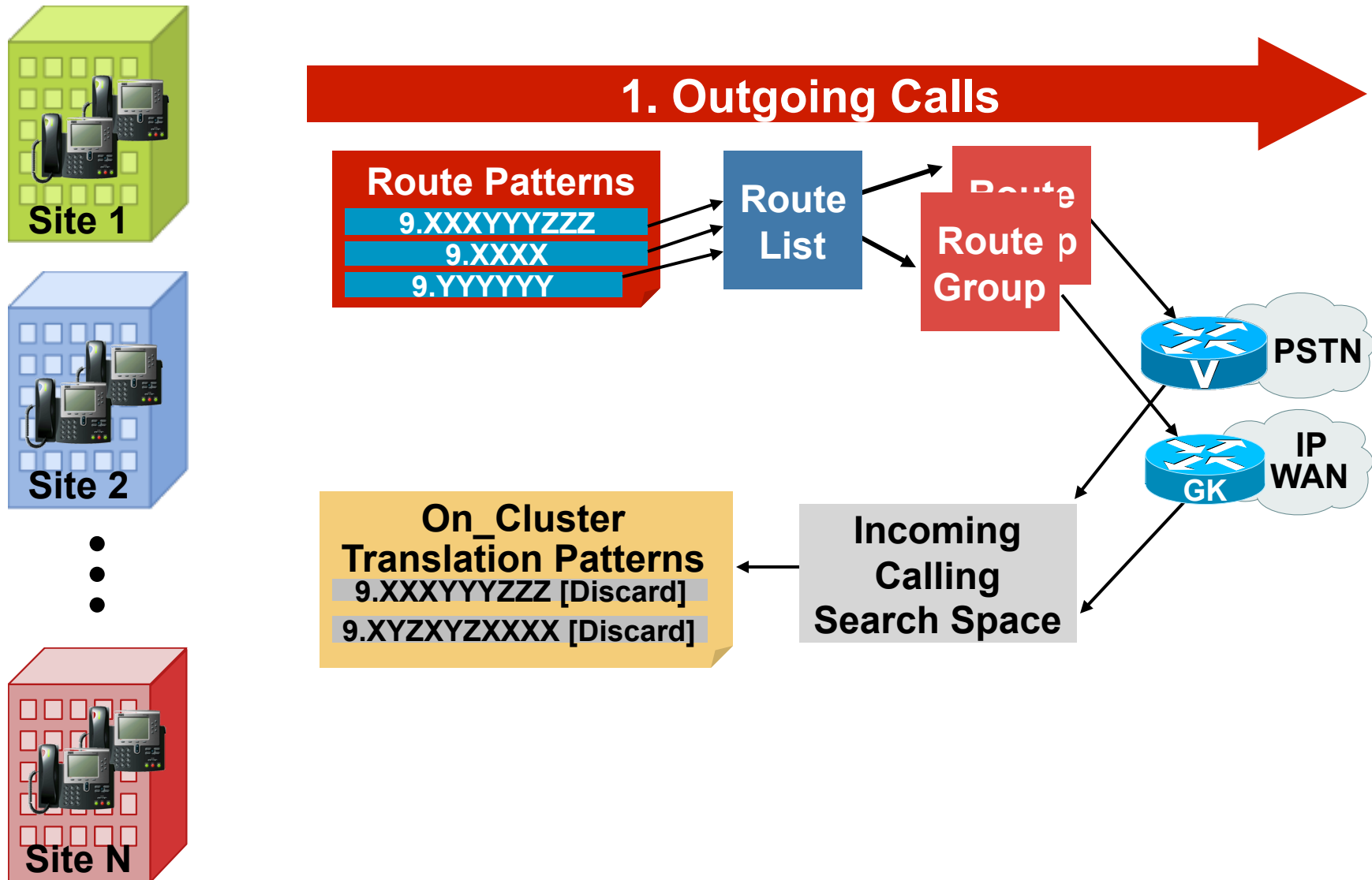
VLOD with Partitioned Addressing

View of Partitions/Calling Search Spaces



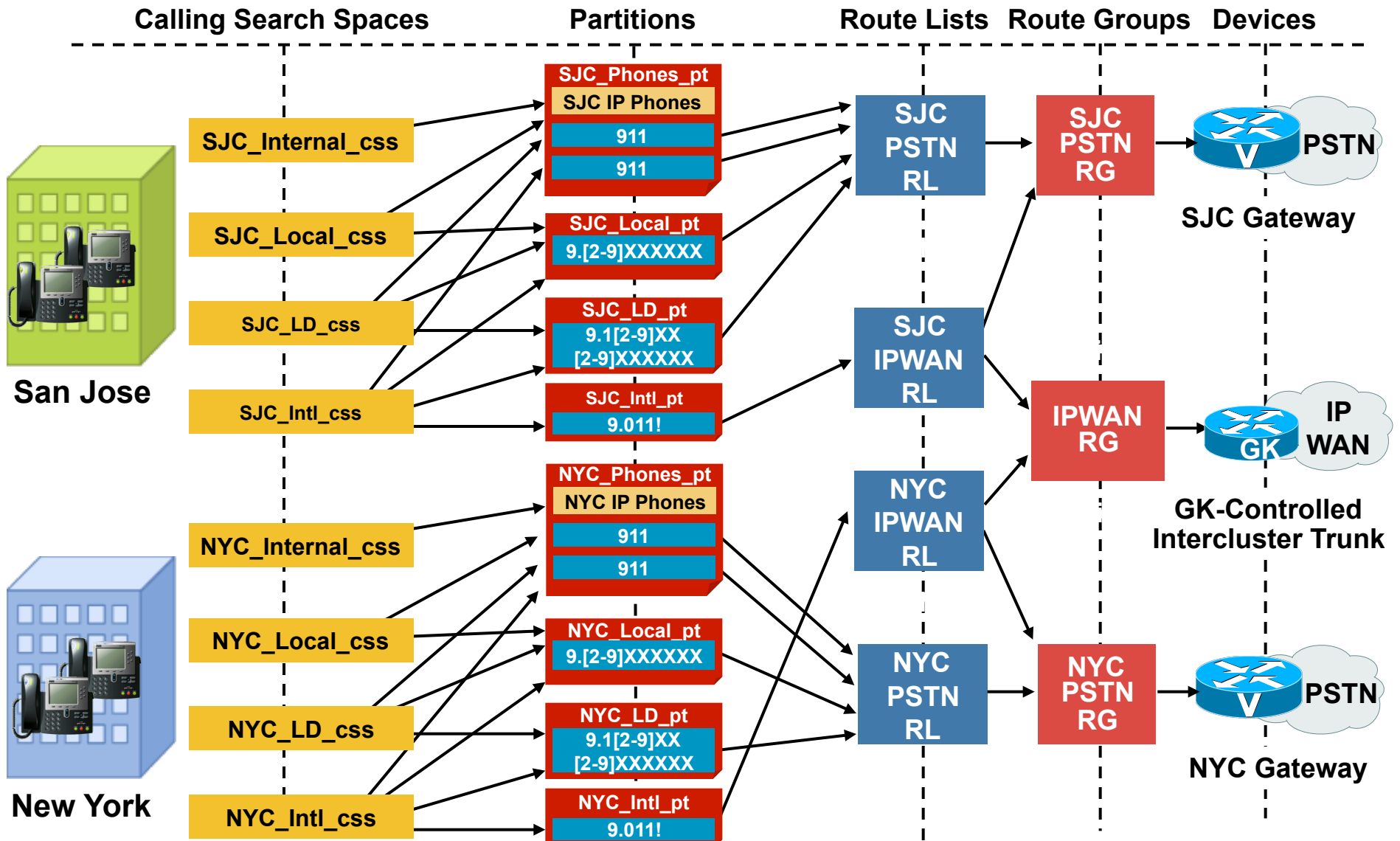
VLOD with Partitioned Addressing

Outgoing PSTN/Gatekeeper Calls



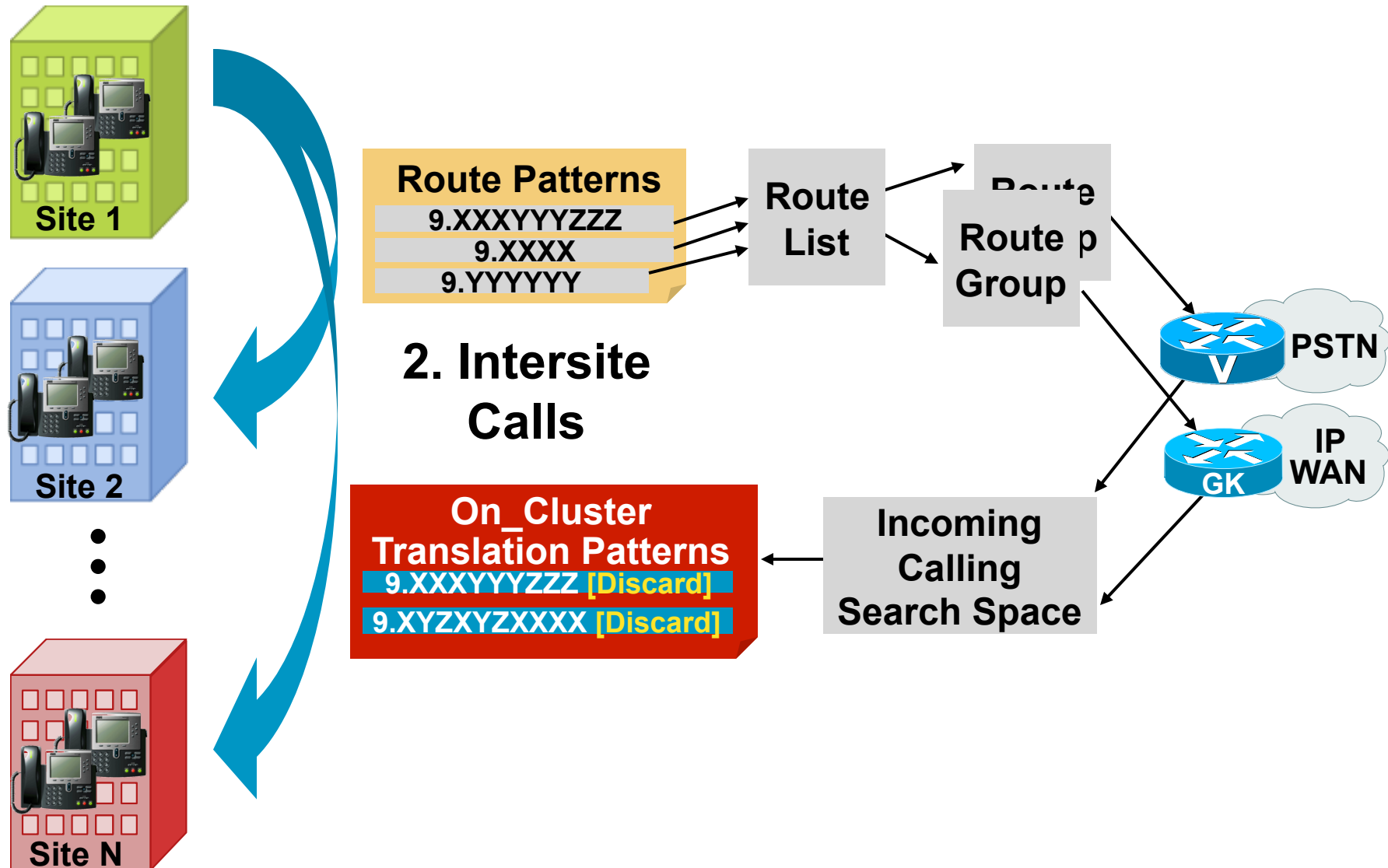
VLOD with Partitioned Addressing

Outgoing PSTN/Gatekeeper Calls



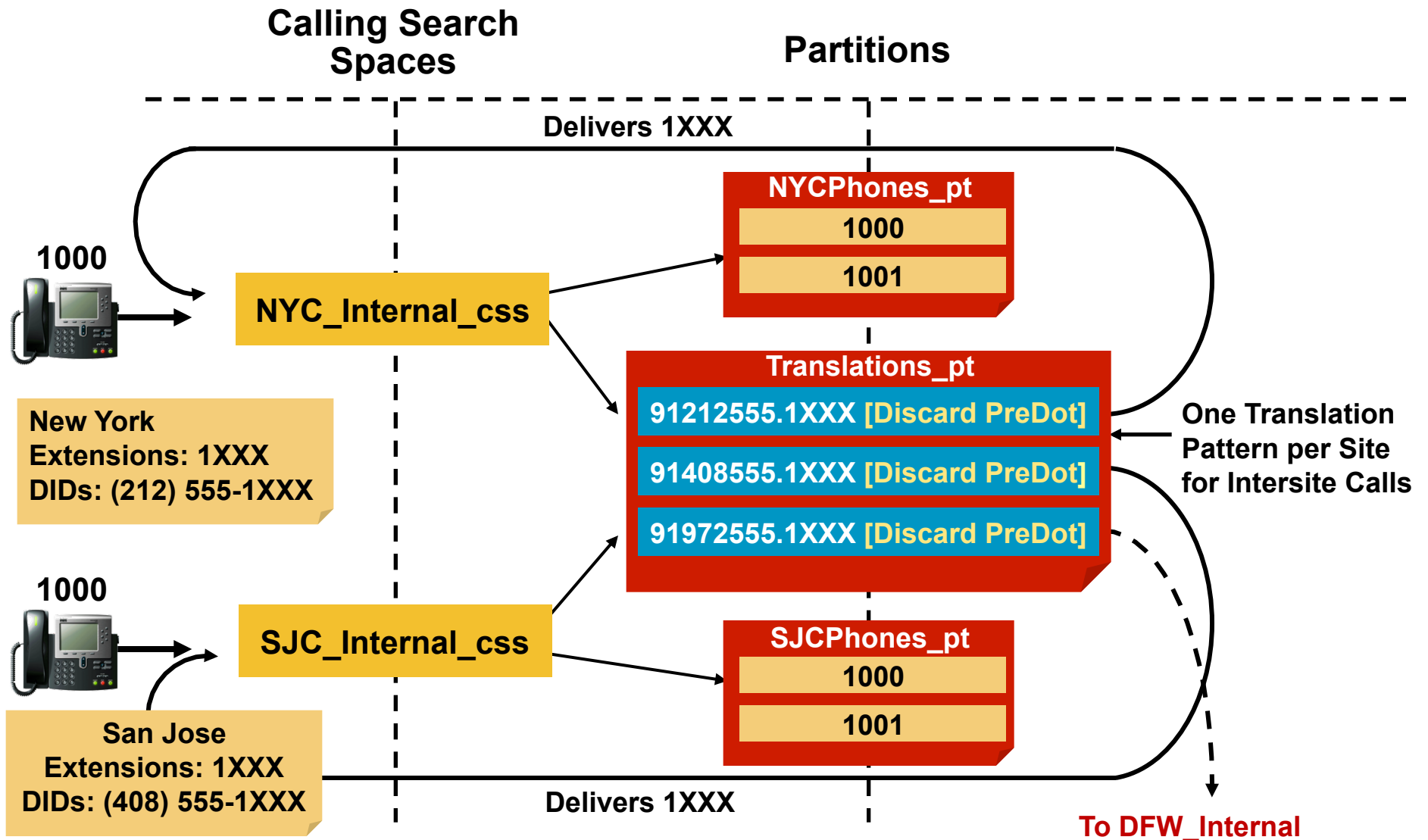
VLOD with Partitioned Addressing

Intersite Calls Within a Cluster



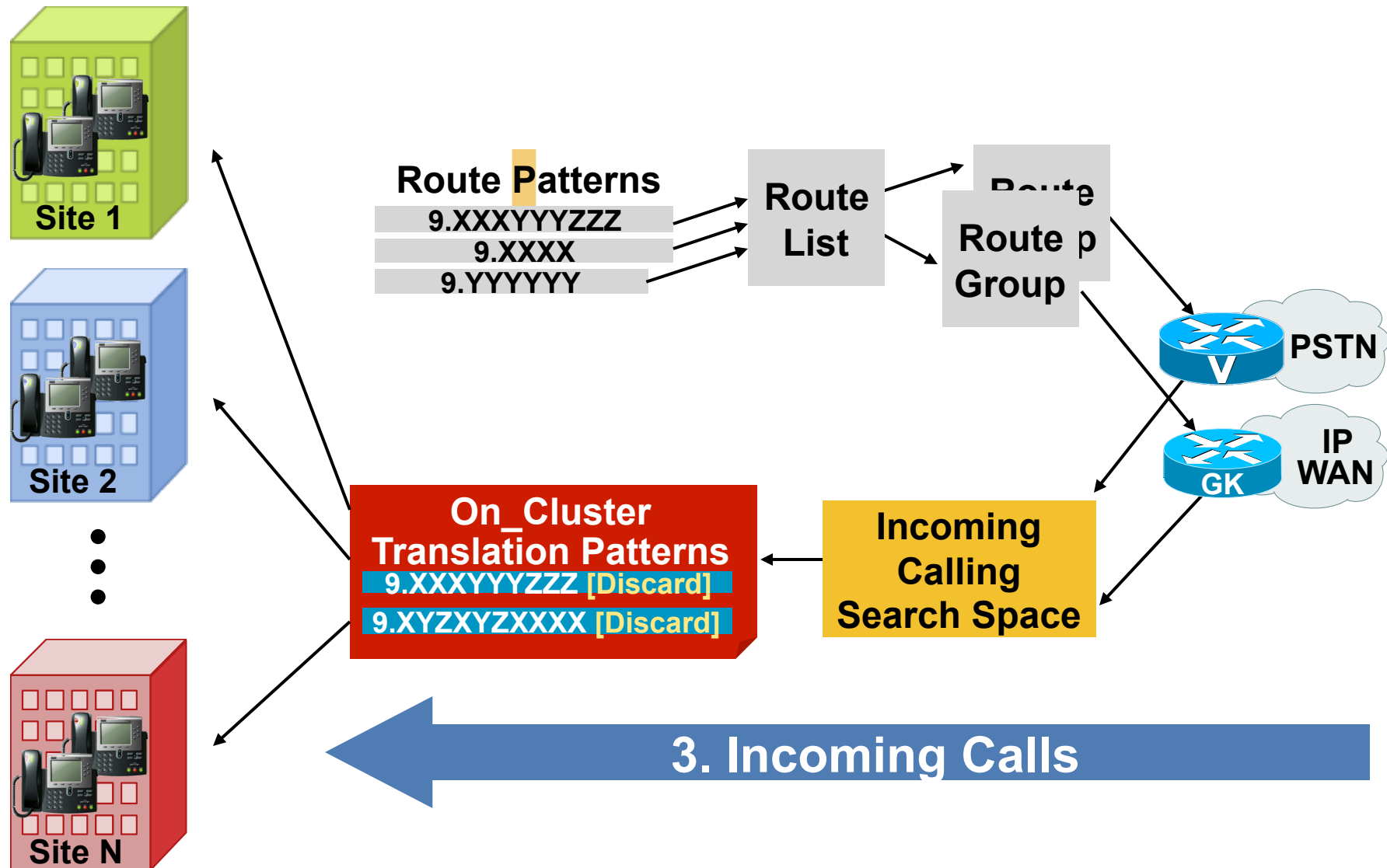
VLOD with Partitioned Addressing

Intersite Calls Within a Cluster



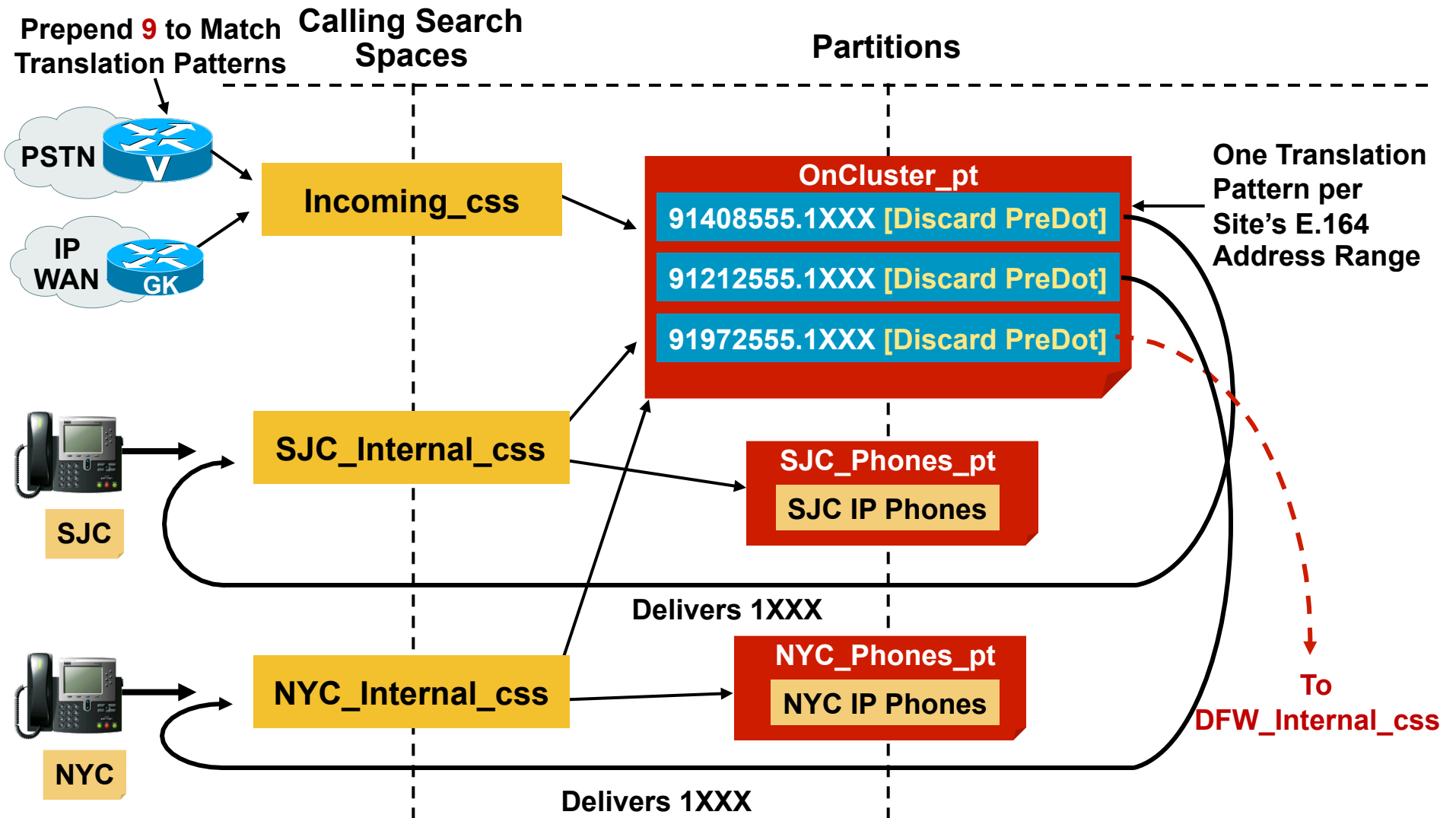
VLOD with Partitioned Addressing

Incoming PSTN/Gatekeeper Calls



VLOD with Partitioned Addressing

Incoming PSTN/Gatekeeper Calls



VLOD with Partitioned Addressing

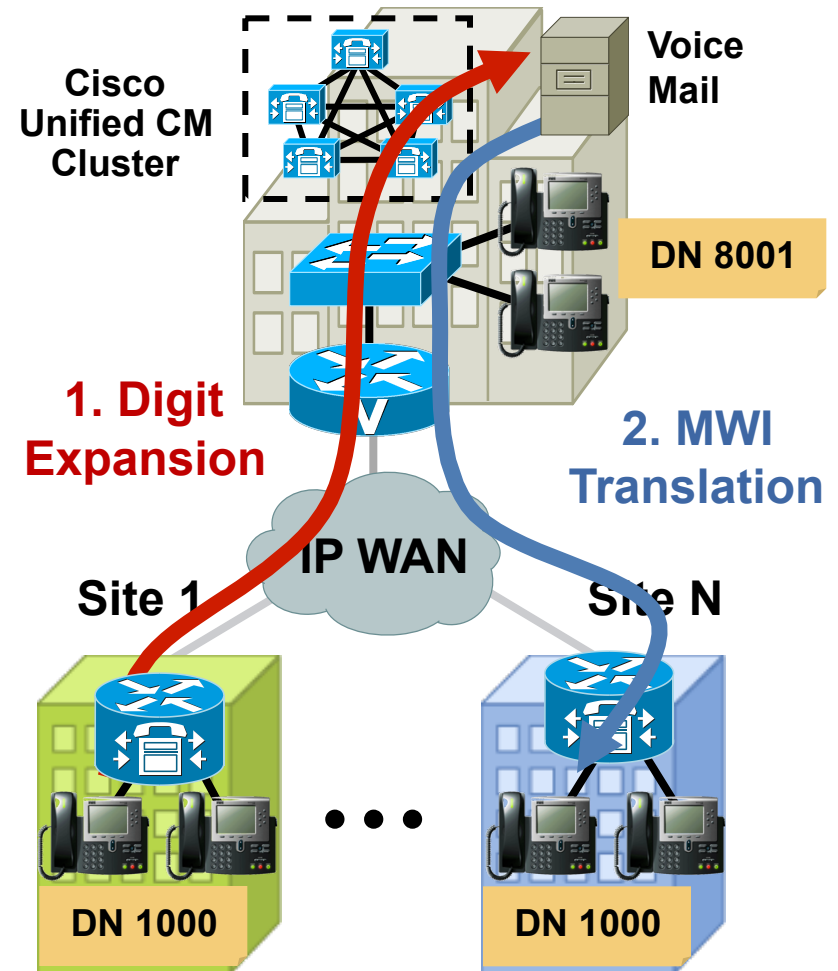
Gatekeeper Configuration

```
gatekeeper
zone local US cisco.com 10.9.11.1
zone local EU cisco.com 10.20.1.1
no zone subnet US default enable
no zone subnet EU default enable
zone subnet US 10.9.11.2/32 enable
zone subnet US 10.9.11.3/32 enable
zone subnet EU 10.20.1.2/32 enable
zone subnet EU 10.20.1.3/32 enable
zone prefix US 14085551...
zone prefix US 12125551...
zone prefix US 19725551...
zone prefix EU 442077881...
zone prefix EU 33144551...
zone prefix EU 390266771...
gw-type-prefix 1#* default-technology
bandwidth interzone zone US 256
bandwidth interzone zone EU 256
arq reject-unknown-prefix
no shutdown
```

VLOD with Partitioned Addressing

Voice Mail Integration

- Both SCCP—(Cisco Unity) and SMDI-based voice mail (VM) systems can be used
- Voice mail boxes need a unique DN
- Need to **expand** DNs when accessing VM
- Message Waiting Indicator (MWI) messages from VM system need to be **translated** to match appropriate DN/partition



VLOD with Partitioned Addressing

Voice Mail Integration: Digit Expansion

Voice Mail Profile Configuration

[Add a New Voice Mail Profile](#)
[Back to Find/List Voice Mail Profiles](#)

Voice Mail Profile: Site1-VMProfile
Status: Ready

Voice Mail Profile Name*

Description

Voice Mail Pilot ** (Choose <None> to use default)

Voice Mail Box Mask

Make this the default Voice Mail Profile for the system

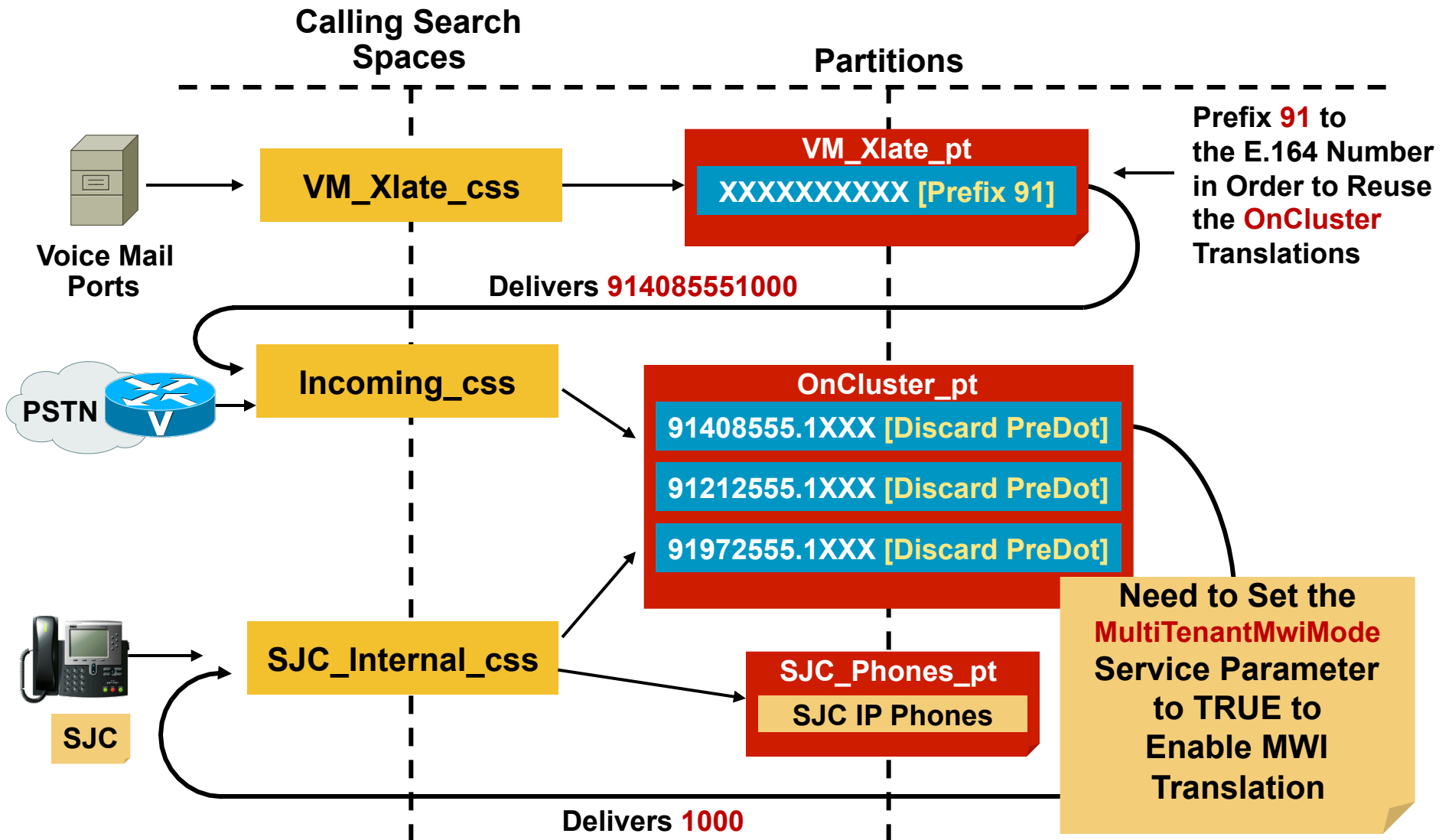
* indicates required item

** The Voice Mail Pilot is comprised of the Voice Mail Pilot Number and it's corresponding Calling Search Space Name (<Voice Mail Pilot Number>/<Calling Search Space>).

Use the **Voice Mail Box Mask** Field in Each Vm Profile to Uniquely Identify the Voice Mail Boxes (e.g., Using the Full E. 164 Number)

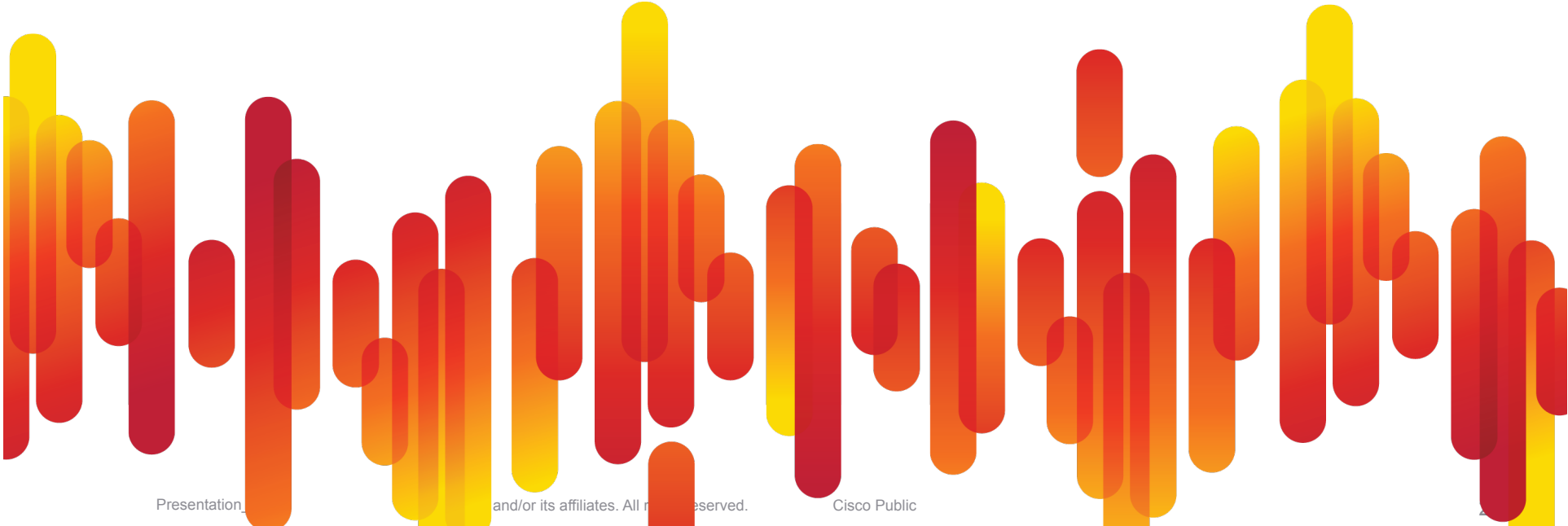
VLOD with Partitioned Addressing

Voice-Mail Integration: MWI Translation



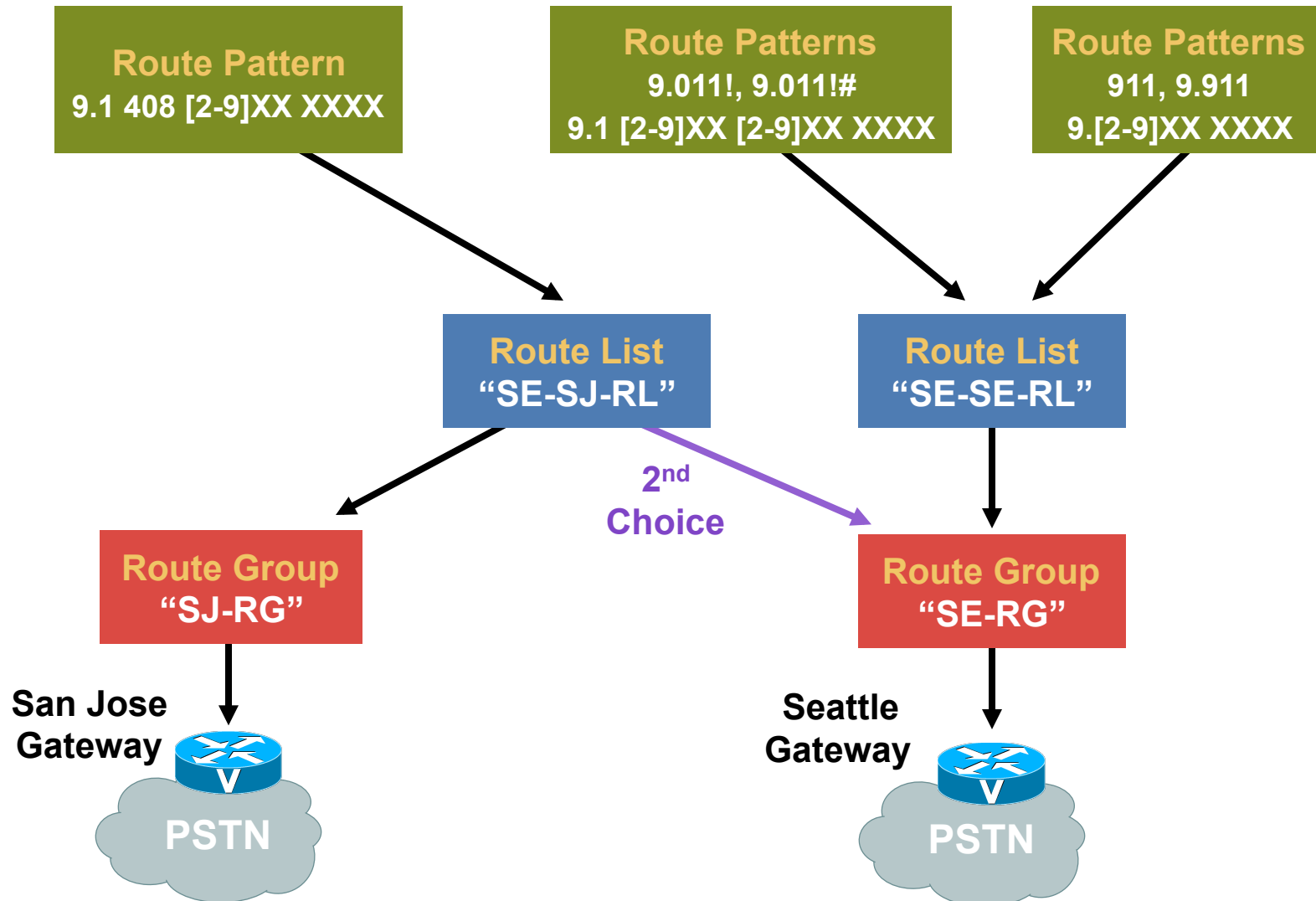
Appendix

Tail End Hop Off: Some pre-LRG Considerations



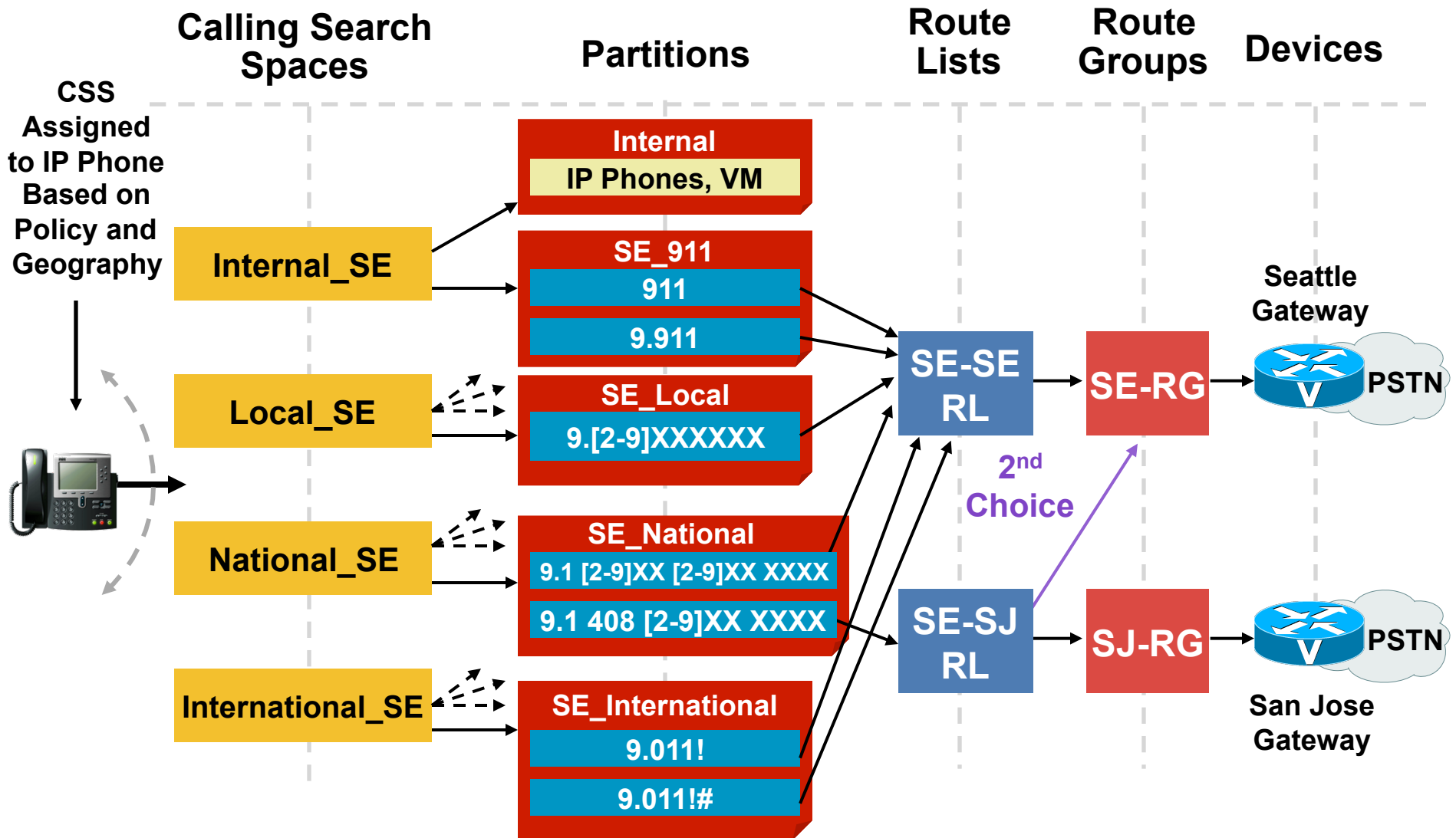
Tail-End Hop-Off (TEHO)

Intracluster: Route Patterns for Seattle



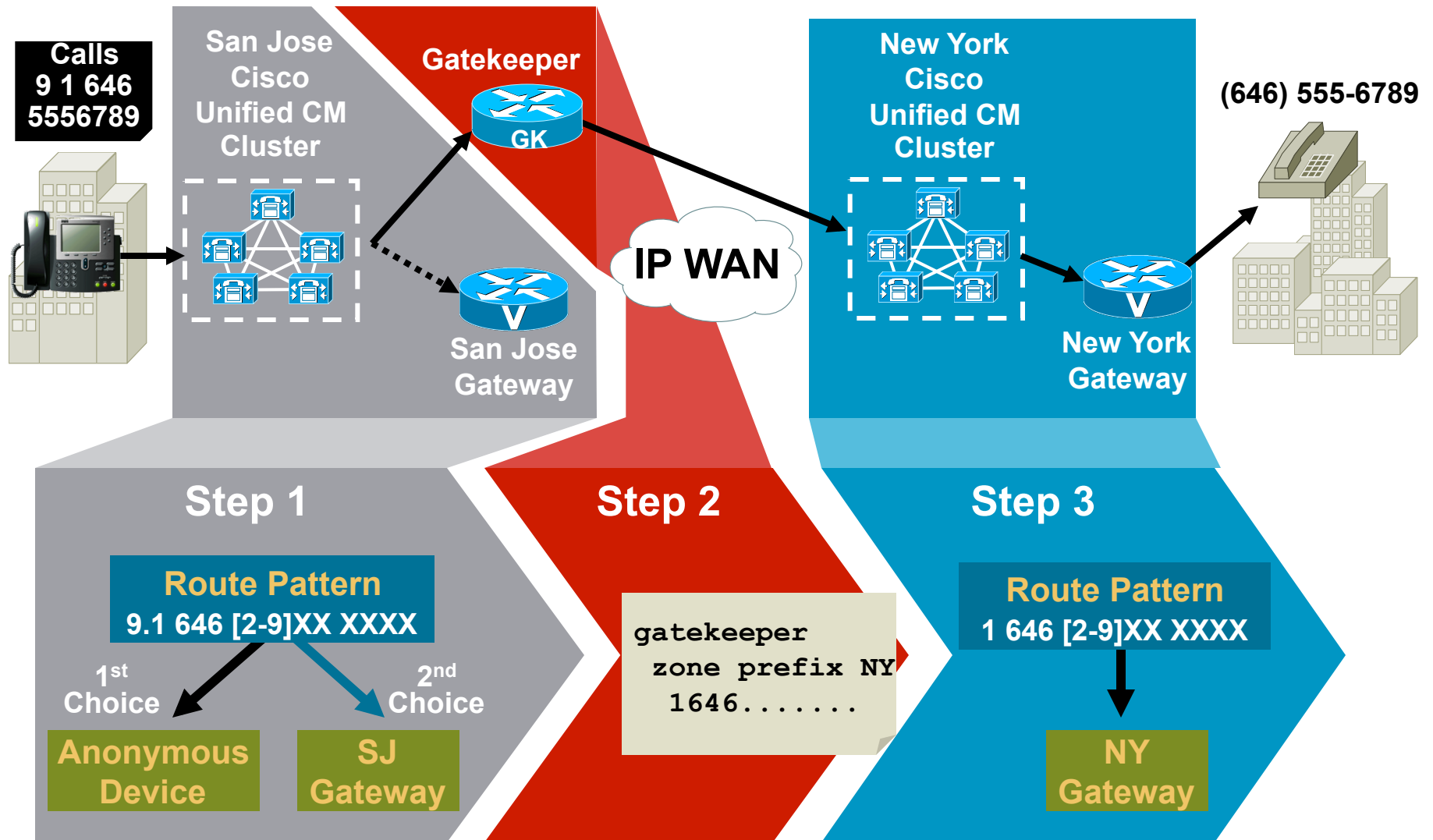
Tail-End Hop-Off (TEHO)

Intracluster: Composite Dial Plan for Seattle



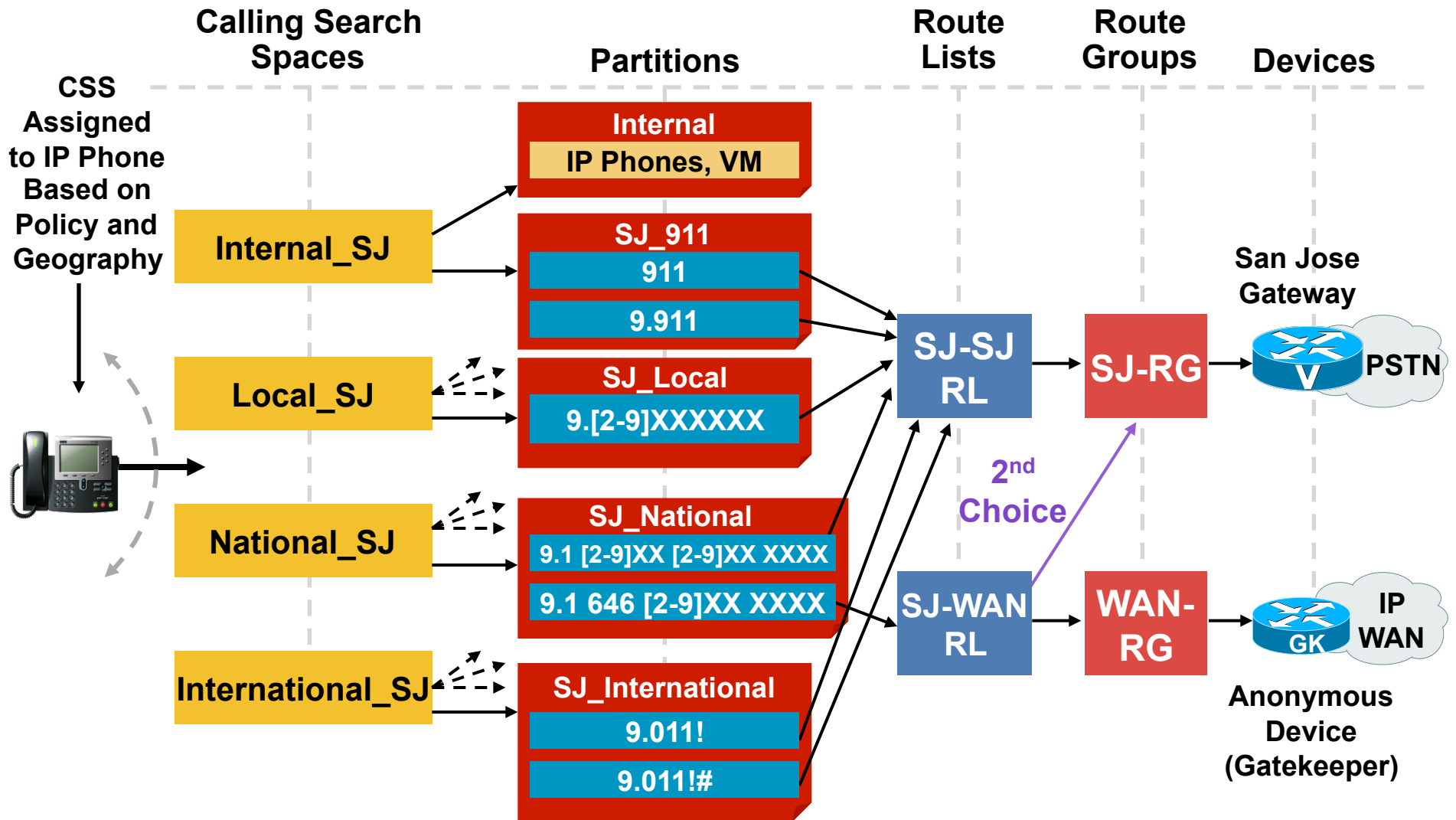
Tail-End Hop-Off (TEHO)

Intercluster: San Jose to New York



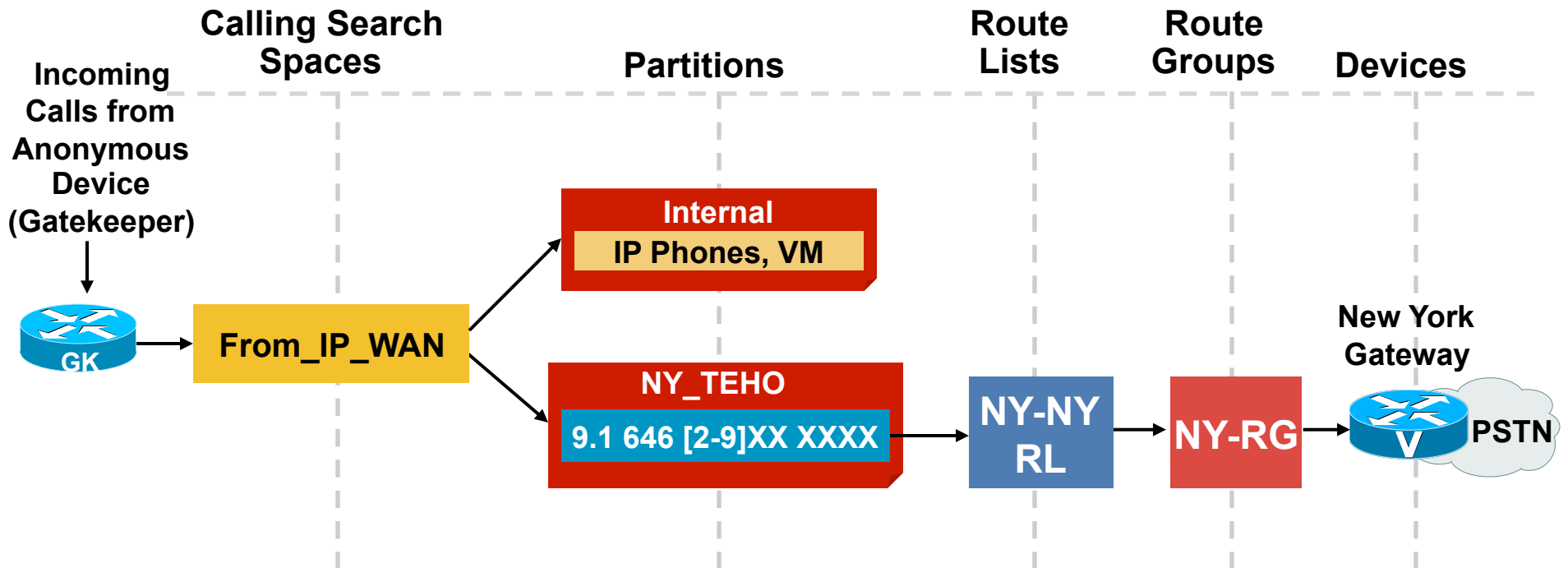
Tail-End Hop-Off (TEHO)

Intercluster: Composite Dial Plan for San Jose



Tail-End Hop-Off (TEHO)

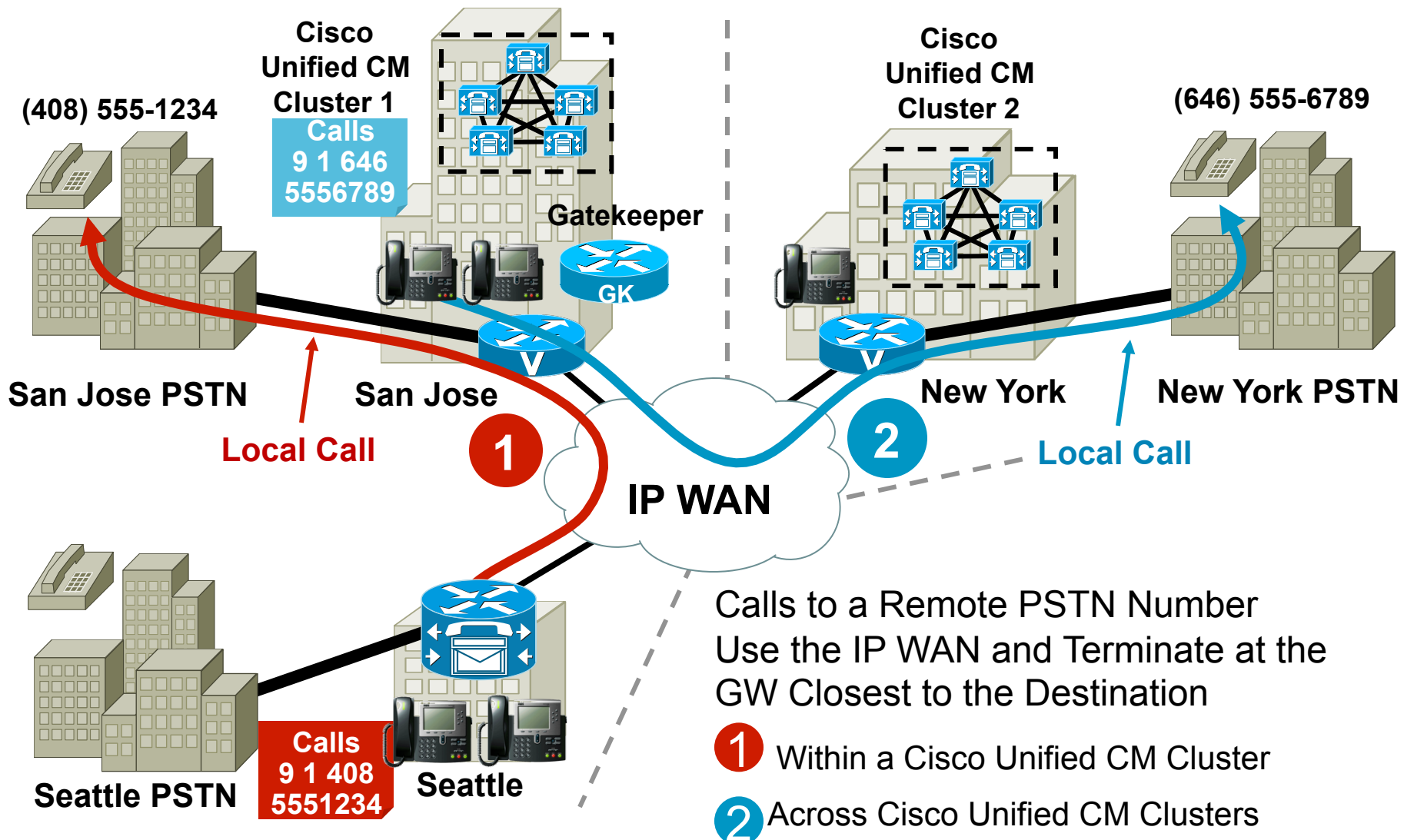
Intercluster: Dial Plan for New York



Note: To Avoid Routing Loops, Do not Include Partitions that Contain IP WAN Routes in the **From_IP_WAN** Calling Search Space

Tail-End Hop-Off (TEHO)

What Is It?

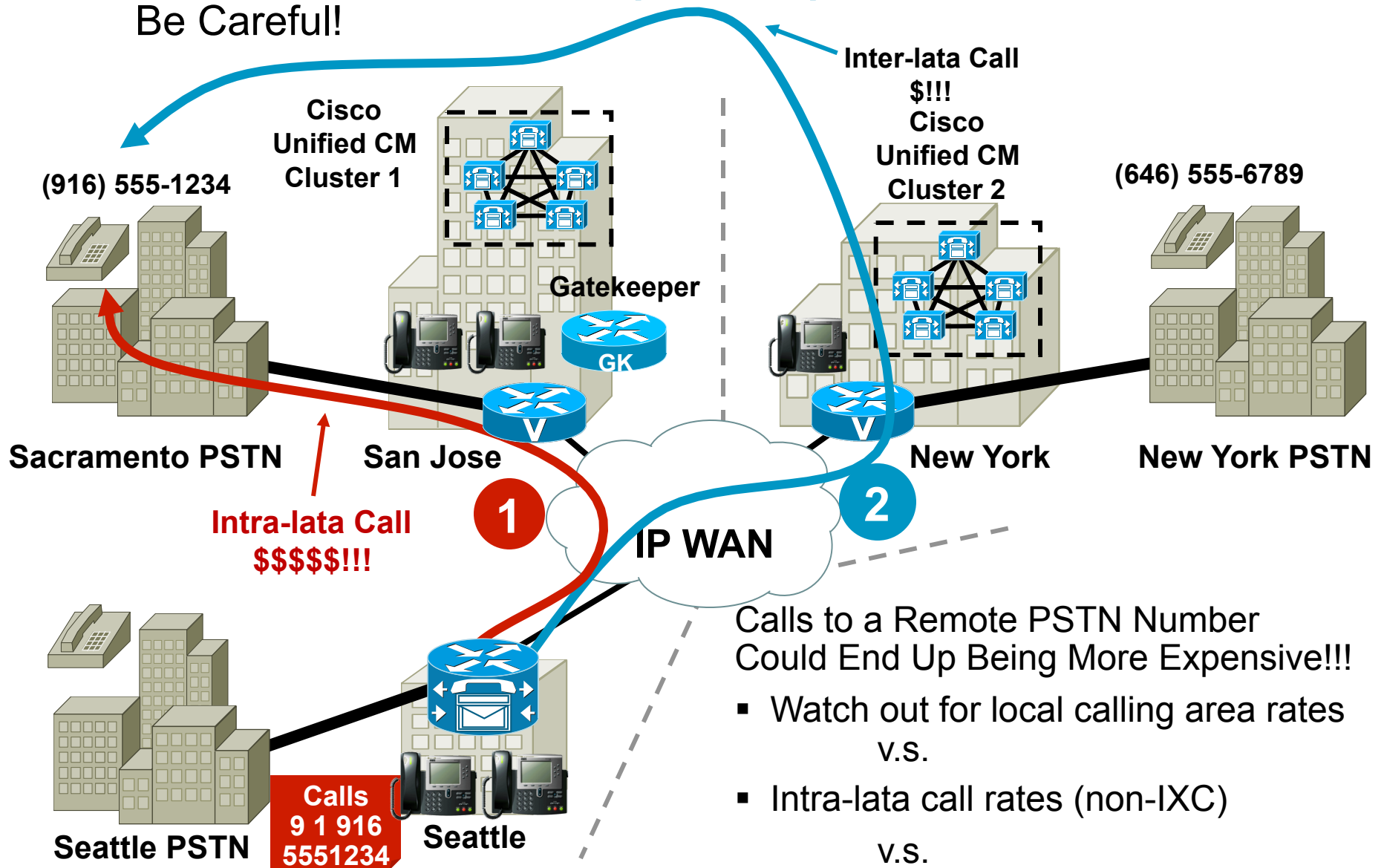


Calls to a Remote PSTN Number
Use the IP WAN and Terminate at the
GW Closest to the Destination

- 1 Within a Cisco Unified CM Cluster
- 2 Across Cisco Unified CM Clusters

Tail-End Hop-Off (TEHO)

Be Careful!

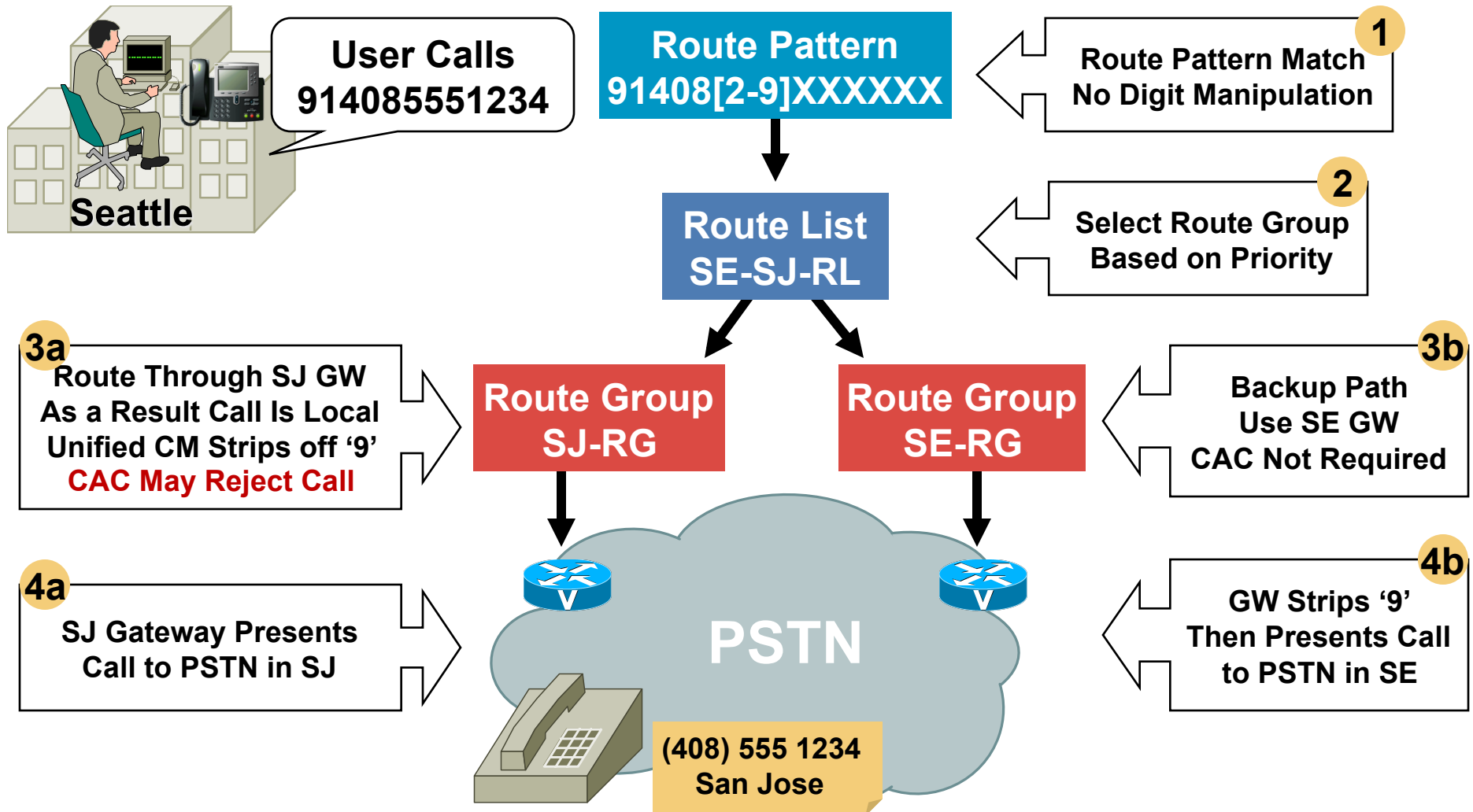


Calls to a Remote PSTN Number
Could End Up Being More Expensive!!!

- Watch out for local calling area rates v.s.
- Intra-lata call rates (non-IXC) v.s.
- Inter-lata call rates (IXC)

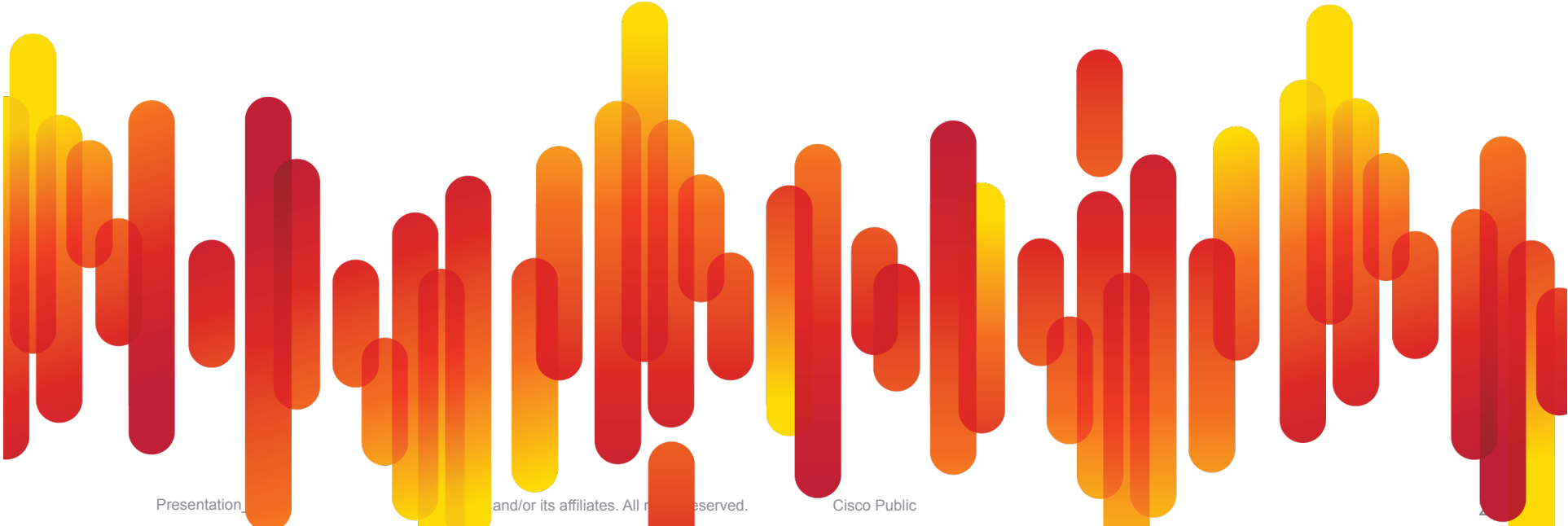
Tail-End Hop-Off (TEHO)

Intracluster: Seattle to San Jose



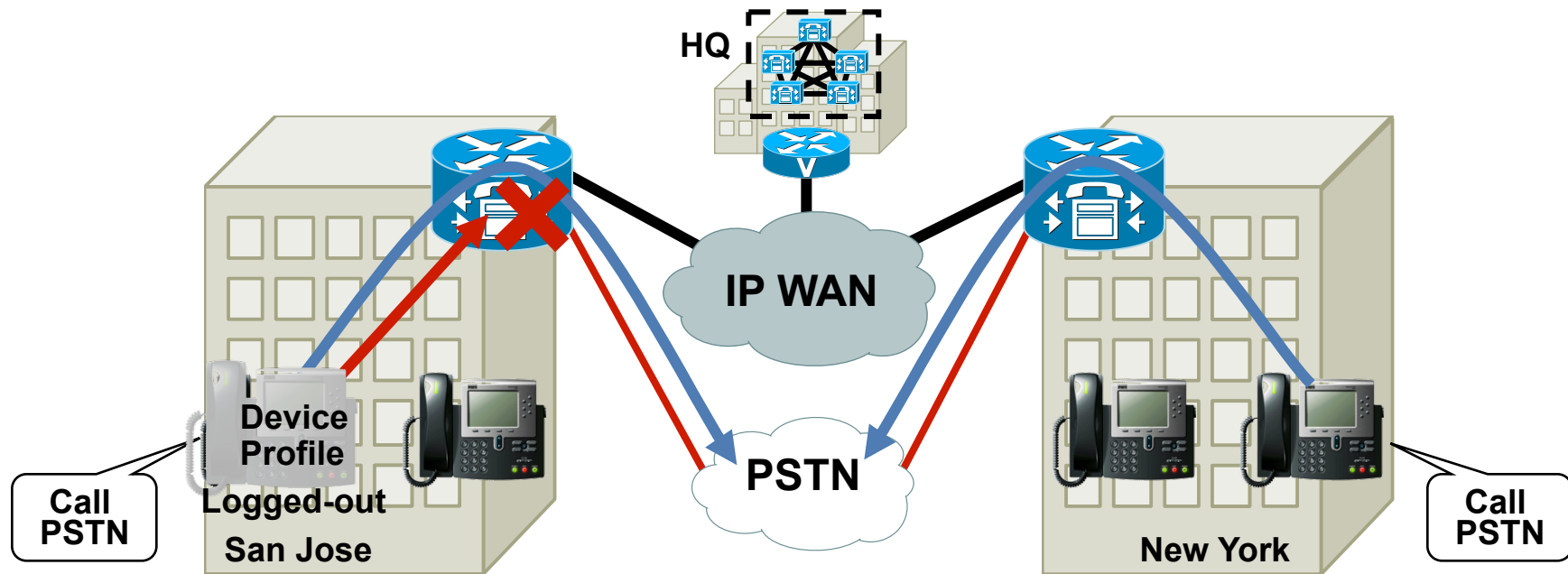
Appendix

Extension Mobility Considerations



Extension Mobility Considerations

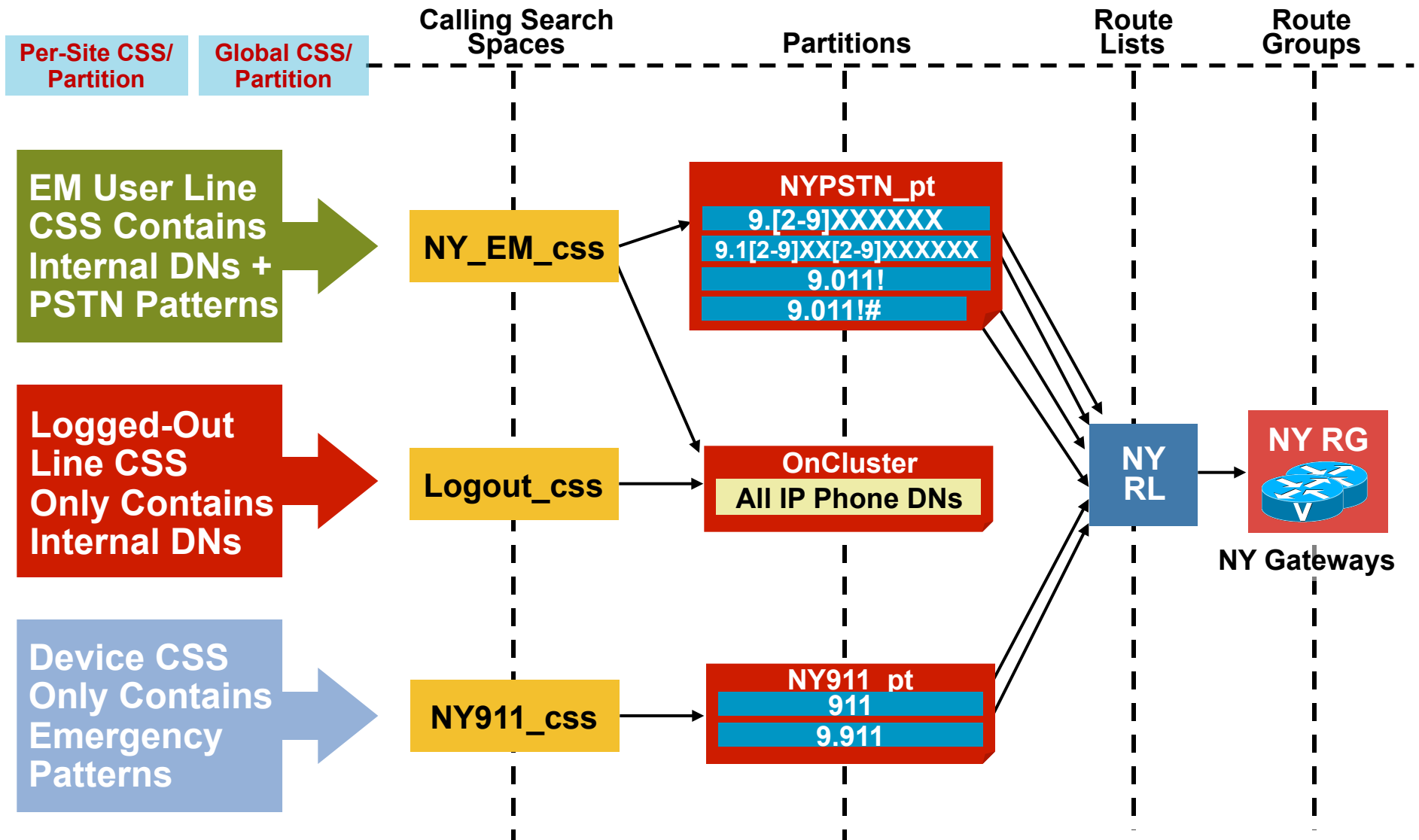
Requirements



- Allow users to log in at different sites with a single device profile
- Restrict PSTN calls when logged out
- Always route emergency calls via local gateway
- **Optional: route all PSTN calls via local gateway**

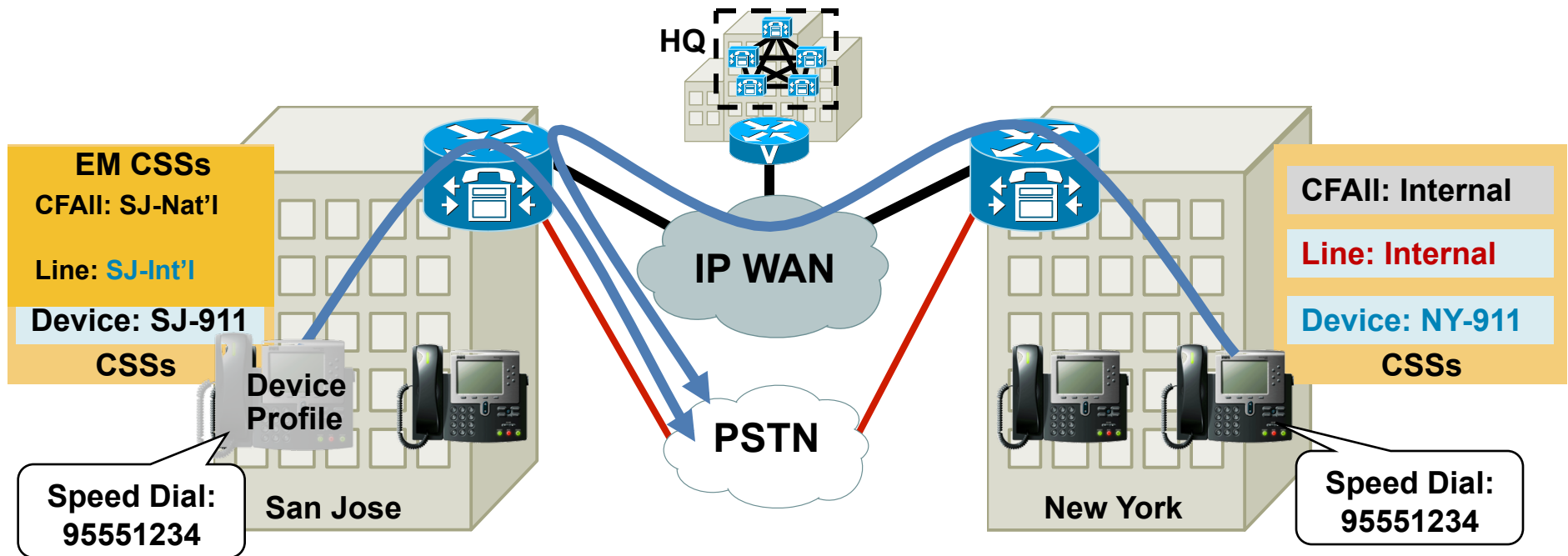
Extension Mobility Considerations

Traditional Dial Plan Approach



Extension Mobility Considerations

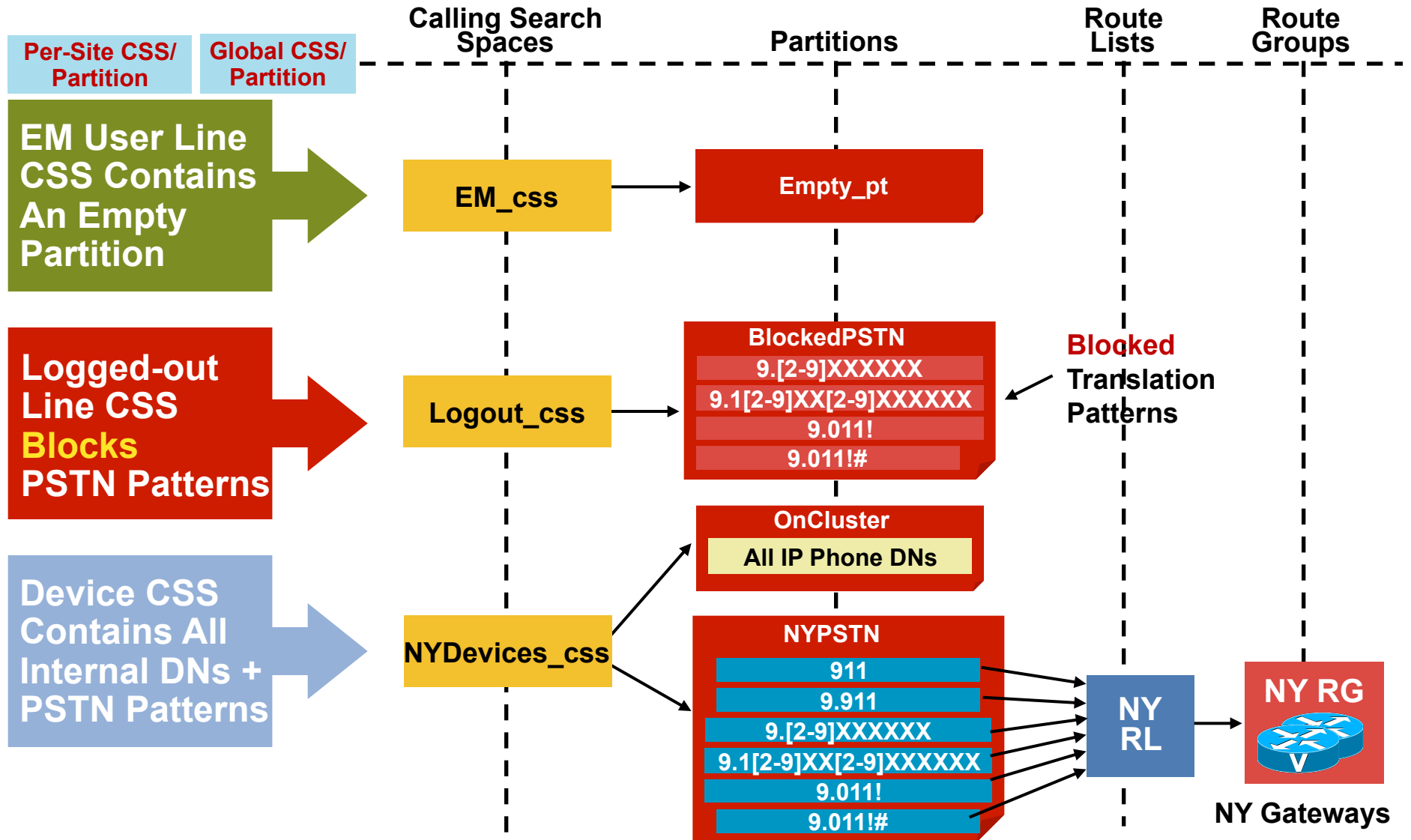
Traditional Dial Plan Approach: Behaviour



- Emergency calls routed via local gateway
- Other PSTN calls routed via **home** gateway
- User dialing habits and speed dials are automatically preserved

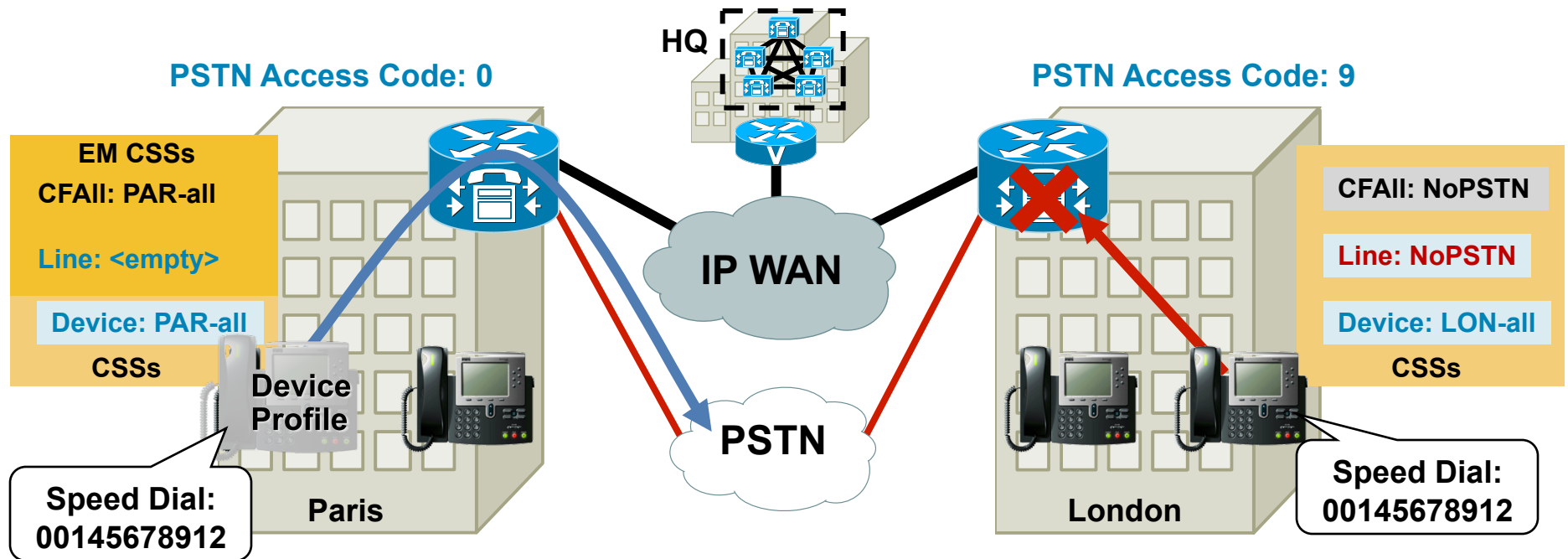
Extension Mobility Considerations

Line/Device Dial Plan Approach



Extension Mobility Considerations

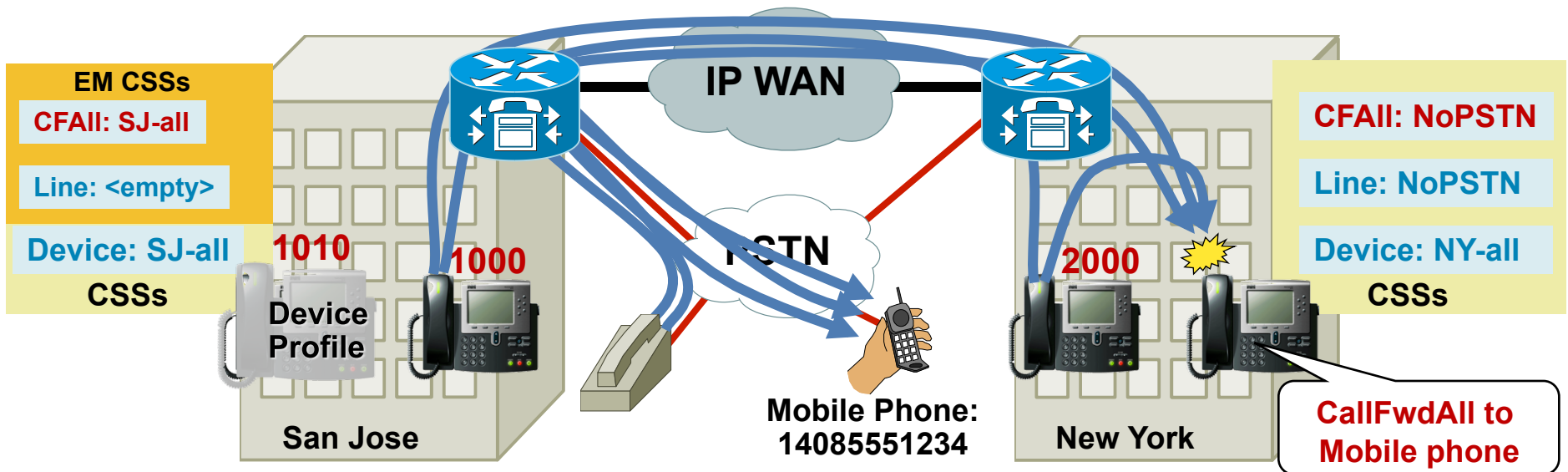
Line/Device Dial Plan Approach: Behaviour



- All PSTN calls are routed via local gateway
- User dialing habits and speed dials are not preserved across different dialing **domains**
- Forwarded calls are routed via **home gateway**

Extension Mobility Considerations

Line/Device Dial Plan Approach: Forwarded Calls

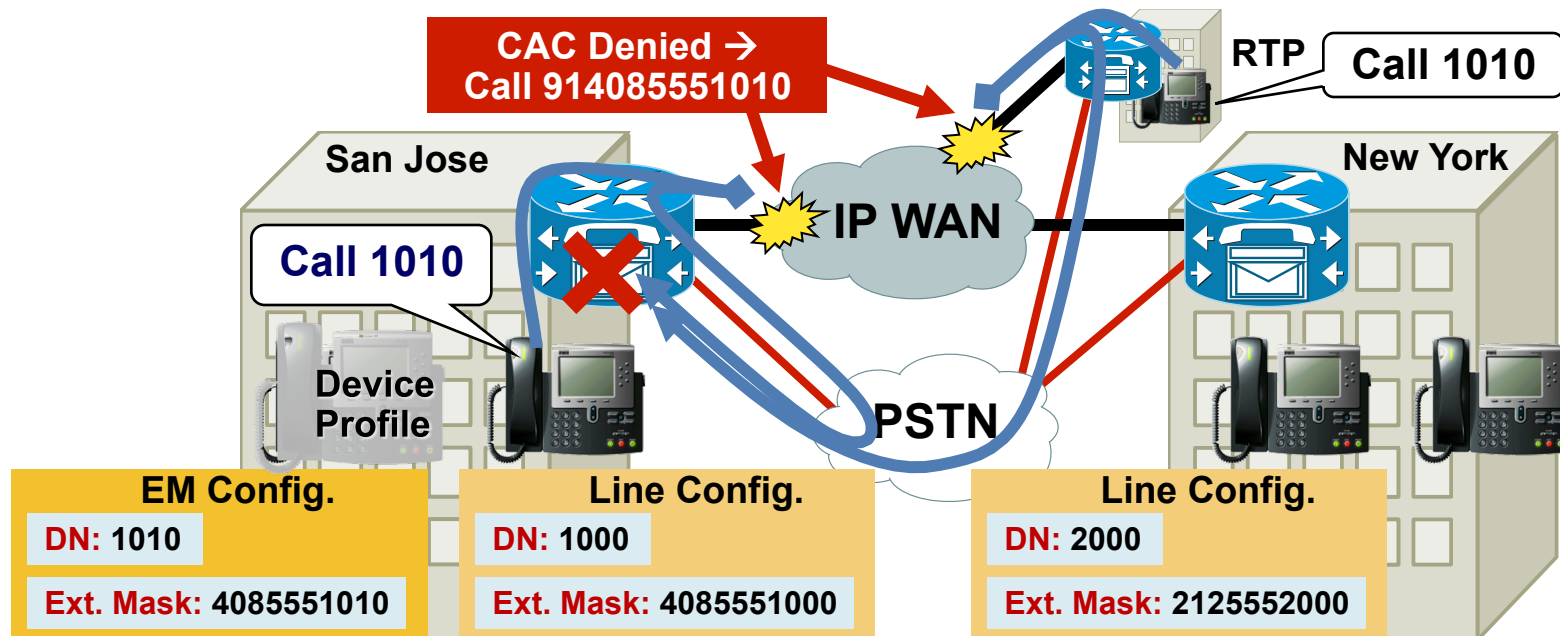


When a SJ User Logs in at NY Site and Forwards His Phone to a PSTN Number:

- Calls from SJ IP phones use SJ PSTN GW
- Calls from PSTN users get hairpinned at the SJ PSTN GW
- **Calls from NY IP phones cross the WAN and use SJ PSTN GW**

Extension Mobility Considerations

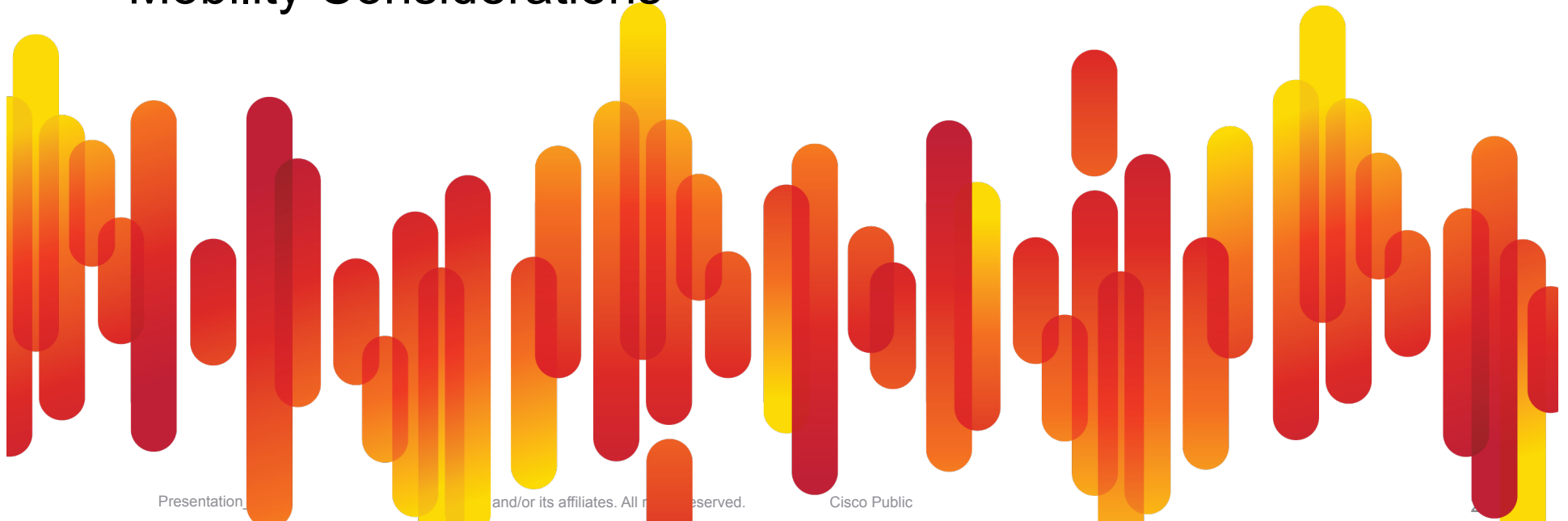
AAR Interaction



- AAR is inherently incompatible with extension mobility users moving across branch sites (regardless of approach)
- When extension mobility users log in at a different site, they cannot be reached via AAR from other sites (DIDs don't move!)
- Ensure that GW CSSs contain internal numbers only to prevent routing loops

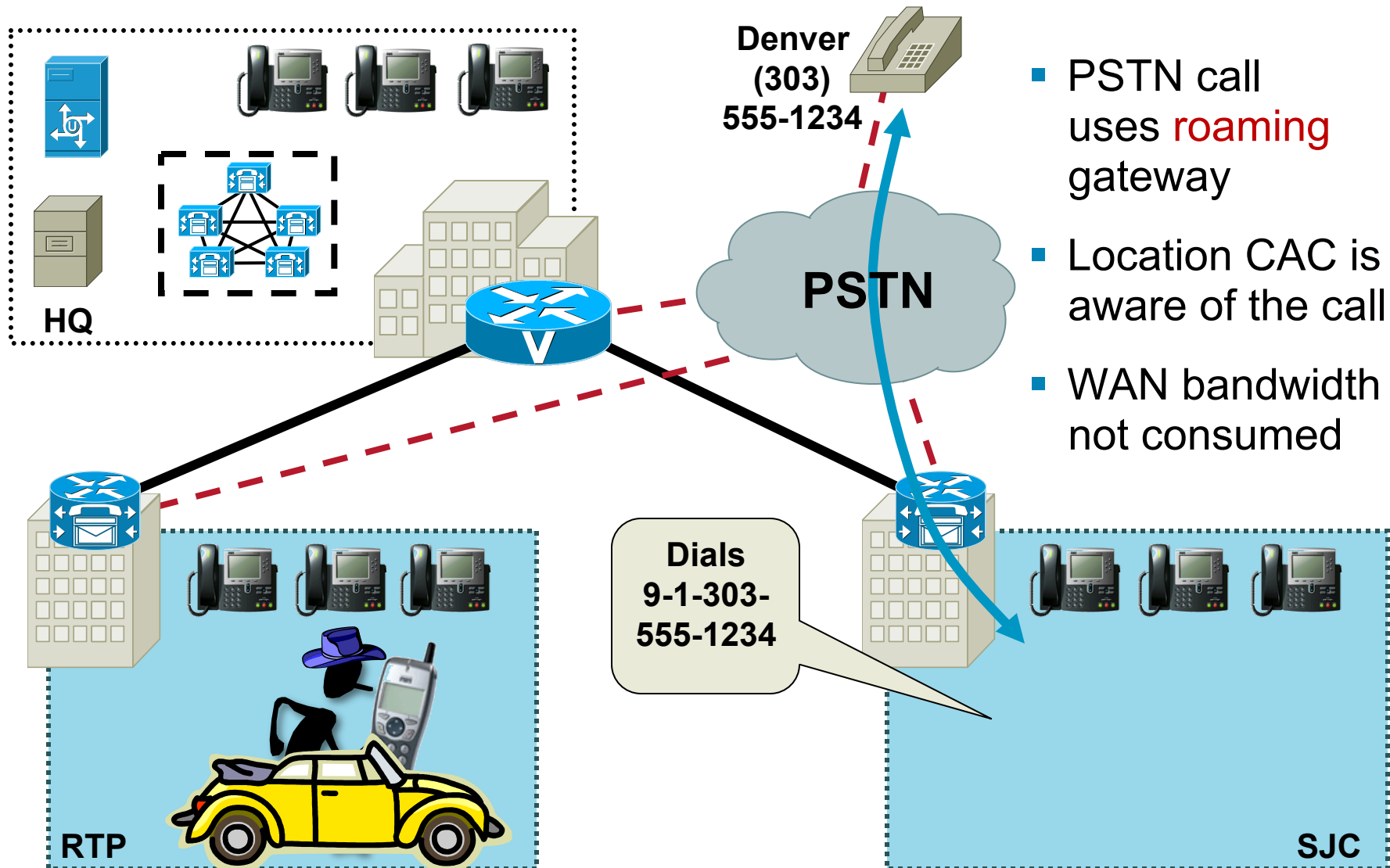
Appendix

Additional Device Mobility Considerations



Device Mobility Considerations

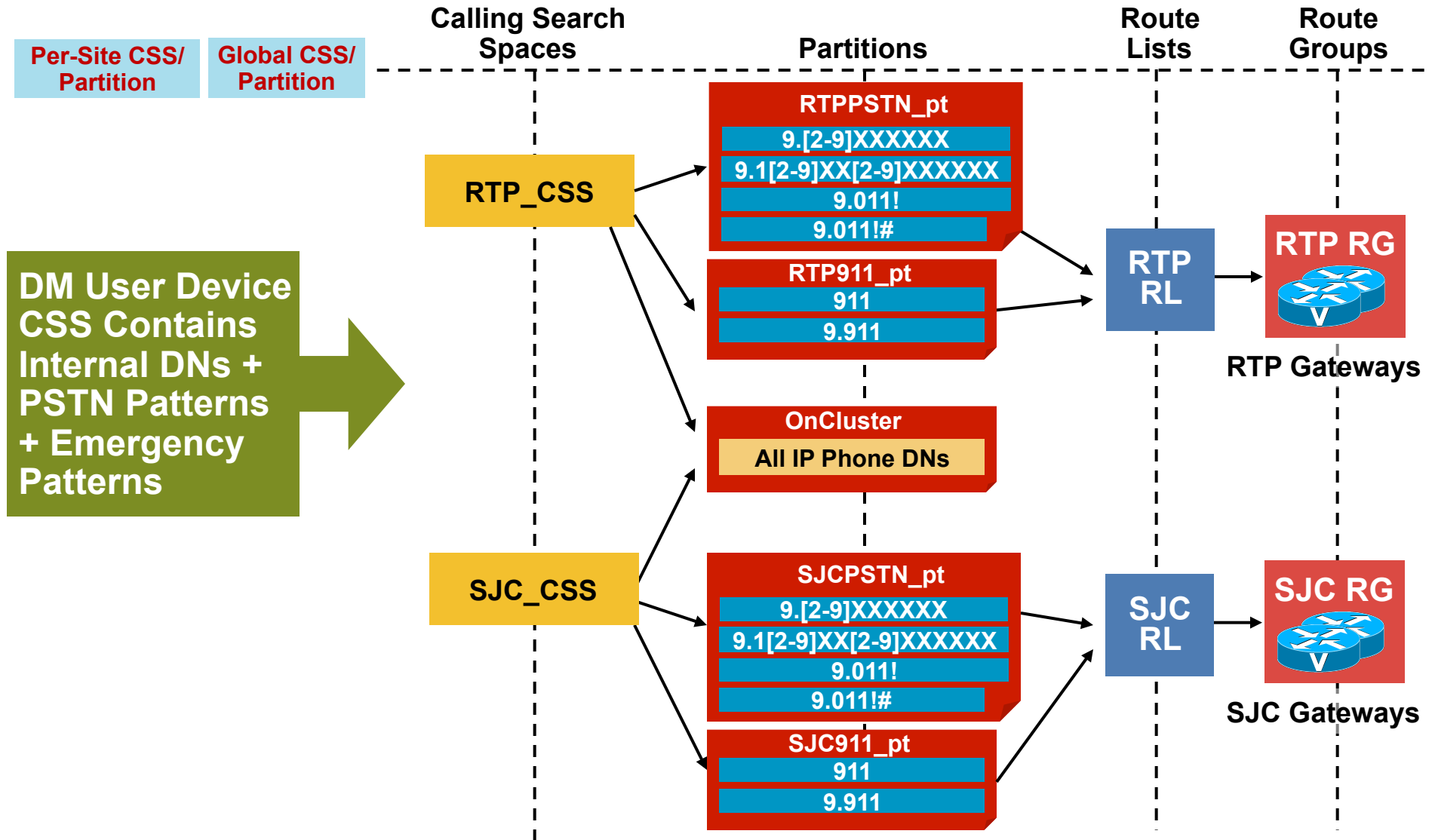
Requirements



- PSTN call uses **roaming gateway**
- Location CAC is aware of the call
- WAN bandwidth not consumed

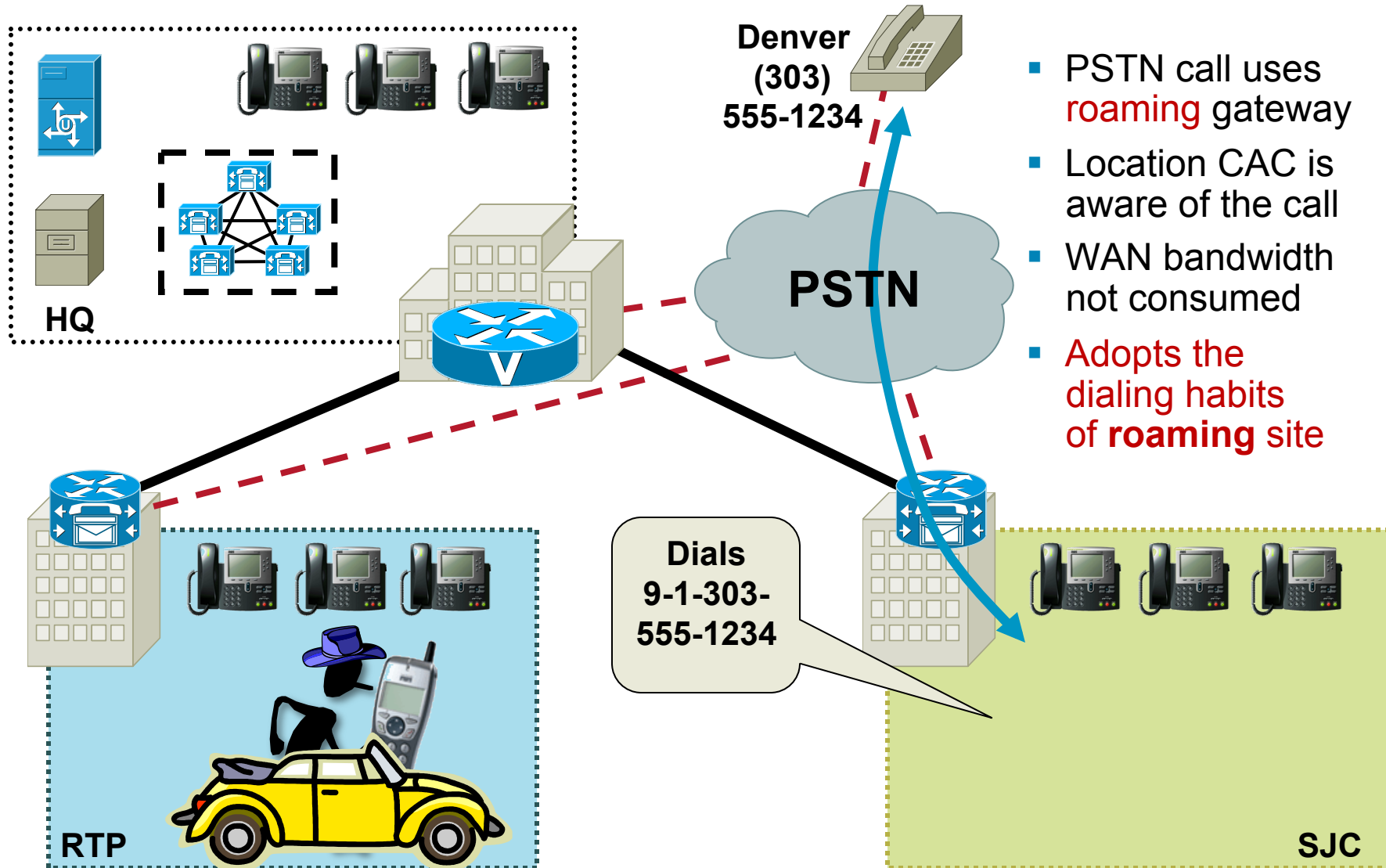
Device Mobility Considerations

Traditional Dial Plan Approach



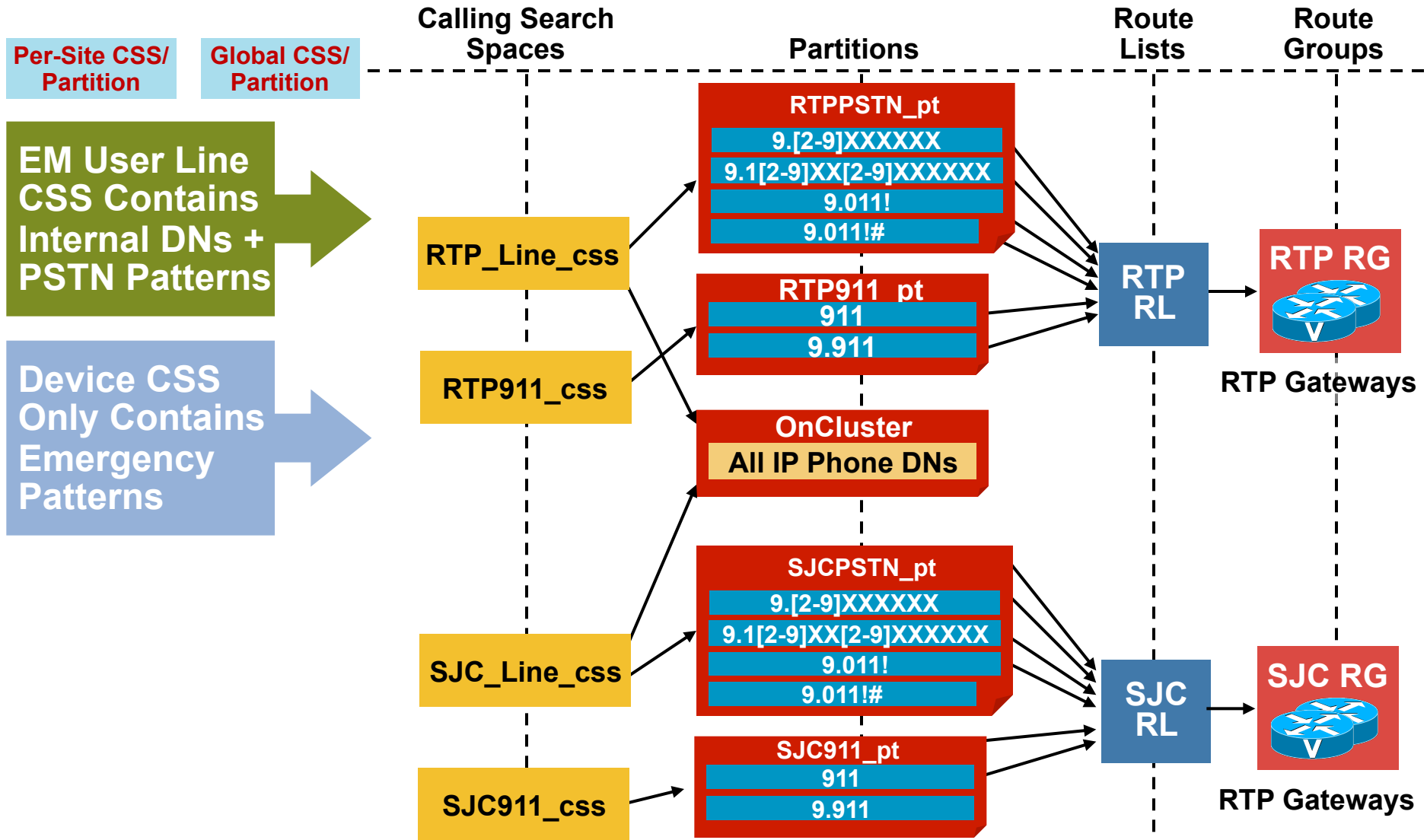
Device Mobility Considerations

Traditional Dial Plan Approach: Behaviour



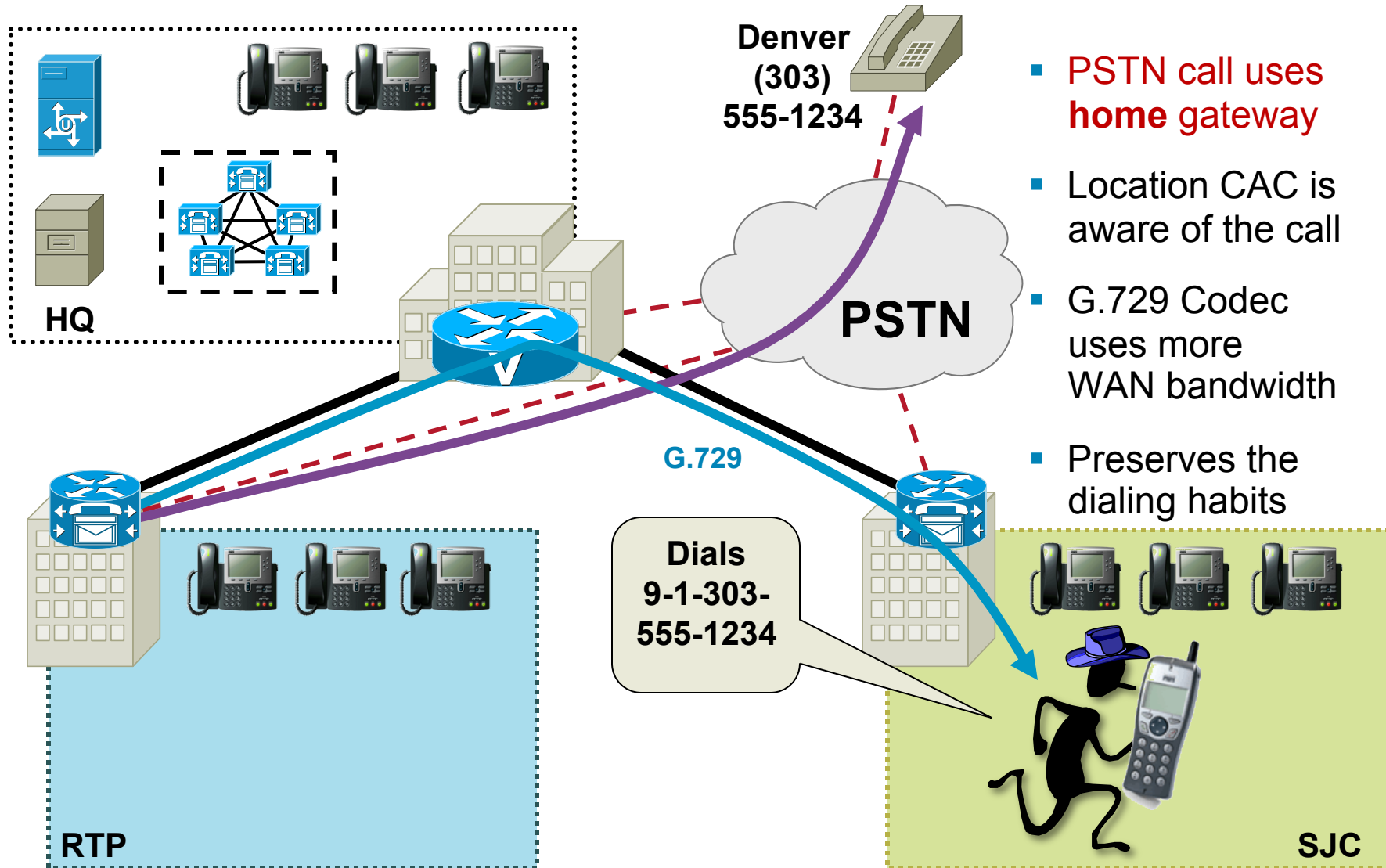
Device Mobility Considerations

Traditional Dial Plan Approach (EM Approach)



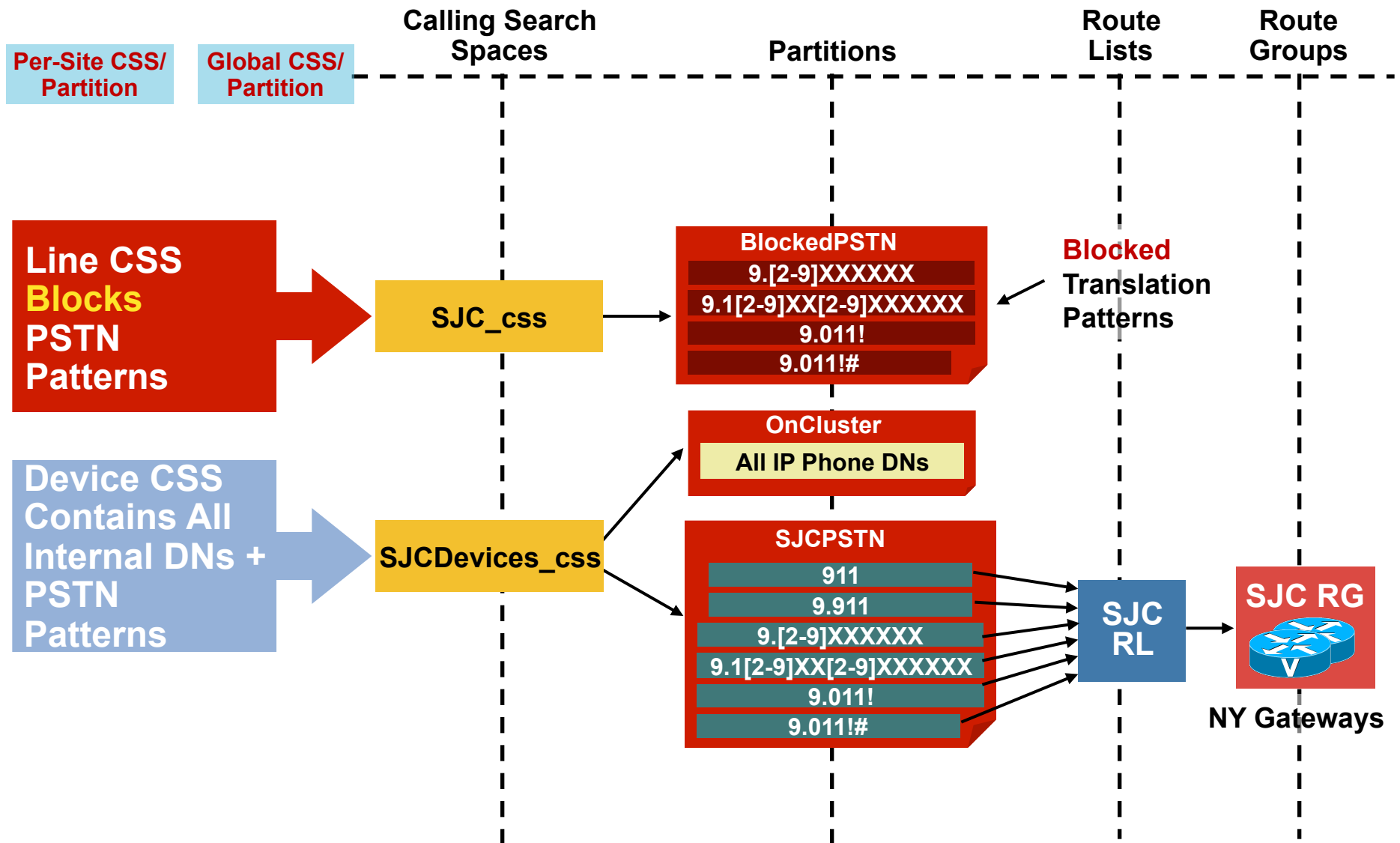
Device Mobility Considerations

Traditional Dial Plan (EM Approach): Behaviour



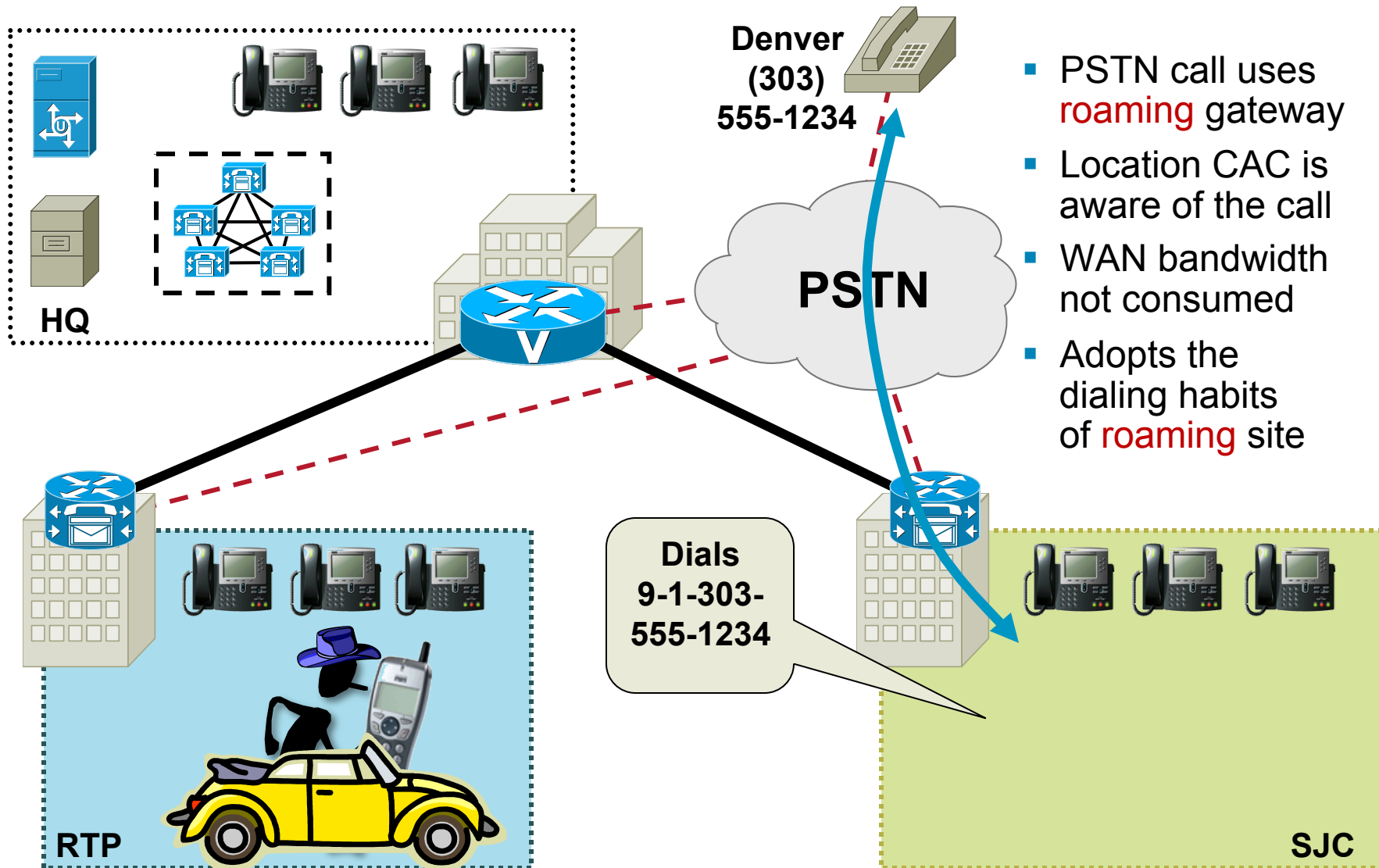
Device Mobility Considerations

Line/Device Dial-Plan Approach



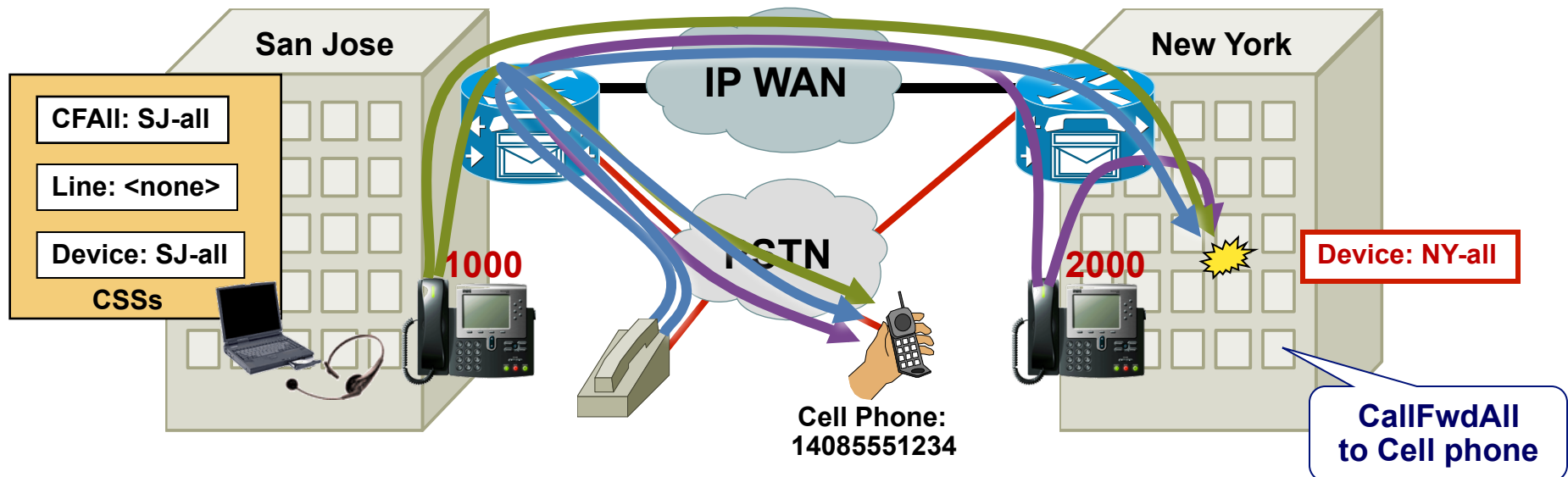
Device Mobility Considerations

Line/Device Dial-Plan Approach: Behaviour



Device Mobility Consideration

Line/Device Dial Plan Approach: Forwarded Calls



- When a SJ user moves to NY site and forwards his phone to a PSTN number:

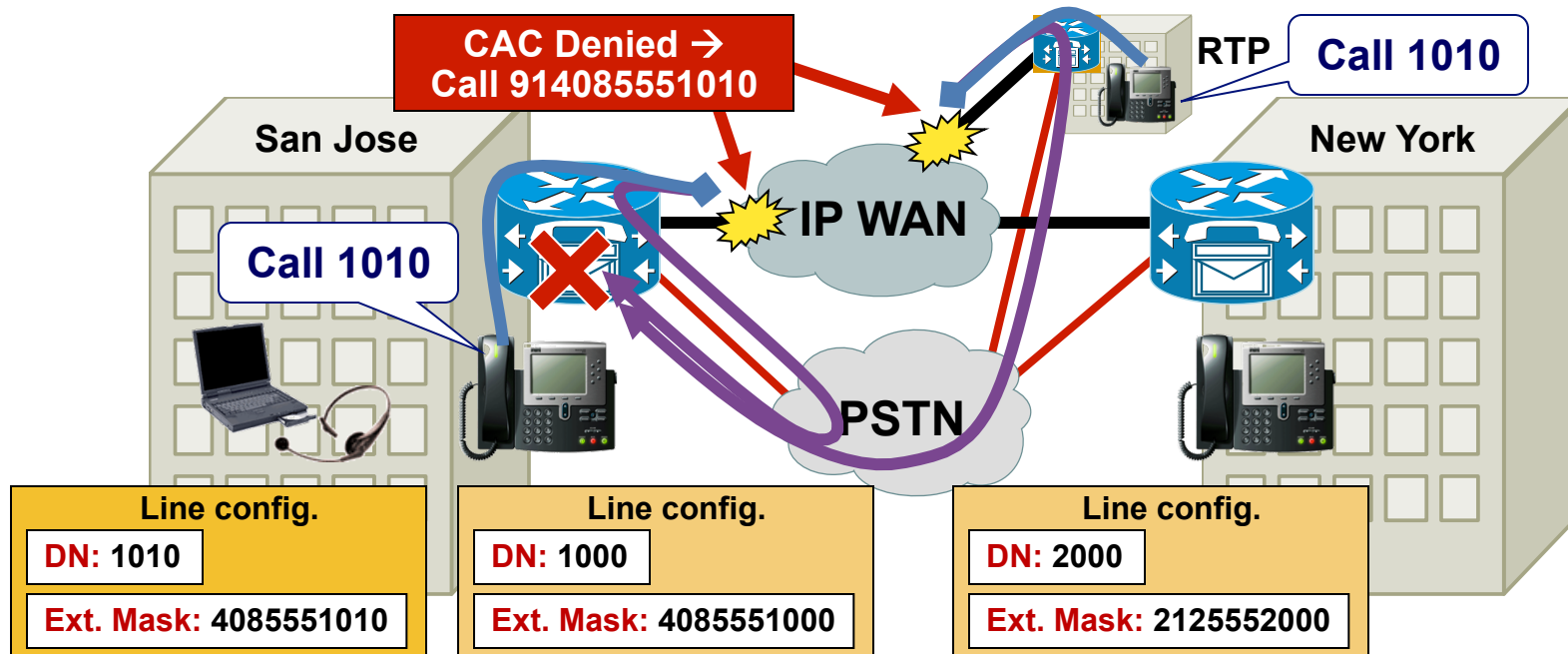
Calls from SJ IP phones use SJ PSTN GW

Calls from PSTN users get hairpinned at the SJ PSTN GW

Calls from NY IP phones cross the WAN and use SJ PSTN GW

Device Mobility Considerations

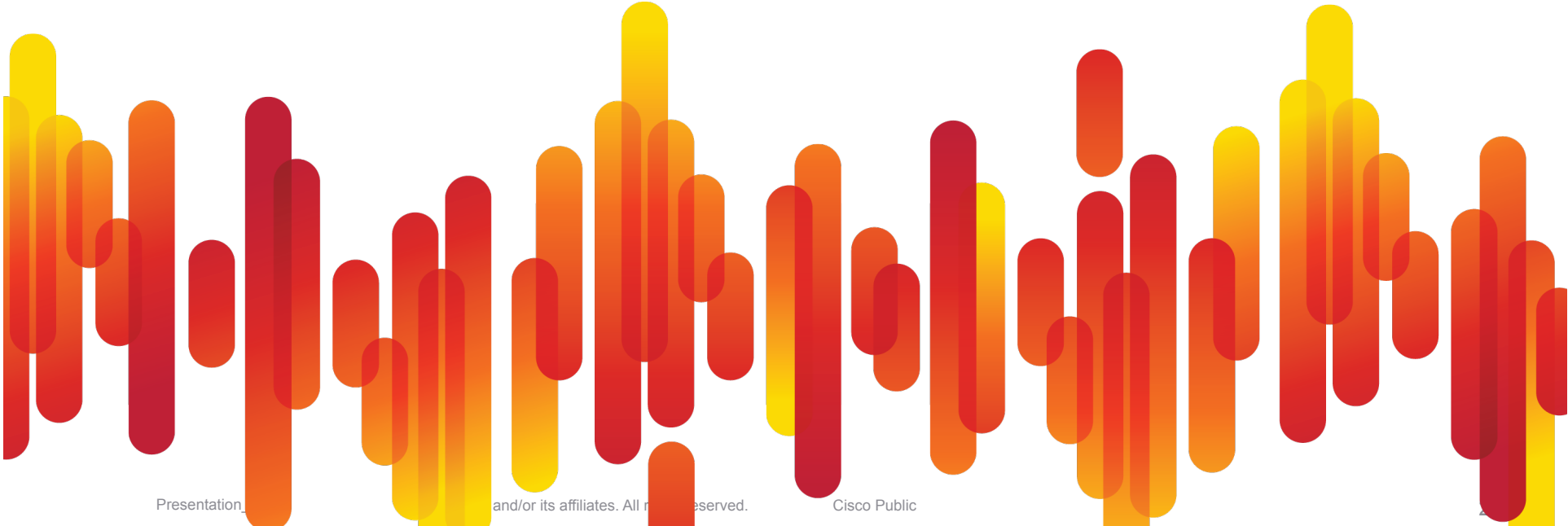
AAR Interactions



- AAR is inherently incompatible with device mobility across sites (same as for extension mobility across sites)
- When DM users move to different site, they cannot be reached via AAR from other sites (DIDs don't move!)
- Ensure that GW CSSs contain internal numbers only to prevent routing loops

Appendix

VoPTSN



What Is Voice over the PSTN (VoPSTN)?

- A variation on the centralised call processing deployment model, where all intersite voice goes over the PSTN (not the WAN)
- We are not **promoting it**: merely setting requirements and expectations
- There are several, fundamental limitations
- Relies on AAR configuration

VoPSTN Using AAR

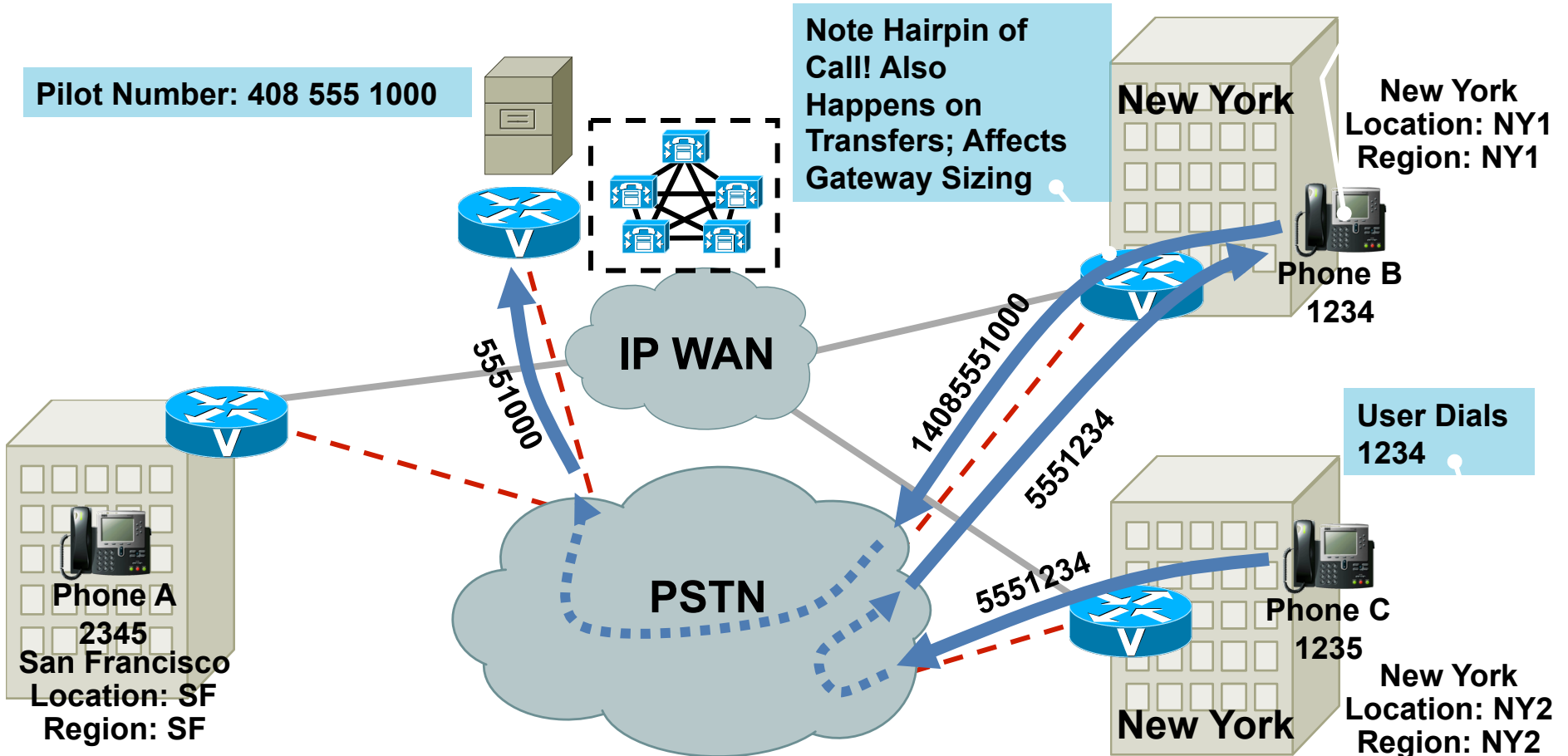
Centralised Voicemail

Note: RDNIS Required End to End for Automated Mail Box Selection!

CFB, CFNA to a PSTN Number (e.g., 1 408 555 1000)

Pilot Number: 408 555 1000

Note Hairpin of Call! Also Happens on Transfers; Affects Gateway Sizing



VoPSTN Using AAR

Summary

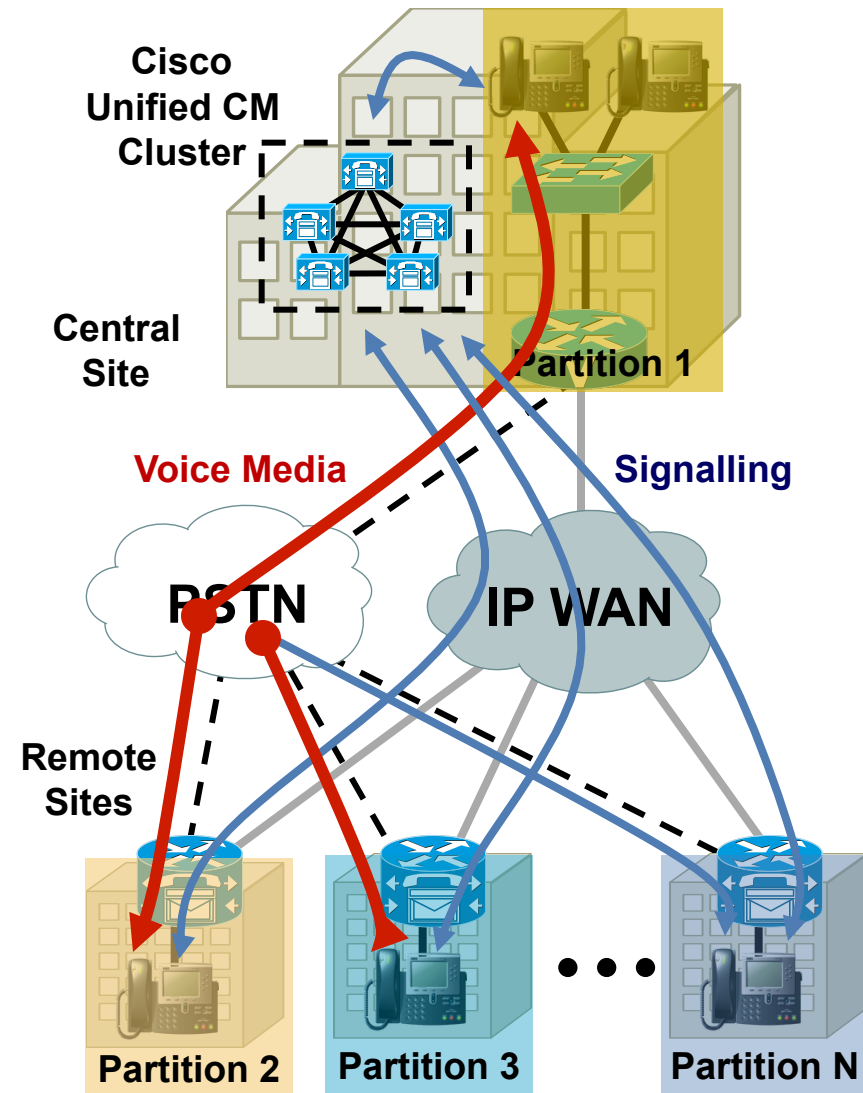
- Only accommodates SCCP destinations
- RDNIS required for centralised VMAIL
- Extension mobility not possible
- No difference between PSTN and Interbranch calls (one ring type)
- Two CDR records for every call (minimum); more if CallFwd invoked
- All intersite calls display network congestion, rerouting
- No shared-line support across branches
- All destinations must be DID
- Does not work during WAN interruption
- No centralised MoH
- No centralised conferencing
- All transferred calls are hairpinned
- All calls forwarded to outside locations are hairpinned
- If you tailor the WAN for signalling only, no attendant console in remote sites, due to directory access BW
- QoS is **required** on the WAN
- High availability is required on the WAN: SRST does not make up for a bad link, only a dead one

VoPSTN Using Dial Plan

Key Points

- Bullet 3 and graphic: changed UC Manager to Unified CM

- DNs at each site are placed in different partitions
- Relies on PSTN route patterns to call other sites
- For Cisco Unified CM, all calls are **external** calls
- No **on-net** features across sites (e.g., CallBack)
- No easy migration to full-blown VoIP
- **Note:** abbreviated dialing possible with translation rules on branch GWs



Design Guidelines Agenda

- 7.0 and 7.1 Updates
- 8.0 updates
- **Building Classes of Service**
- Multisite Deployments
- Mobility Considerations

Building Classes of Service

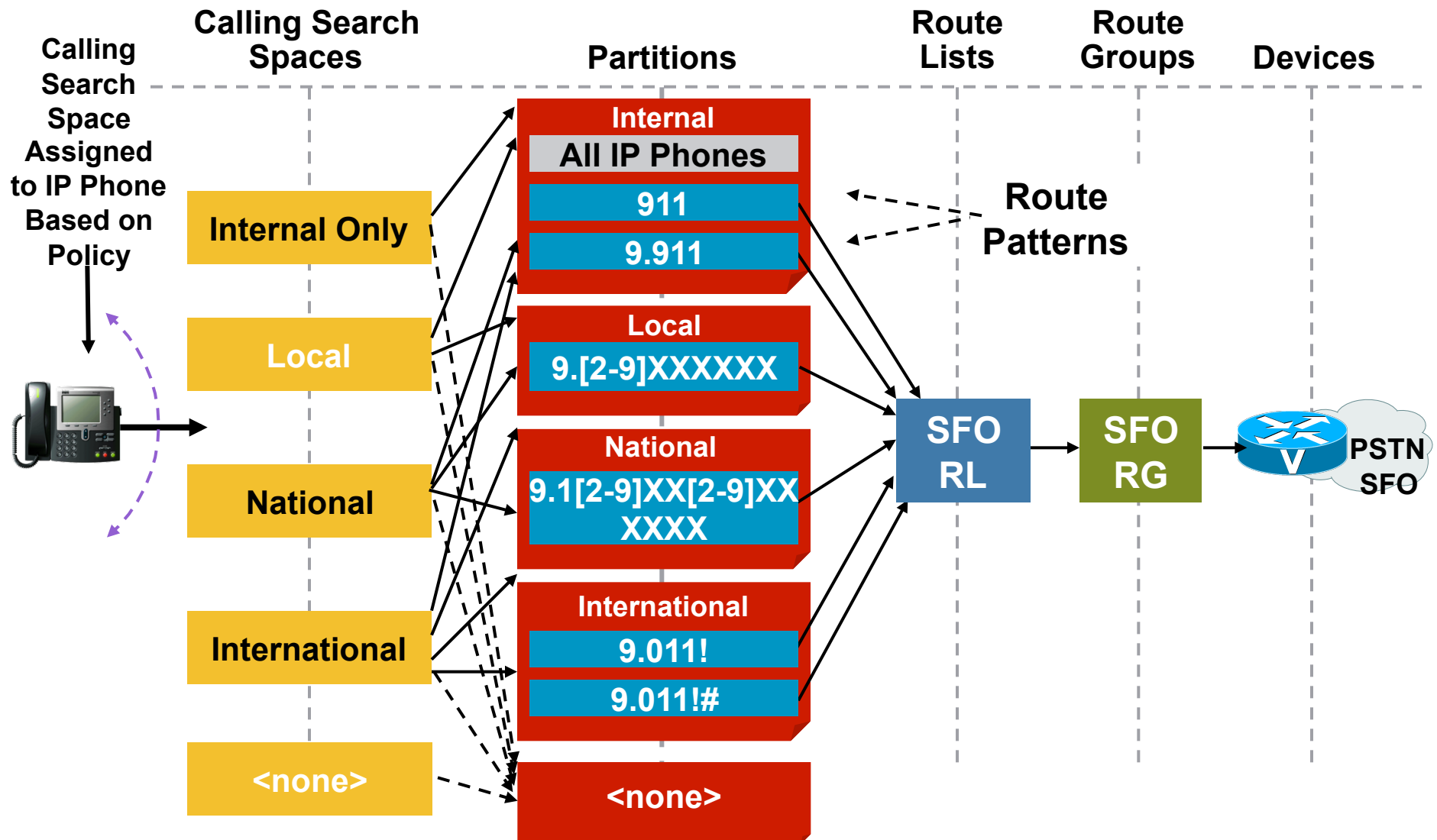
- When we need to restrict the calls a user can place, typically based on the call type and/or the costs associated with the call.
- There are some boundaries that are easy to enforce:
 - No international calls: block calls made to 9011!, or to +[^1]; the generic idea is to block calls made to a country code other than that of the originator's location.
 - 900 numbers, as the pattern is easily identifiable, and is independent of the location of the caller
- There are other boundaries more intricate to enforce: Intra-country area calls are tricky!!! No one-size-fits-all solution there!!!
 - In the NANP, local calls can be 7 digits. So if a user is to be restricted to local calls, then we can restrict calls to those matching 7 digit patterns (e.g.: 9.[2-9]XX XXXX)
 - BUT
 - This is site-specific: you need to enforce this type of restriction on a local level: "local" calls in San Francisco may be 7 digits, but in Ottawa, they are ten digit calls.
 - AND
 - What if someone dials the fully qualified number of a local destination?
 - From San Francisco, you place a call from the missed calls list to +14155551234. It does not look like 95551234, but the intent is the same?
 - From San Francisco, you place a call from a contact in your smartphone (dual mode), on the WiFi network
 - From San Francisco, you click to call on a number listed in a web page

Building Classes of Service

- The tactical challenge of implementing intra-country (e.g.: local, long distance) classes of service in the *context of a globalized dial plan* is this: you cannot rely on a fixed, recognizable **form** of the called number to apply policies IF AT THE SAME TIME you want to allow calls to the E.164 or the +E.164 form of the same numbers.
- Example:
 - In San Francisco, you restrict a user to 9.[2-9]XX XXXX calls.
 - The user then receives a call from a customer. The Received calls list is populated with +14155551234.
 - The user presses the “dial” key to reach +14155551234; it will not match 9.[2-9]XXXXXX. You can let it match +1!, but then you allow call back calls to any destination in country code 1. You can then restrict calls to +1415 [2-9]XXXXXX.
 - BUT
 - Not all of the 415 area code is part of the local calling area...
- Bottom line: restricting calls placed to fully qualified destinations is trickier than relying on the *form* of the number (e.g.: 7 digit, 10 digit, 1 plus 10 digits)...

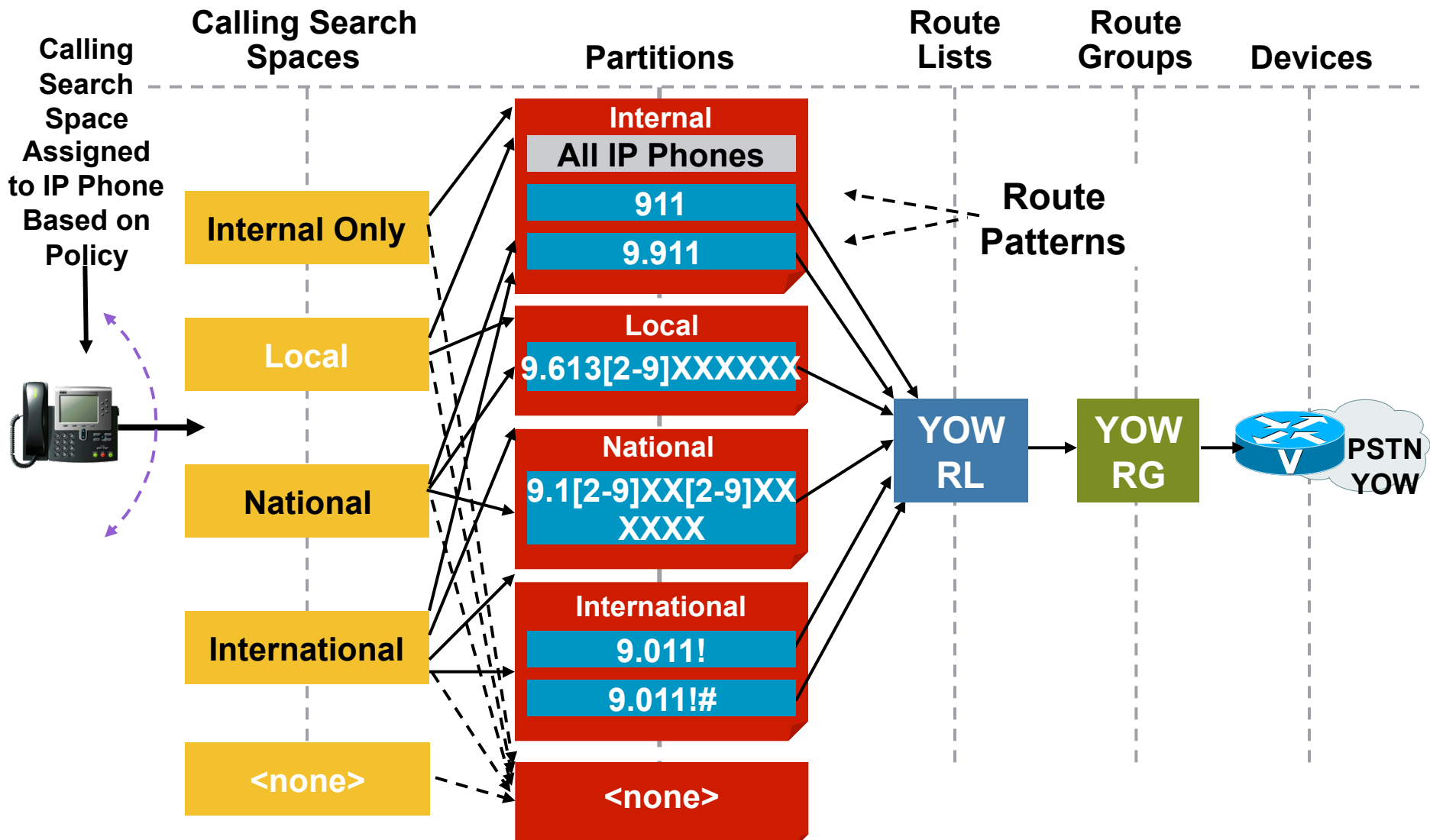
Traditional CSS Approach

Example — San Francisco – note 7 digit local call form



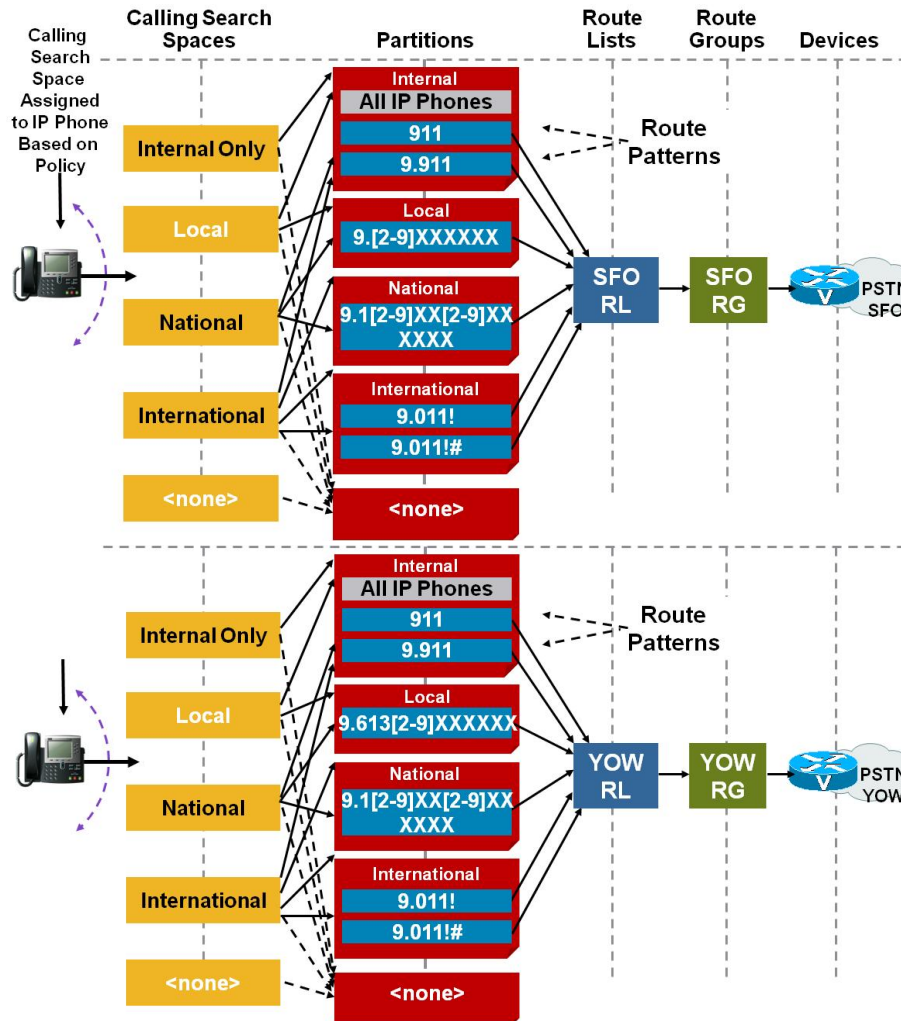
Traditional CSS Approach

Example — Ottawa — note 10 digit local call form



Traditional CSS Approach

Example of Composite View— NANP 2 site system



You need to have site-specific off-cluster route patterns, such that route selection is based on the calling device's location.

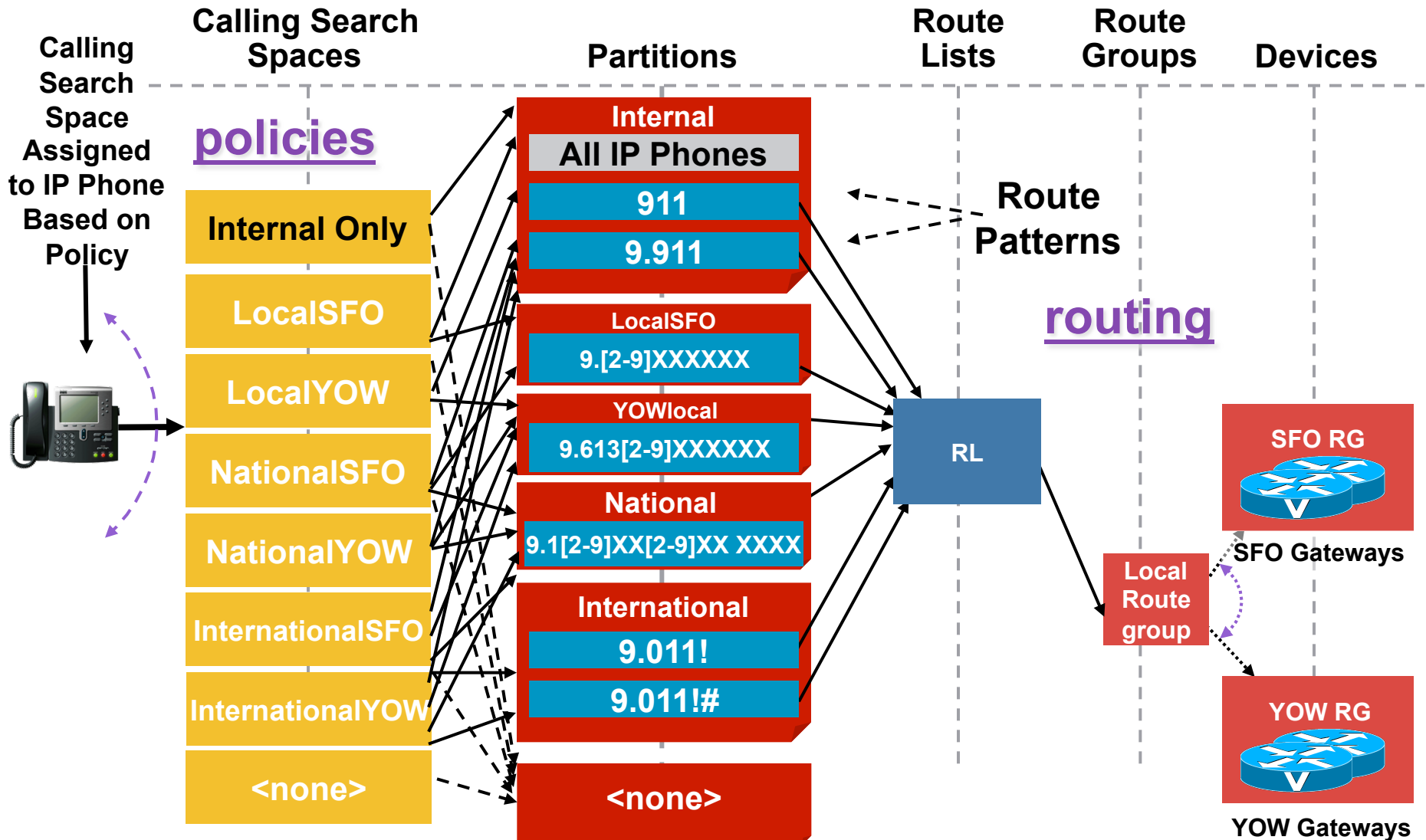
Traditional CSS Approach

Example of Composite View— NANP 2 site system

- Can we do better than that now?
- How about the Local Route Group?
- The local route group allows us to create site-specificity of off-cluster routing
- It does NOTHING to alleviate the site-specificity of call policies.

New (old???) CSS Approach

Example — San Francisco AND Ottawa

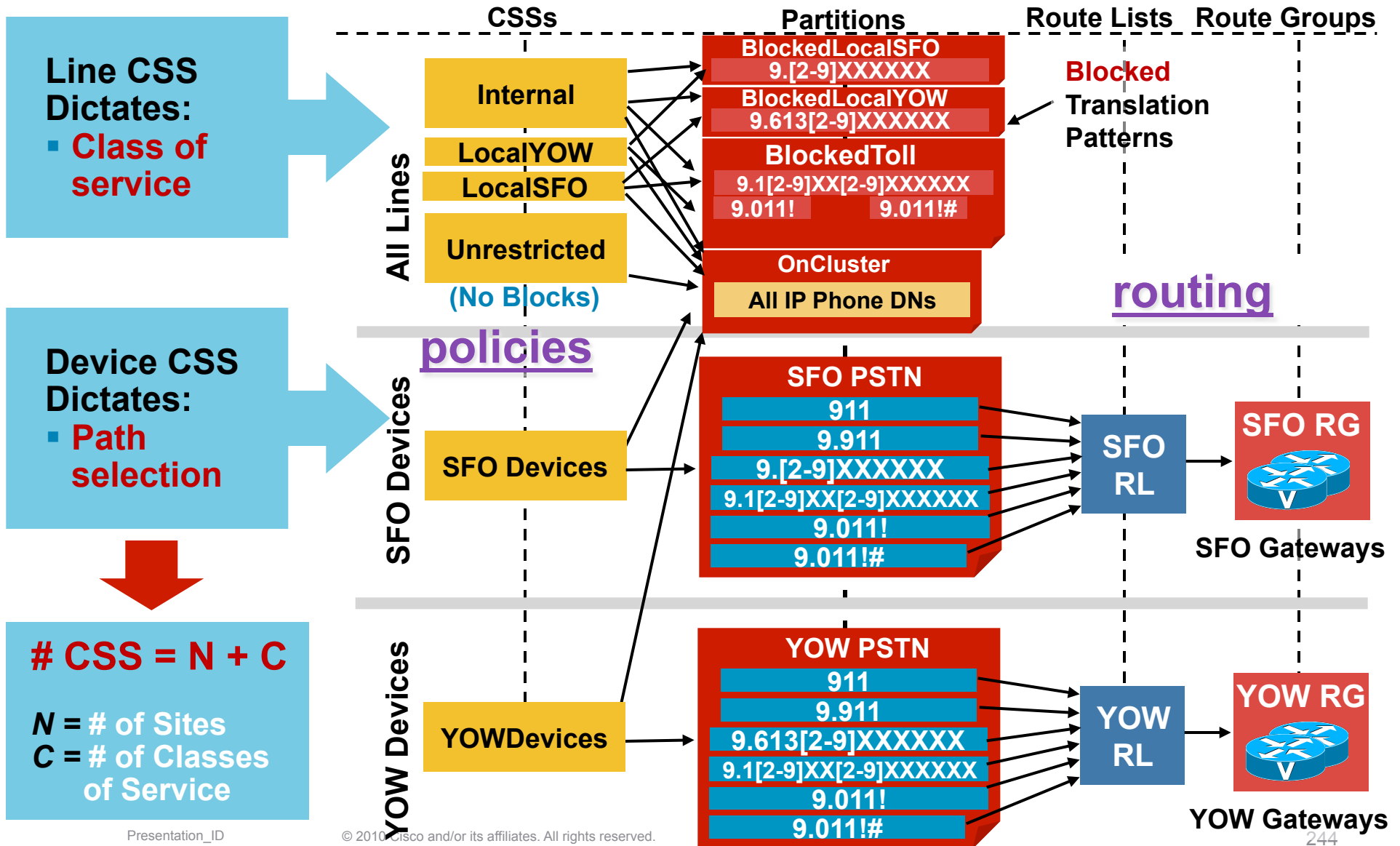


Line/Device CSS Approach

- The “original” way of efficiently combining line-based policy with device-based site-specificity of the routing.
- It offers (still) a great reduction on the quantity of Calling Search Spaces needed to implement site-specific routing of calls, when compared to the pre-Local Route Group traditional CSS approach.
- The site-specificity of the routing of calls was (is) compatible with the device mobility feature, where the phone will inherit its device calling search space from the device pool, which itself is inherited from the IP subnet it is detected to be in.
- BUT
- The policies applied on the line are not changing as the device moves around
 - When not trying to restrict calls in a site-specific manner, that is ok.
 - BUT
 - If you want a user to be restricted to local calls as they move through site boundaries, then you need to inherit a site-specific policy.
- NOTE: for clarity, the following example does not normalize user input using translation patterns, nor does it include +; all call matching is done on routed patterns. An actual ***implementation*** would be need normalization translation patterns (see earlier TUI examples earlier in this presentation)

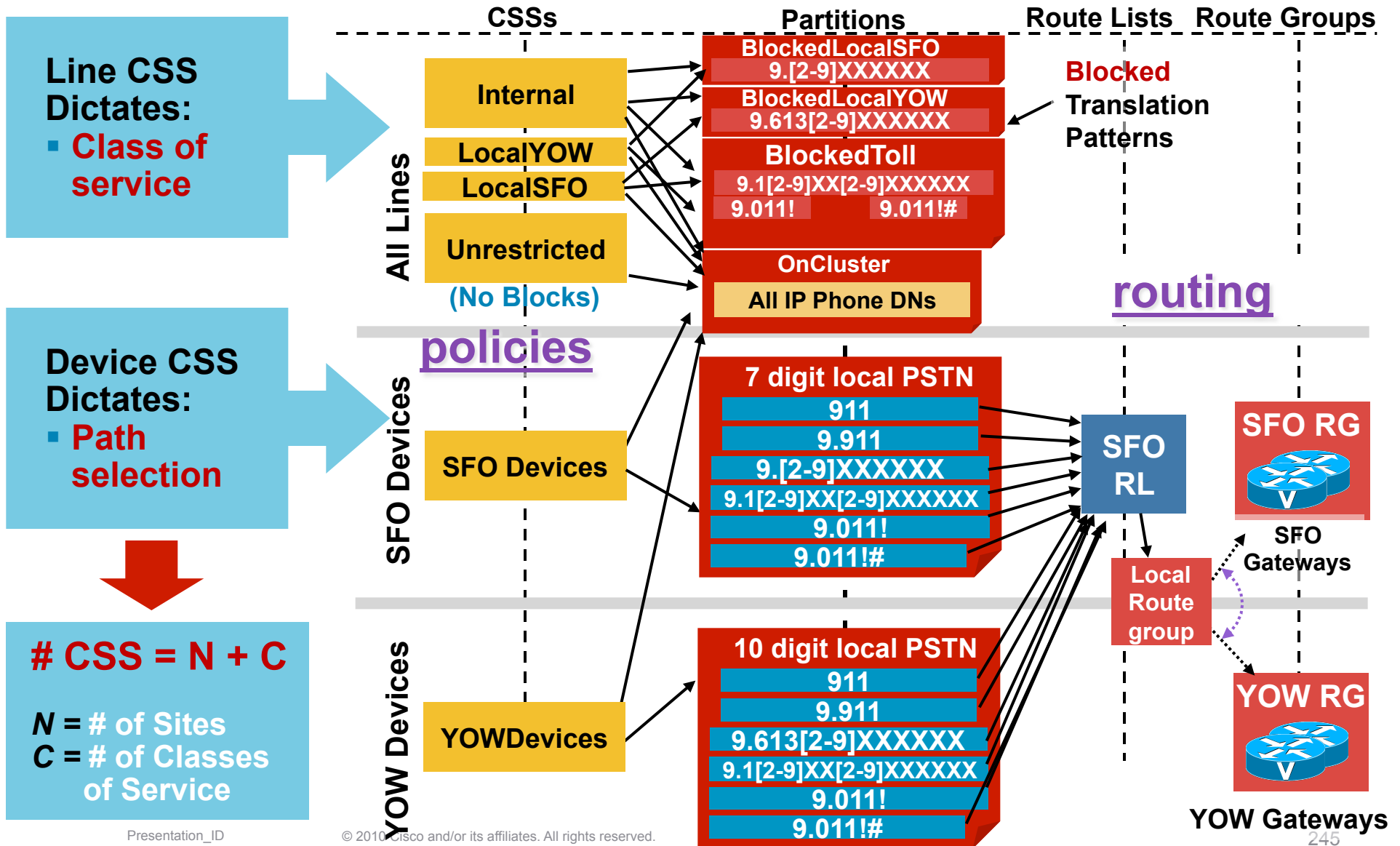
The Line/Device CSS Approach

Scalability for Centralised Deployments



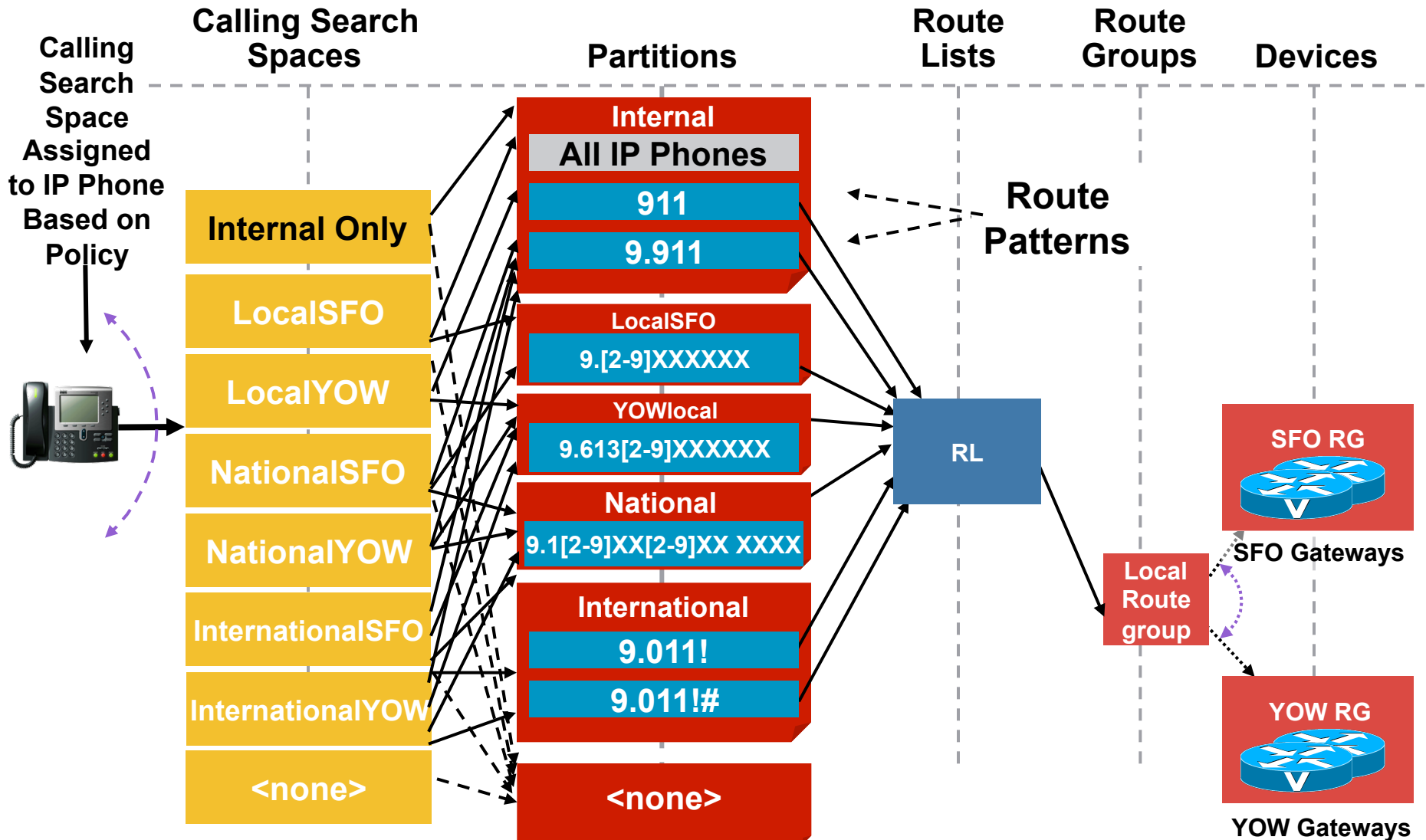
The Line/Device CSS Approach

Scalability for Centralised Deployments



New (old???) CSS Approach

Example — San Francisco AND Ottawa



Building Classes of Service

Luc's Dial Plan: How to prevent toll calls?

