



**Cisco Networkers**  
**2008**

# Advanced Dial Plan Design

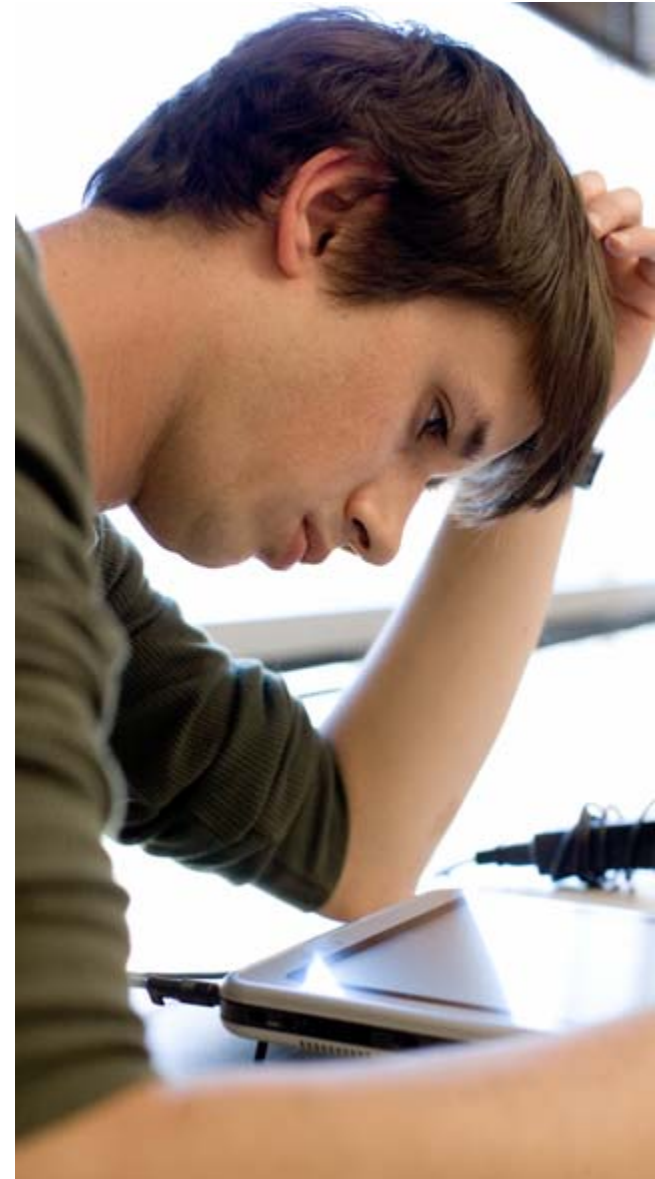


**BRKUCT-3007**

**Luc Bouchard**

# 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, PBX's have hundreds of sites**)
- To explore the design and implementation possibilities of Cisco's IP telephony system
  - Design based on Cisco Unified Communications Manager 4.X, 5.X and 6.X
- Aspects we will cover:
  - Design guidelines (**Classes of service, multisite deployments, extension mobility...**)
  - Integration of multiple UCMs in a single system (**e.g. inter-UCM call routing, device mobility**)



# Overall Agenda

- Planning Considerations



- Design Guidelines



- Conclusions



# Planning Considerations



# 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 "9" or "0" 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



# Planning Considerations

## Uniform Dial Plans Are Simple

Q: Could this system use a **uniform** three digit dial plan?

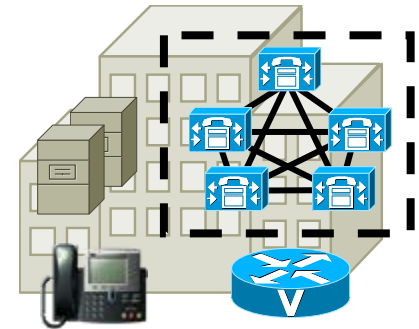
A: No! Chicago and Dallas' DID ranges overlap in the last three digits

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

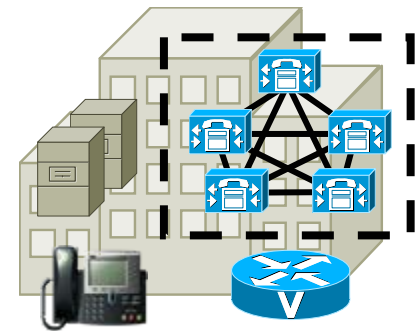
## Uniform Dial Plans Are Simple (2)

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 9 or 0, 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 (3)

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

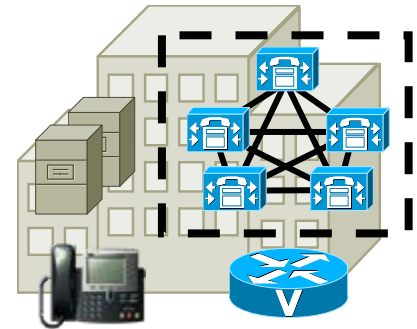
A: No! Because calls to New York (e.g. 57540) will sometimes overlap with calls to Hawaii's phones e.g. 575403), forcing the interdigit timeout to occur before the call is routed (and a few other reasons: can you spot them?)

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 9!



**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

## Uniform Dial Plans Are Simple (or So We Hoped)

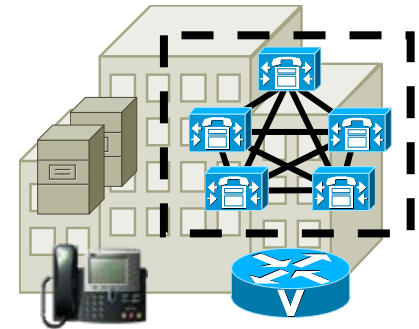
Q: Eight digits?

A: Ok for now: but you'll never open an office in Raleigh (area code 919)

Q: Nine digits? Oops. Forget about it!  
That 0 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 0 for operator, 9 for outside line, and 8 for intersite calls?

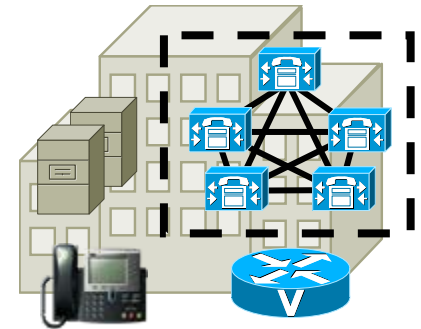
A: Great idea

Q: How many digits for intrasite calls, though?

A: Not 3 (4XX and 1XX overlap)

Not 4 either (911!)

5 would work!



**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

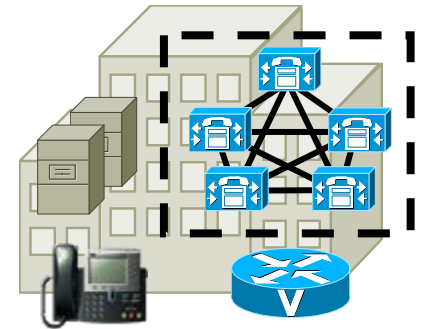
9 + 7 digits, 9 + 10 digits , 9 + 1 + 10 digits,  
9 + 011... all off-net patterns

And then any five digits that begin with 1 though 7  
is an on-net, intrasite call

Am I good to go?

A: Yes

...for now



**Anchorage**  
907 507 18XX



**New York**  
212 555 75XX



**Chicago**  
708 552 91XX



**Birmingham**  
205 937 54XX



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**Hawaii**  
808 557 54XX

# Planning Considerations

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

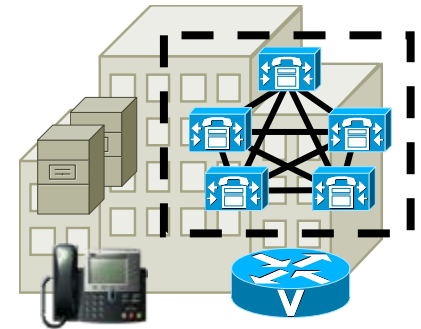
Q: I have 250 branches, with over 90 with 100+ users, and a dozen with more than 1000 users, and a headquarter with 12000 users. Can I still use eight + five digits for on-net, intersite calls?

A: No!

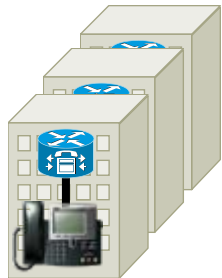
You essentially have the following to play with:

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

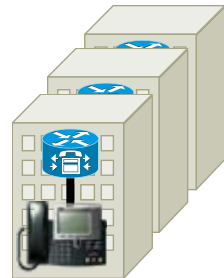
250 phone companies' DID ranges, the need for more than a whole five digit range for a single site, and dividing the rest into 250 unequal parts. Future planning, area code splits, new office codes, etc...



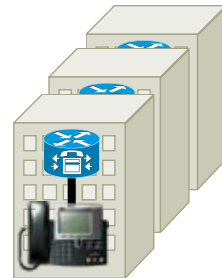
**San Jose**  
**408 526 XXXX**  
**408 853 XXXX**  
**Site Codes 123 and 124**



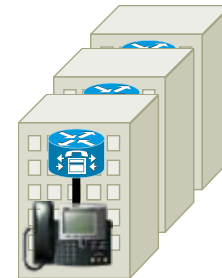
**Baltimore**  
**240 555 XXXX**  
**Site Code 012**



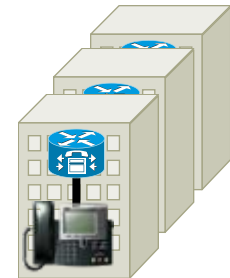
**Oakland**  
**510 555 51XX**  
**Site Code 345**



**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? (2)

Q: What to do?

A: Site codes are a good idea

0 = operator

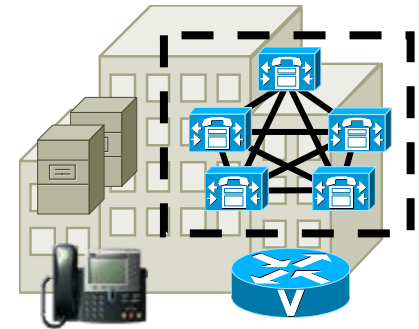
9 = outside line, all combinations

8 + site code (three digits would work up to 1000 sites), followed by a four digit extension

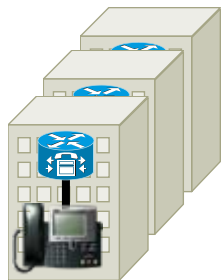
[1-7]XXX: on-net, intrasite dialing

Q: But I have a site with more than 10000 users?

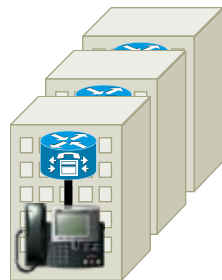
A: Would you be OK with using two site codes for that site?  
And having that site use five digit on-net?



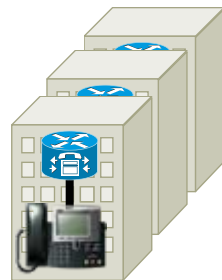
**San Jose**  
408 526 XXXX  
408 853 XXXX  
**Site Codes 123 and 124**



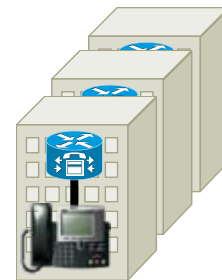
**Baltimore**  
240 555 XXXX  
**Site Code 012**



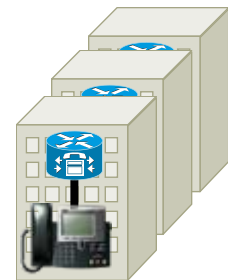
**Oakland**  
510 555 51XX  
**Site Code 345**



**New Orleans**  
504 555 5XXX  
**Site Code 256**

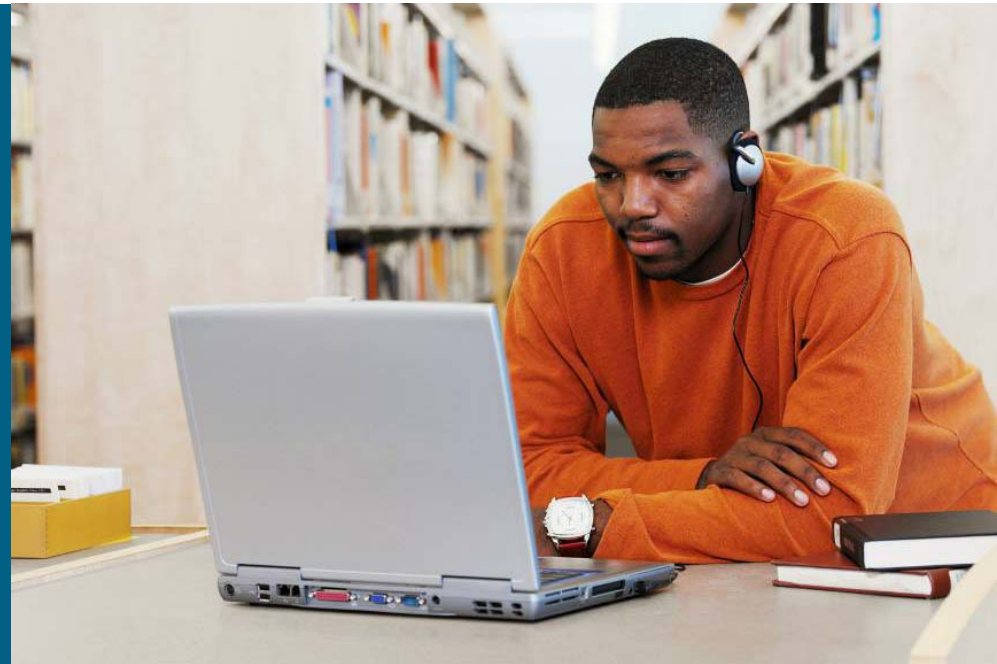


**Philadelphia**  
267 555 1XXX  
**Site Code 390**



**Hawaii**  
808 557 54XX  
**Site Code 822**

# Design Guidelines



# Design Guidelines Agenda

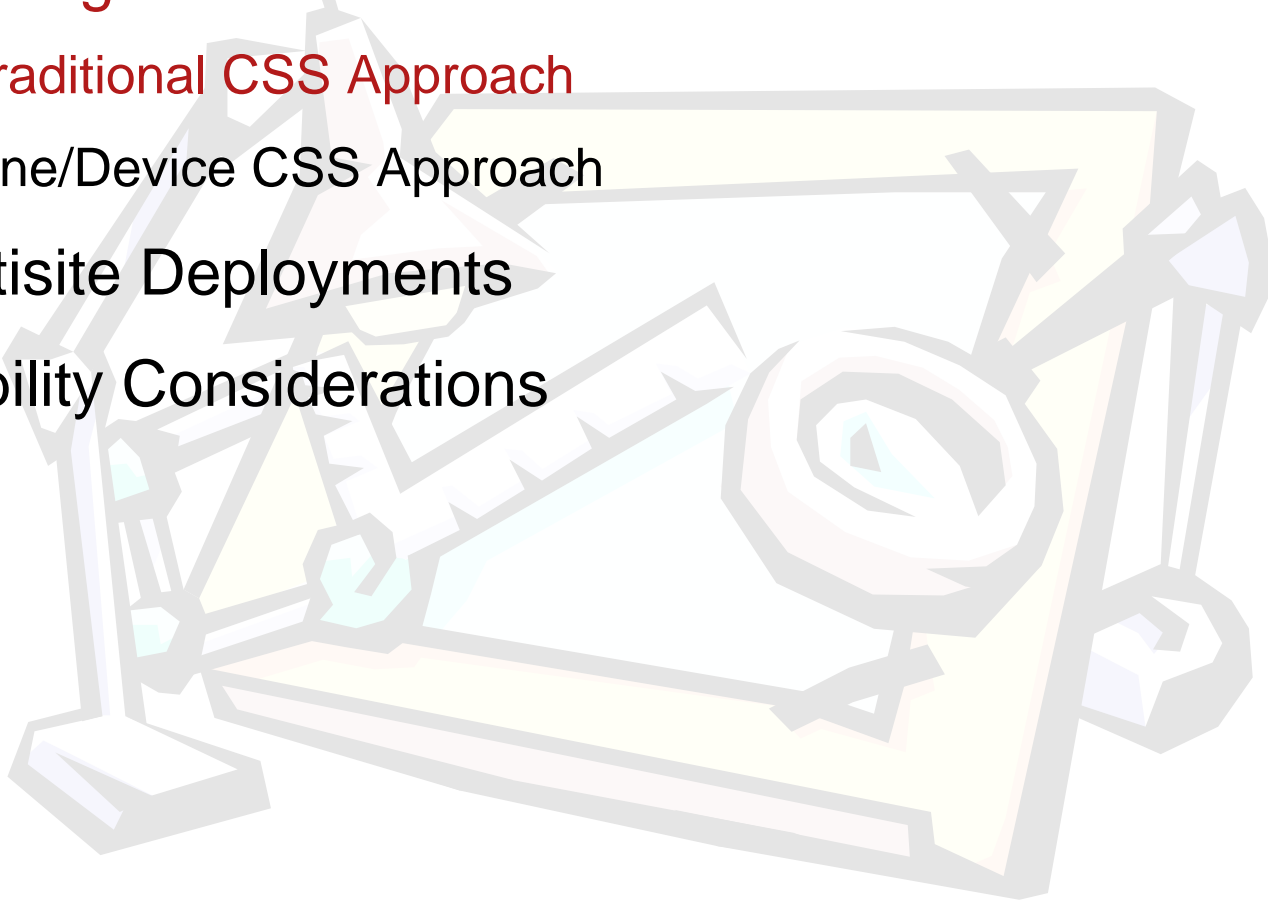
- **Building Classes of Service**

  - Traditional CSS Approach

  - Line/Device CSS Approach

- **Multisite Deployments**

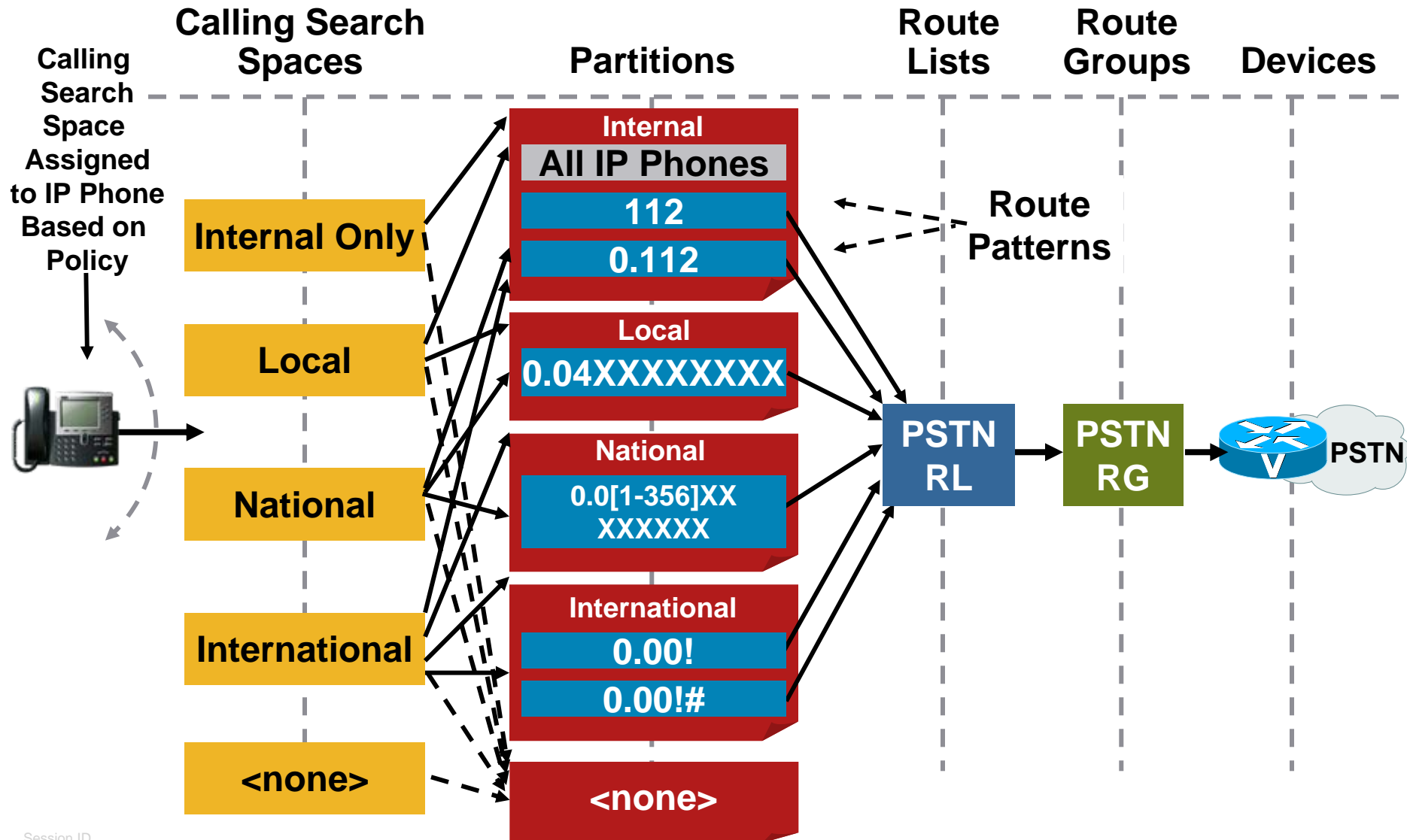
- **Mobility Considerations**





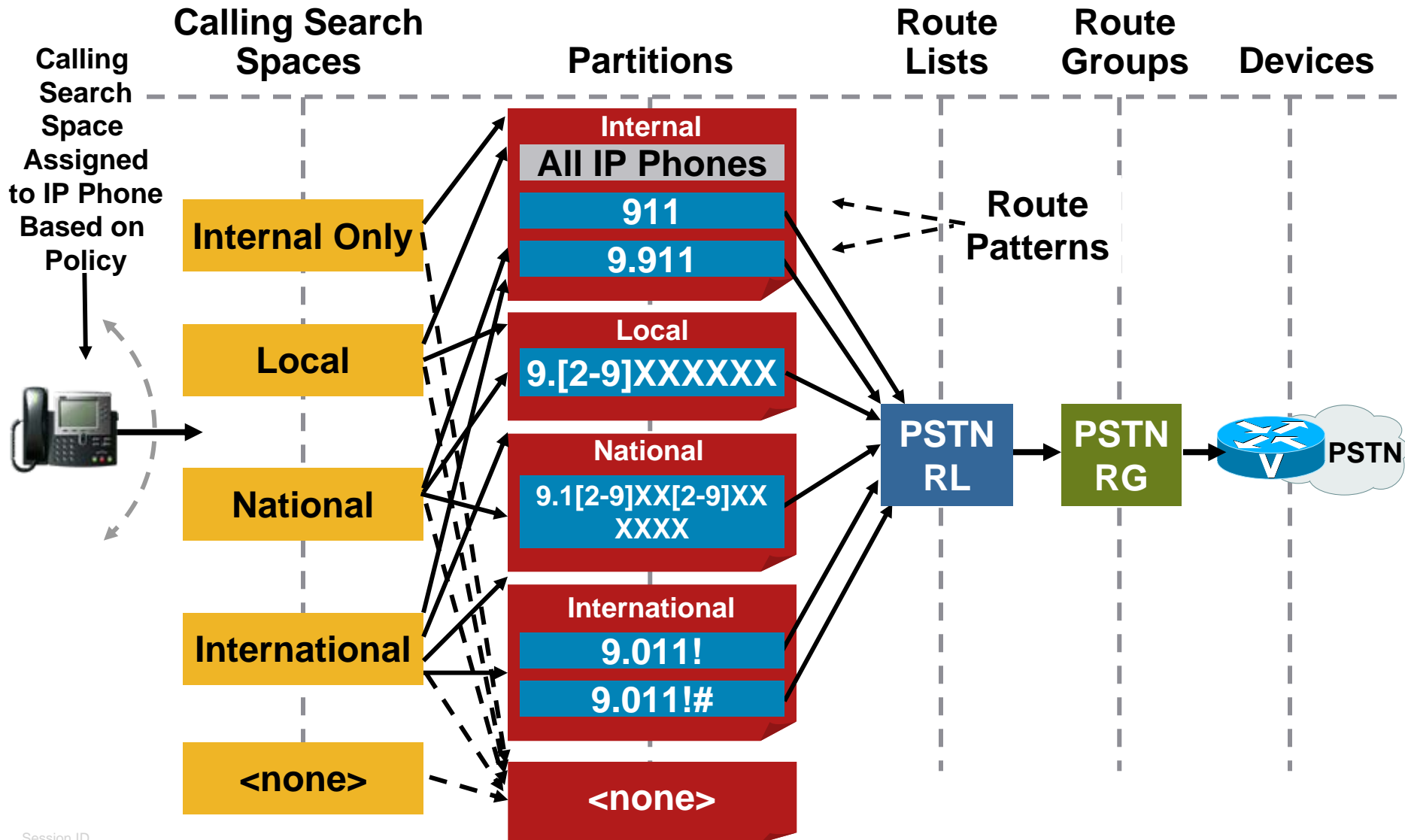
# Traditional CSS Approach

## Example of Composite View—France



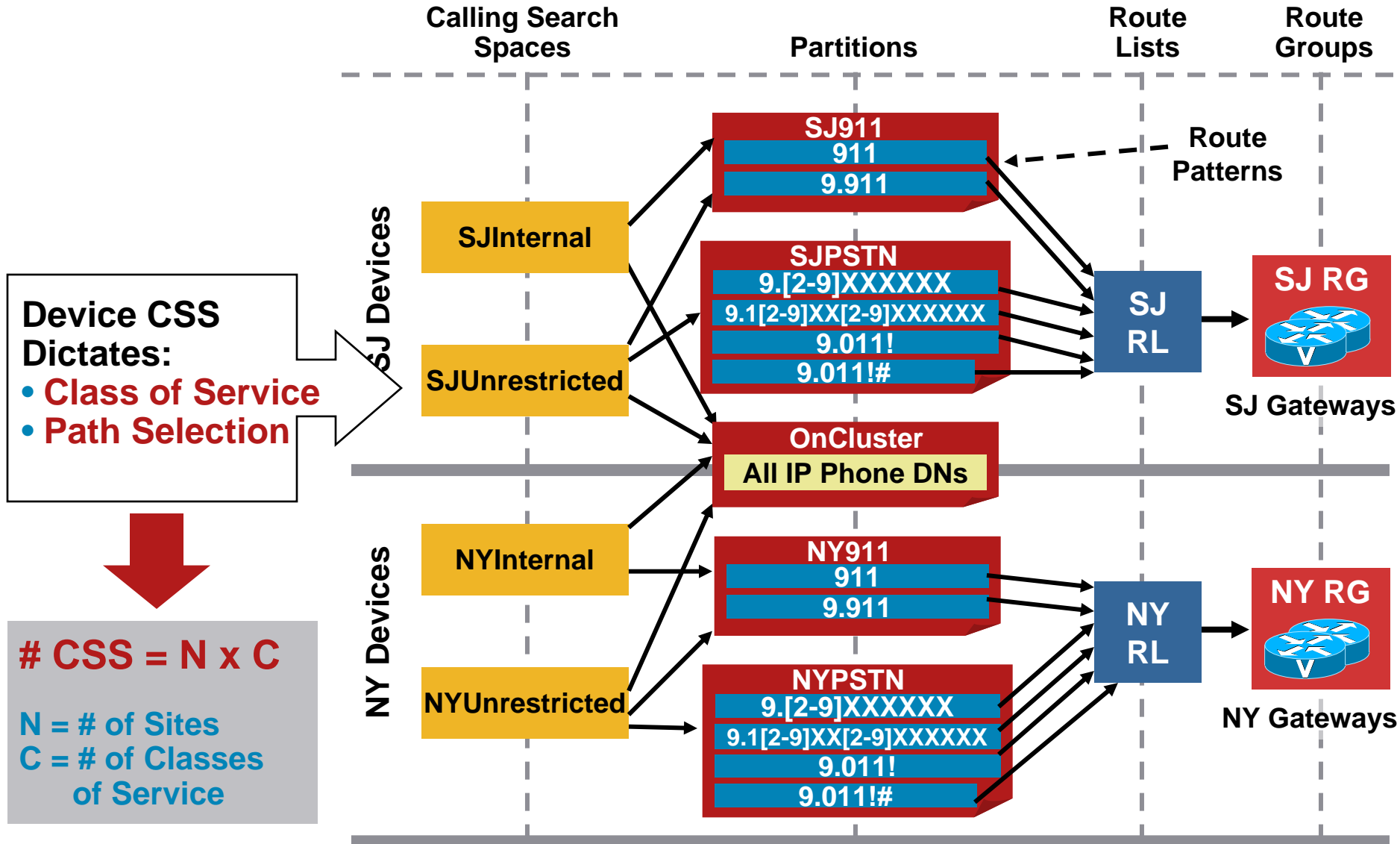
# Traditional CSS Approach

Example of Composite View—North America



# Traditional CSS Approach

## Scalability for Centralized Deployments



**Device CSS Dictates:**

- **Class of Service**
- **Path Selection**

$$\# \text{ CSS} = N \times C$$

**N = # of Sites**  
**C = # of Classes of Service**

# Design Best Practices Agenda

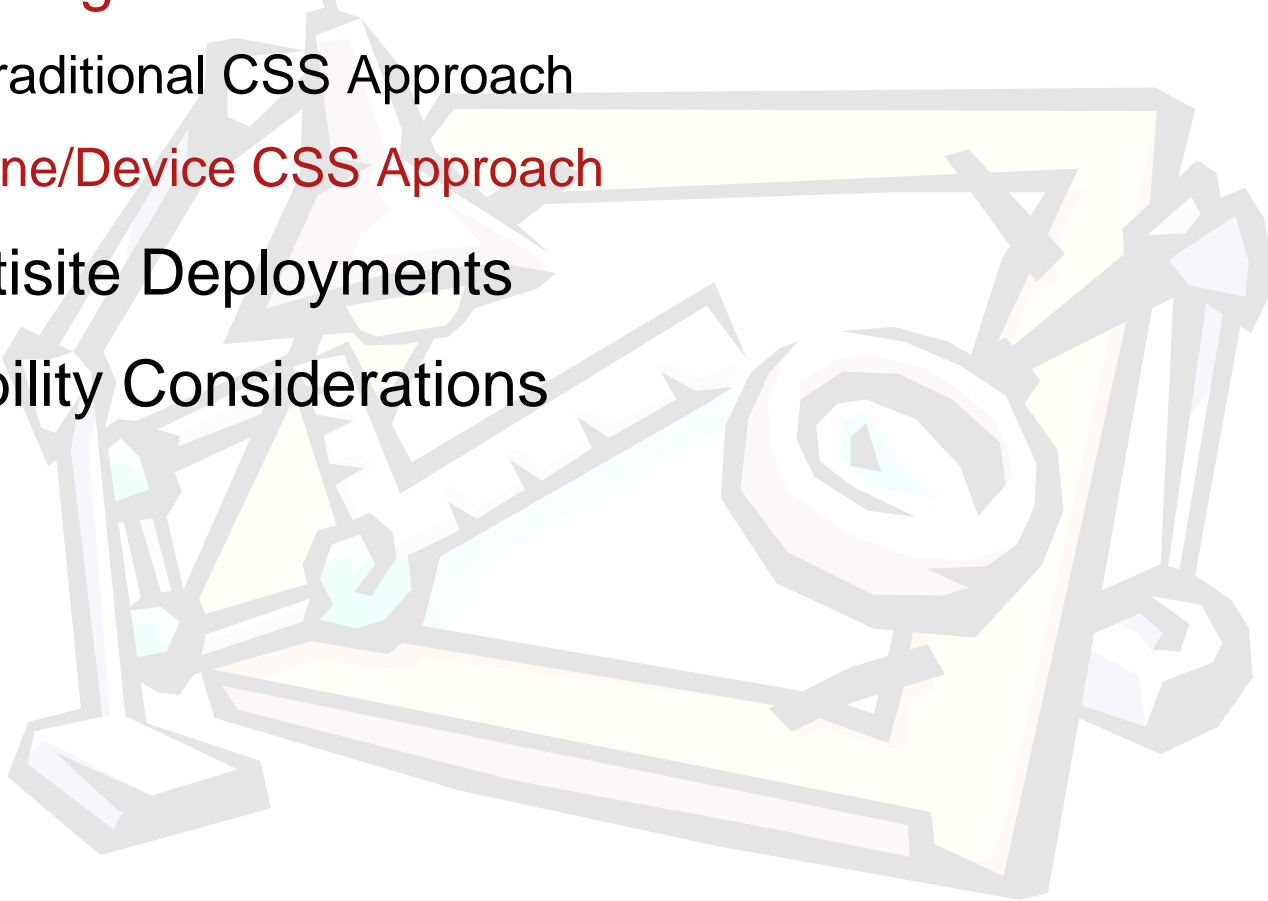
- **Building Classes of Service**

  - Traditional CSS Approach

  - Line/Device CSS Approach

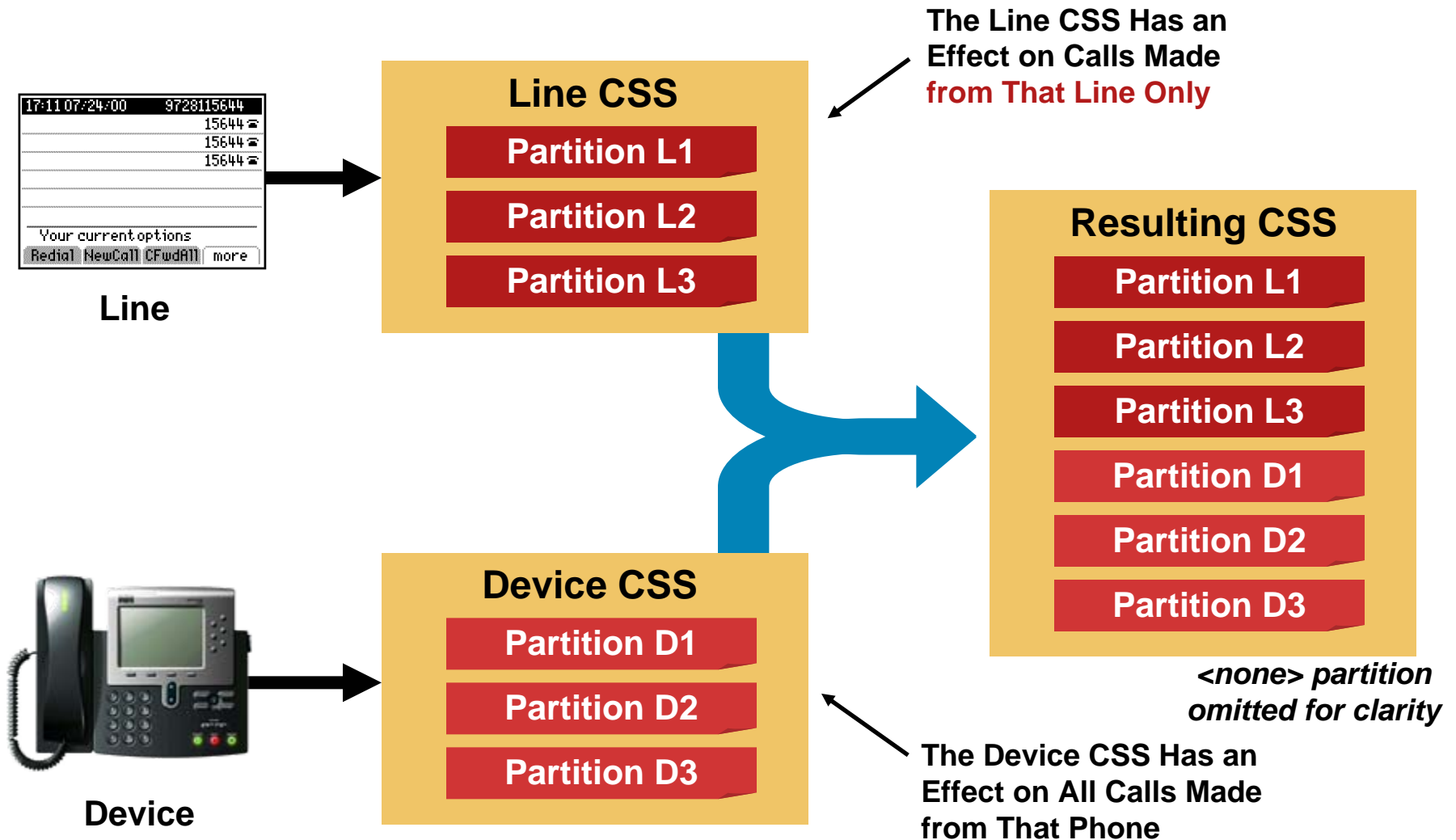
- **Multisite Deployments**

- **Mobility Considerations**



# The Line/Device CSS Approach

## Line CSS vs. Device CSS

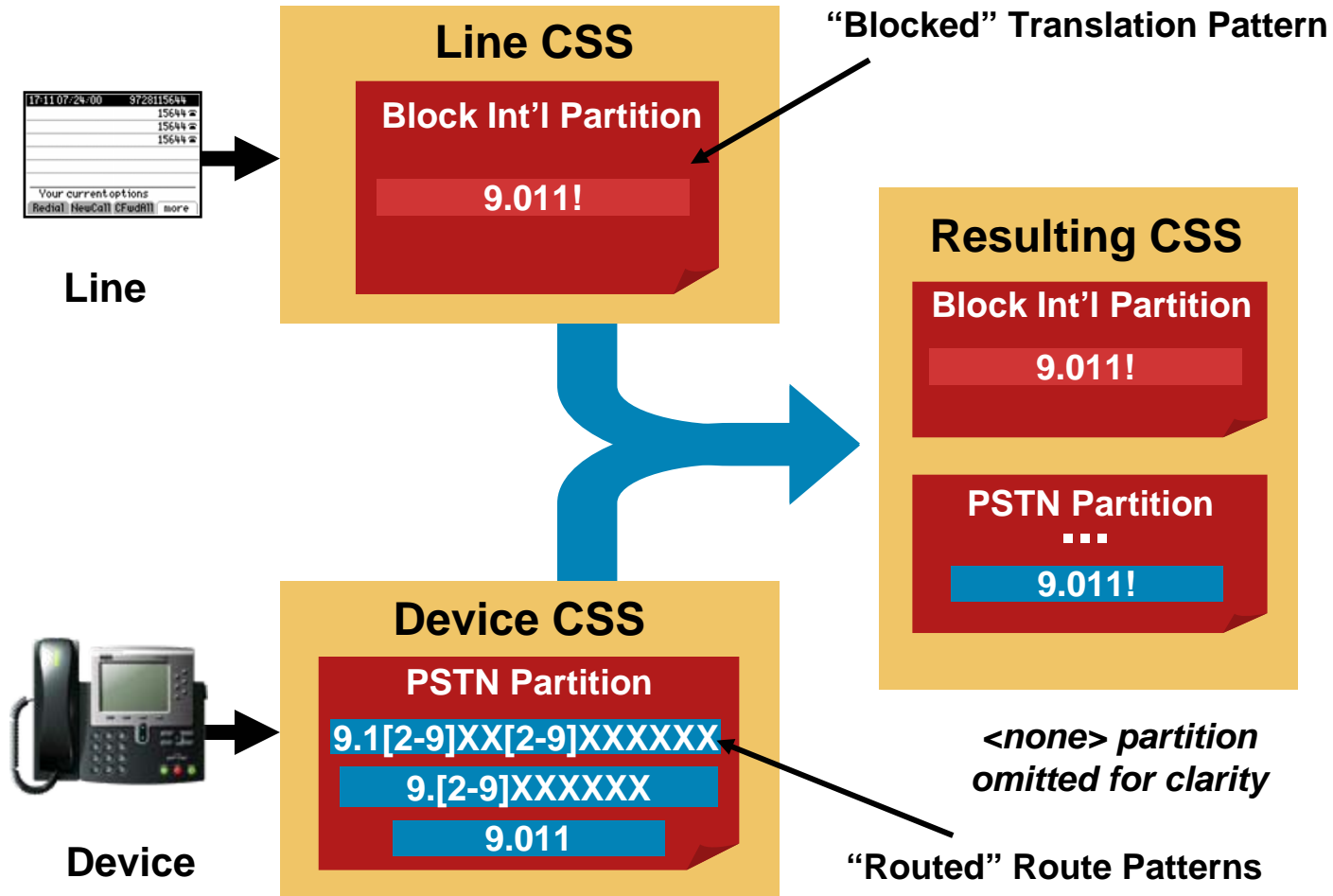


# The Line/Device CSS Approach

## 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 Centralized Deployments

**Line CSS Dictates:**

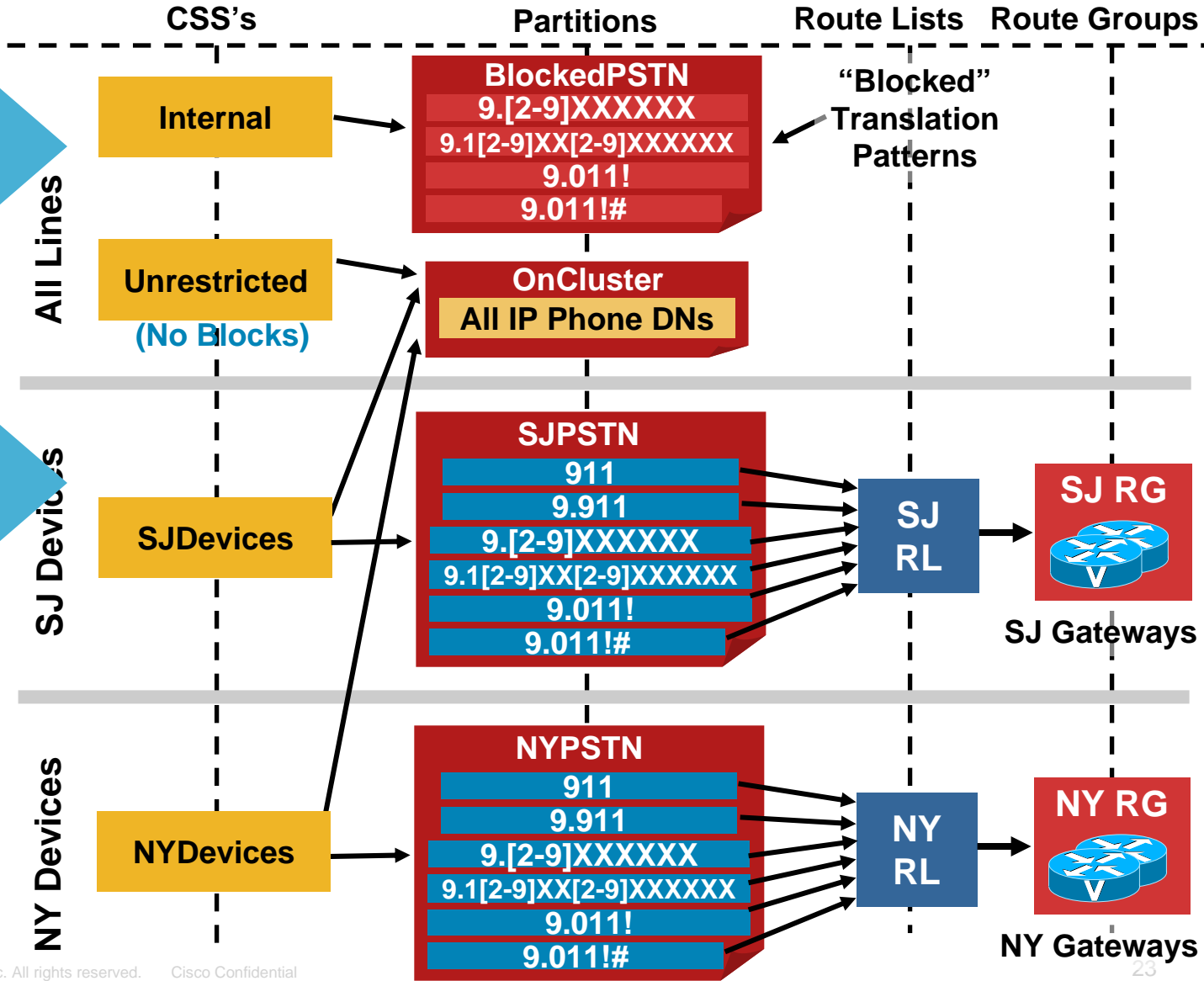
- **Class of Service**

**Device CSS Dictates:**

- **Path Selection**

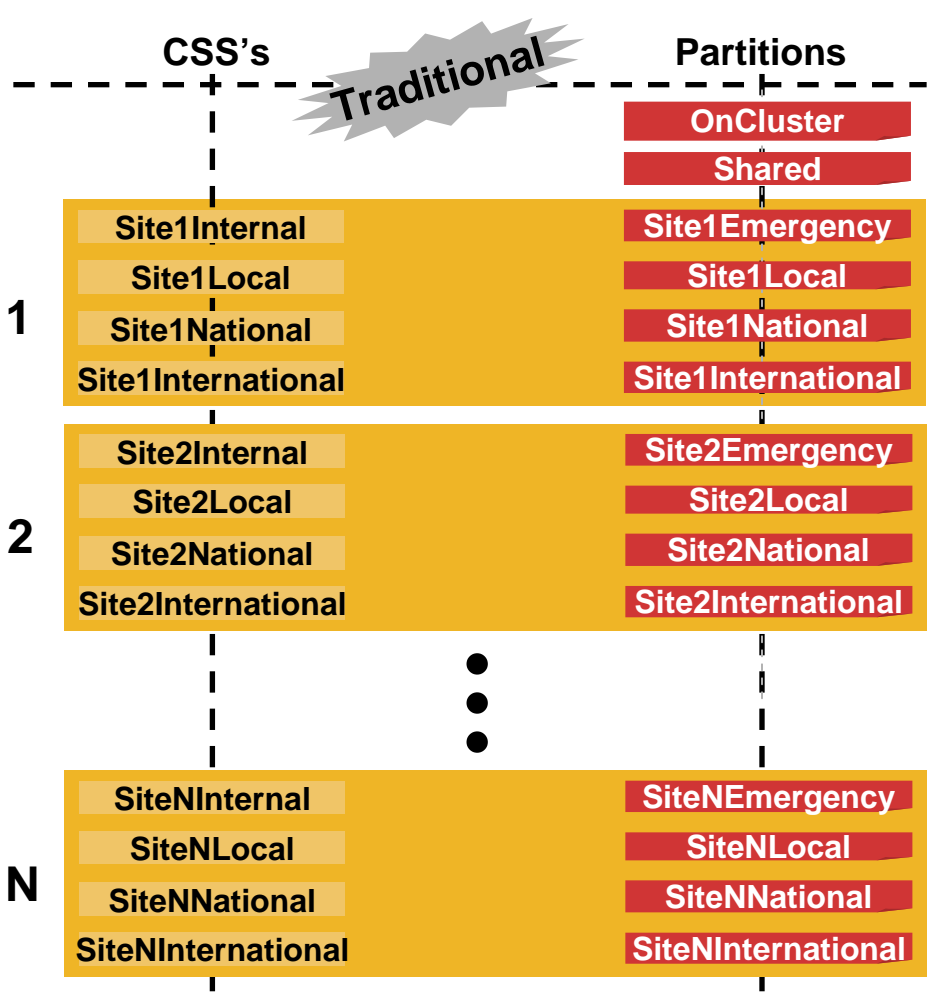
**# CSS = N + C**

**N** = # of sites  
**C** = # of classes of service

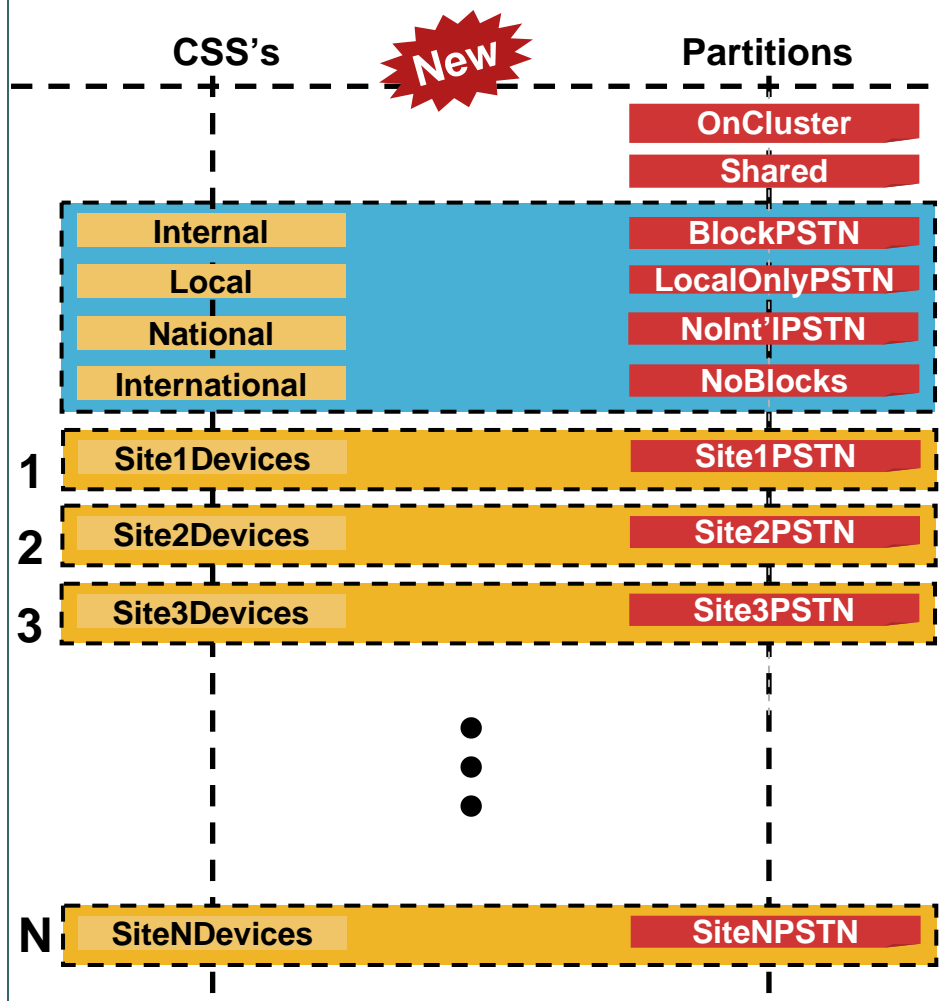


# The Line/Device CSS Approach

## Comparison of the Two Methods



**(N\*4) CSS's      ((N\*4) + 2) Partitions**



**(N + 4) CSS's      (N + 6) Partitions**



# The Line/Device CSS Approach

## CallForward Caveats (1 of 2)

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



In CUCM 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 (2 of 2)

**New**

- 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 configured CFAll and Secondary CSS for CFAll 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/Device CSS approach **cannot be \*directly\* applied to CTI devices**, such as Softphone (not 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)

# Design Best Practices Agenda

- Building Classes of Service

- **Multisite Deployments**

  - **Choosing a Dial Plan Approach**

    - Uniform On-Net Dialing

    - Variable-Length On-Net Dialing with Partitioned Addressing

    - Variable-Length On-Net Dialing with Flat Addressing

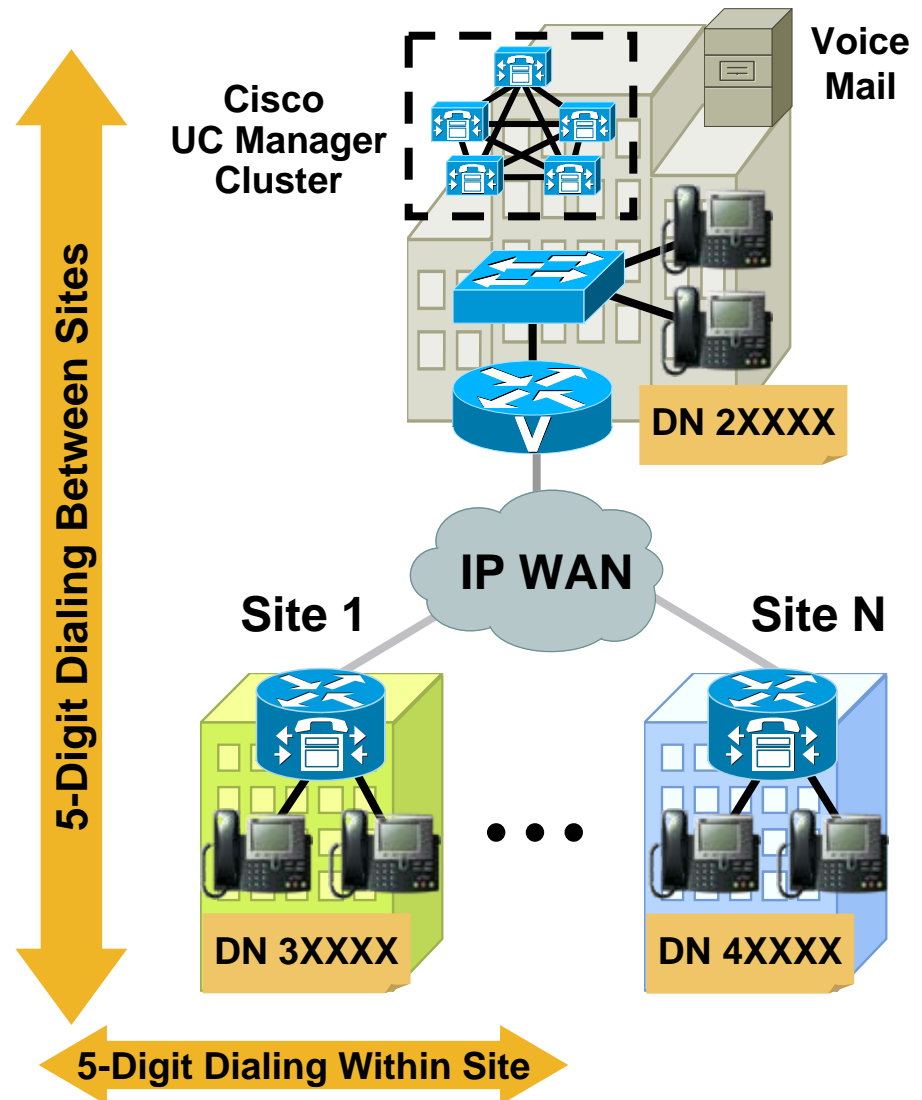
    - Tail End Hop Off (a.k.a. toll bypass)

- Mobility Considerations

# Choosing a Dial Plan Approach

## Uniform On-Net Dialing

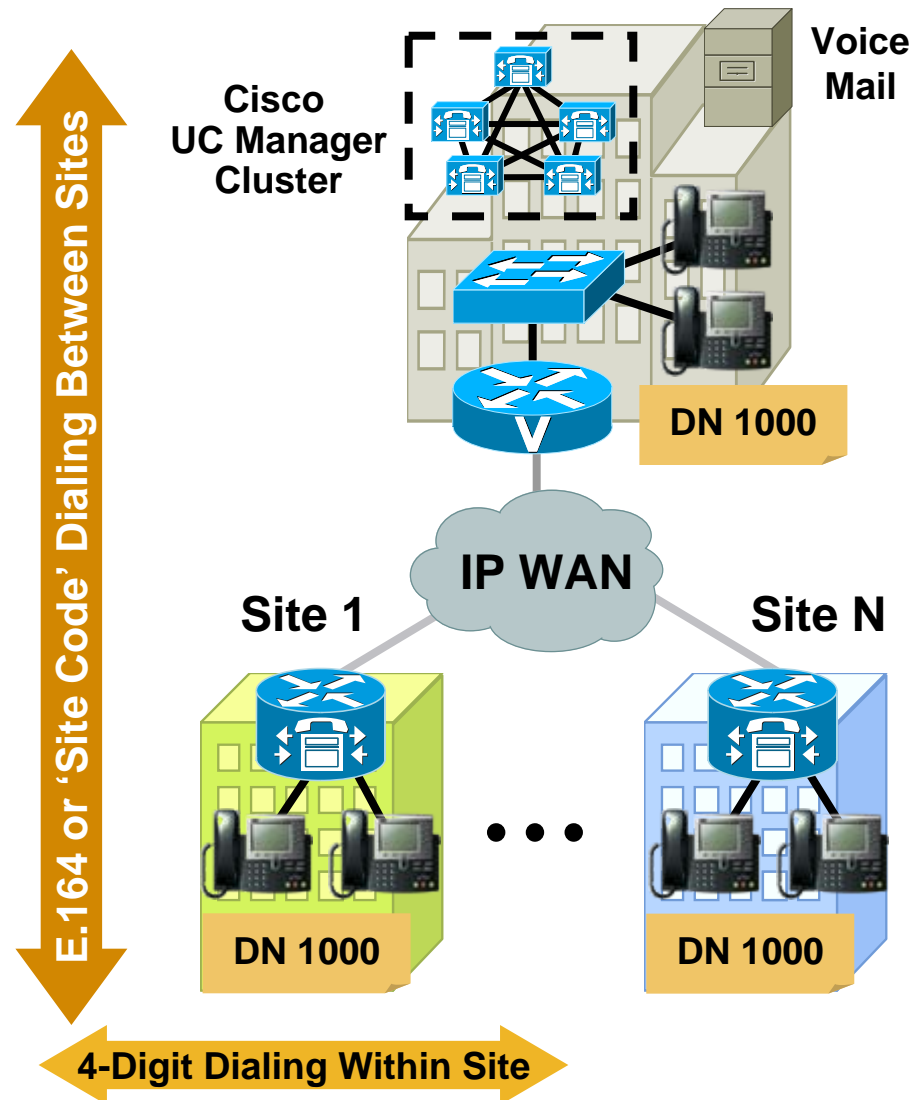
- Dialing within a site and across sites with same number of digits (e.g., 5)
- Extensions are globally unique
- Easy to design and configure
- Limited scalability of the addressing method (**number of sites, number of extensions**)



# Choosing a Dial Plan Approach

## Variable-Length On-Net Dialing (VLOD)

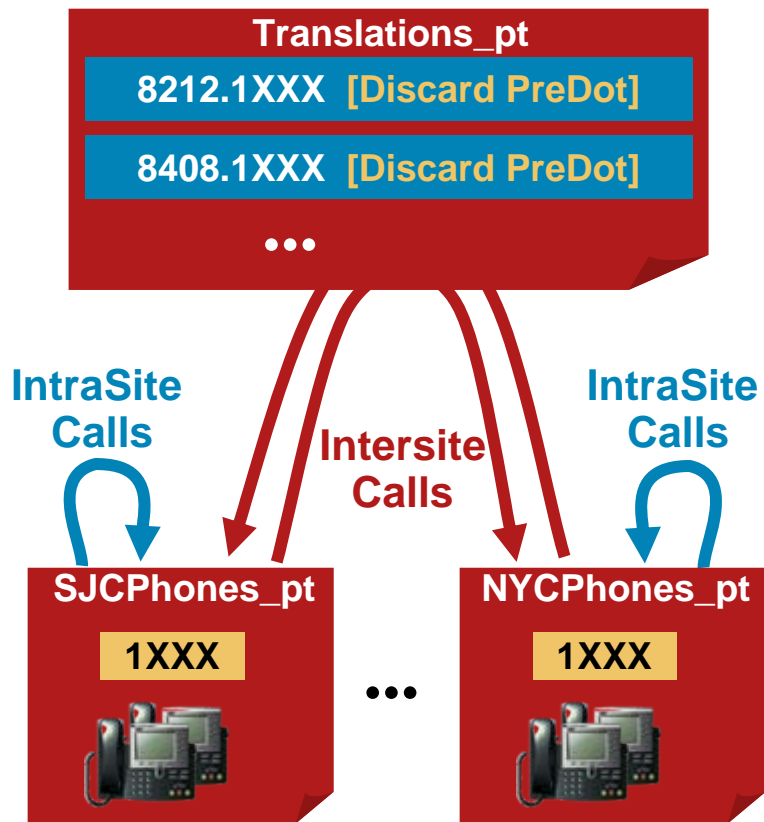
- Abbreviated dialing within a site (four or five digits)
- Identical extensions (e.g., 1000) may appear at different sites
- Intersite calls use an “escape code” (e.g., “9 + full E.164”, or “8 + site code + extension”)
- Easier scalability for large numbers of extensions and sites



# Choosing a Dial Plan Approach

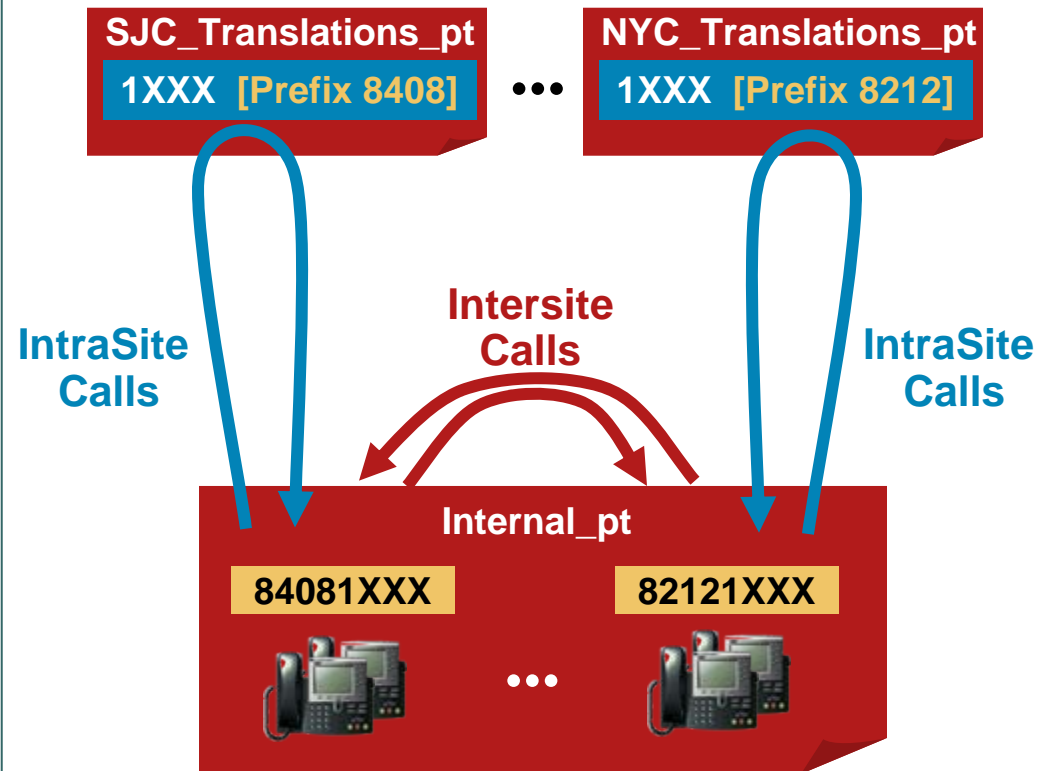
## Addressing Methods for VLOD

### Partitioned Addressing



- Phone DN's in different partitions
- Global Xlations for intersite calls

### Flat Addressing



- Phone DN's in same global partition
- Per-site translations for intrasite calls

# Choosing a Dial Plan Approach

## Preliminary Design Questions

- How many sites are going to be part of the system?
- What are the calling patterns between sites?
- What do users dial within a site and to reach another site?
- What transport network is going to be used for intersite calls (PSTN or IP WAN)?
- What (if any) CTI applications are being used?
- Is there a desire for a standardized on-net dialing structure (e.g., using site codes)?



# Design Best Practices Agenda

- Building Classes of Service

- **MultiSite Deployments**

  - Choosing a Dial Plan Approach

  - Uniform On-Net Dialing**

  - Variable-Length On-Net Dialing with Partitioned Addressing

  - Variable-Length On-Net Dialing with Flat Addressing

  - Tail End Hop Off (a.k.a. toll bypass)

- Mobility Considerations

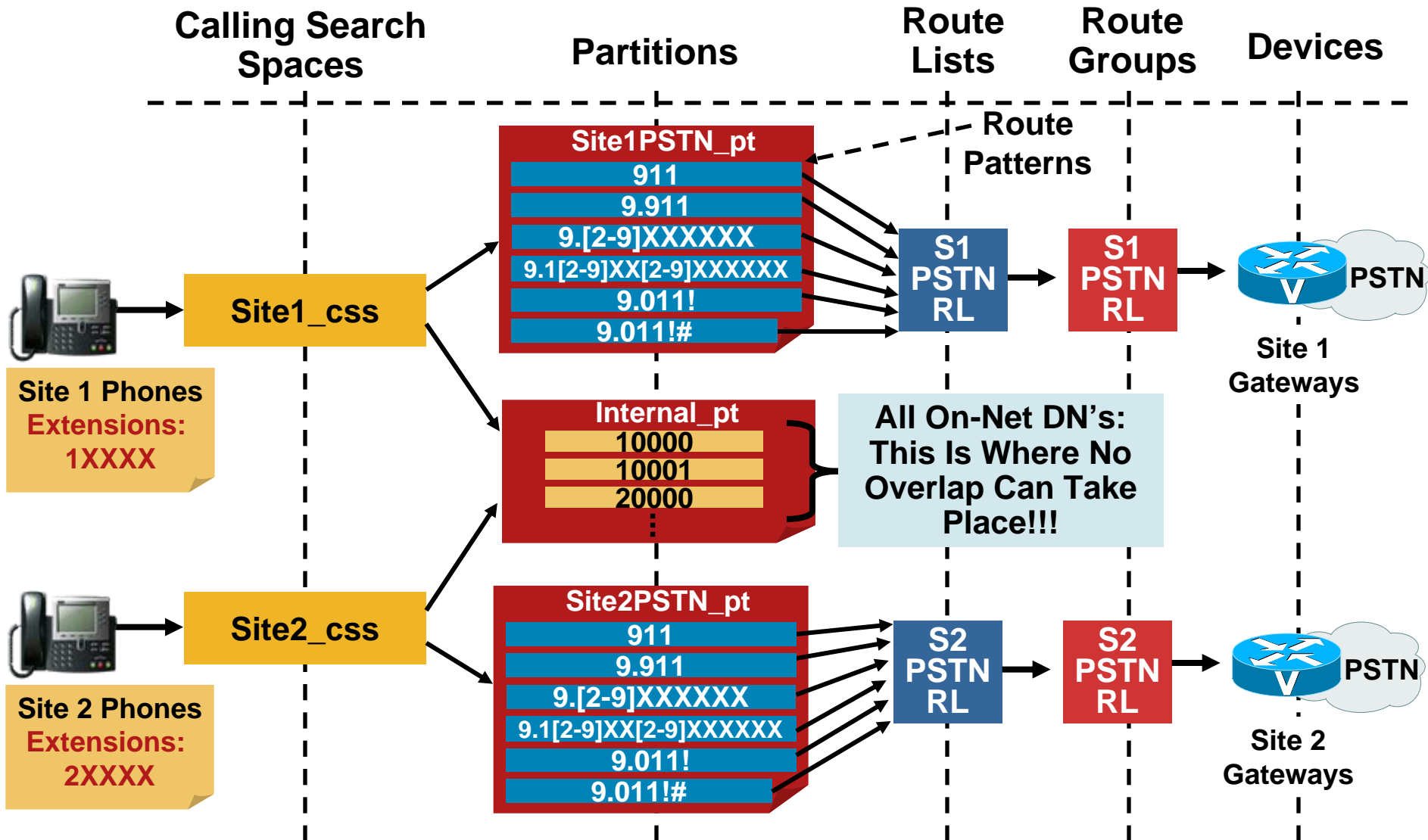
# Uniform On-Net Dialing

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



# Design Best Practices Agenda

- Building Classes of Service

- **MultiSite Deployments**

  - Choosing a Dial Plan Approach

  - Uniform On-Net Dialing

  - Variable-Length On-Net Dialing with Partitioned Addressing**

  - Variable-Length On-Net Dialing with Flat Addressing

  - Tail End Hop Off (a.k.a. toll bypass)

- Mobility Considerations

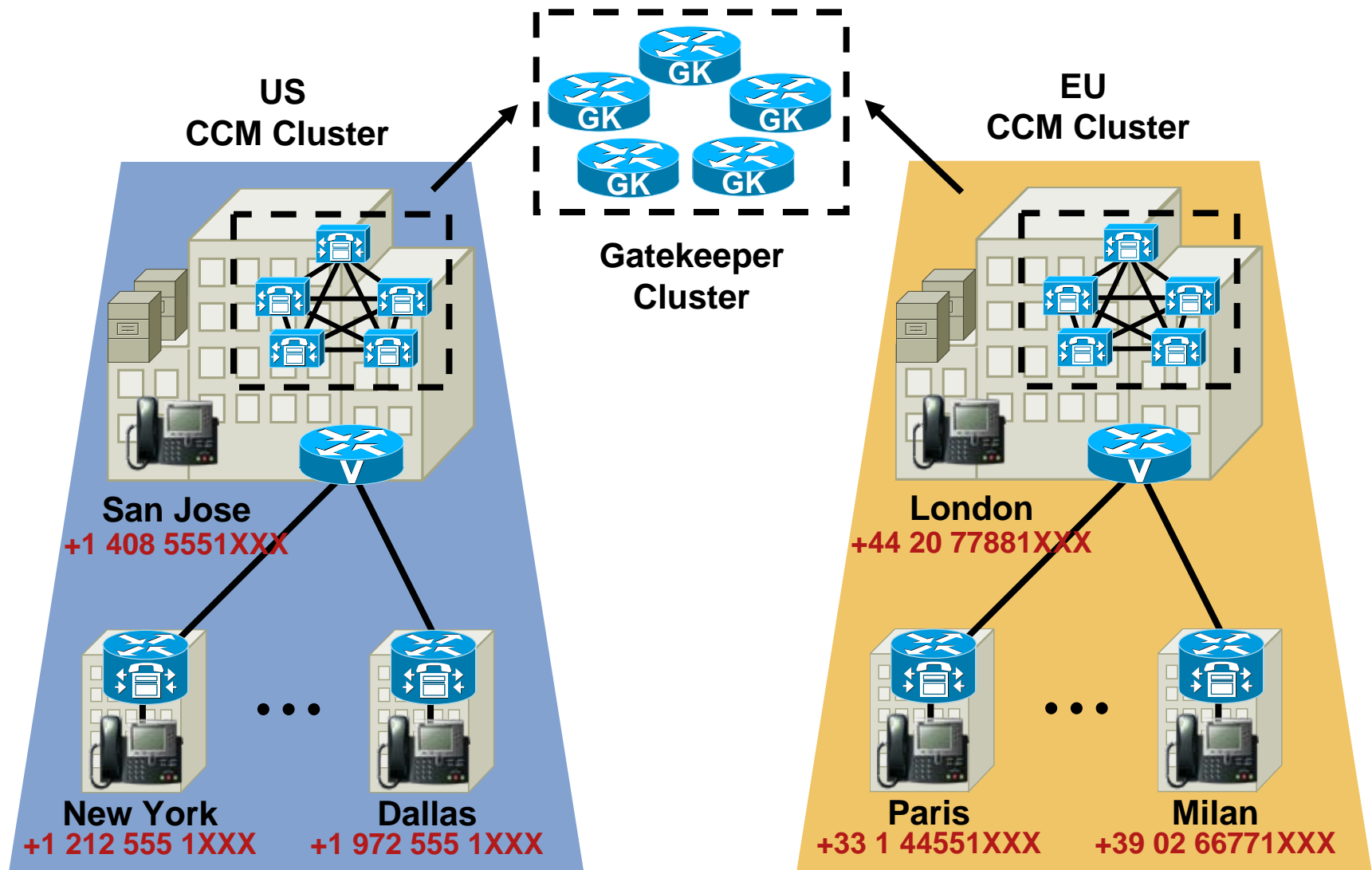
# VLOD with Partitioned Addressing

## 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...

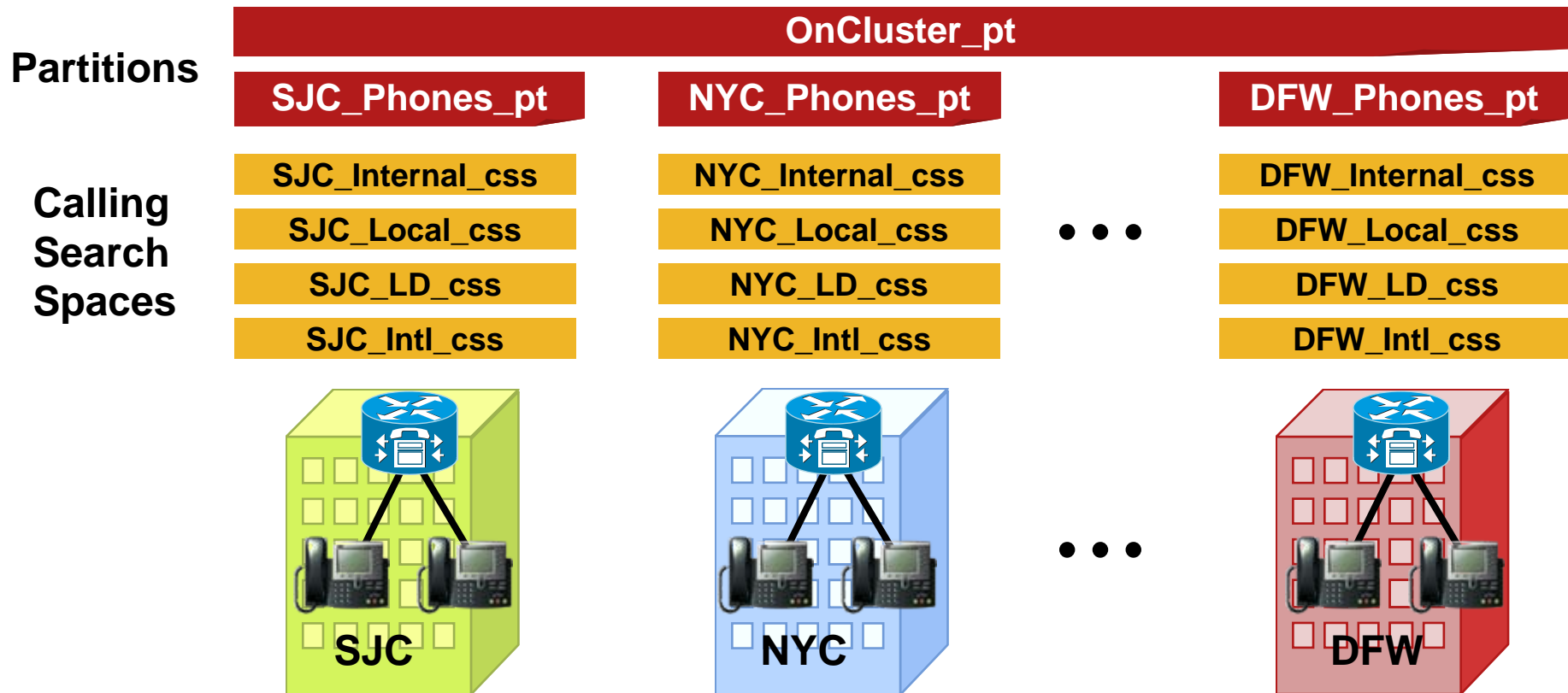
# VLOD with Partitioned Addressing

## Hypothetical Customer Example



# VLOD with Partitioned Addressing

## Partitions and Calling Search Spaces



**\* Note: If Using the Line/Device CSS Approach, the Number of CSS's 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

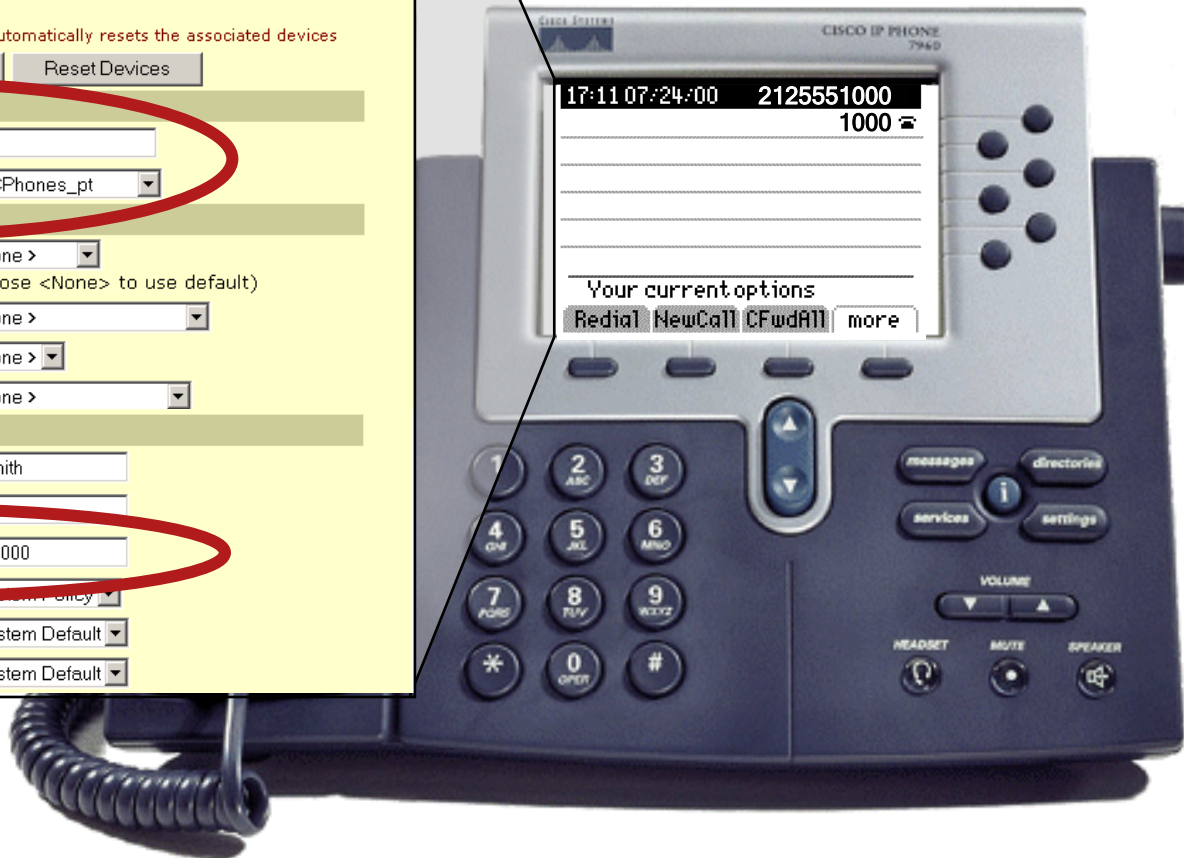
Line Text Mask

External Phone Number Mask 2125551000

Message Waiting (Phone Idle) Use System Default

Ring Setting (Phone Idle) Use System Default

Ring Setting (Phone Active)\*\* Use System Default

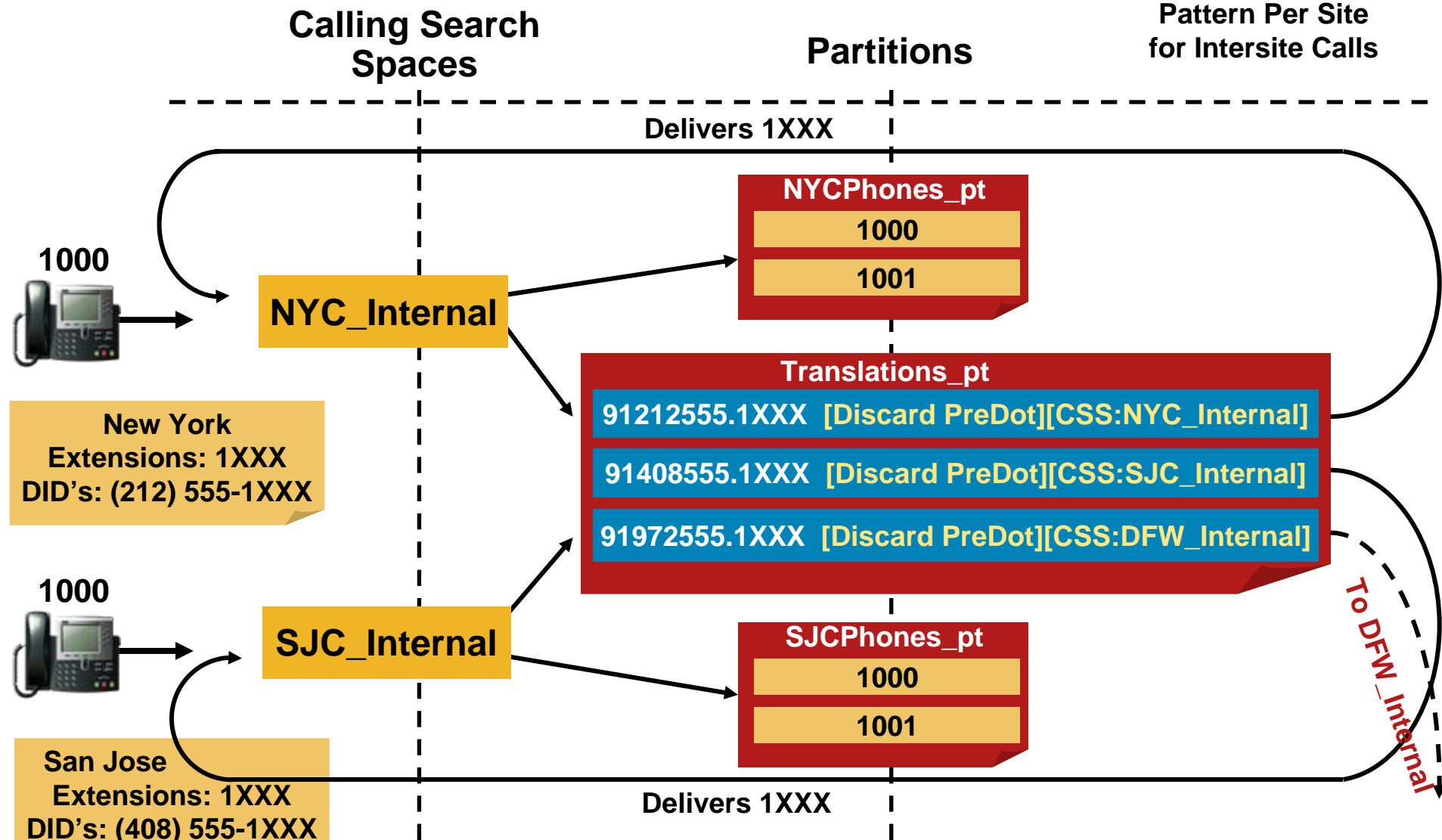




# VLOD with Partitioned Addressing

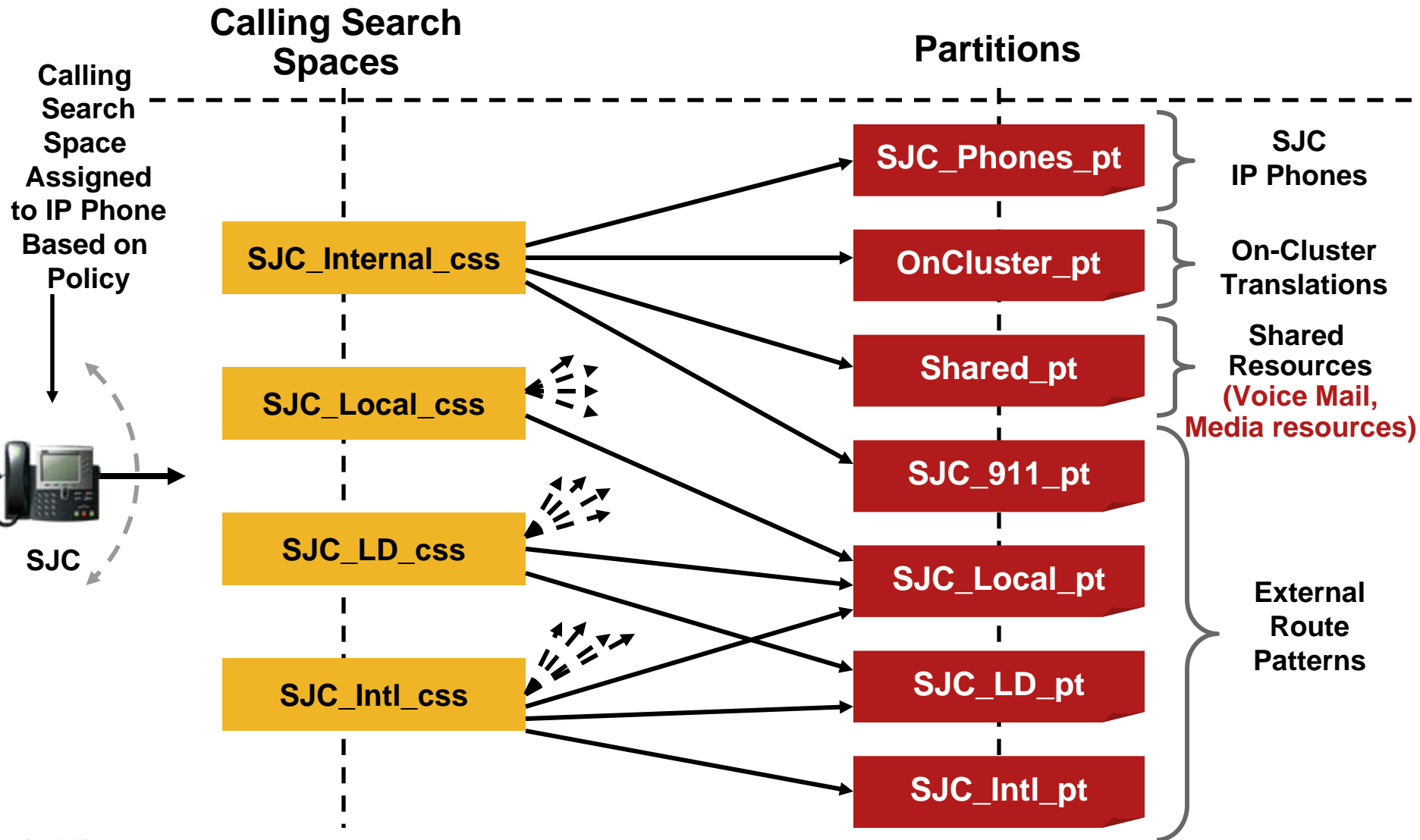
## Intersite Calls Within a Cluster

One Translation Pattern Per Site for Intersite Calls



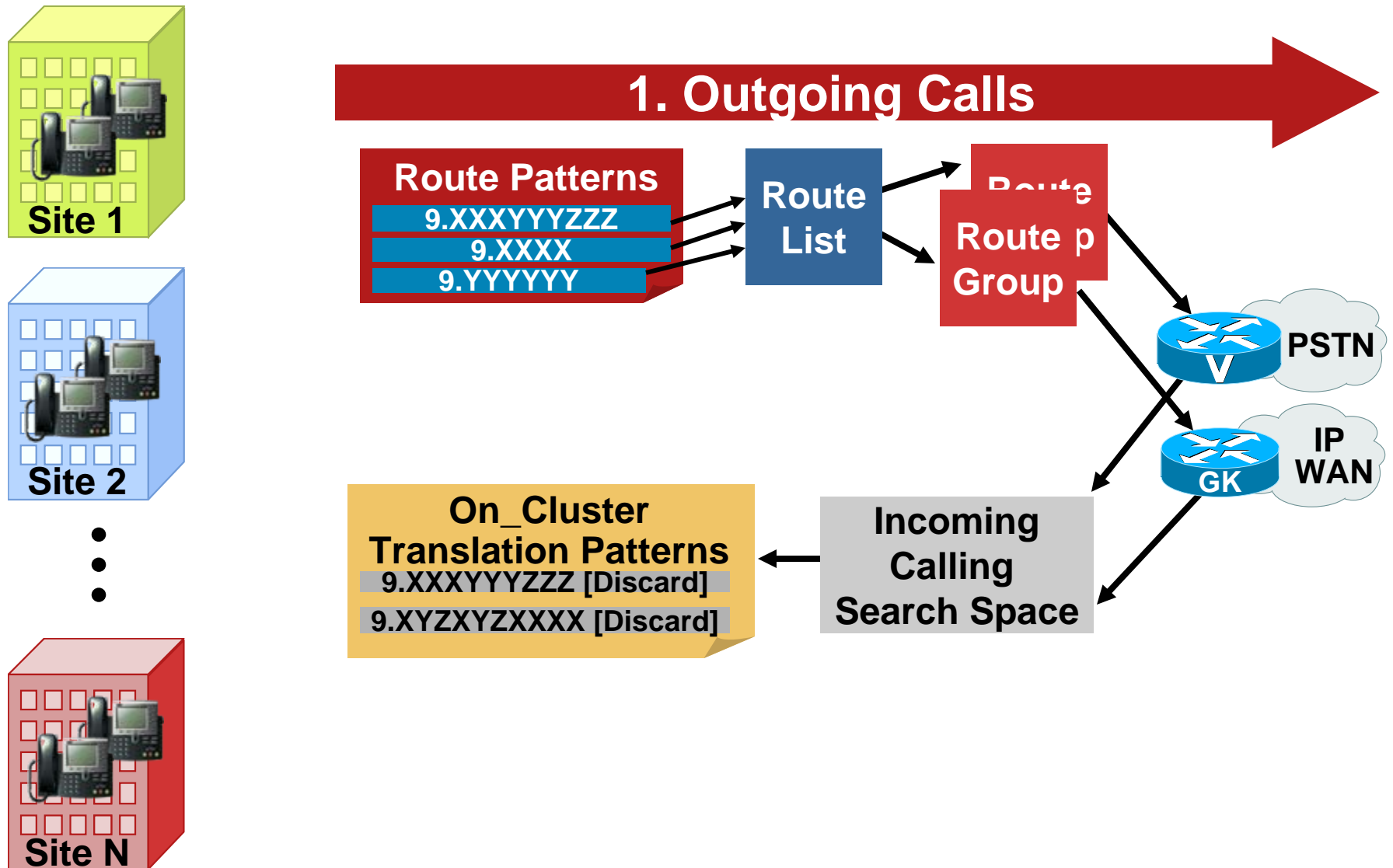
# VLOD with Partitioned Addressing

## View of Partitions/Calling Search Spaces



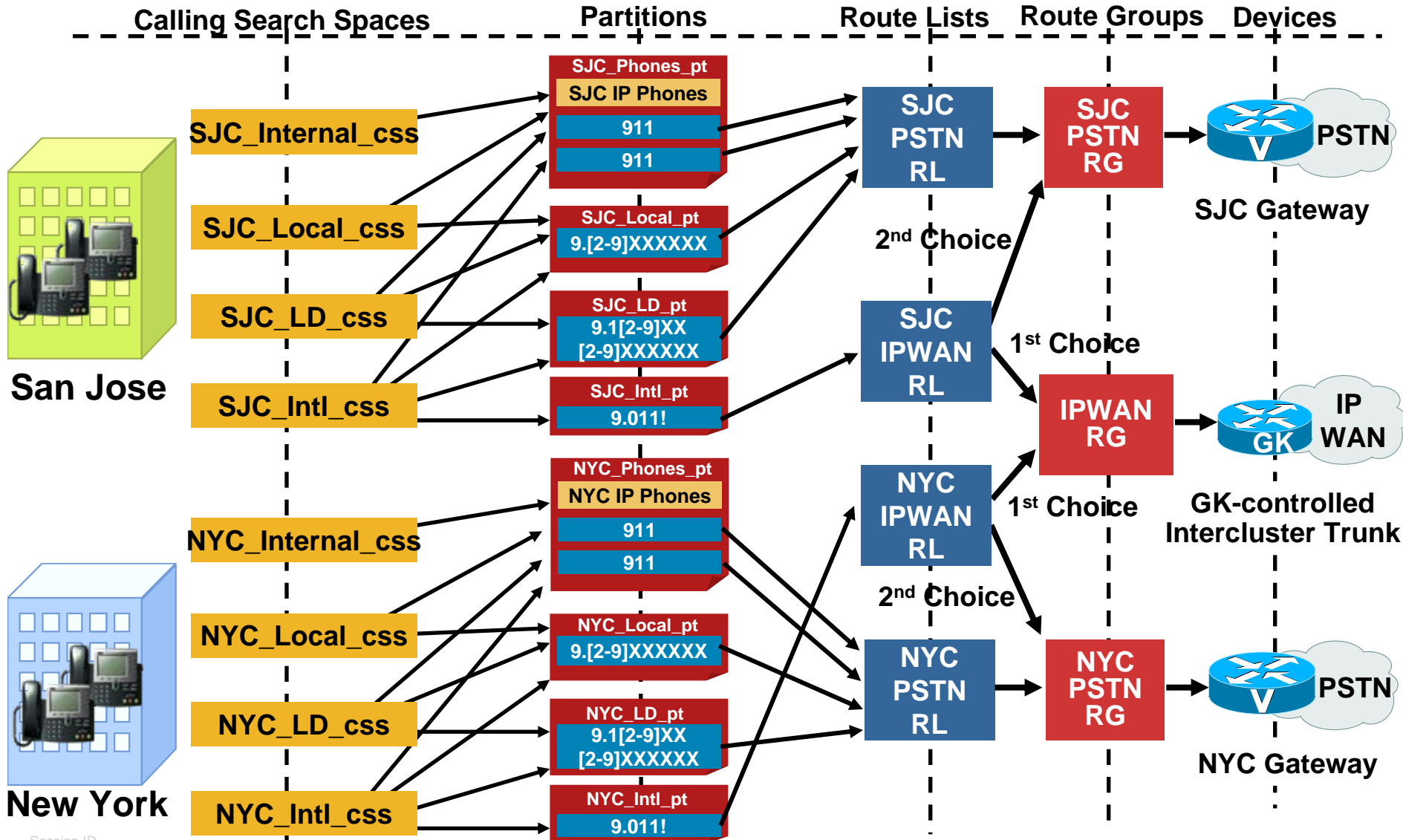
# VLOD with Partitioned Addressing

## Outgoing PSTN/Gatekeeper Calls



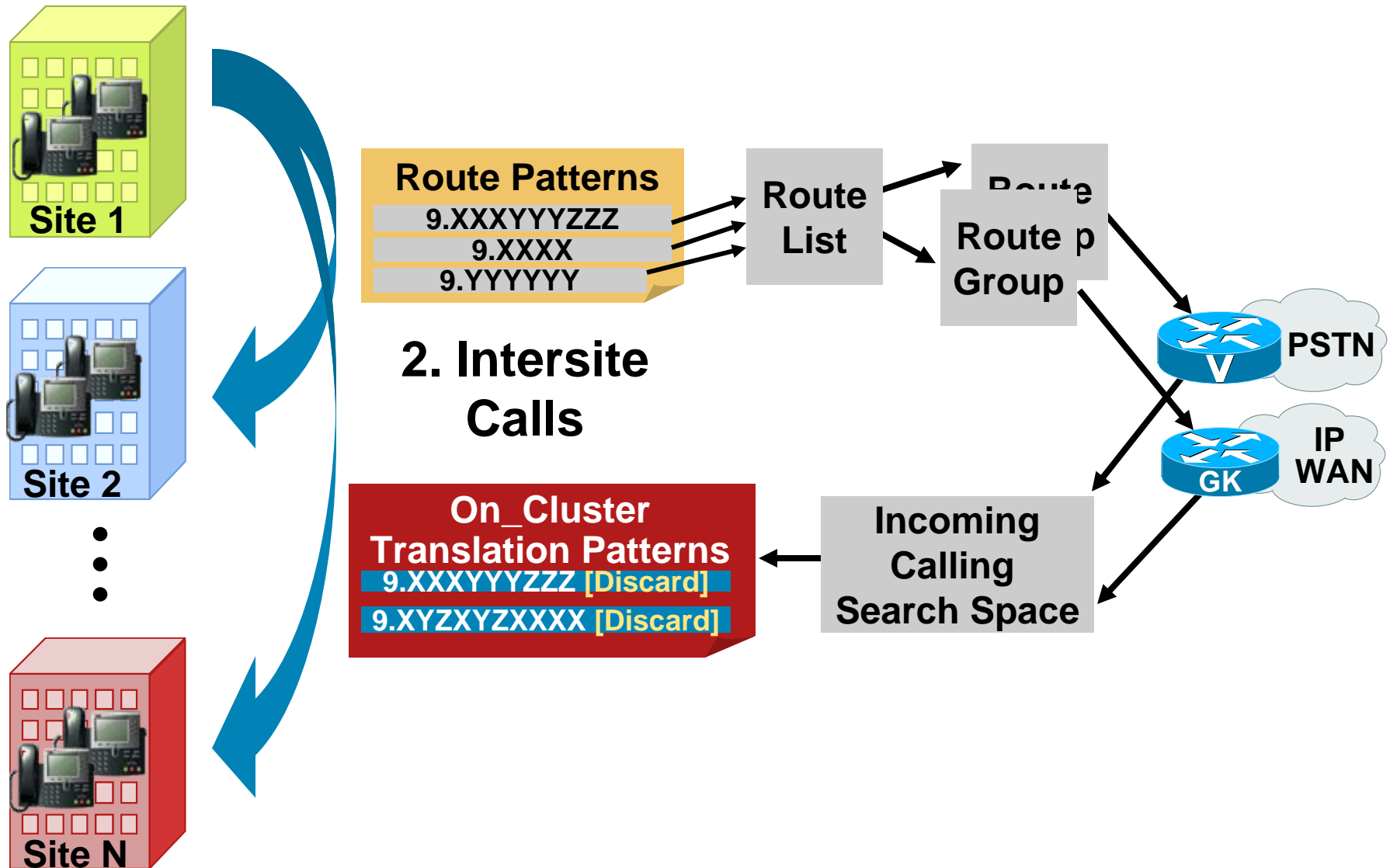
# VLOD with Partitioned Addressing

## Outgoing PSTN/Gatekeeper Calls



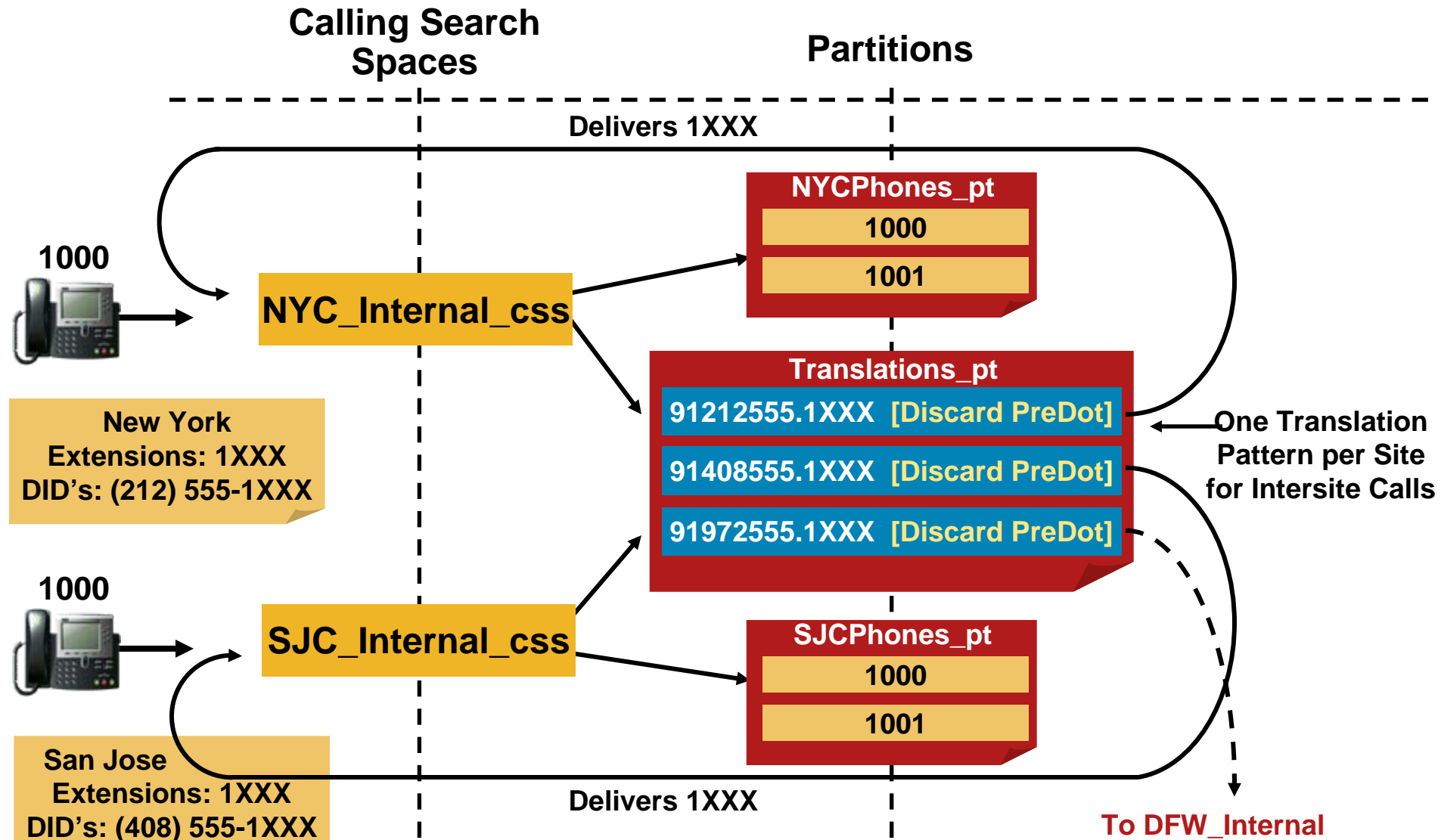
# VLOD with Partitioned Addressing

## Intersite Calls Within a Cluster



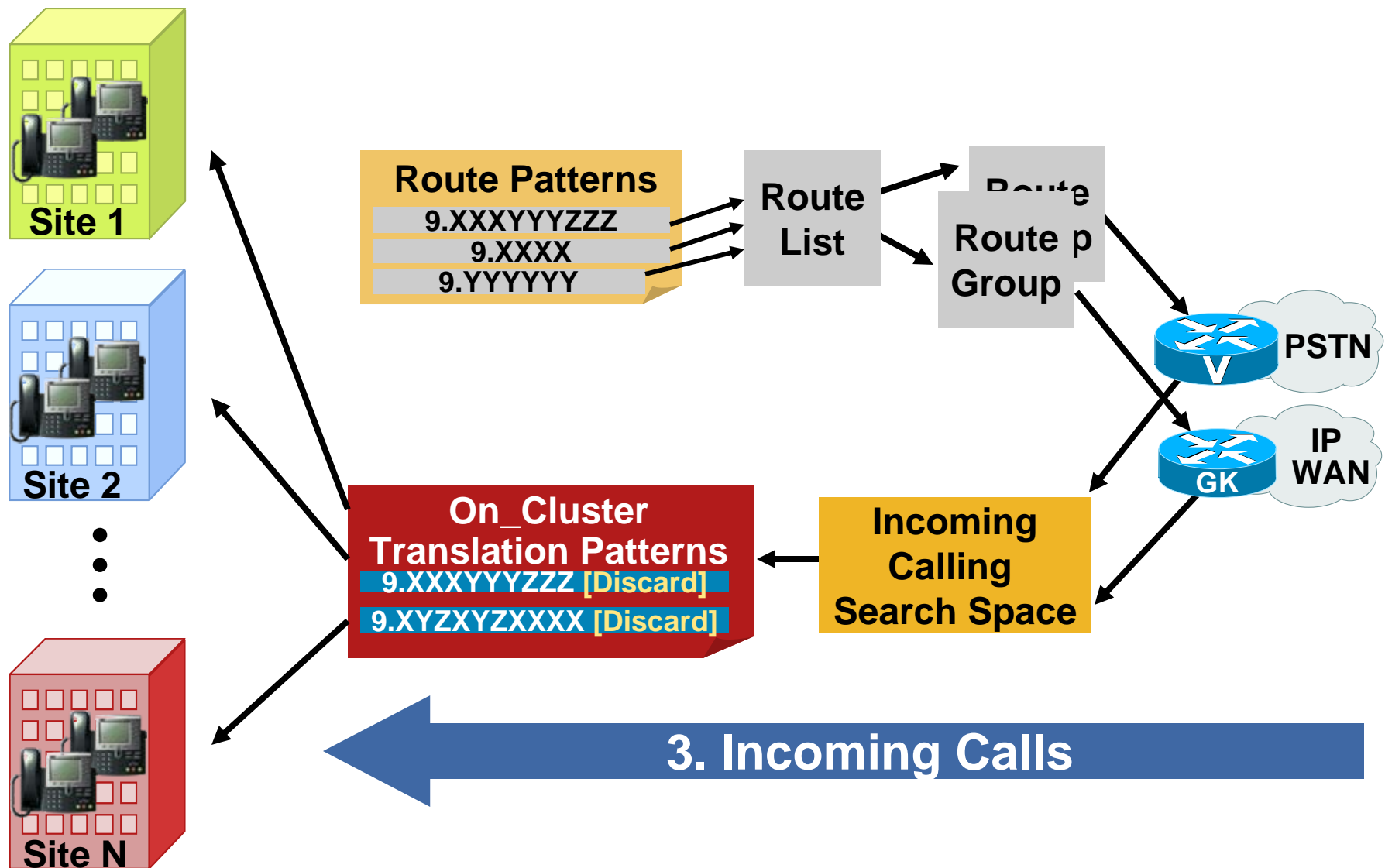
# VLOD with Partitioned Addressing

## Intersite Calls Within a Cluster



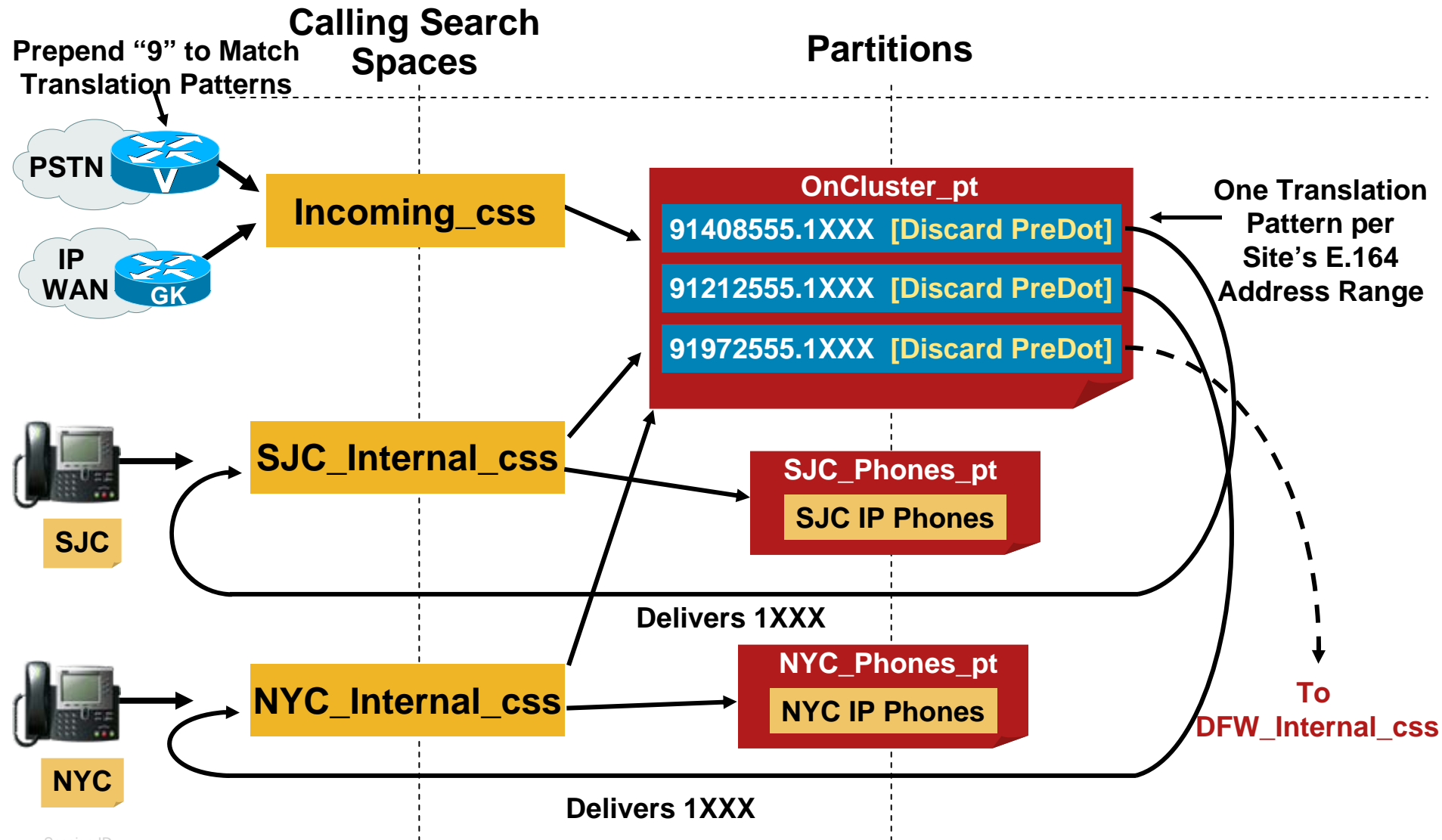
# VLOD with Partitioned Addressing

## Incoming PSTN/Gatekeeper Calls



# VLOD with Partitioned Addressing

## Incoming PSTN/Gatekeeper Calls





# VLOD with Partitioned Addressing

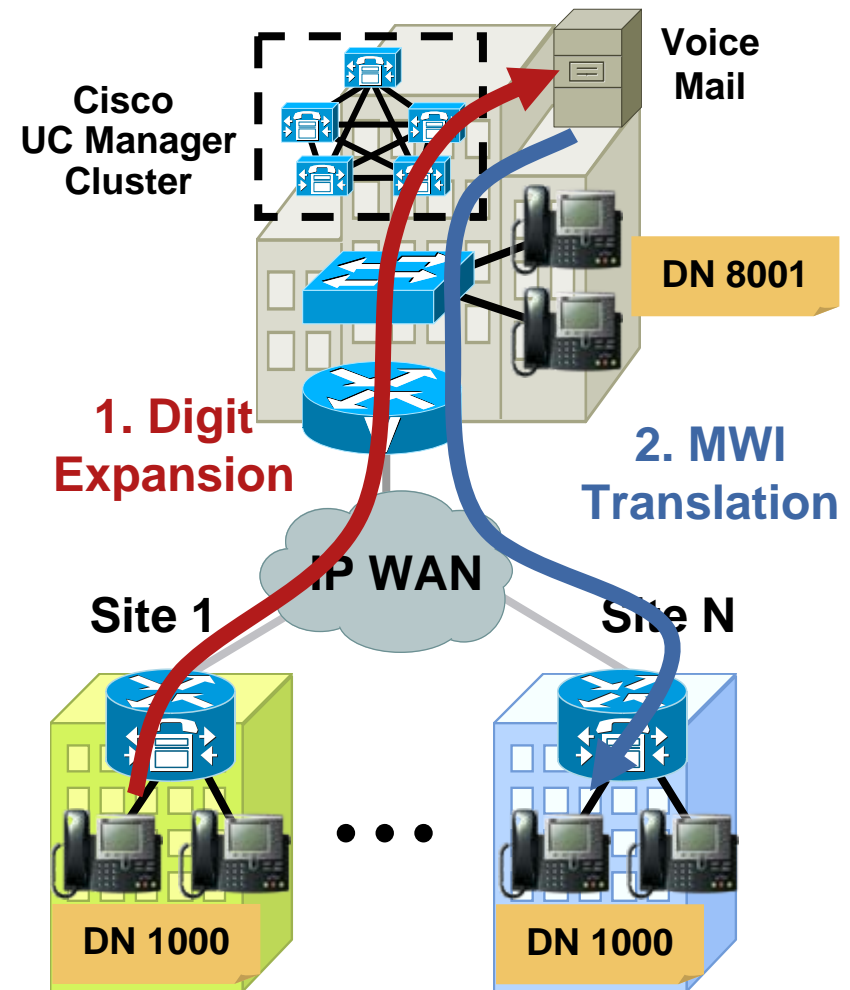
## Gatekeeper Configuration

```
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—(**Unity**) and SMDI-based Voice Mail systems can be used
- Voice mail boxes need a unique DN
- Need to “expand” DNs when accessing VM
- 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

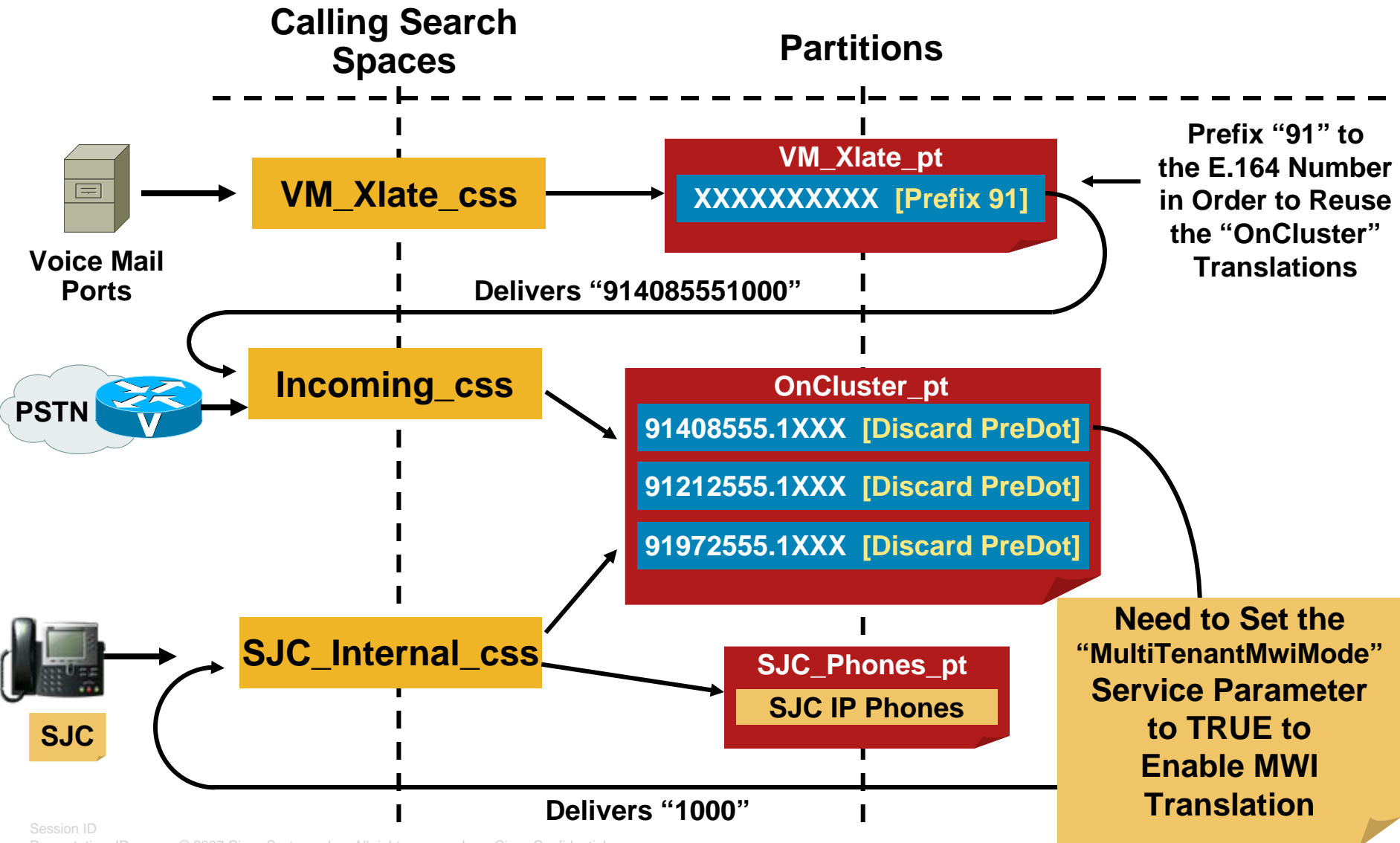
\* indicates required item

\*\* 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



# Design Best Practices Agenda

- Building Classes of Service

- **MultiSite Deployments**

  - Choosing a Dial Plan Approach

  - Uniform On-Net Dialing

  - Variable-Length On-Net Dialing with Partitioned Addressing

  - Variable-Length On-Net Dialing with Flat Addressing**

  - Tail End Hop Off (a.k.a. toll bypass)

- Mobility Considerations

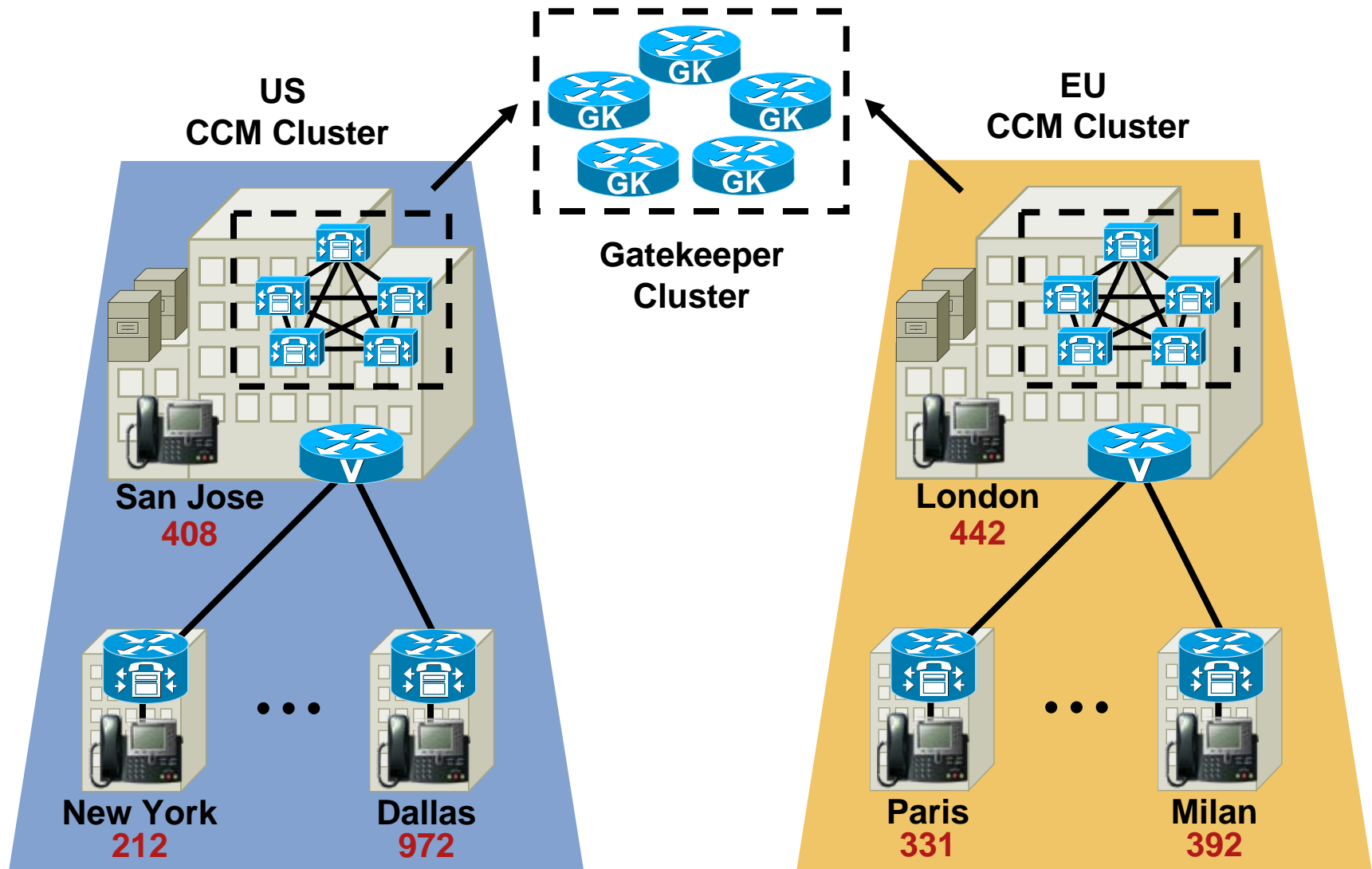
# VLOD with Flat Addressing

## 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

Partitions

Internal\_pt (contains all the phones)

SJC\_Xlations\_pt      NYC\_Xlations\_pt      DFW\_Xlations\_pt

Calling Search Spaces

SJC_Internal_css	NYC_Internal_css	DFW_Internal_css
SJC_Local_css	NYC_Local_css	DFW_Local_css
SJC_LD_css	NYC_LD_css	DFW_LD_css
SJC_Intl_css	NYC_Intl_css	DFW_Intl_css



...



**\* Note: If Using the Line/Device CSS Approach, the Number of CSS's 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 Name (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 1: Eight digit only**

  - Simple, easy to maintain

  - No automatic PSTN failover (manual redial)

- **Option 2: Eight digit + E.164 with centralized 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

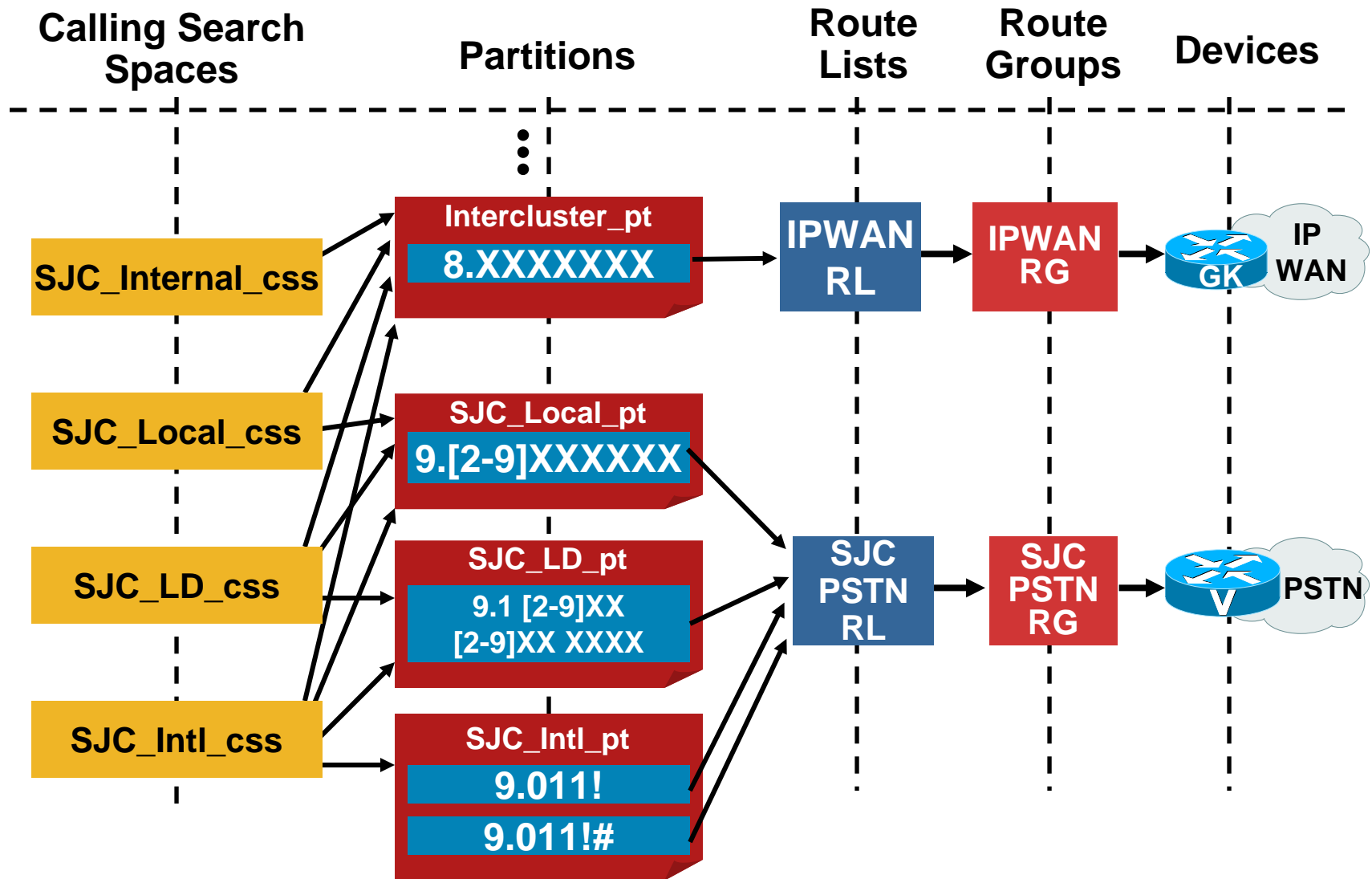
- **Option 3: Eight digit + E.164 with distributed PSTN failover**

  - A lot more configuration and maintenance

  - Automatic PSTN failover using local gateway

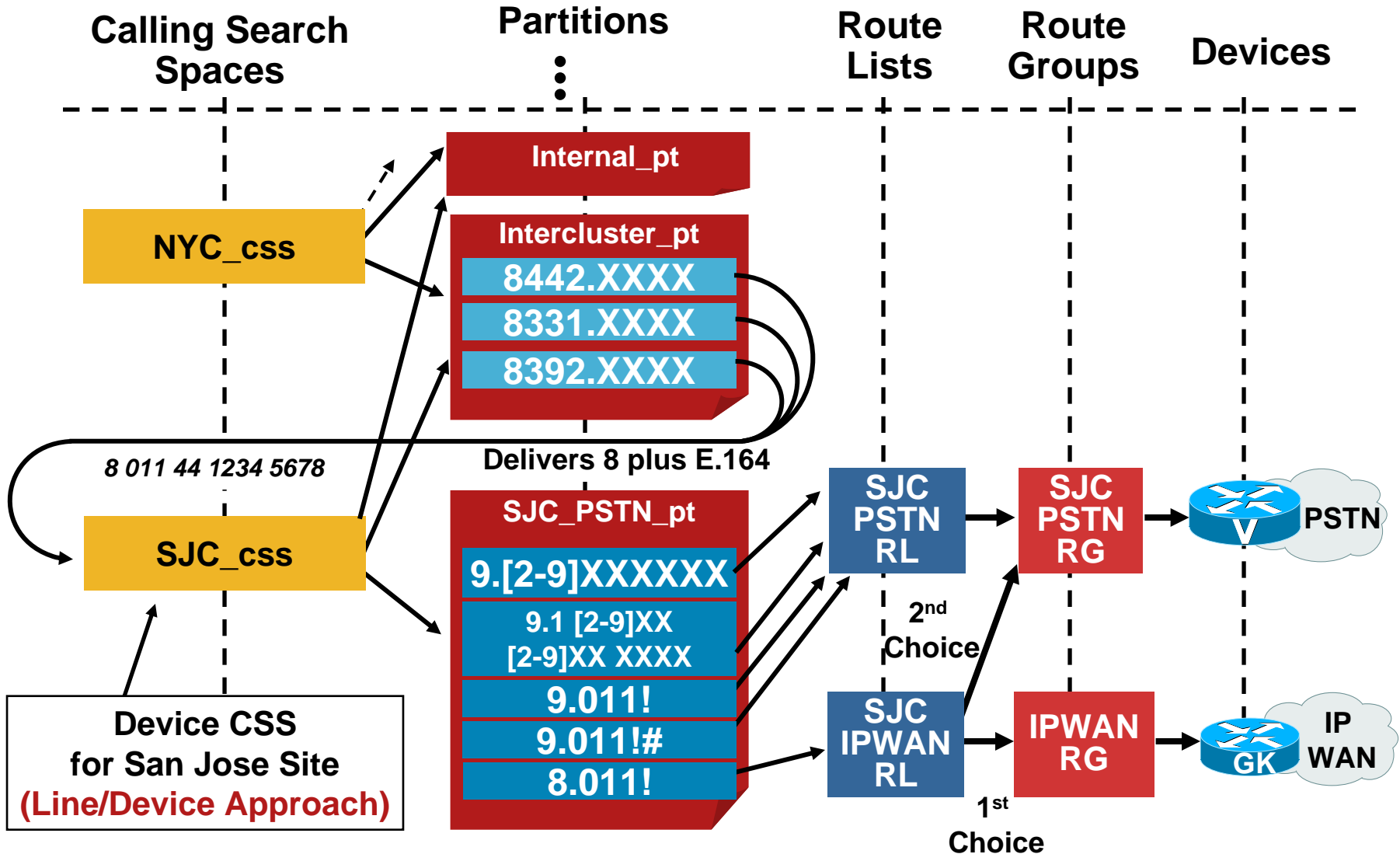
# VLOD with Flat Addressing

## Outgoing PSTN/IP WAN Calls: Option 1



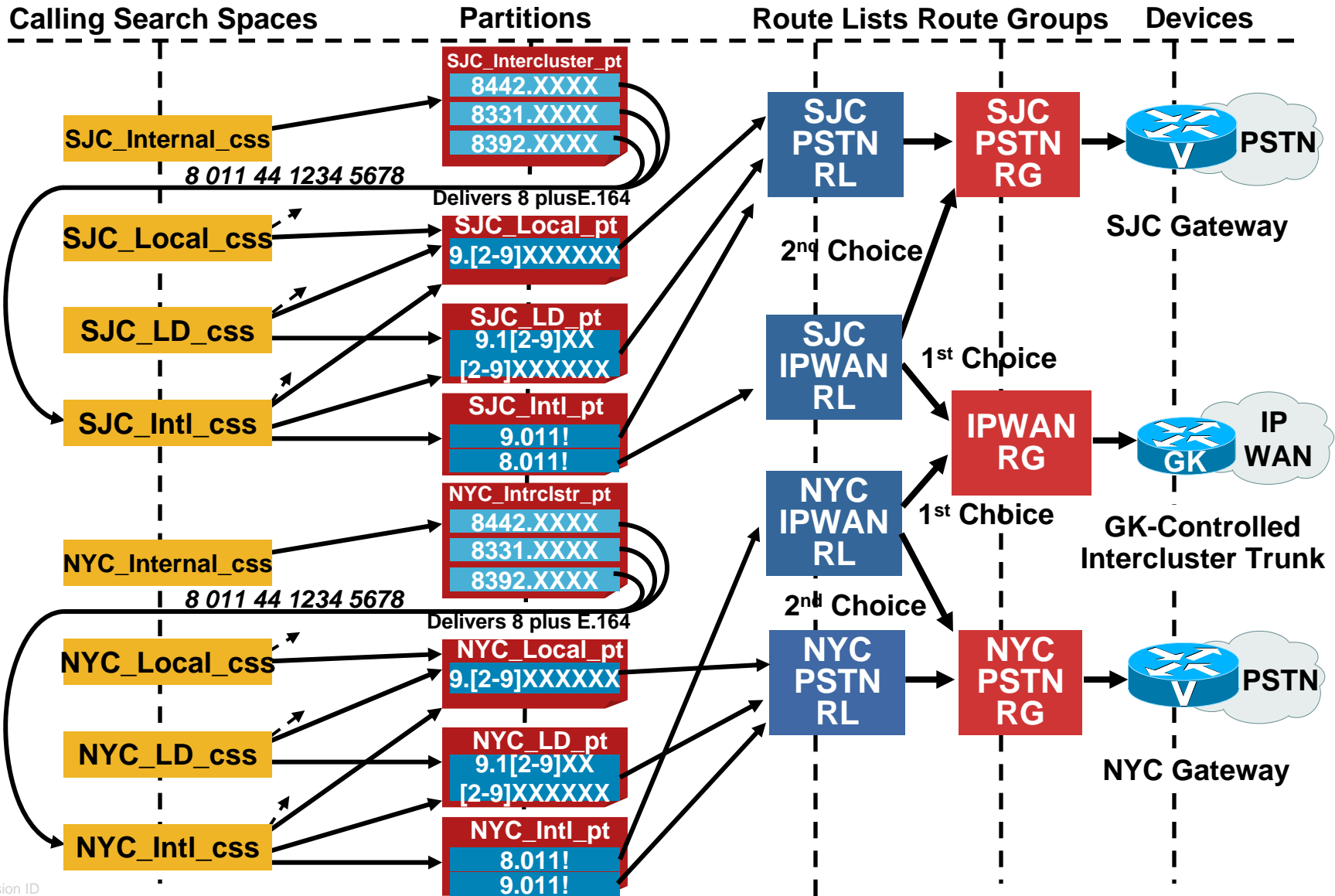
# VLOD with Flat Addressing

## Outgoing PSTN/IP WAN Calls: Option 2



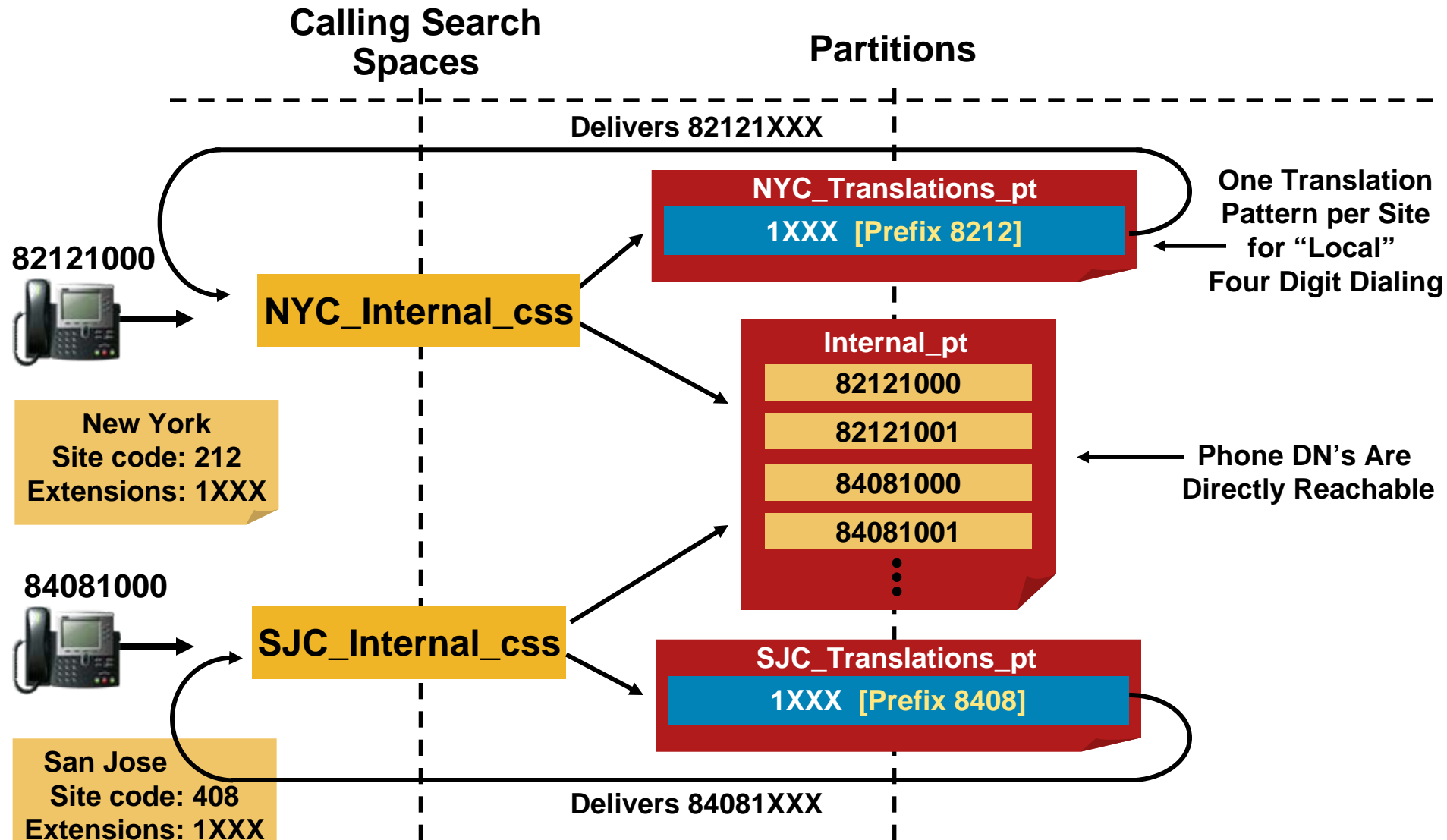
# VLOD with Flat Addressing

## Outgoing PSTN/IP WAN Calls: Option 3



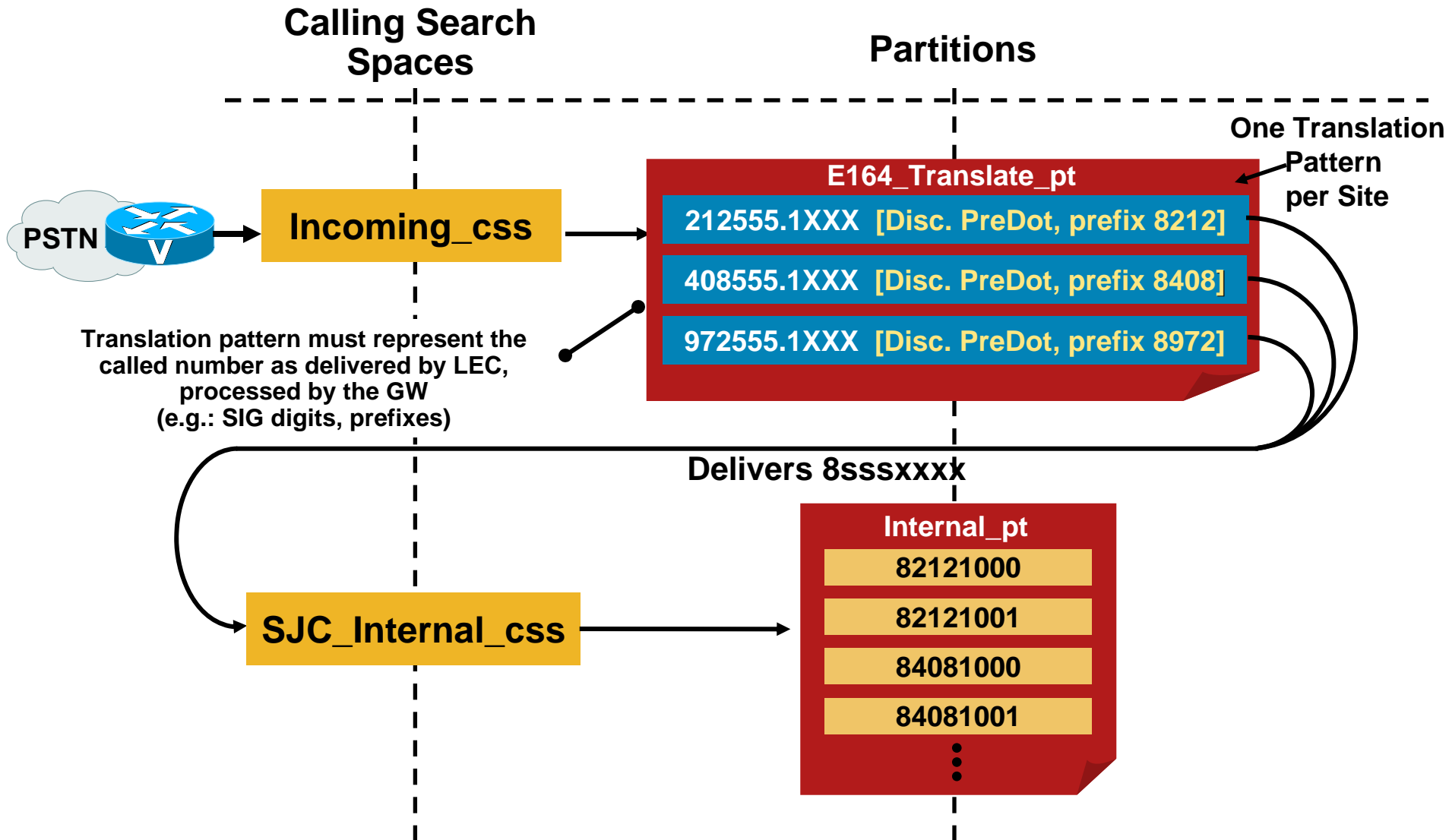
# VLOD with Flat Addressing

## Intra/Inter-site Calls Within a Cluster



# VLOD with Flat Addressing

## Incoming PSTN/IP WAN Calls



# VLOD with Flat Addressing

## Incoming PSTN/ IP WAN Calls

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

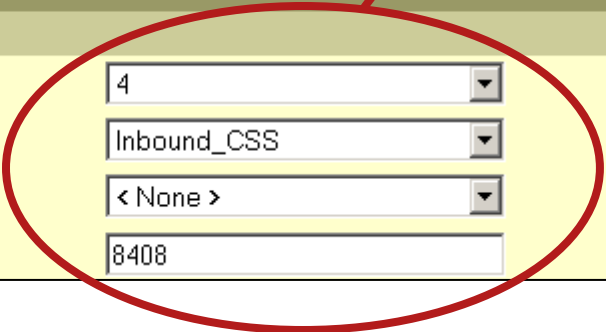
##### Inbound Calls

Significant Digits\*

Calling Search Space

AAR Calling Search Space

Prefix DN



**Configure GW to Strip and Prefix Relevant Digits**



# VLOD with Flat Addressing

## 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.164's 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...**

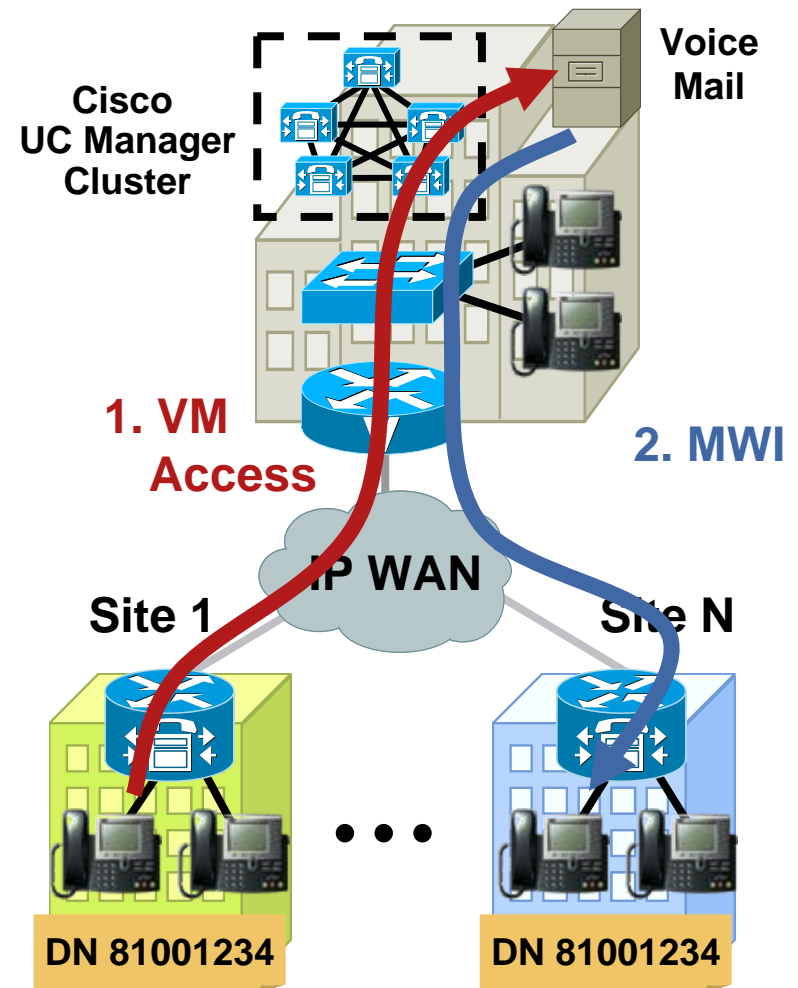
**zone prefix EU 83921...**

**!**

# 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

- Building Classes of Service

- **MultiSite Deployments**

  - Choosing a Dial Plan Approach

  - Uniform On-Net Dialing

  - Variable-Length On-Net Dialing with Partitioned Addressing

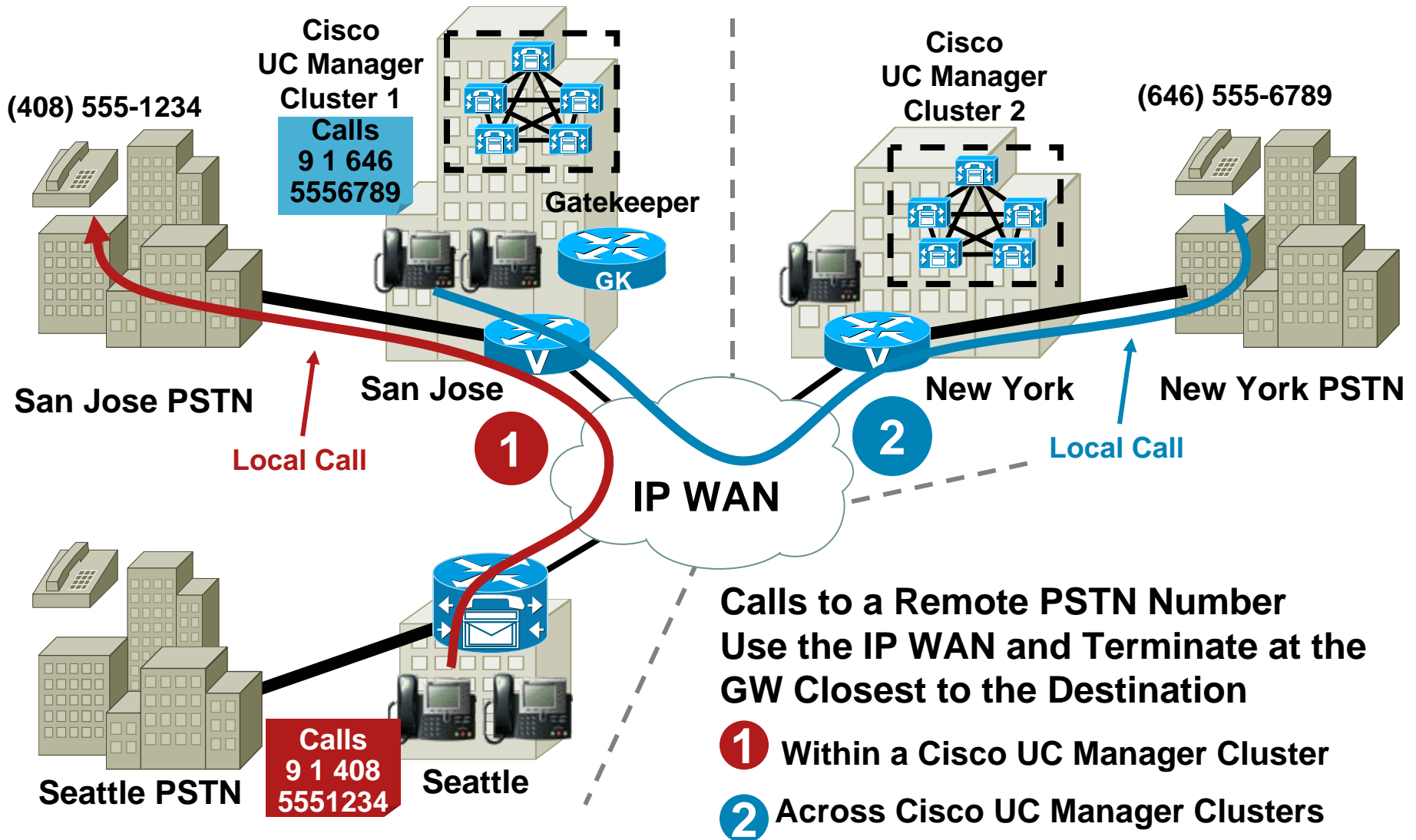
  - Variable-Length On-Net Dialing with Flat Addressing

  - Tail End Hop Off (a.k.a. toll bypass)**

- Mobility Considerations

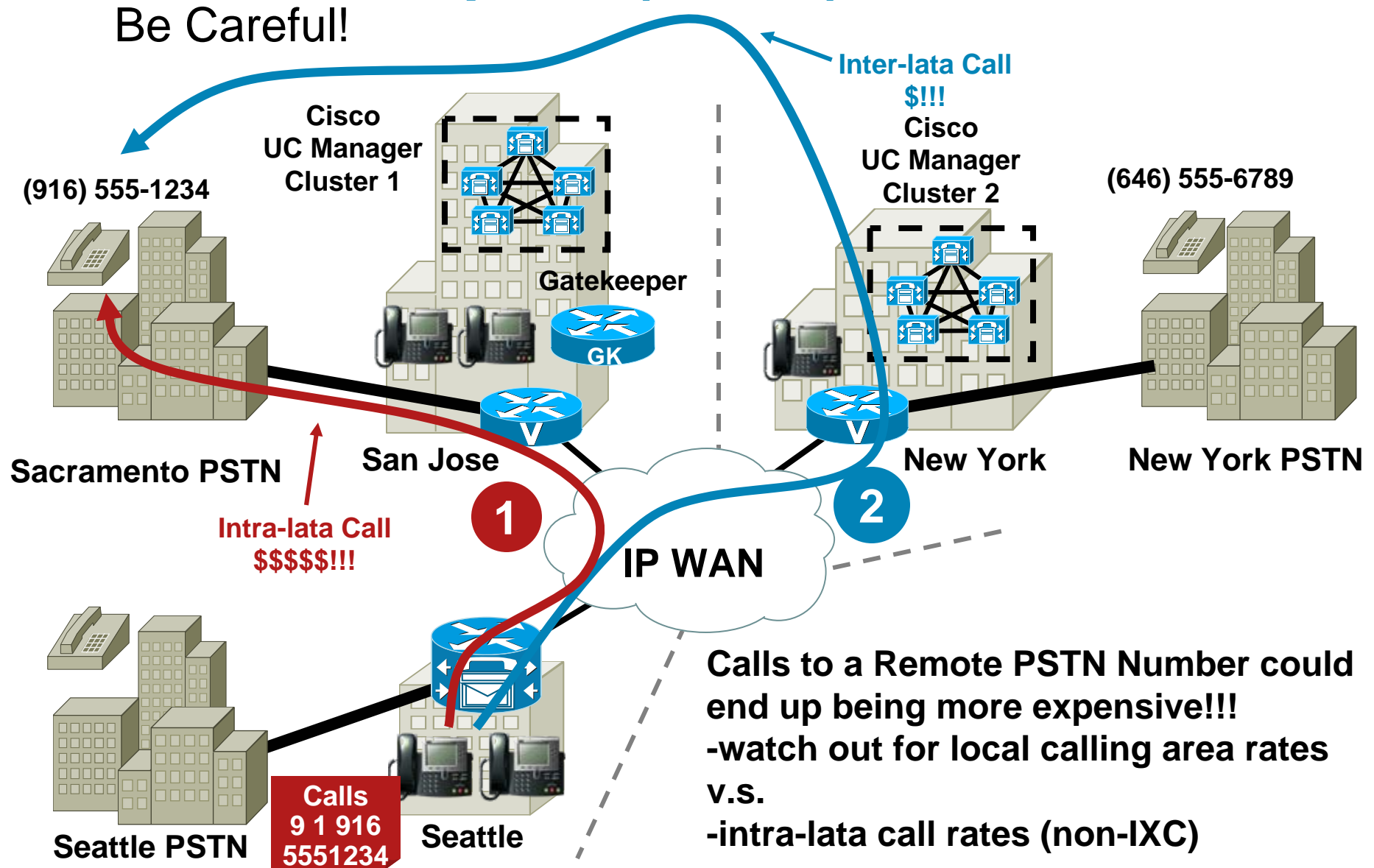
# Tail-End Hop-Off (TEHO)

What Is It?



# Tail-End Hop-Off (TEHO)

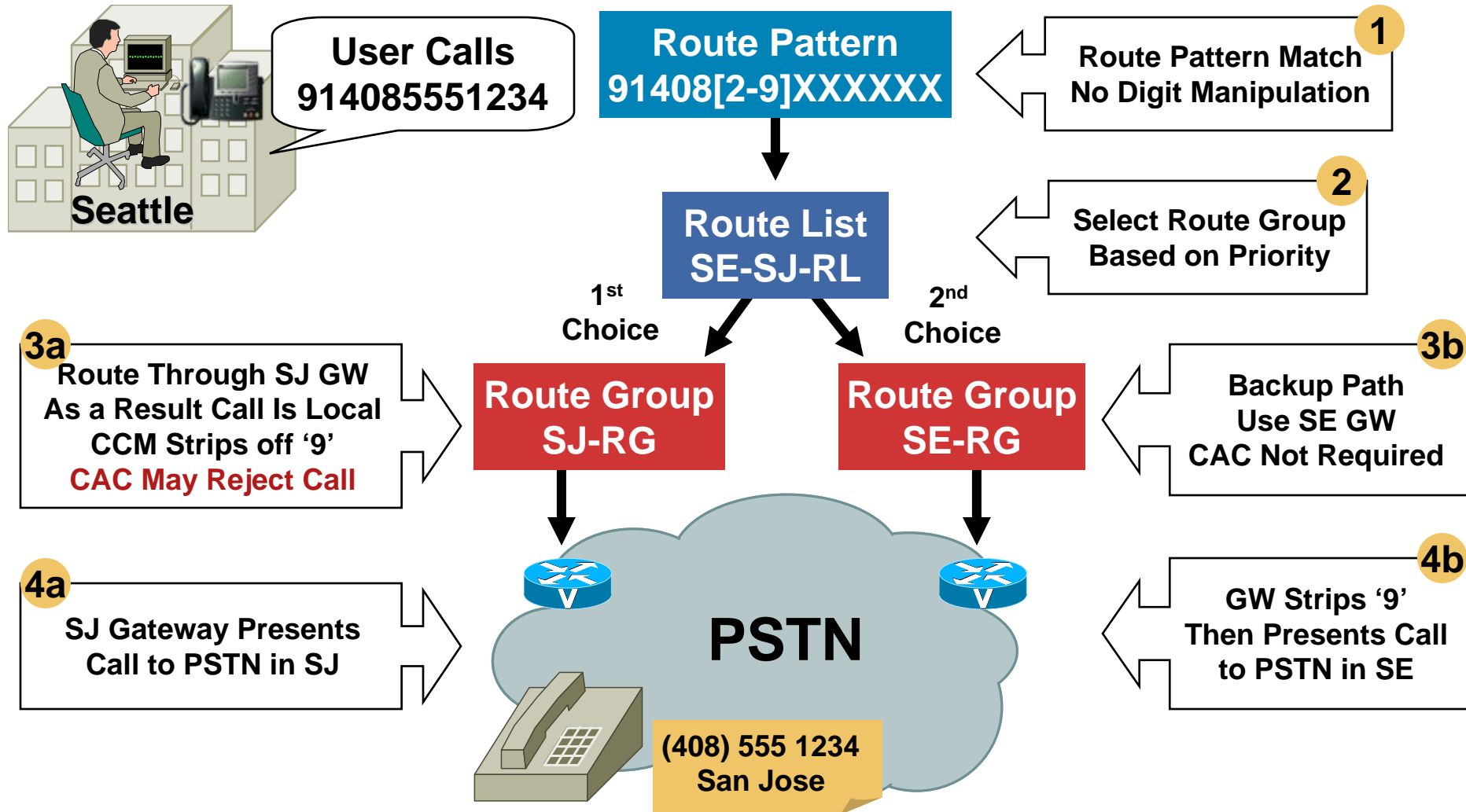
Be Careful!



**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)

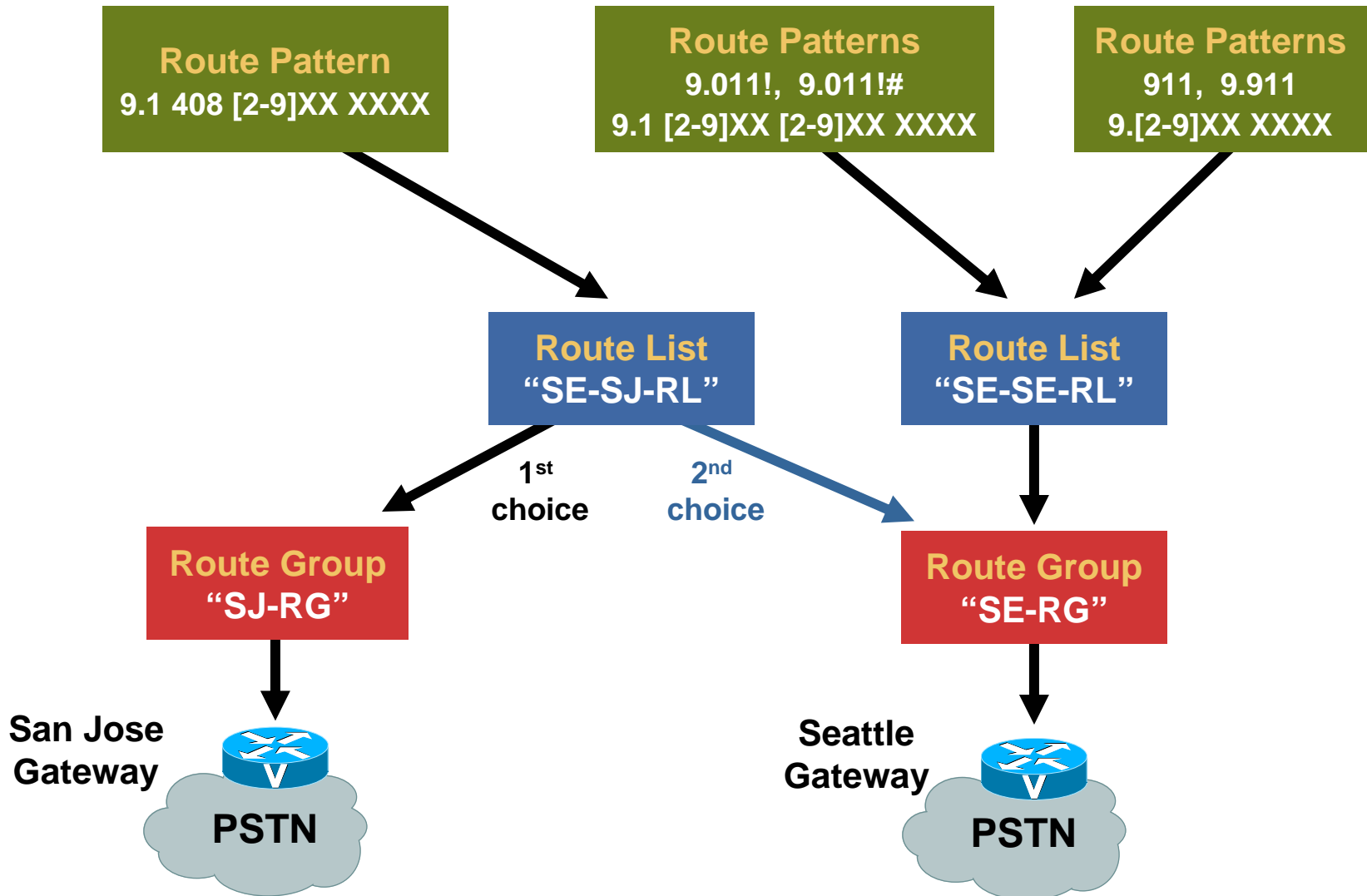
# Tail-End Hop-Off (TEHO)

Intracluster: Seattle to San Jose



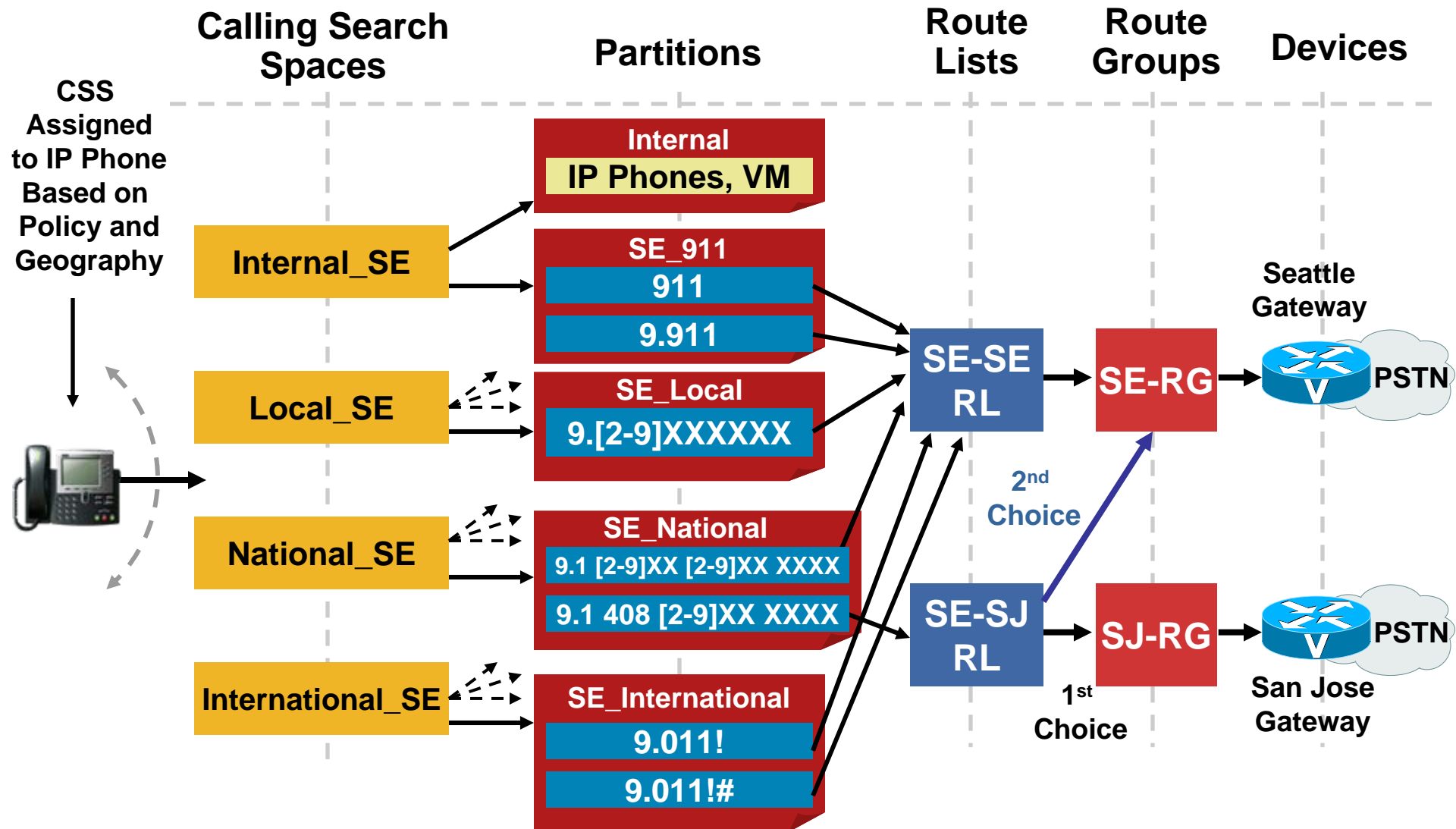
# Tail-End Hop-Off (TEHO)

## Intracluster: Route Patterns for Seattle



# Tail-End Hop-Off (TEHO)

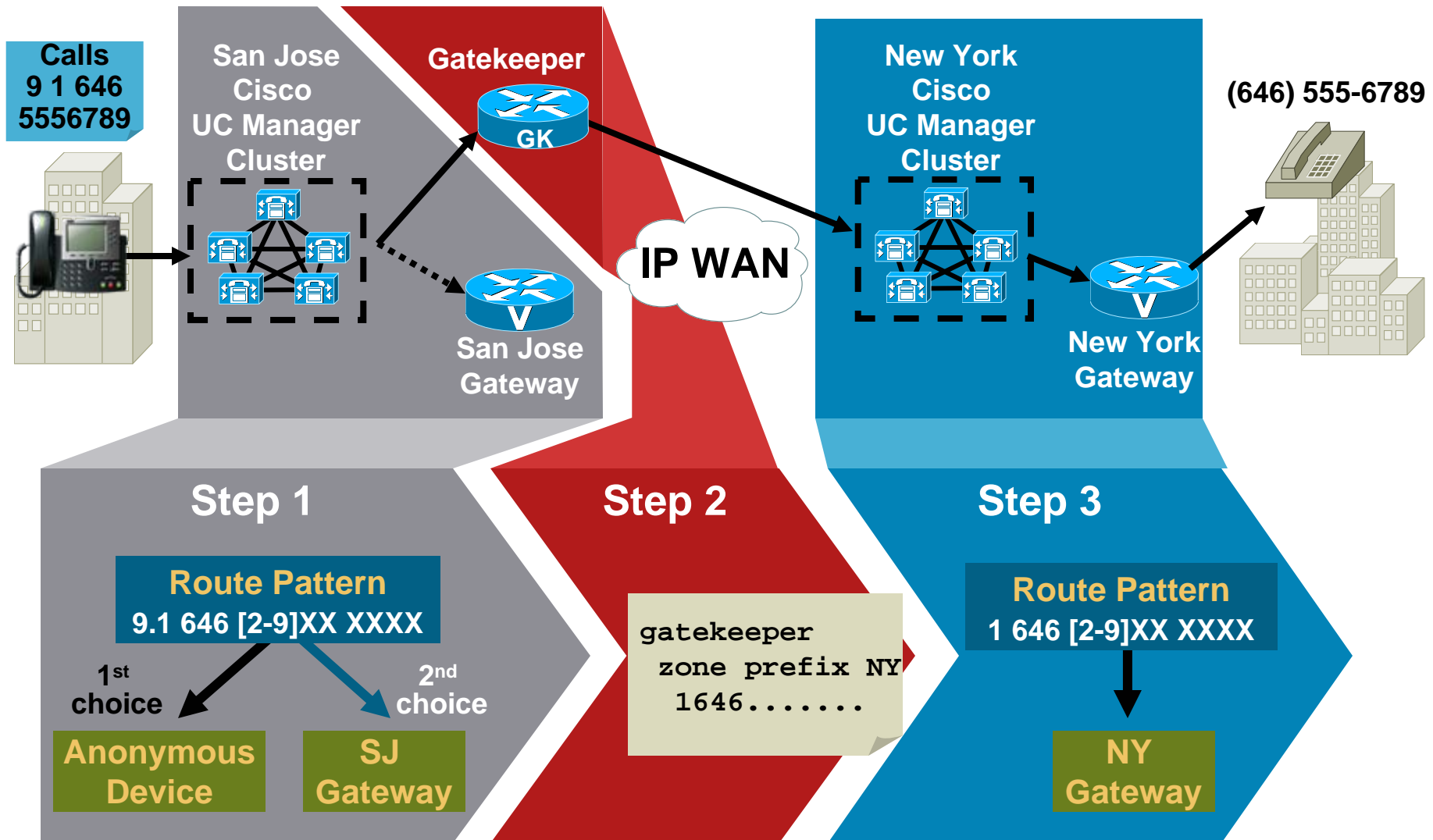
Intracluster: Composite Dial Plan for Seattle





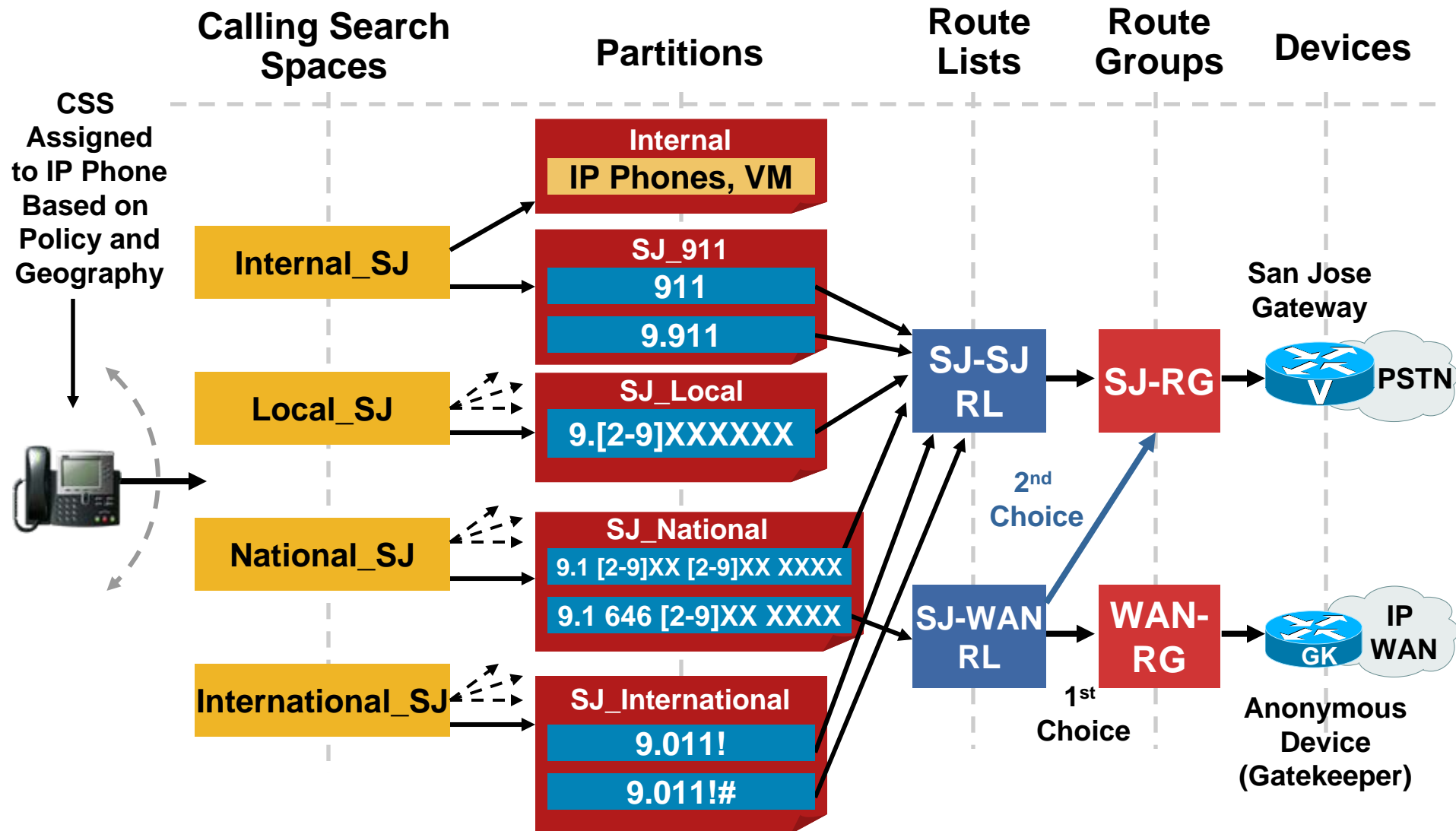
# Tail-End Hop-Off (TEHO)

Intercluster: San Jose to New York



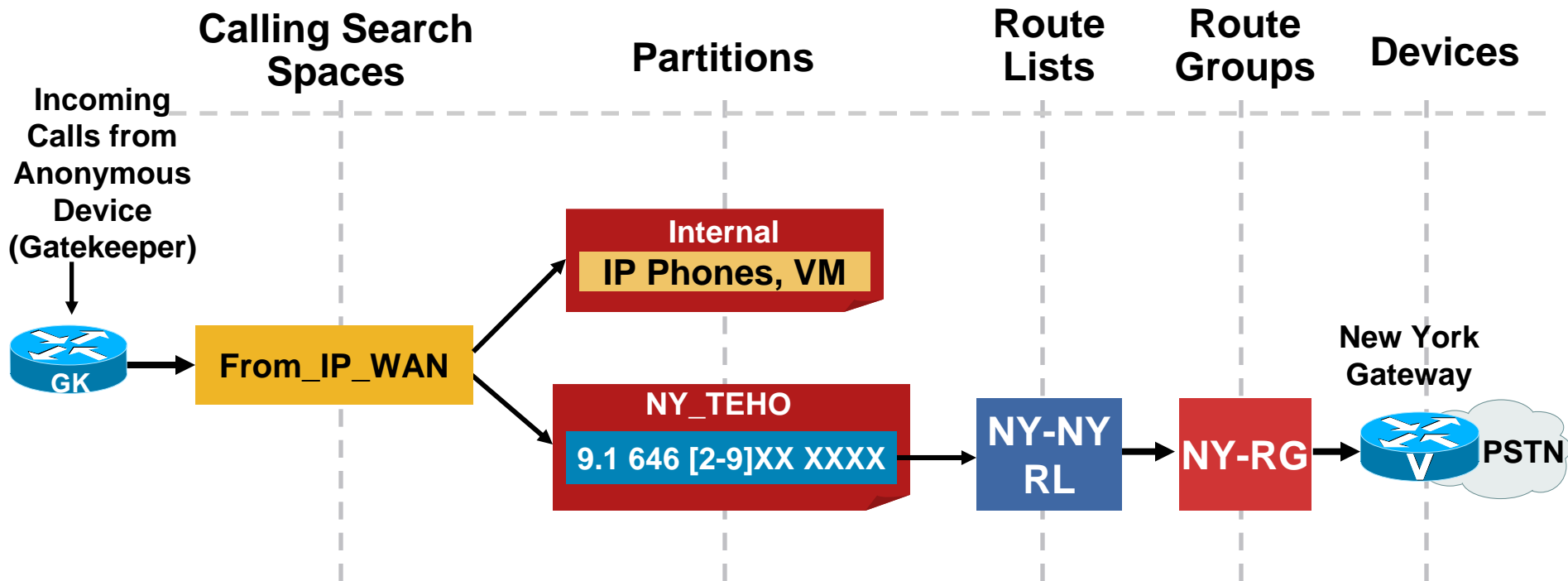
# Tail-End Hop-Off (TEHO)

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

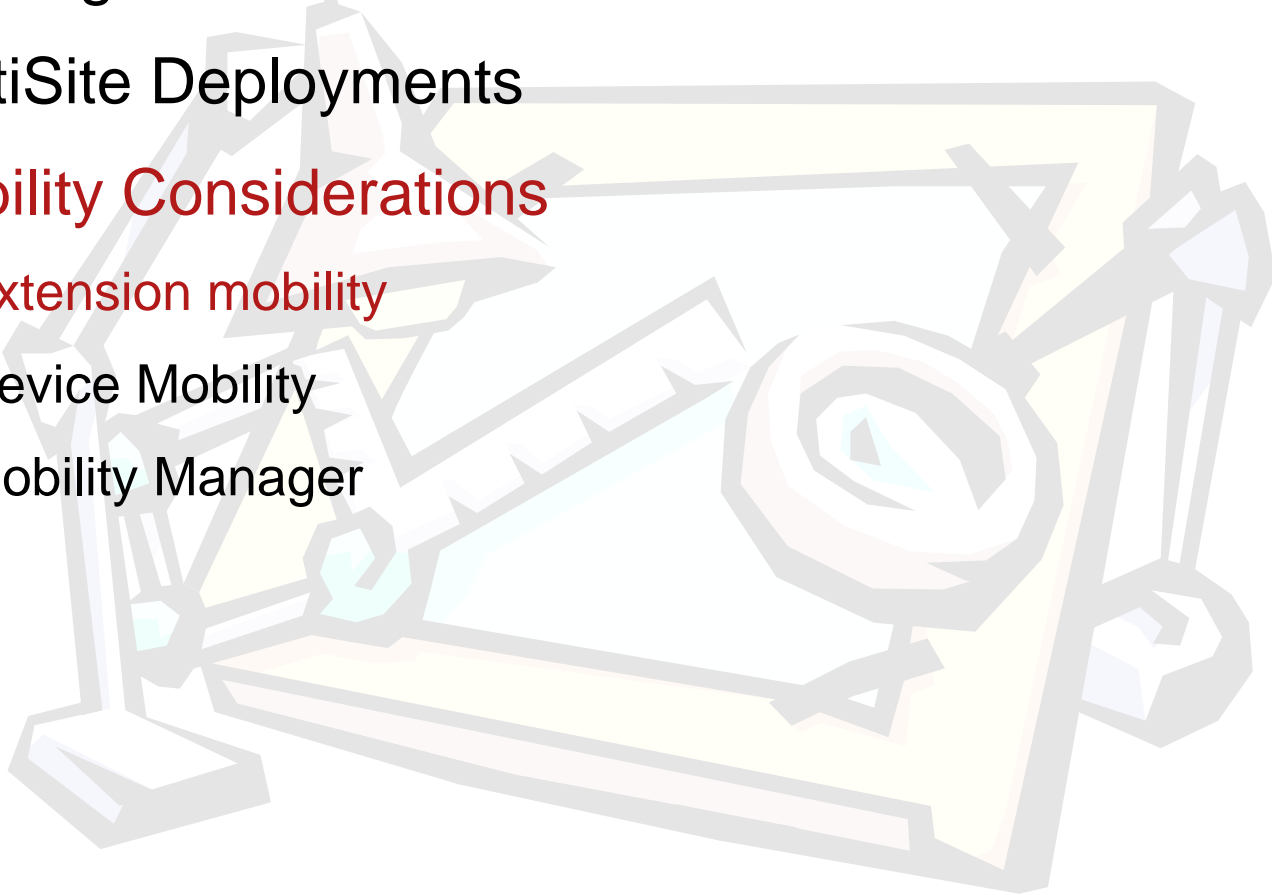
# Design Best Practices Agenda

- Building Classes of Service
- MultiSite Deployments
- **Mobility Considerations**

Extension mobility

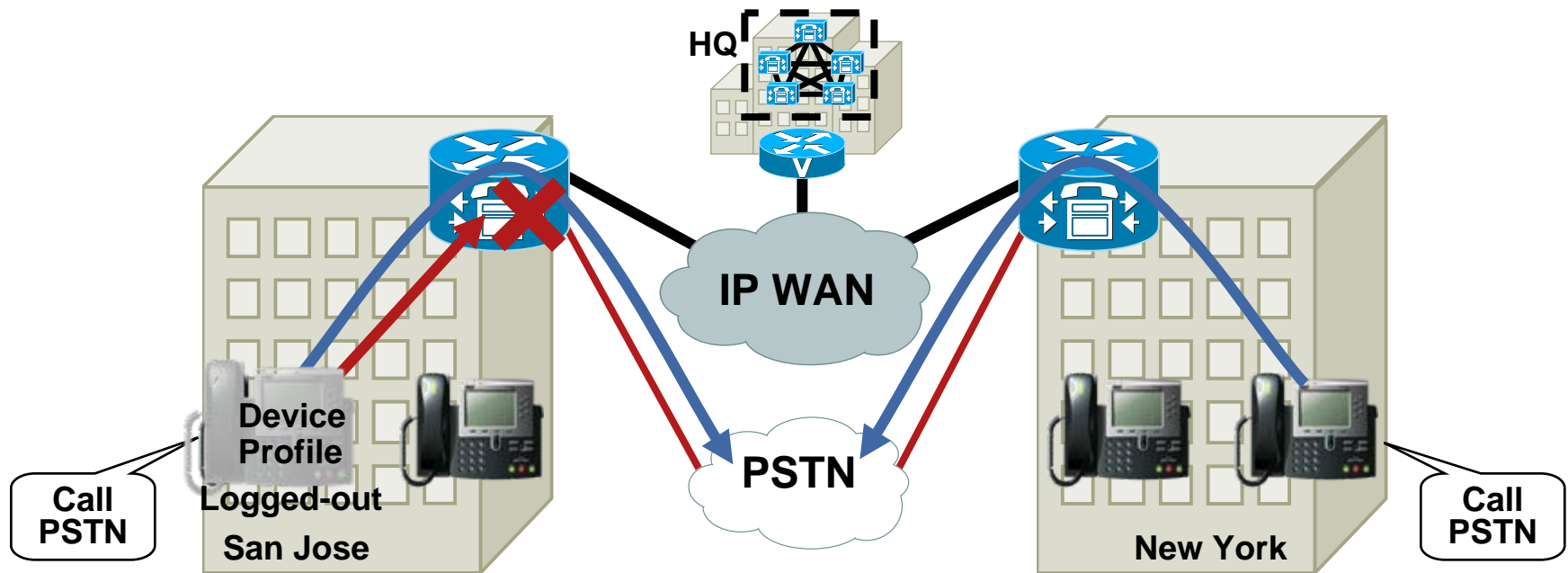
Device Mobility

Mobility Manager



# Extension Mobility Considerations

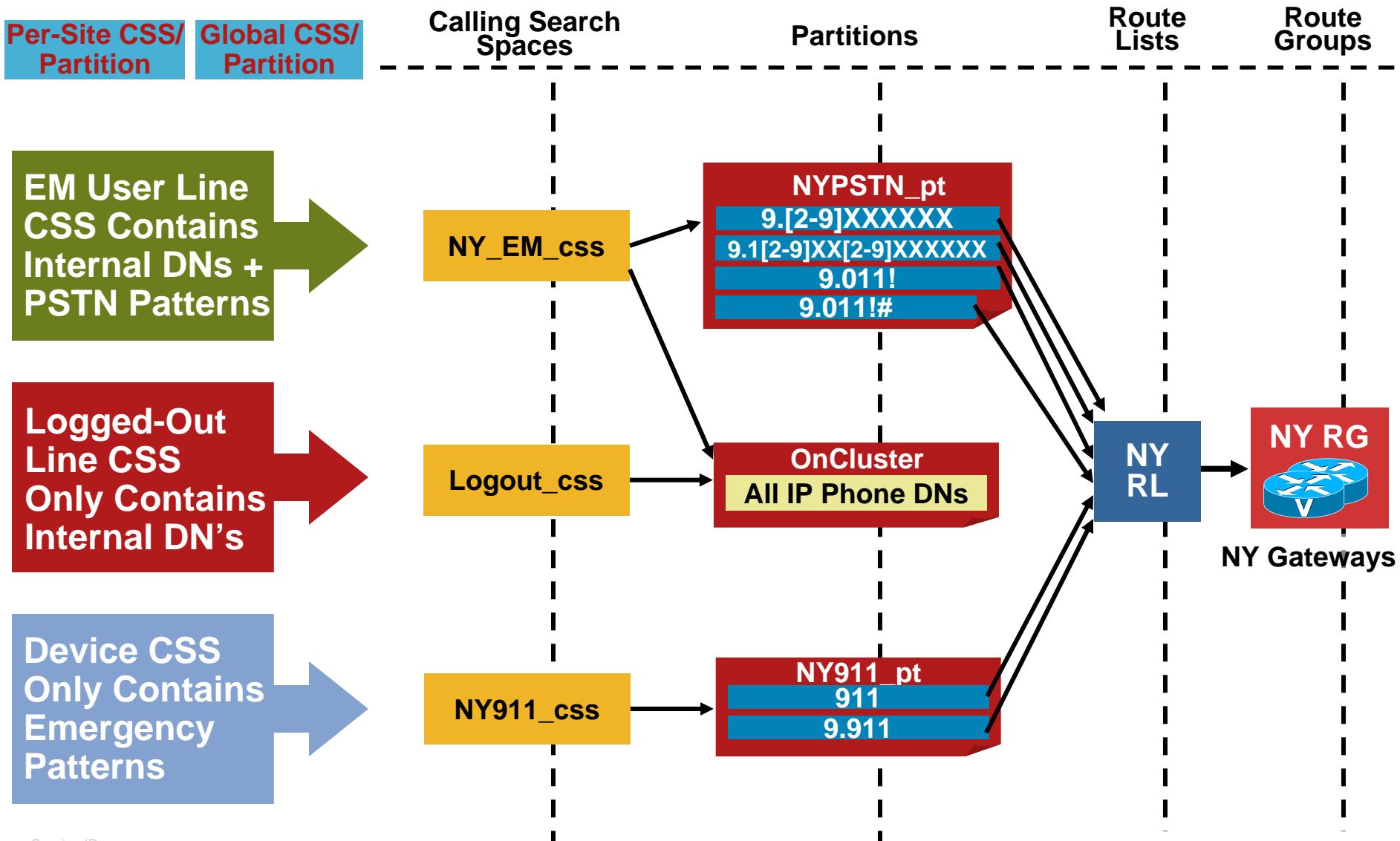
## Requirements



- 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**

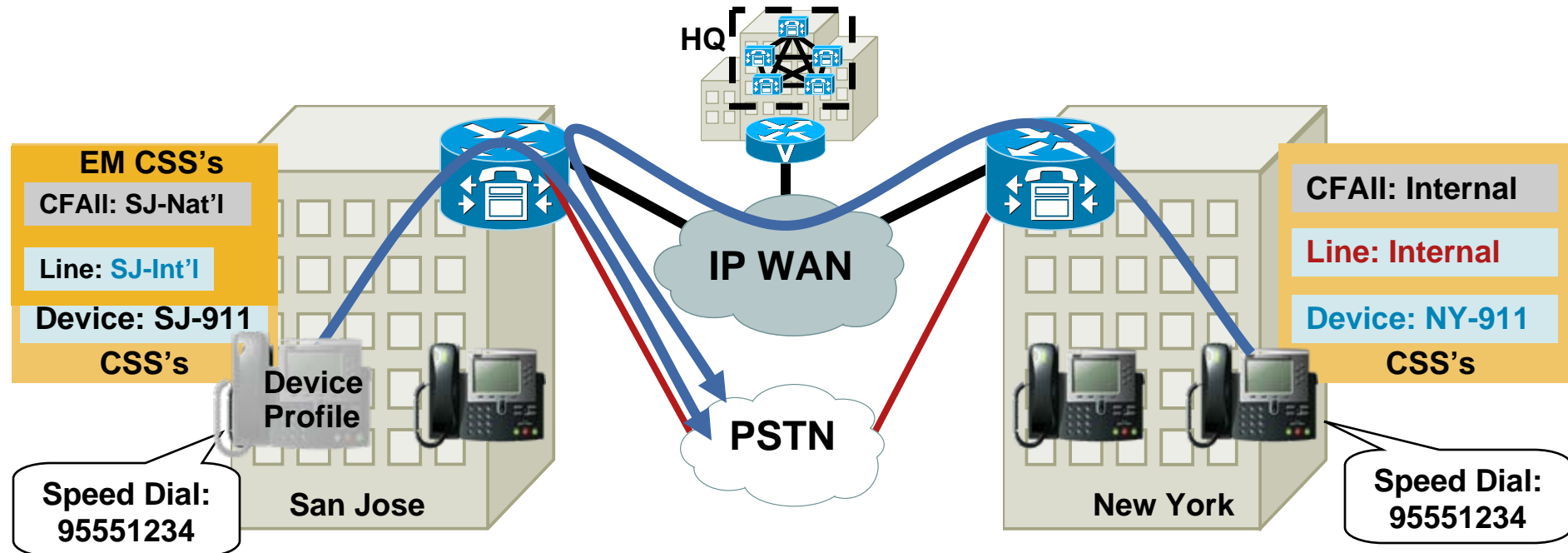
# Extension Mobility Considerations

## Traditional Dial Plan Approach



# Extension Mobility Considerations

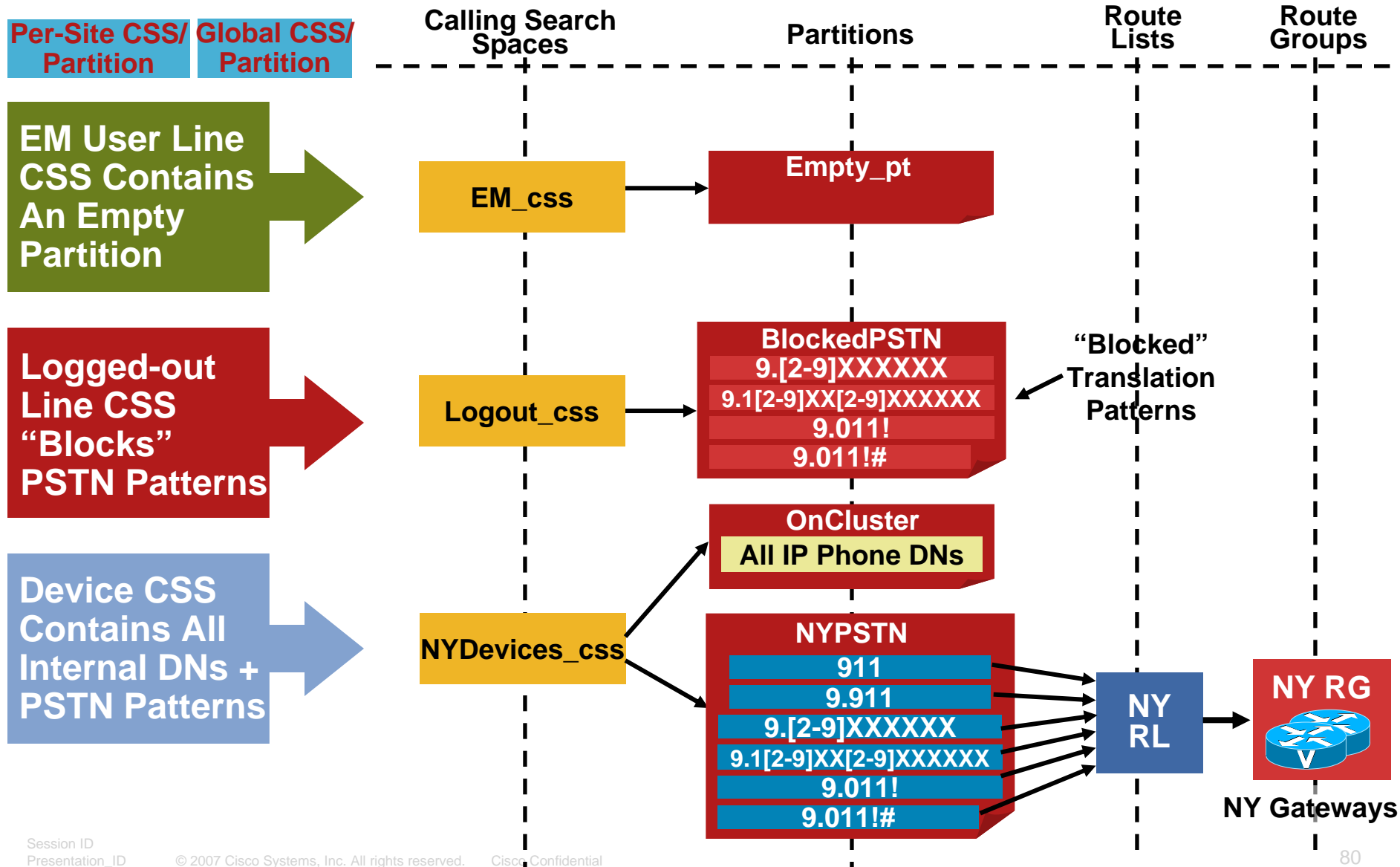
## Traditional Dial Plan Approach: Behavior



- 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

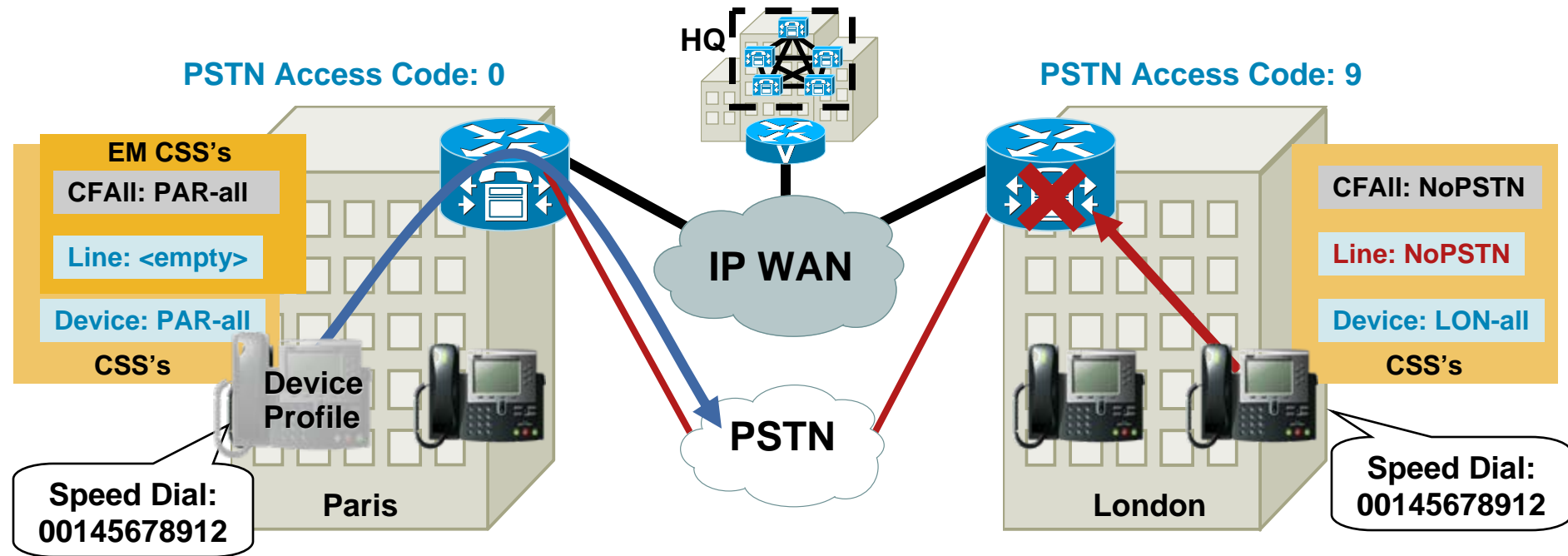
## Line/Device Dial Plan Approach





# Extension Mobility Considerations

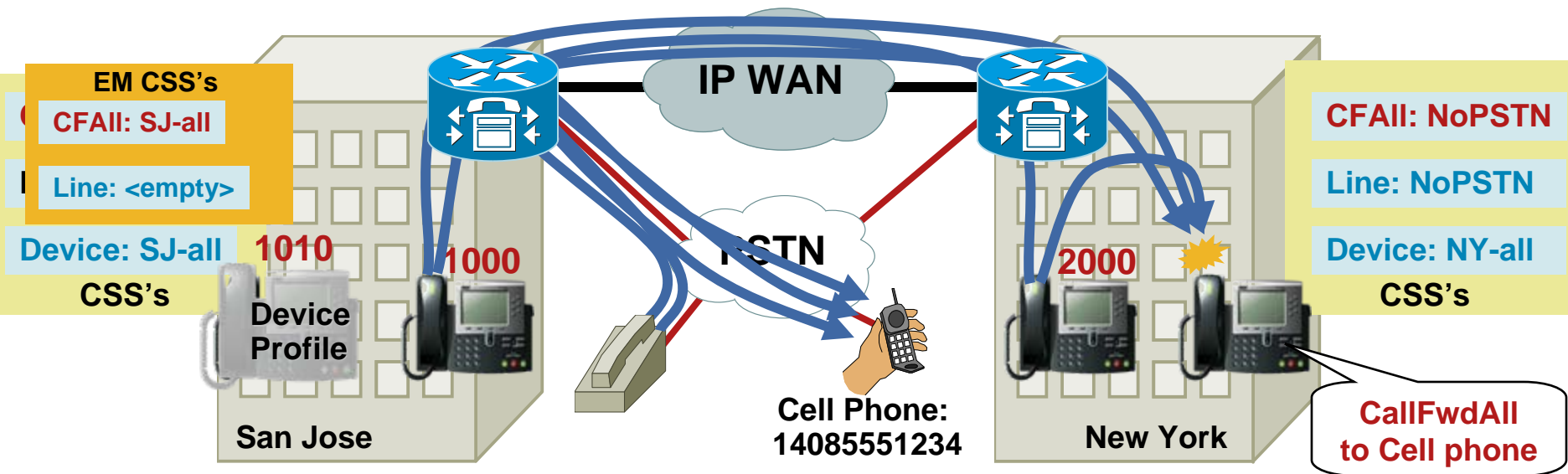
## Line/Device Dial Plan Approach: Behavior



- 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

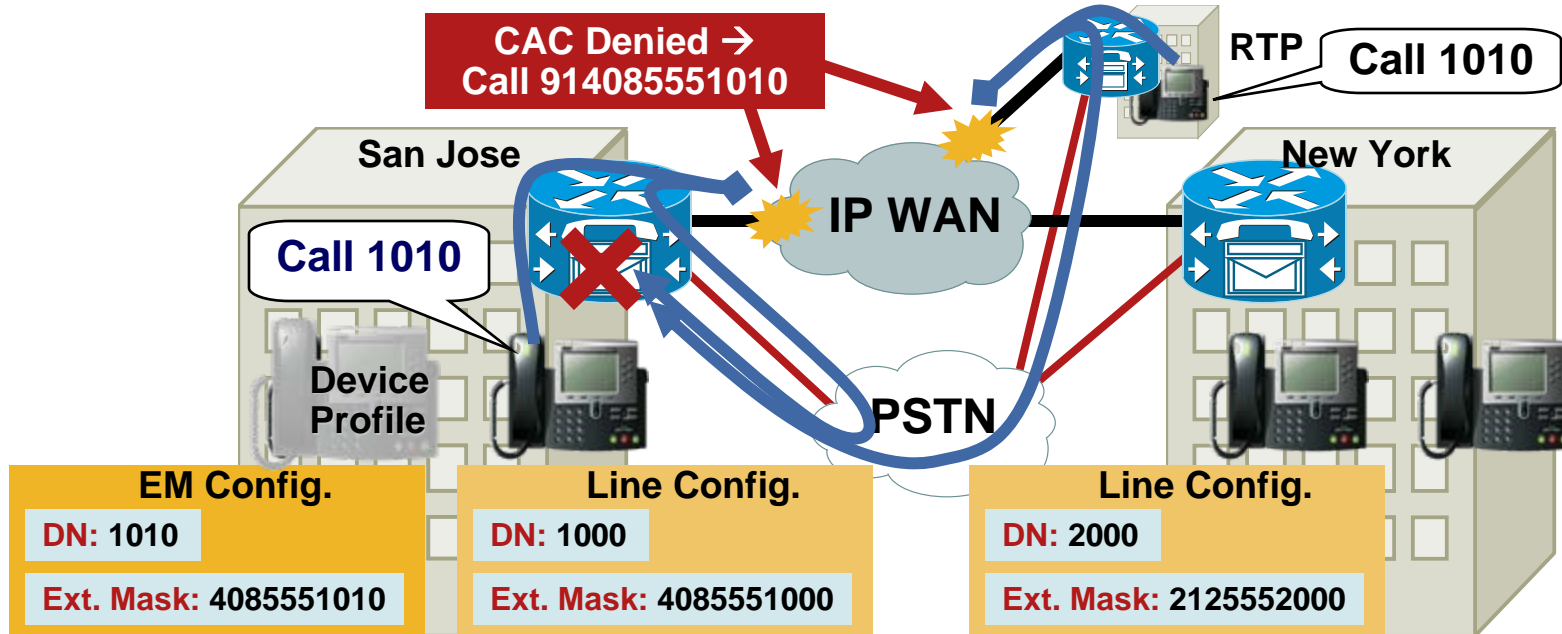


**When a SJ User Logs in at NY Site and Forwards His Phone to a PSTN Number:**

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

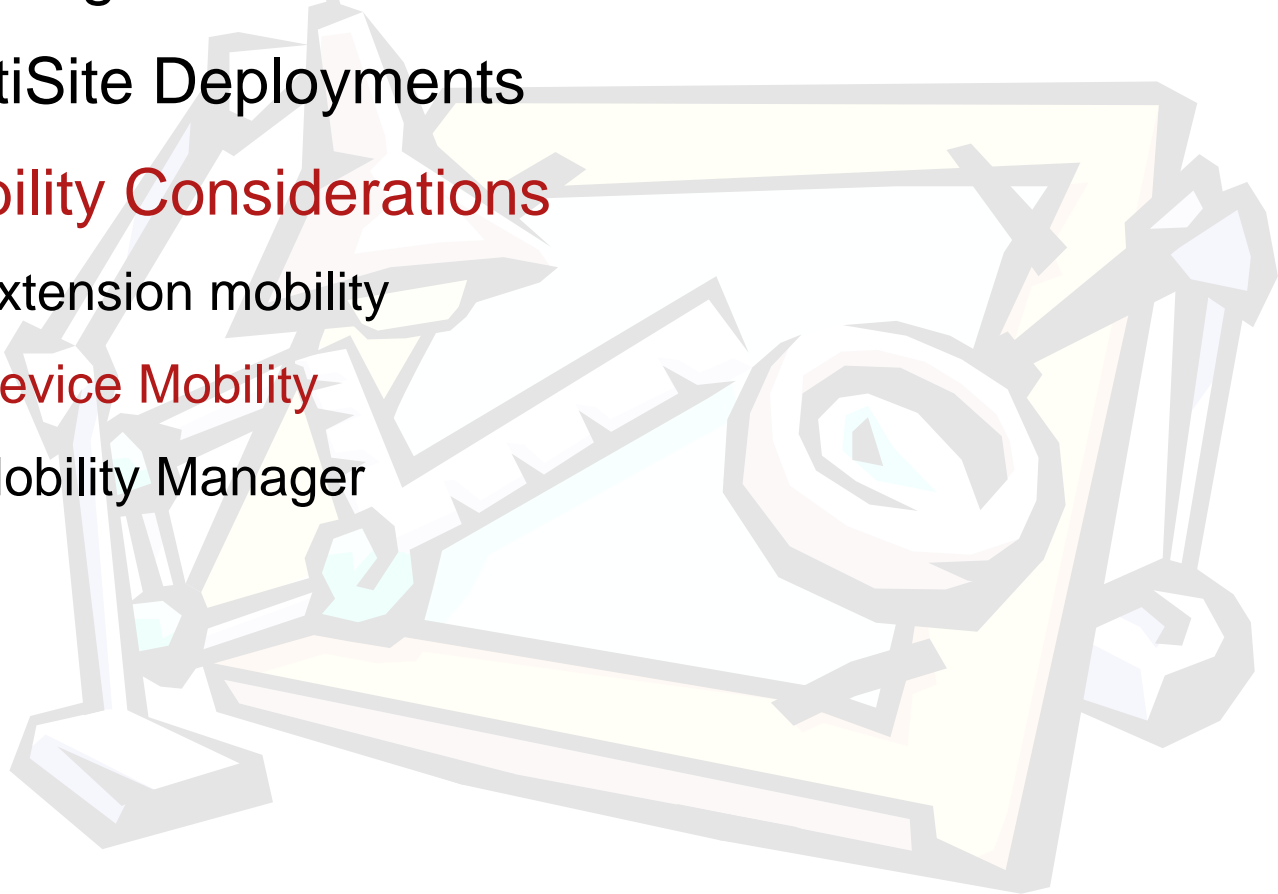
# Design Best Practices Agenda

- Building Classes of Service
- MultiSite Deployments
- **Mobility Considerations**

Extension mobility

**Device Mobility**

Mobility Manager



# Device Mobility Considerations

High-Level Behavior—UCM 4.2 and 6.0 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 UC Manager:

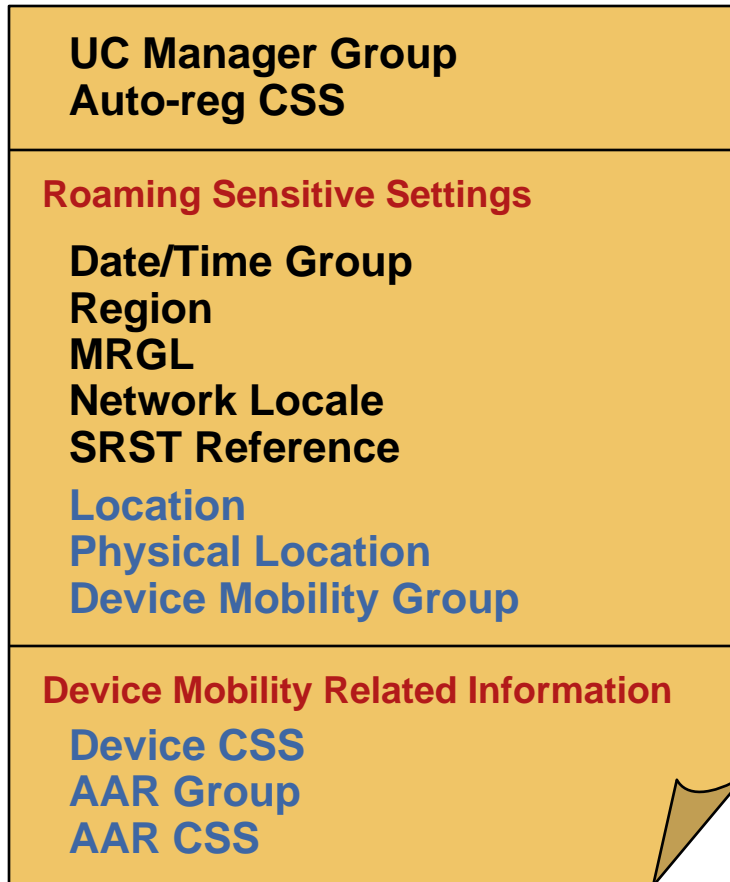
Device in Home Location

Device in Roaming Location

# Device Mobility

## Device Pool Changes

### Device Pool



### Common Profile (new)

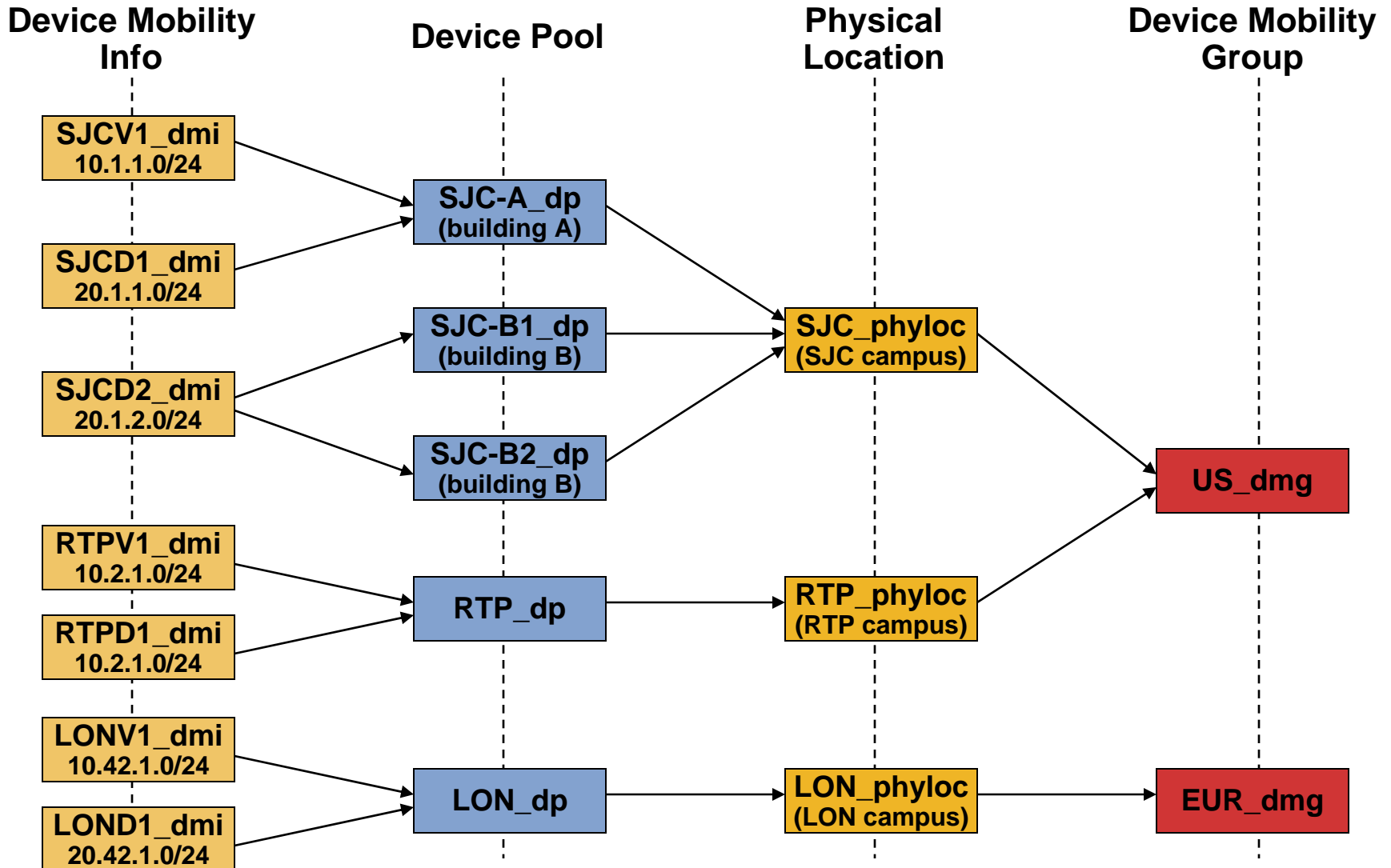
**Softkey Template**  
**Network Hold MoH Audio Source**  
**User Hold MoH Audio Source**  
**MLPP Indication**  
**MLPP Preemption**  
**MLPP Domain**



Device

# Device Mobility

## New Concepts



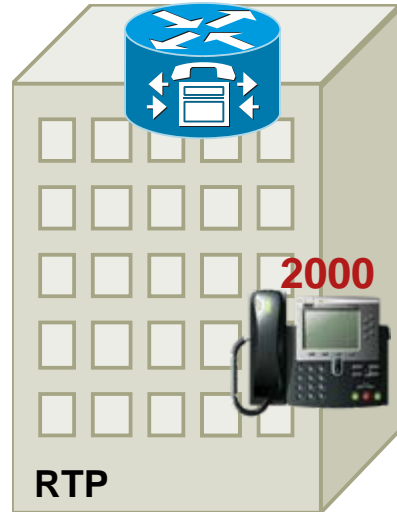
# Device Mobility Considerations

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



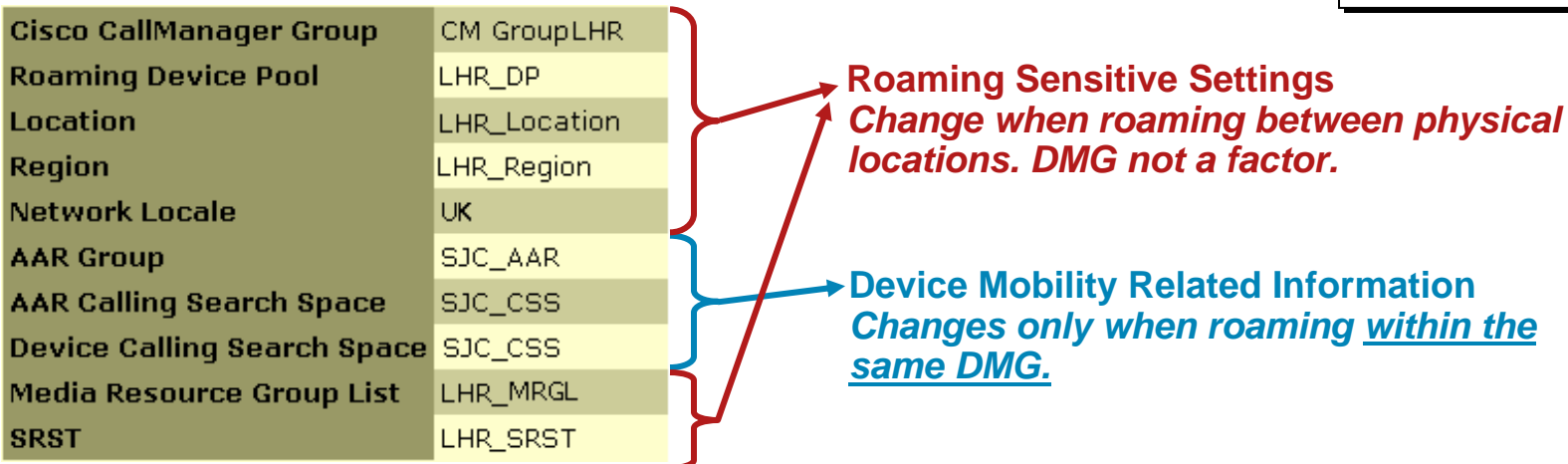
voice subnet: 10.2.1.0/24  
data subnet: 20.2.1.0/24



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.





# 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)

Device security mode only takes effect if the parameter Cluster Security Mode is set to On

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



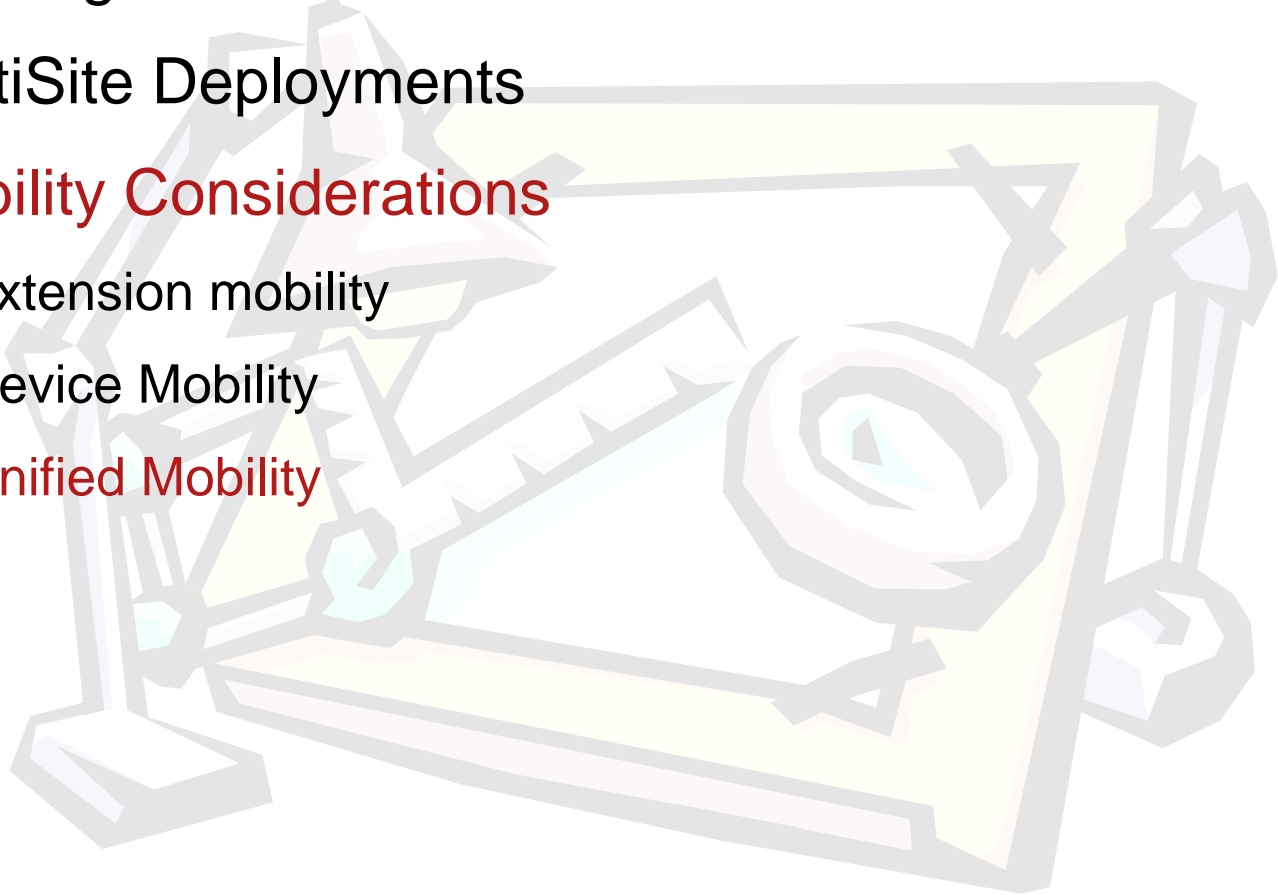
# Design Best Practices Agenda

- Building Classes of Service
- MultiSite Deployments
- **Mobility Considerations**

Extension mobility

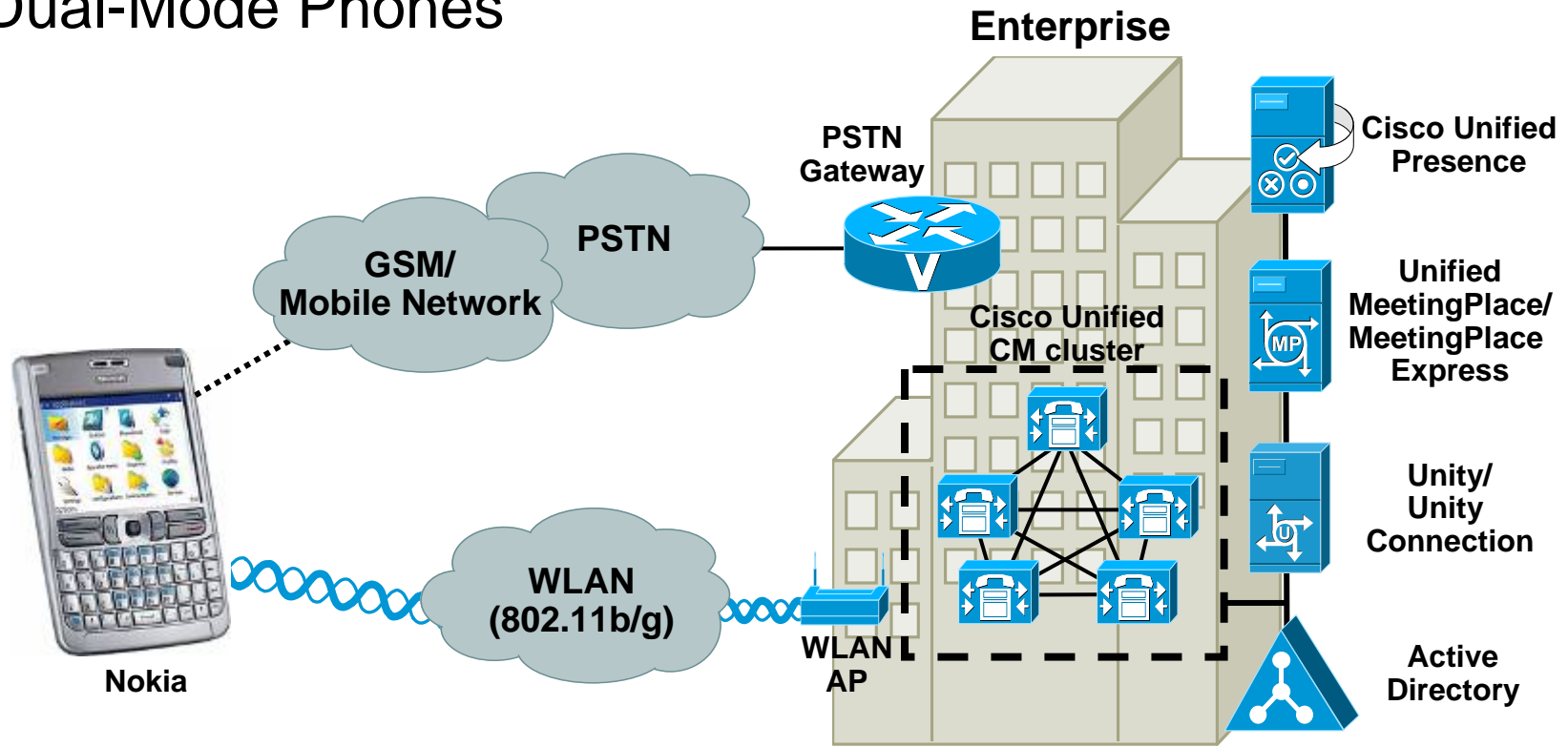
Device Mobility

**Unified Mobility**



# Campus Mobility

## Dual-Mode Phones

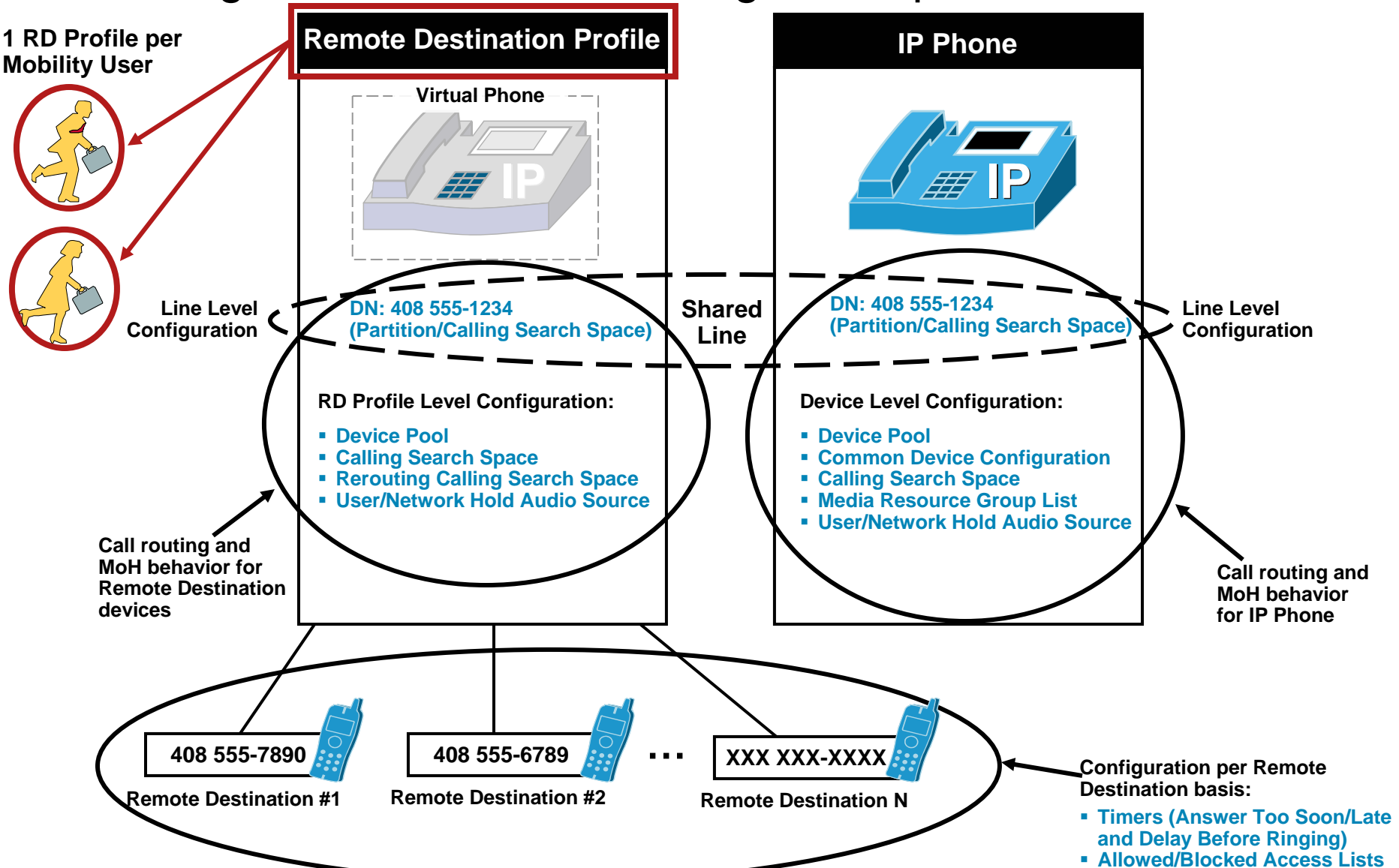


**Dual-mode phones provide the ability to use either PSTN/GSM or WLAN connectivity for making and receiving calls**

- When on the WLAN, the mobile phone uses SCCP or SIP Cisco client to register with CUCM as a phone
- When the WLAN is unavailable, the mobile phone uses PSTN/GSM for calls
- Manual handoff of calls between the PSTN/GSM and WLAN network is possible

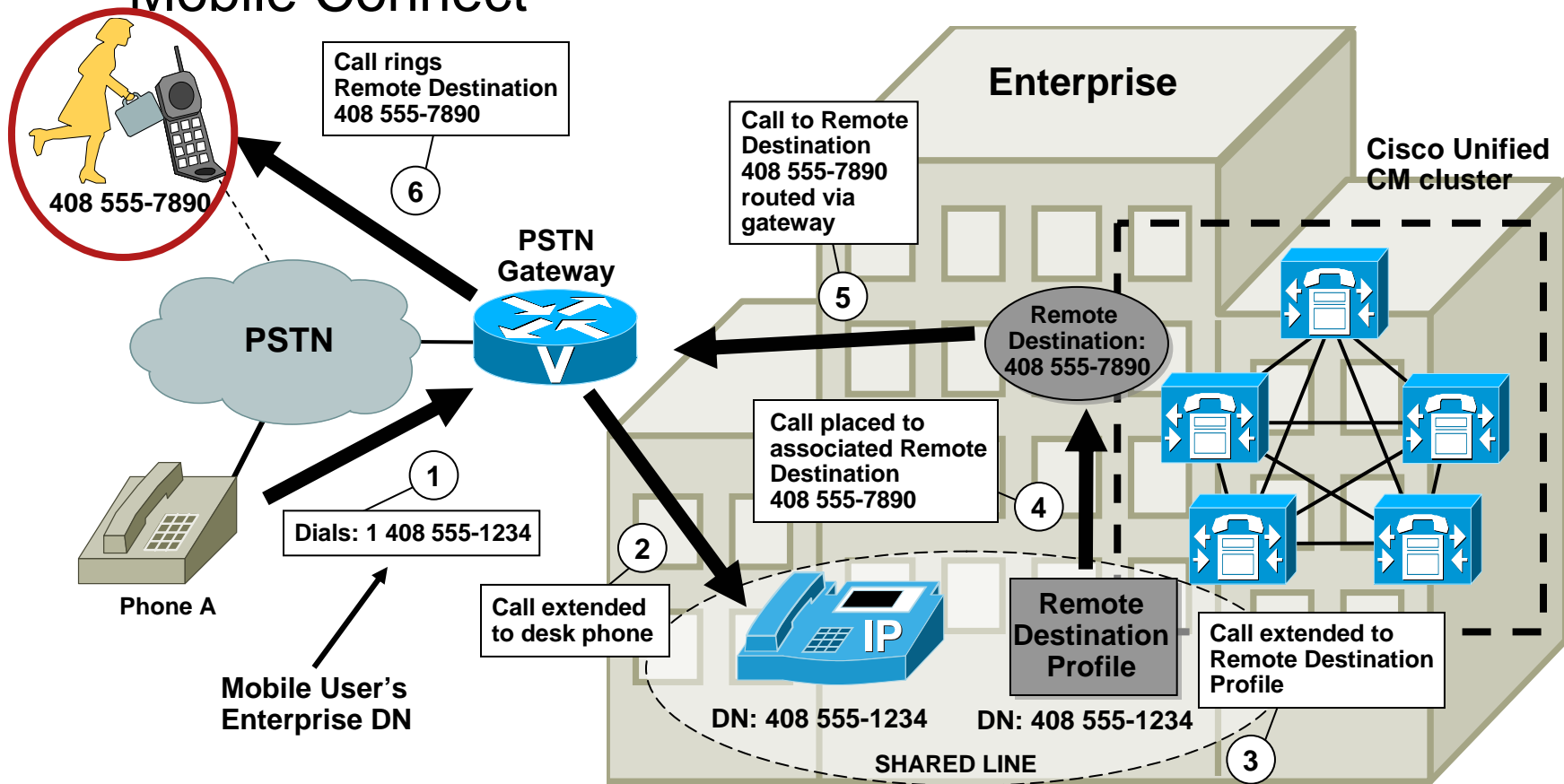
# Unified Mobility

## Configuration and Call Routing Concept



# Unified Mobility

## Mobile Connect



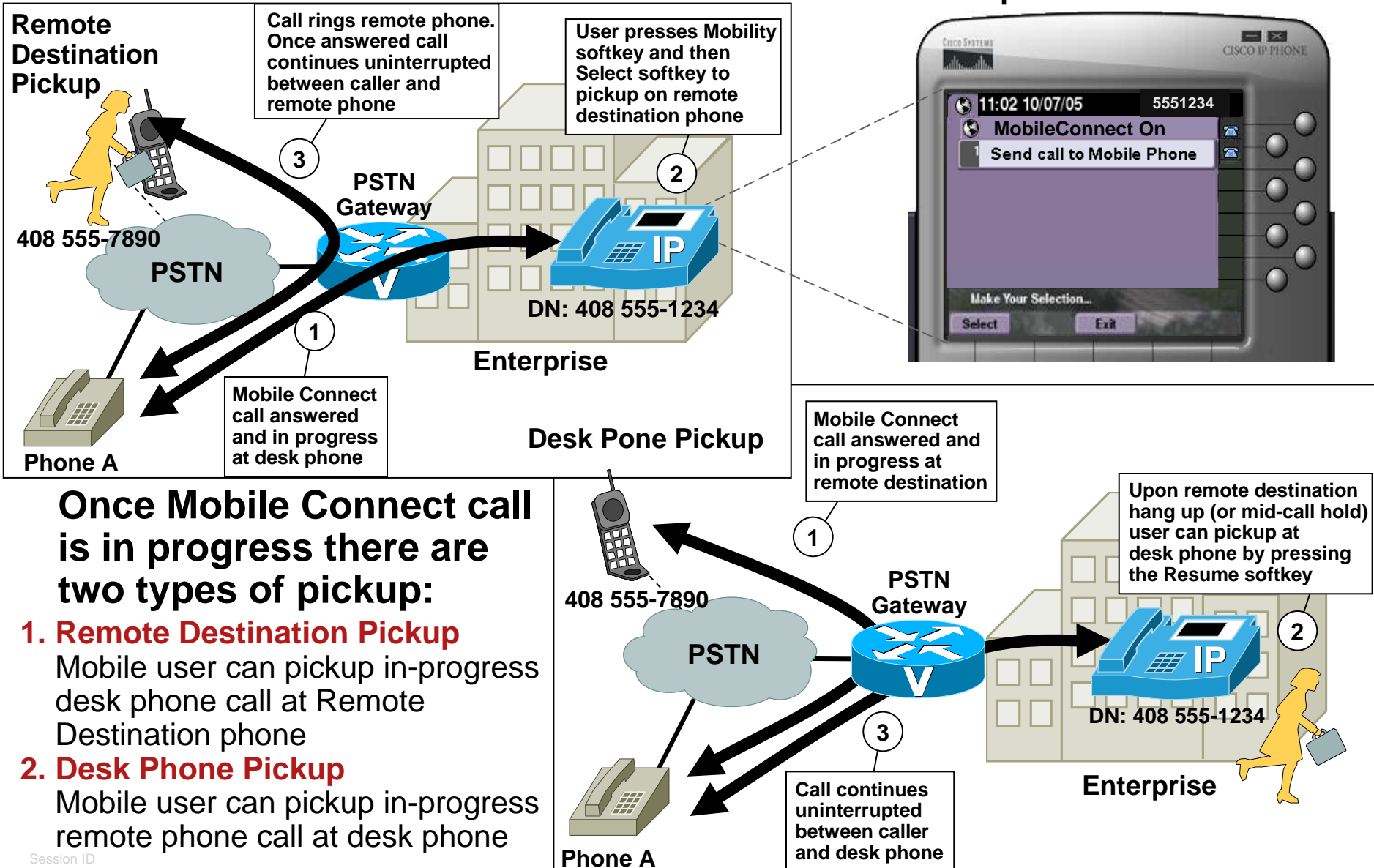
### Call to mobile user's Enterprise directory number rings at desk phone and Remote Destination phone:

- Call can be answered at either phone
- Once answered all other call legs are cleared

**Note: No changes are required on mobility user's Remote Destination phone**

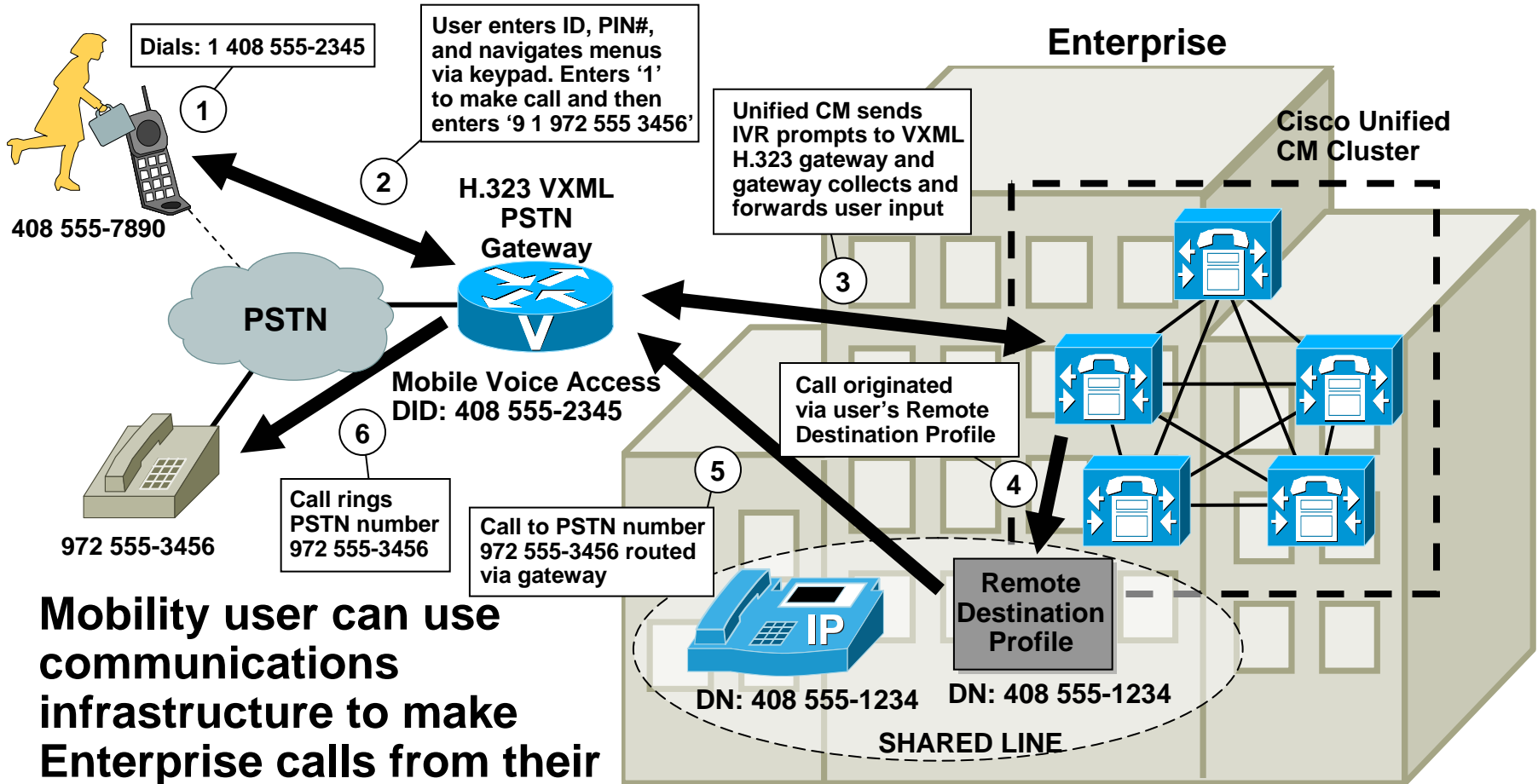
# Unified Mobility

## Remote Destination and Desk Phone Pickup



# Unified Mobility

## Mobile Voice Access



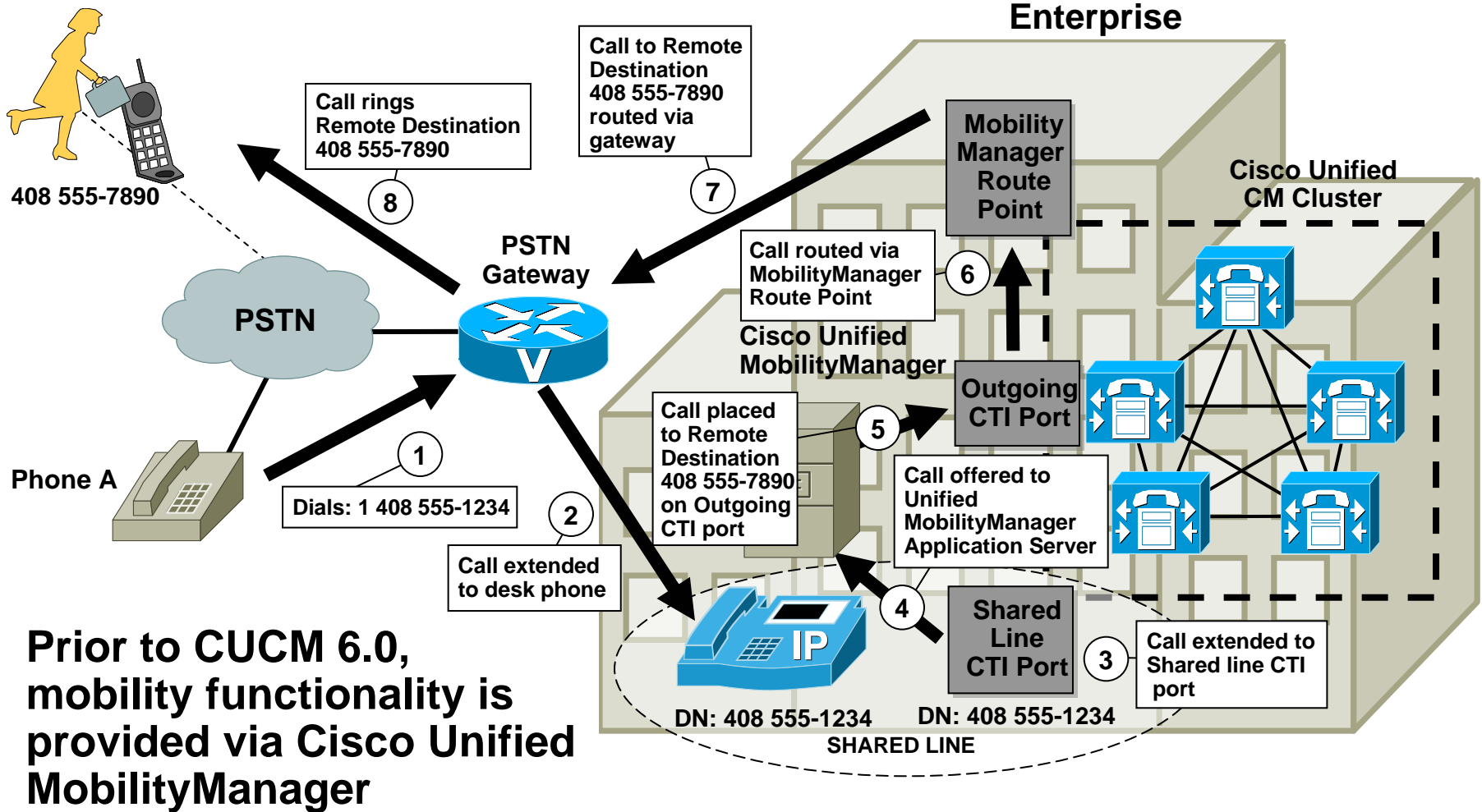
**Mobility user can use communications infrastructure to make Enterprise calls from their remote destination phone**

- Call made to Enterprise Mobile Voice Access number
- User follows IVR prompts and enters information to make call
- User can also disable and enable Mobile Connect on a per remote destination basis



# Unified Mobility

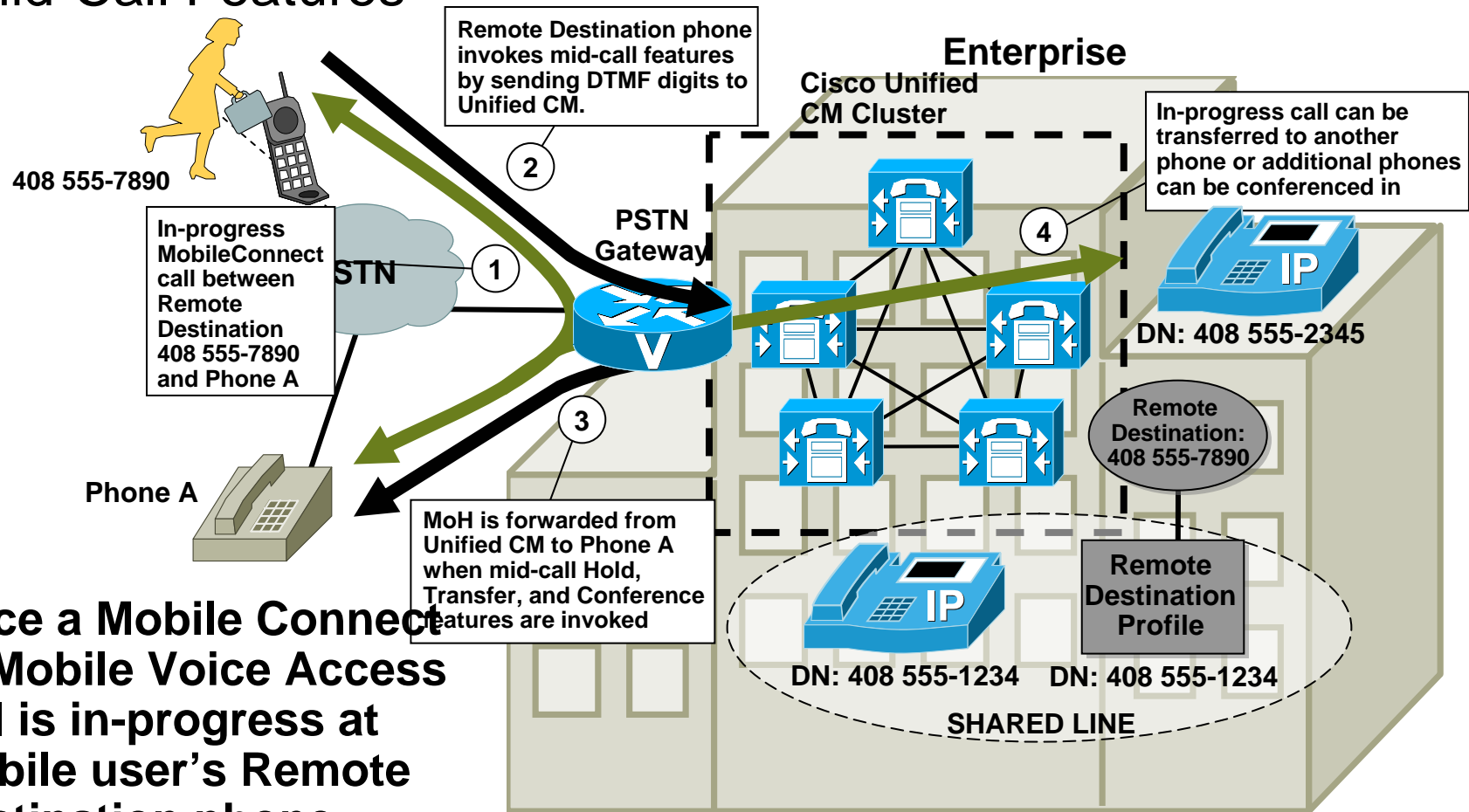
## Off-Box Mobility with MobilityManager (CUCM 4.X/5.X)



**CTI is required for interaction between MobilityManager application server and Cisco Unified CM, but behavior is the same**

# Unified Mobility

## Mid-Call Features

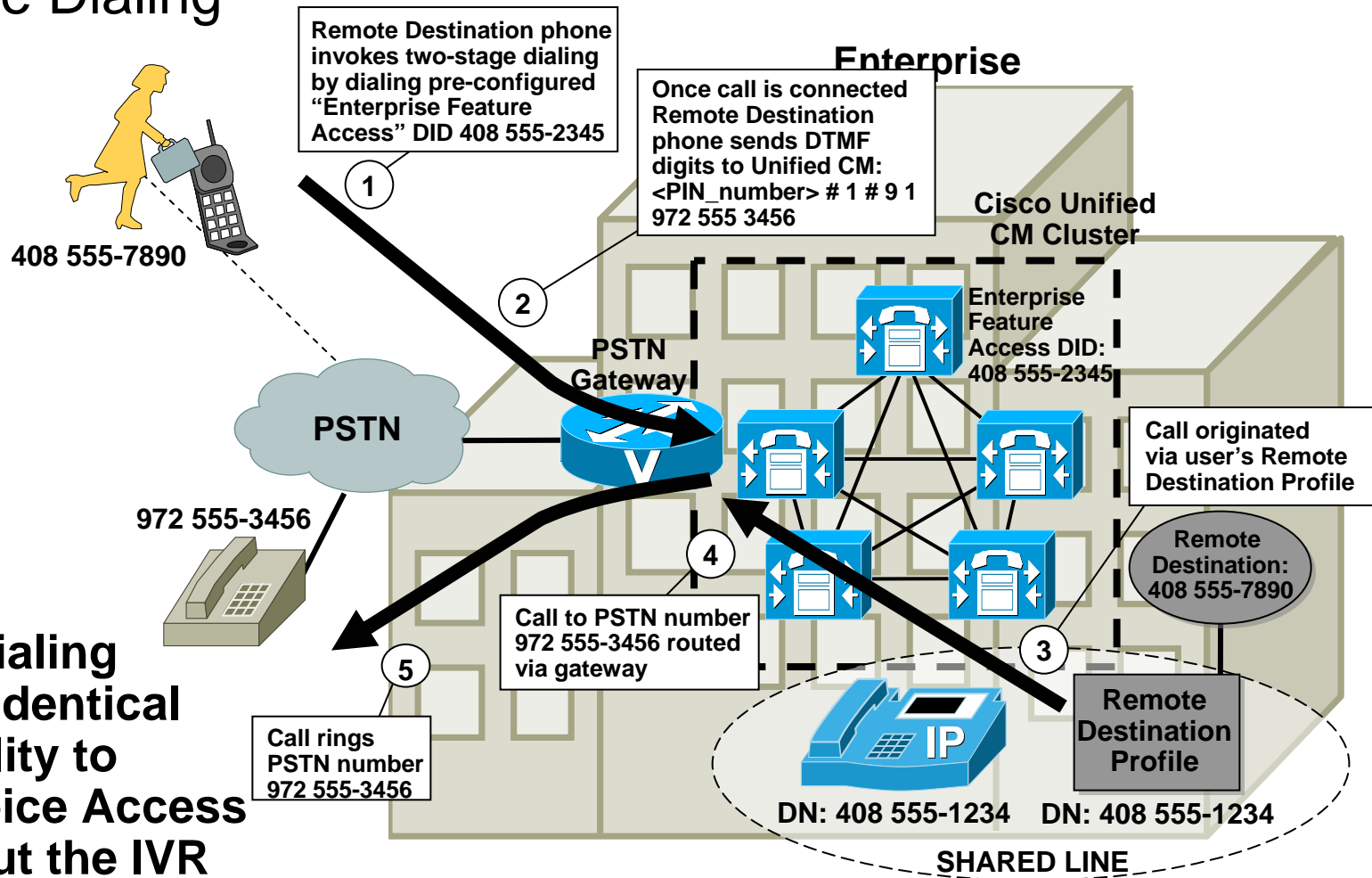


**Once a Mobile Connect or Mobile Voice Access call is in-progress at mobile user's Remote Destination phone:**

- Mid-call features like Hold, Transfer, and Conference can be invoked via Smart Phone softkeys or manual key presses
- DTMF tones are sent from the Remote Destination phone to the CUCM via the Enterprise PSTN gateway

# Unified Mobility

## 2 Stage Dialing

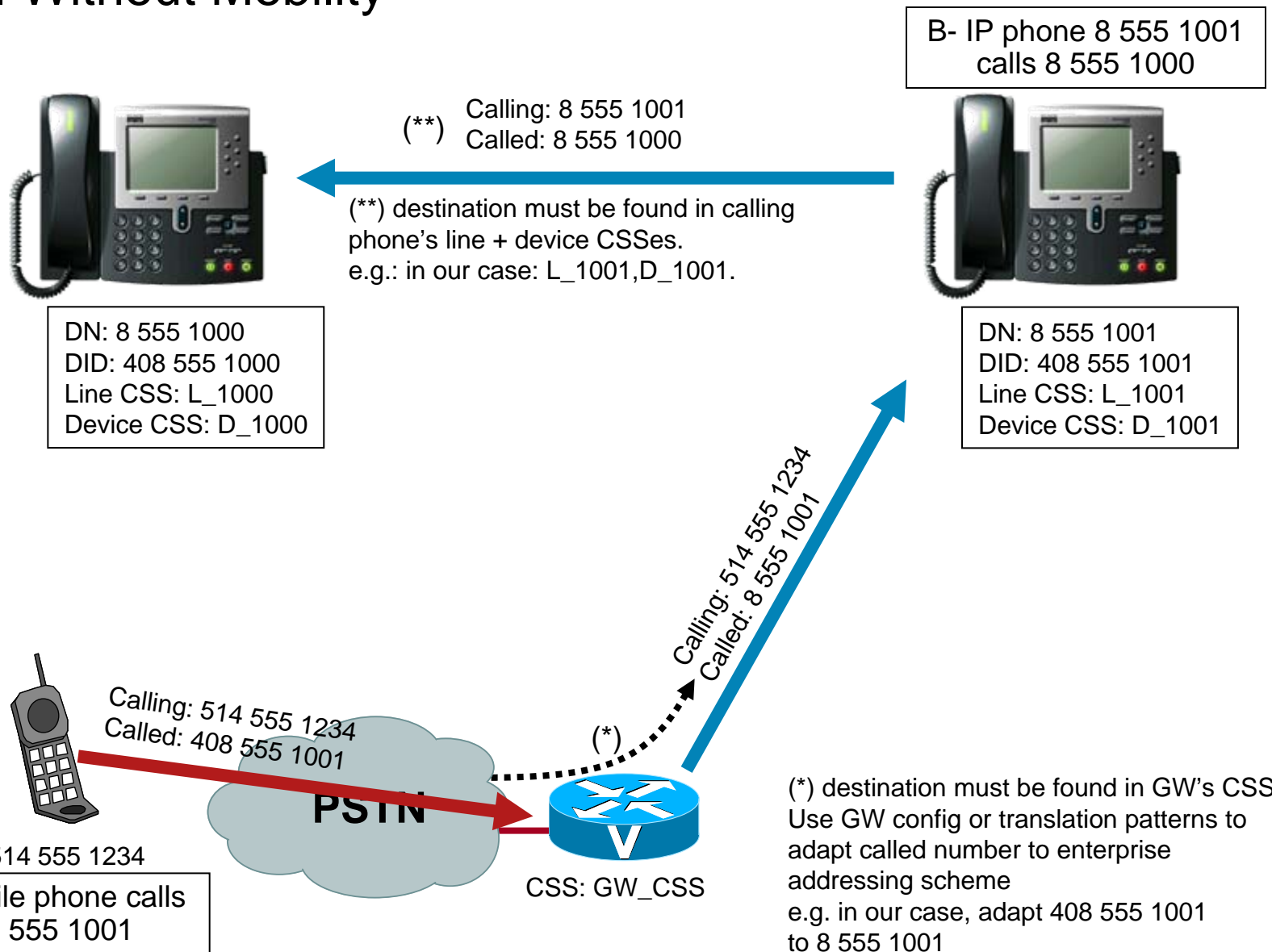


**2 Stage Dialing provides identical functionality to Mobile Voice Access but without the IVR**

- Call made to Enterprise Feature Access number
- User presses Smart Phone softkeys or manually keys digits (sent via DTMF) to make call
- User can also disable and enable Mobile Connect on a per remote destination basis

# Mobility: Dial Plan Implications

## 1. Without Mobility



# Mobility: Dial Plan Implications

## 2. New Configuration



DN: 8 555 1000  
DID: 408 555 1000  
Line CSS: L\_1000  
Device CSS: D\_1000

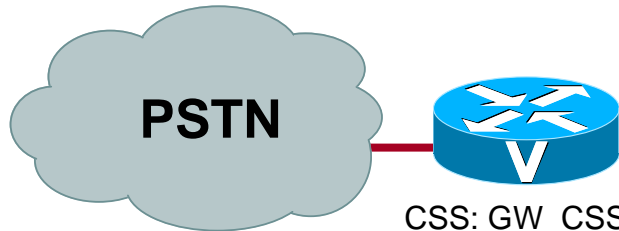
### Remote Destination Profile

CSS: css\_mrk\_1  
Reroute CSS: css\_mrk\_2  
Calling party transformation css: css\_mrk\_3

Remote Destination Number: 5145551234



514 555 1234



DN: 8 555 1001  
DID: 408 555 1001  
Line CSS: L\_1001  
Device CSS: D\_1001

**New configuration is tied to the DN of the phone**

# Mobility: Dial Plan Implications

## RDP and Remote Destination Number Associated to DN

**Directory Number Information**

Directory Number\* 85551000

Route Partition mrk\_1

Description

Alerting Name

ASCII Alerting Name

Allow Control of Device from CTI

Associated Devices SEP003094C26112  
rdp\_john\_doe

[Edit Device](#)

[Edit Line Appearance](#)

Dissociate Devices

---

**Directory Number Settings**

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

Calling Search Space css\_mrk\_1

Presence Group\* Standard Presence group

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

---

**Associated Remote Destinations**

Name	Destination Number
<a href="#">john_doe_cell</a>	5145551234

# 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

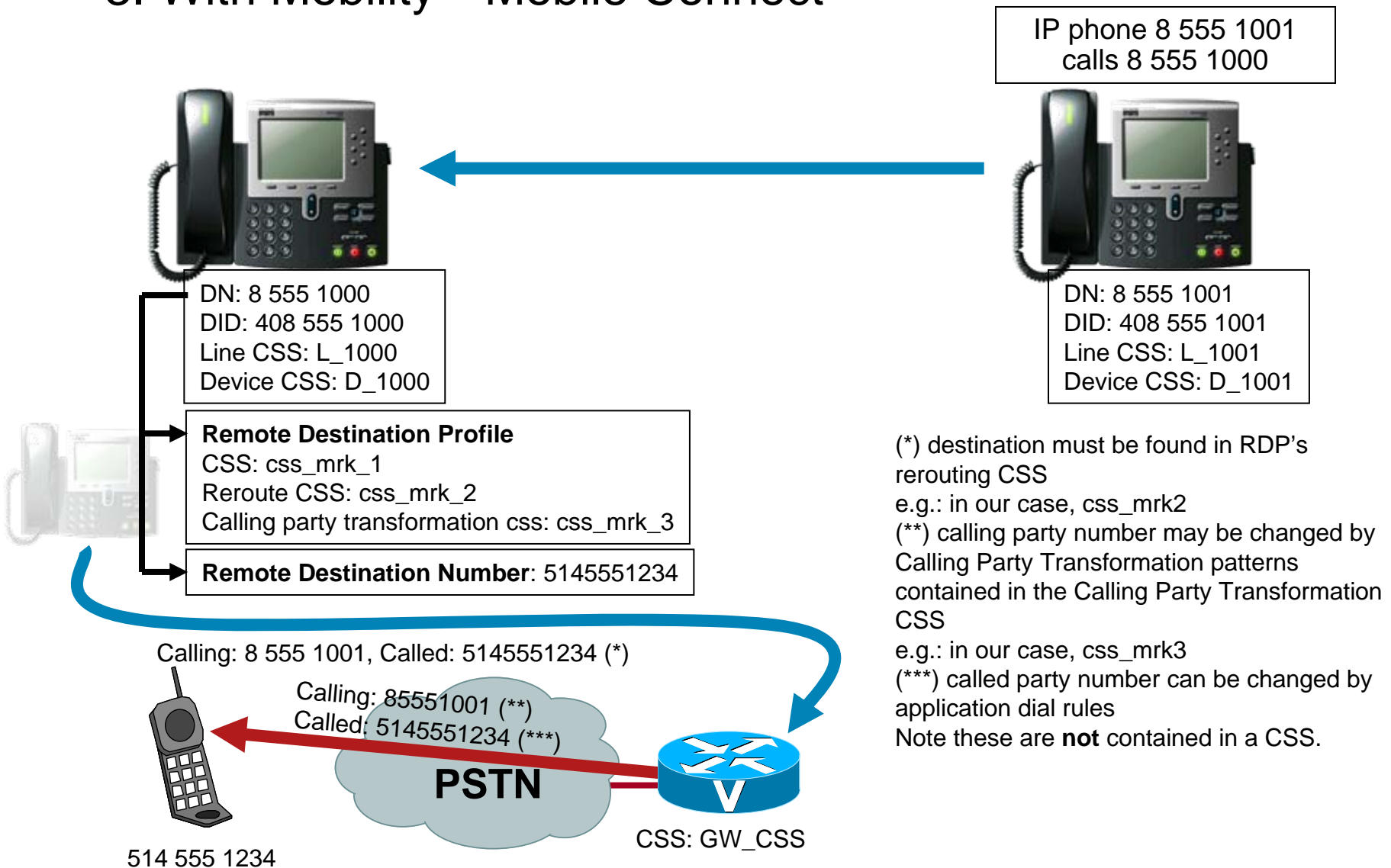
Ignore Presentation Indicators (internal calls only)

**Associated Remote Destinations**

Name	Destination Number
john_doe_cell	5145551234
<a href="#">Add a New Remote Destination</a>	

# Mobility: Dial Plan Implications

## 3. With Mobility—Mobile Connect



(\*) destination must be found in RDP's rerouting CSS  
e.g.: in our case, css\_mrk2  
(\*\*) 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.



# 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

# 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 Removed
<a href="#">NPA415_NXX555</a>	514555	10	0
<a href="#">CC1NPA514NXX555</a>	1514555	11	1

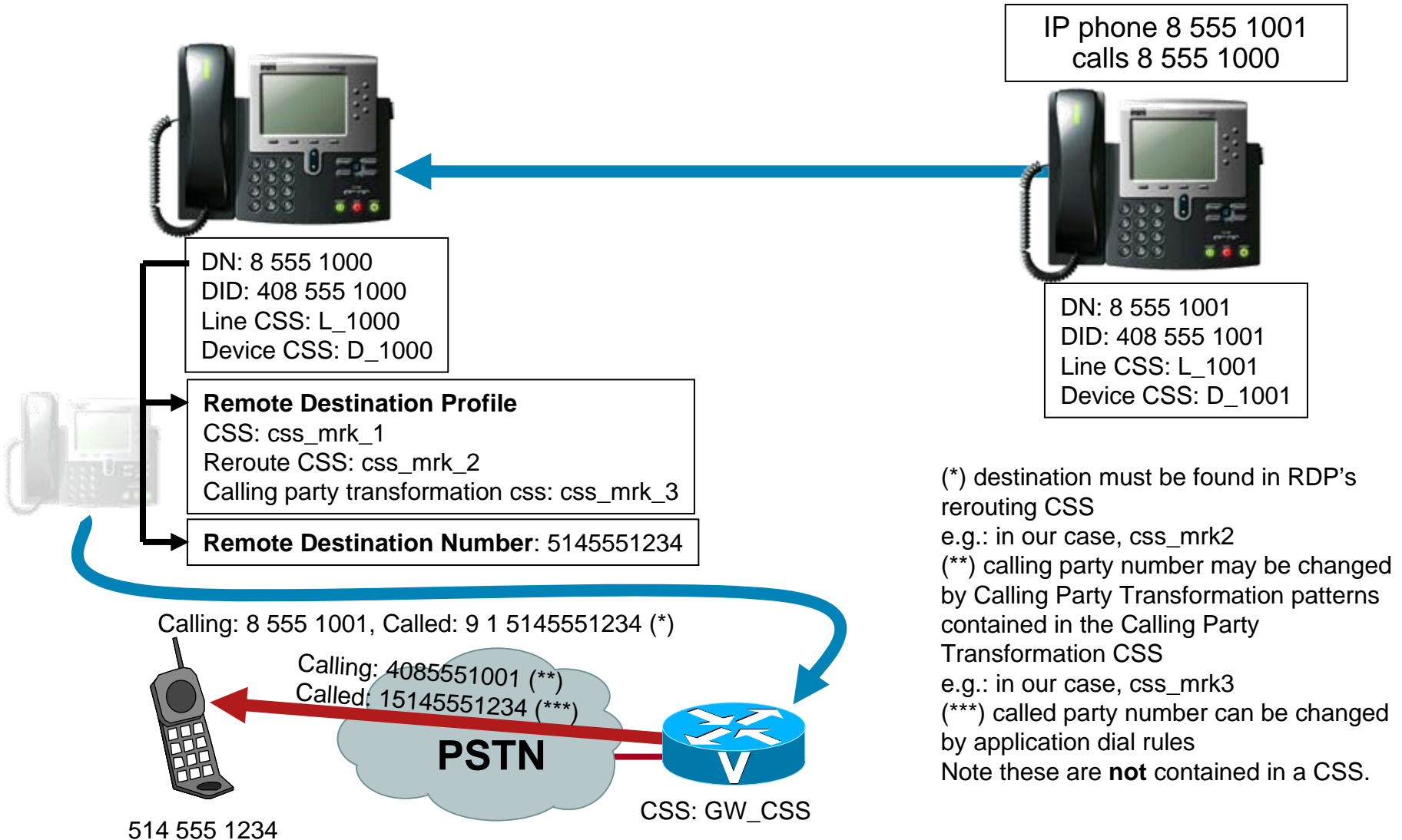
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Save Delete Add New

\*- indicates required item.

# Mobility: Dial Plan Implications

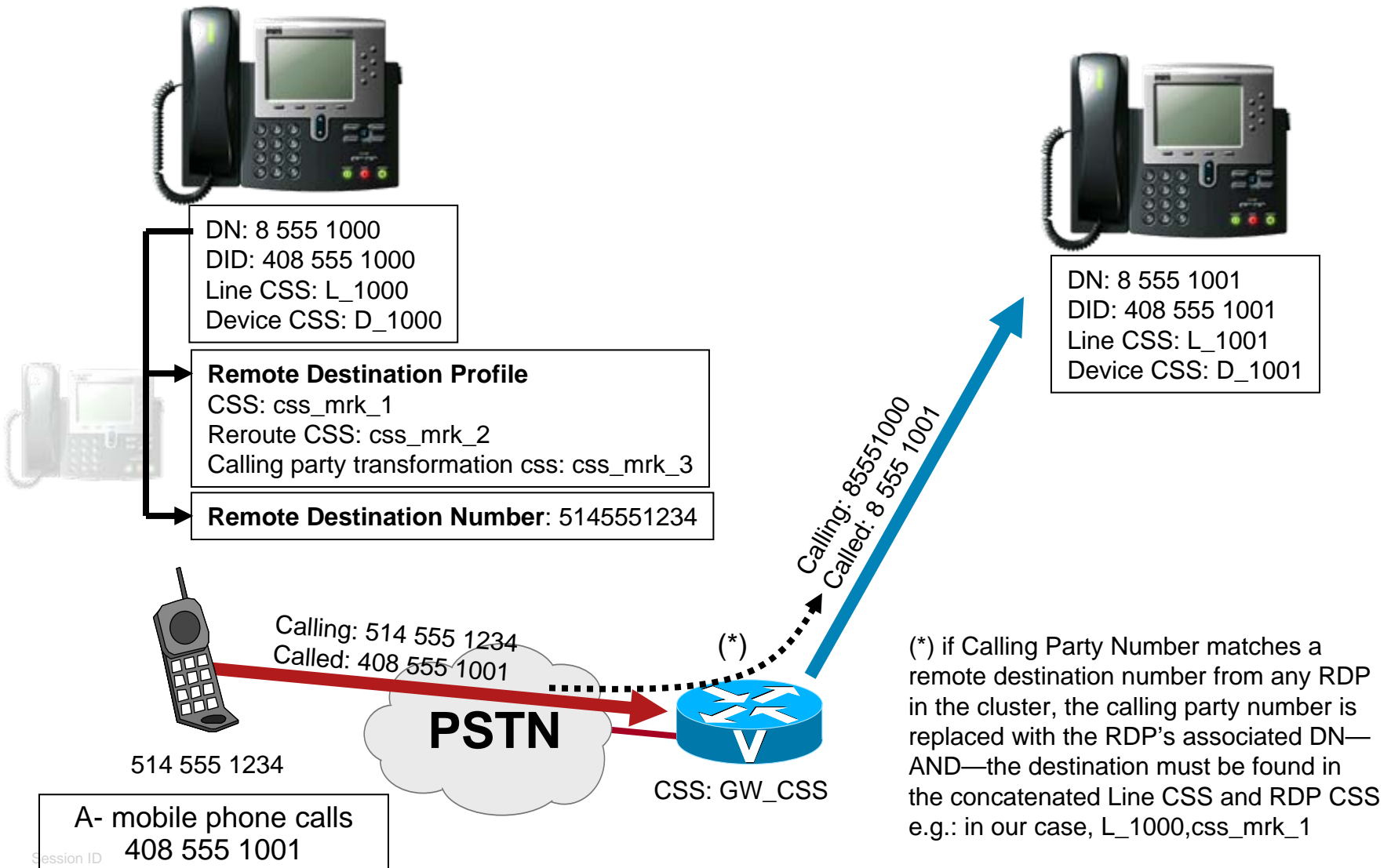
## 4. With Mobility—Mobile Connect Enhanced



(\*) destination must be found in RDP's rerouting CSS  
e.g.: in our case, css\_mrk2  
(\*\*) 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.

# Mobility: Dial Plan Implications

## 5. With Mobility—Inbound Calls



# Conclusions

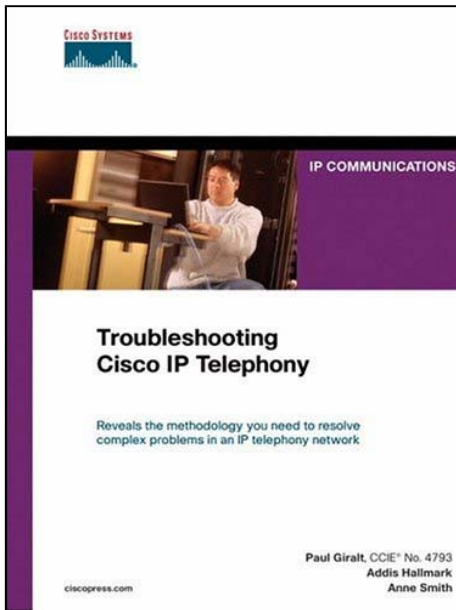
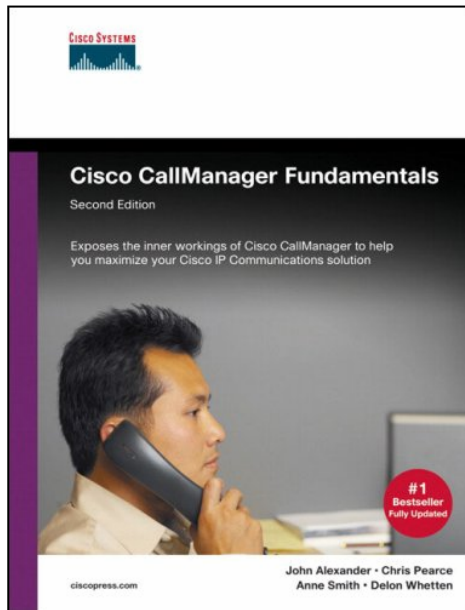


# Conclusions

## General Recommendations

- **Keep It Simple!**

# Recommended Reading



- Continue your Networkers at Cisco Live learning experience with further reading from Cisco Press
- Check the Recommended Reading flyer for suggested books
- A few suggestions:
  - Cisco CallManager Fundamentals, Second edition
  - Troubleshooting Cisco IP Telephony

**Available Onsite at the Cisco Company Store**

# Complete Your Online Session Evaluation

- Win fabulous prizes; Give us your feedback
- Receive ten Passport Points for each session evaluation you complete
- Go to the Internet stations located throughout the Convention Center to complete your session evaluation
- Drawings will be held in the World of Solutions

Tuesday, June 20 at 12:15 p.m.

Wednesday, June 21 at 12:15 p.m.

Thursday, June 22 at 12:15 p.m. and  
2:00 p.m.







# Appendix



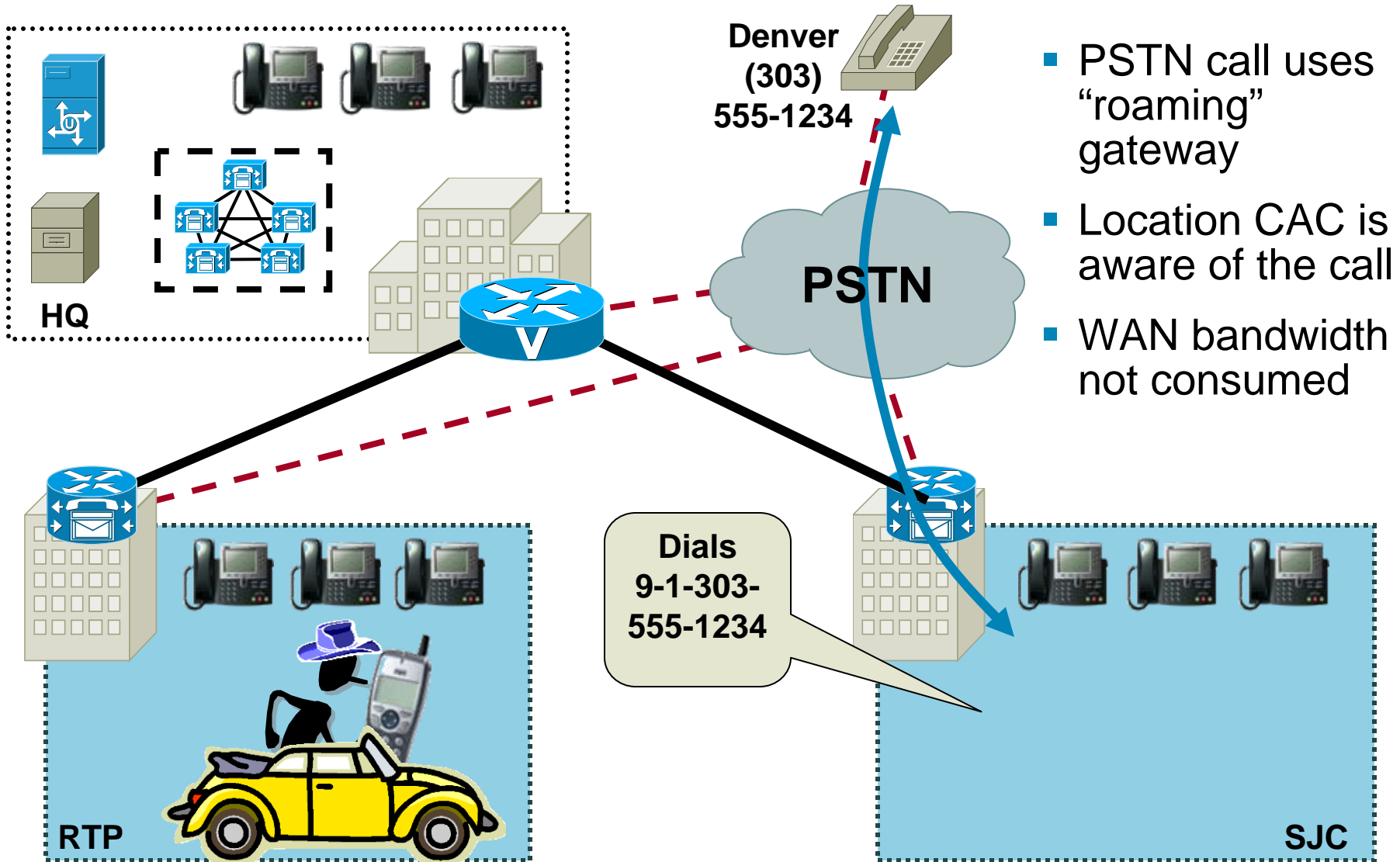
# Appendix

- Additional Device Mobility considerations



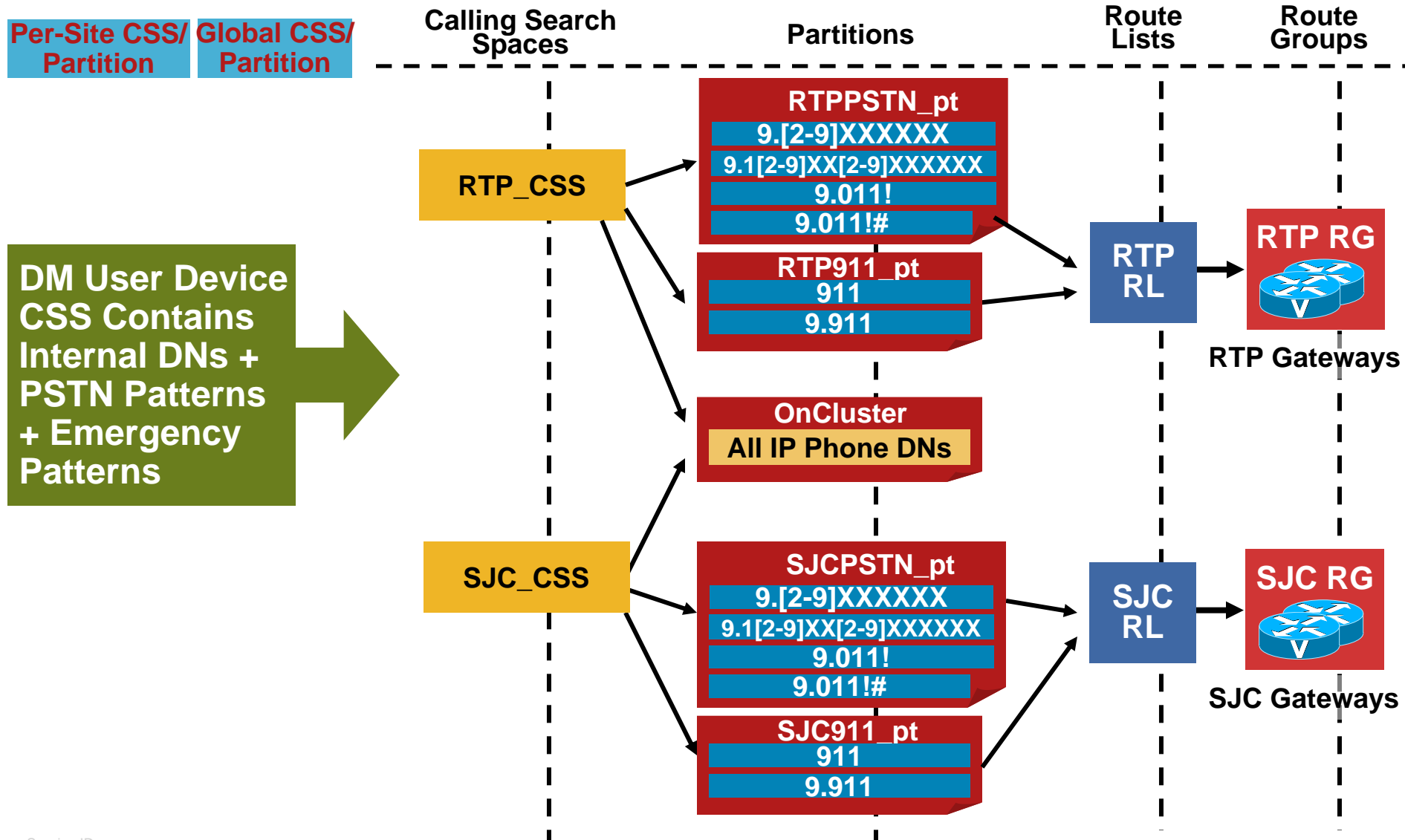
# Device Mobility Considerations

## Requirements



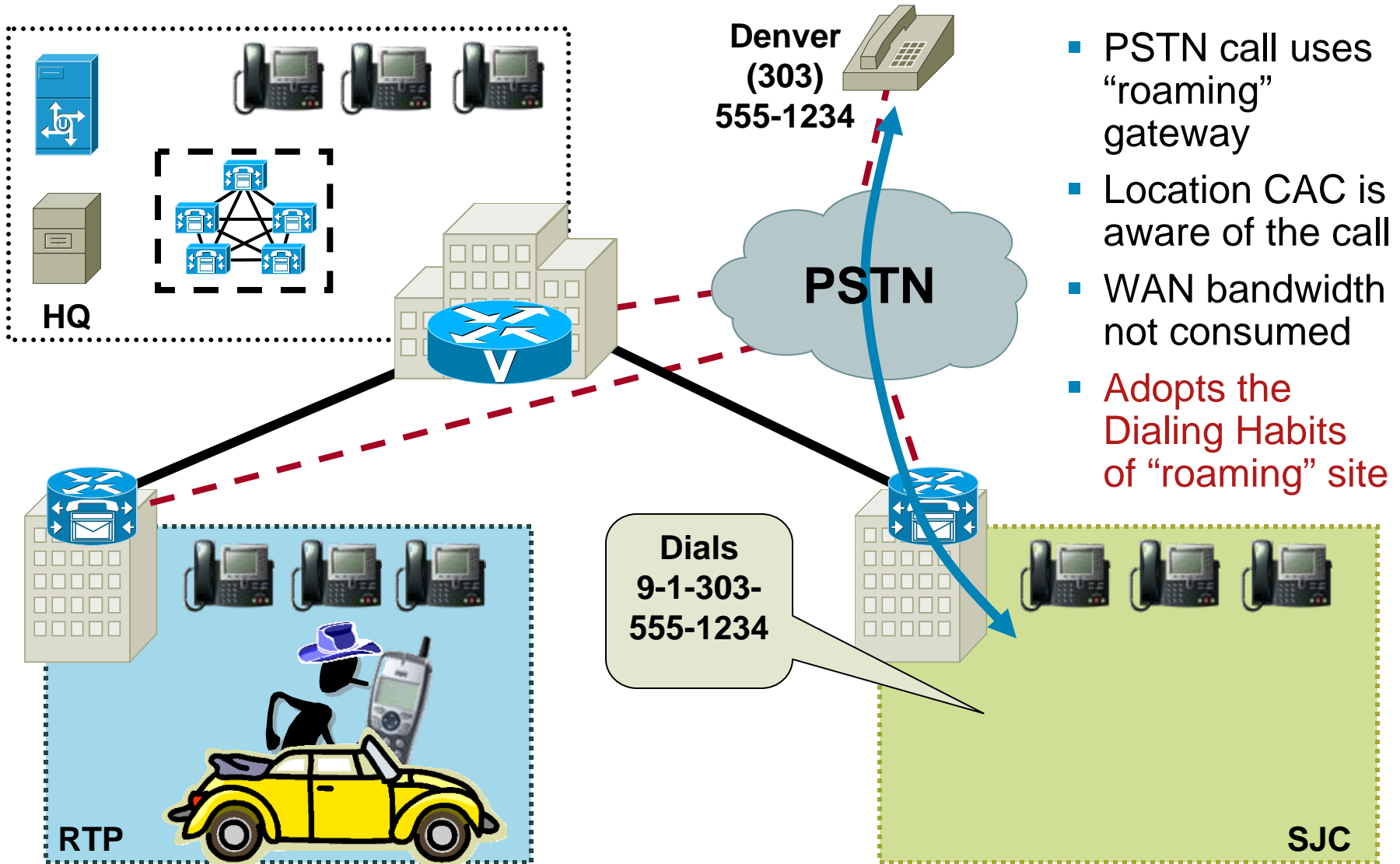
# Device Mobility Considerations

## Traditional Dial Plan Approach



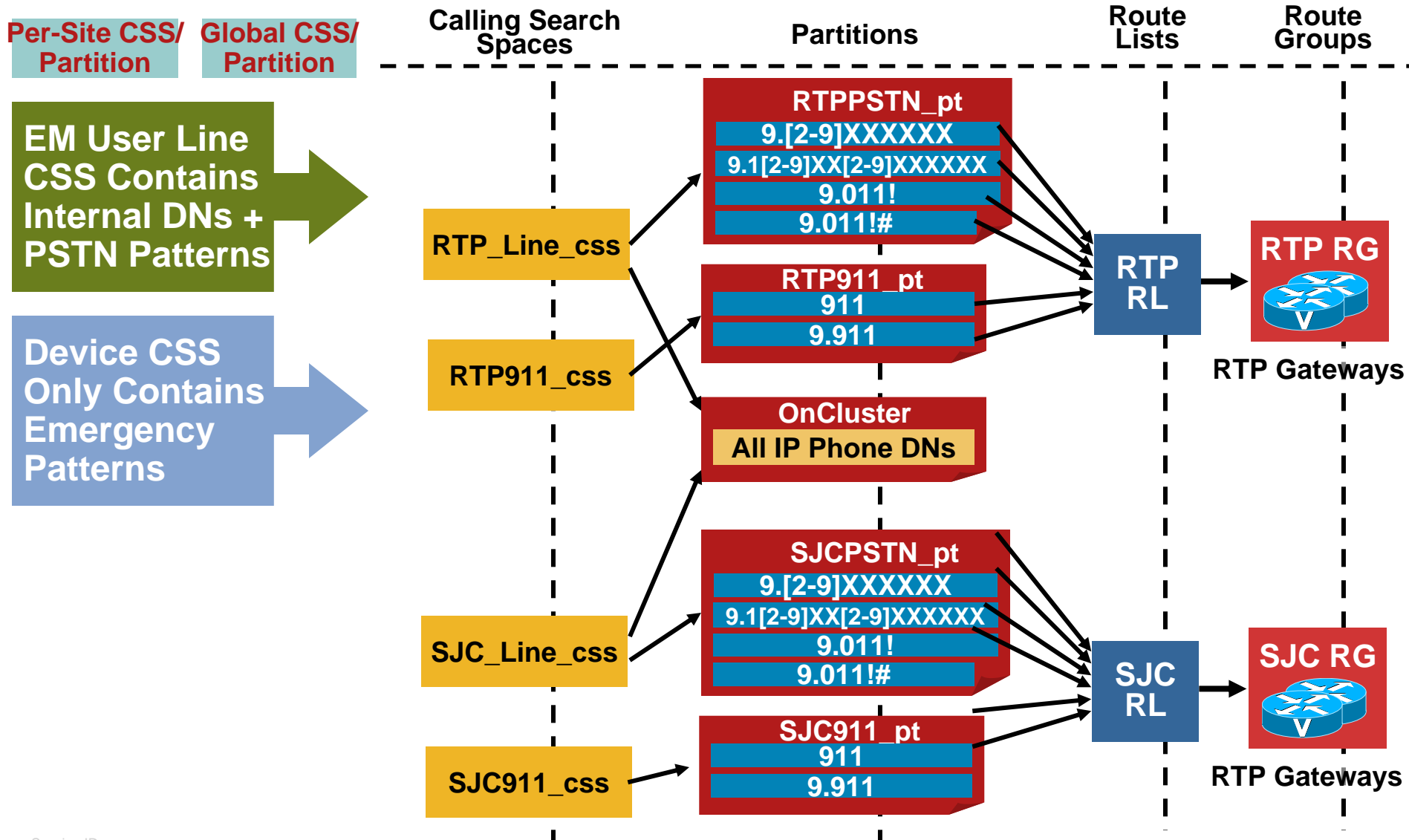
# Device Mobility Considerations

## Traditional Dial Plan Approach: Behavior



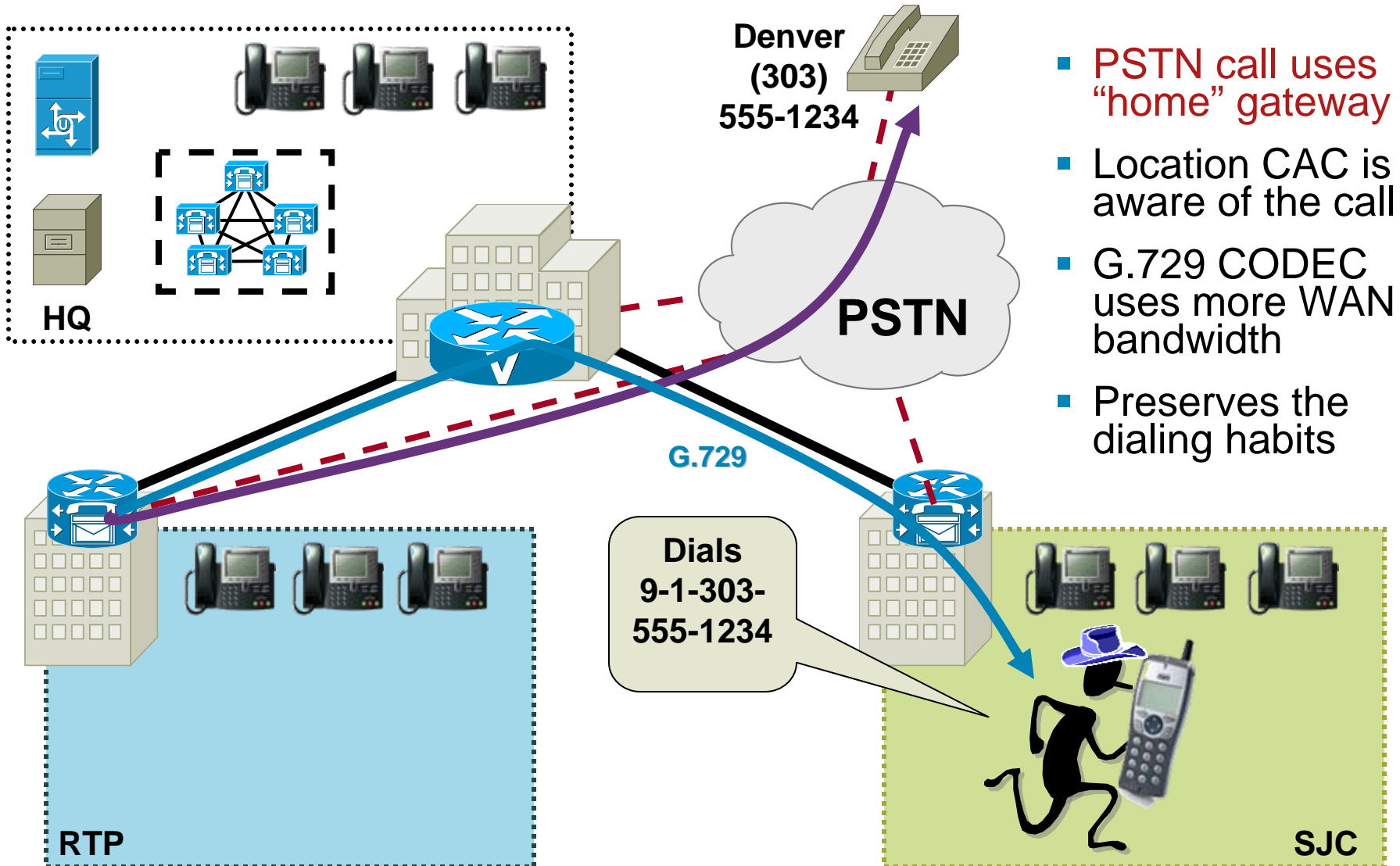
# Device Mobility Considerations

## Traditional Dial Plan Approach (EM Approach)



# Device Mobility Considerations

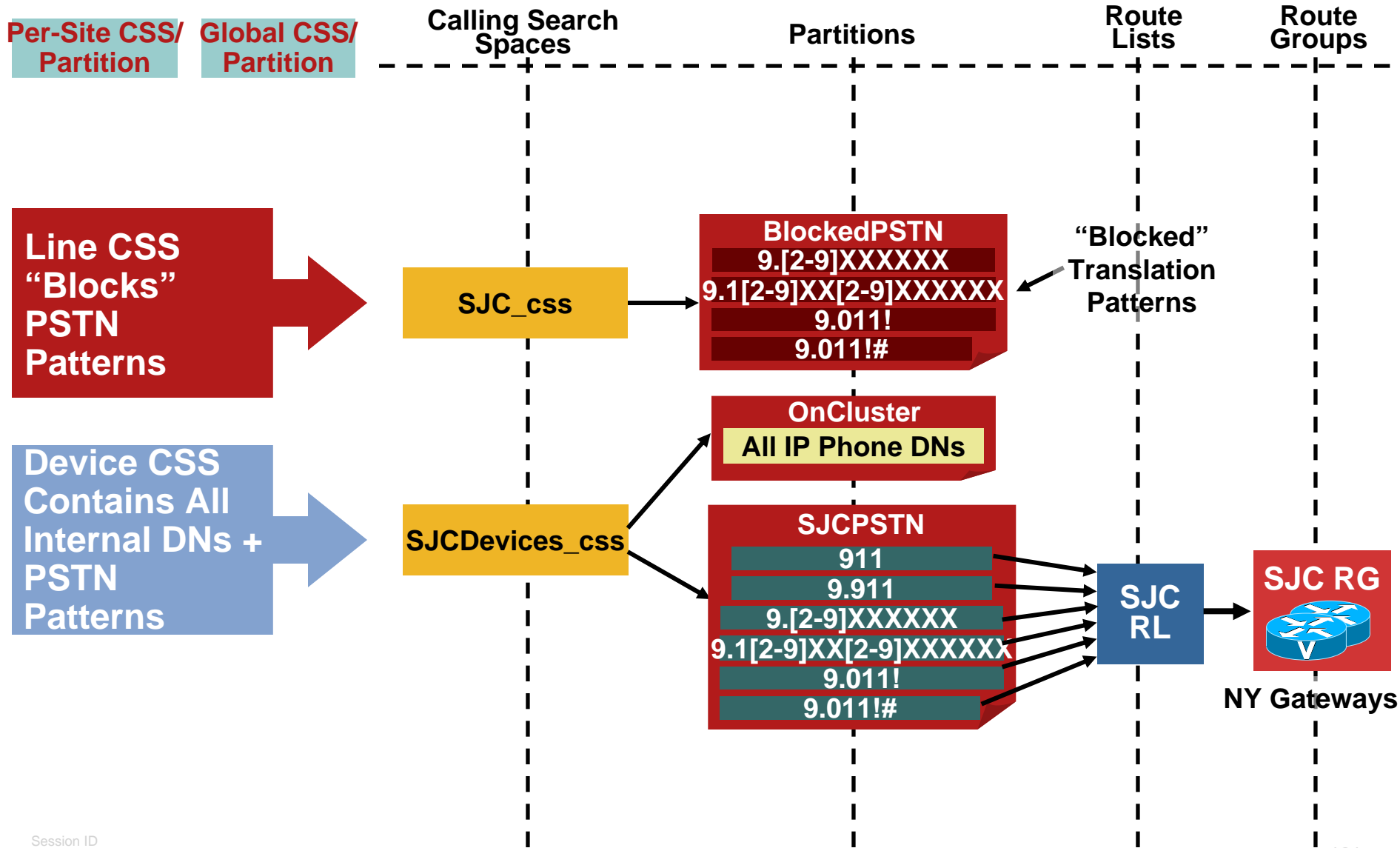
## Traditional Dial Plan (EM Approach): Behavior





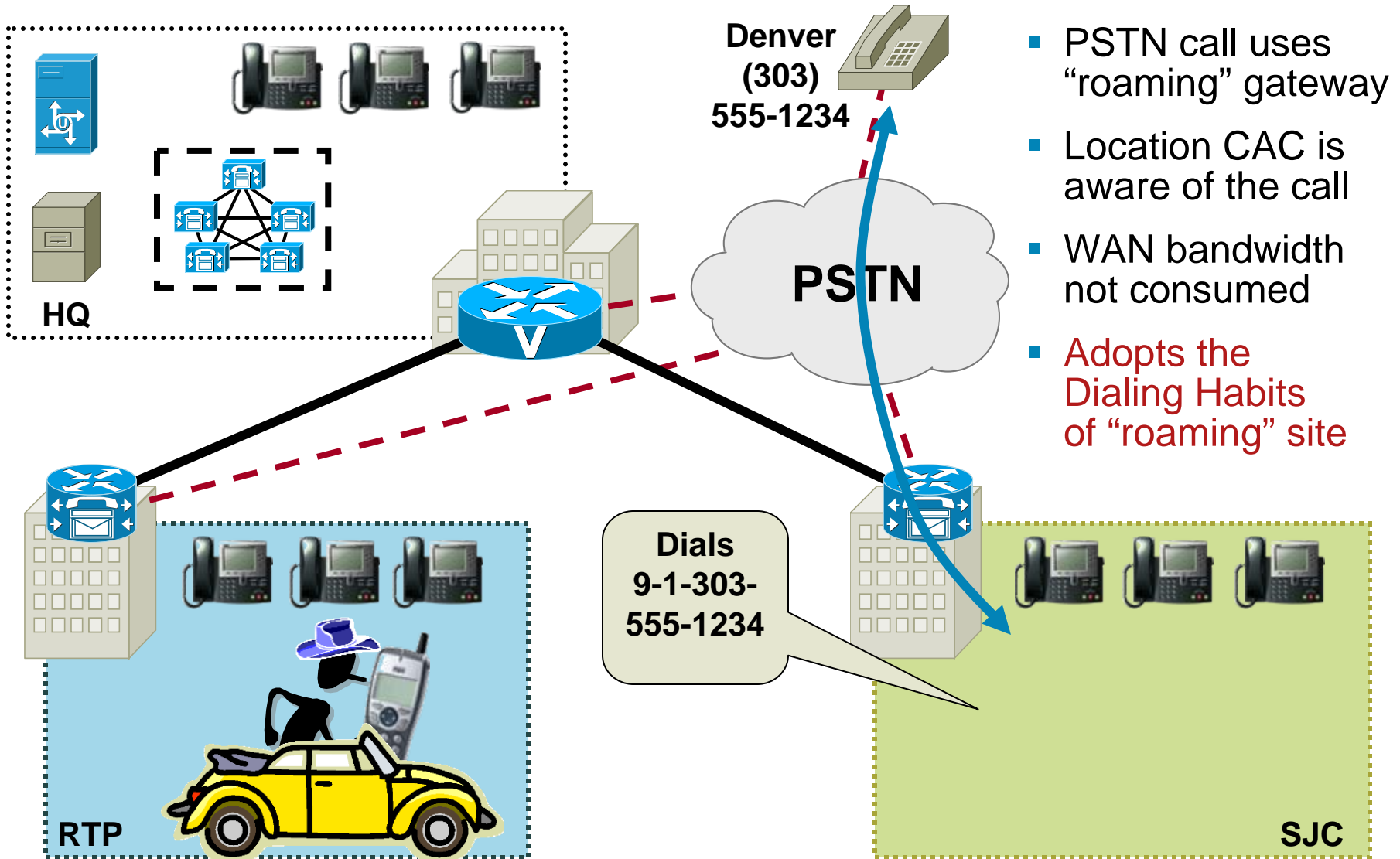
# Device Mobility Considerations

## Line/Device Dial Plan Approach



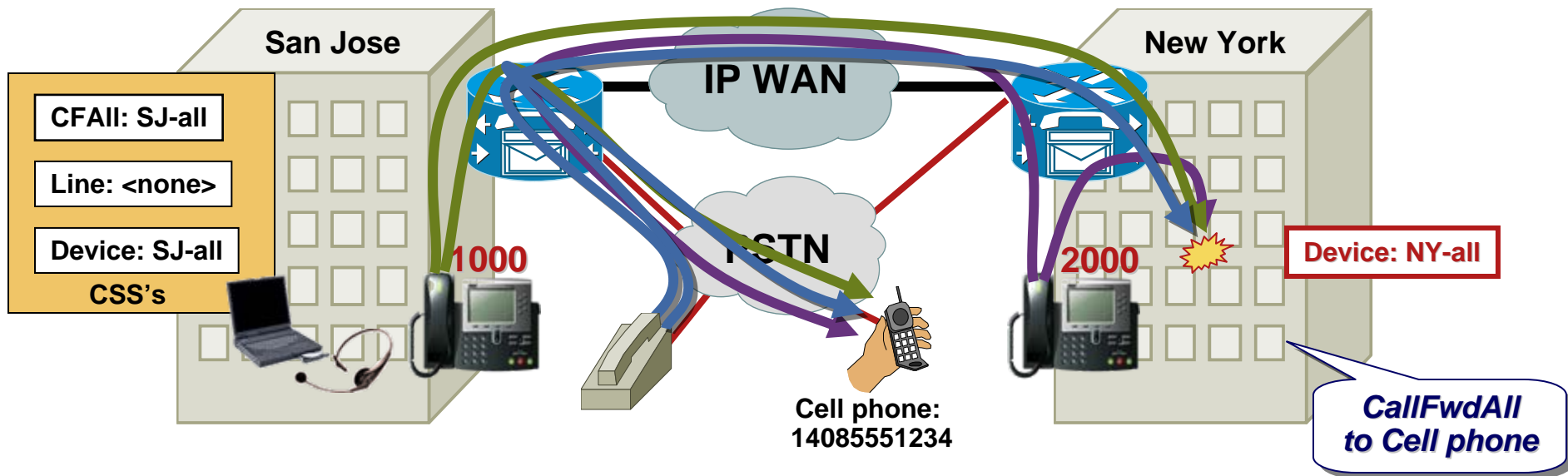
# Device Mobility Considerations

## Line/Device Dial Plan Approach: Behavior



# 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:

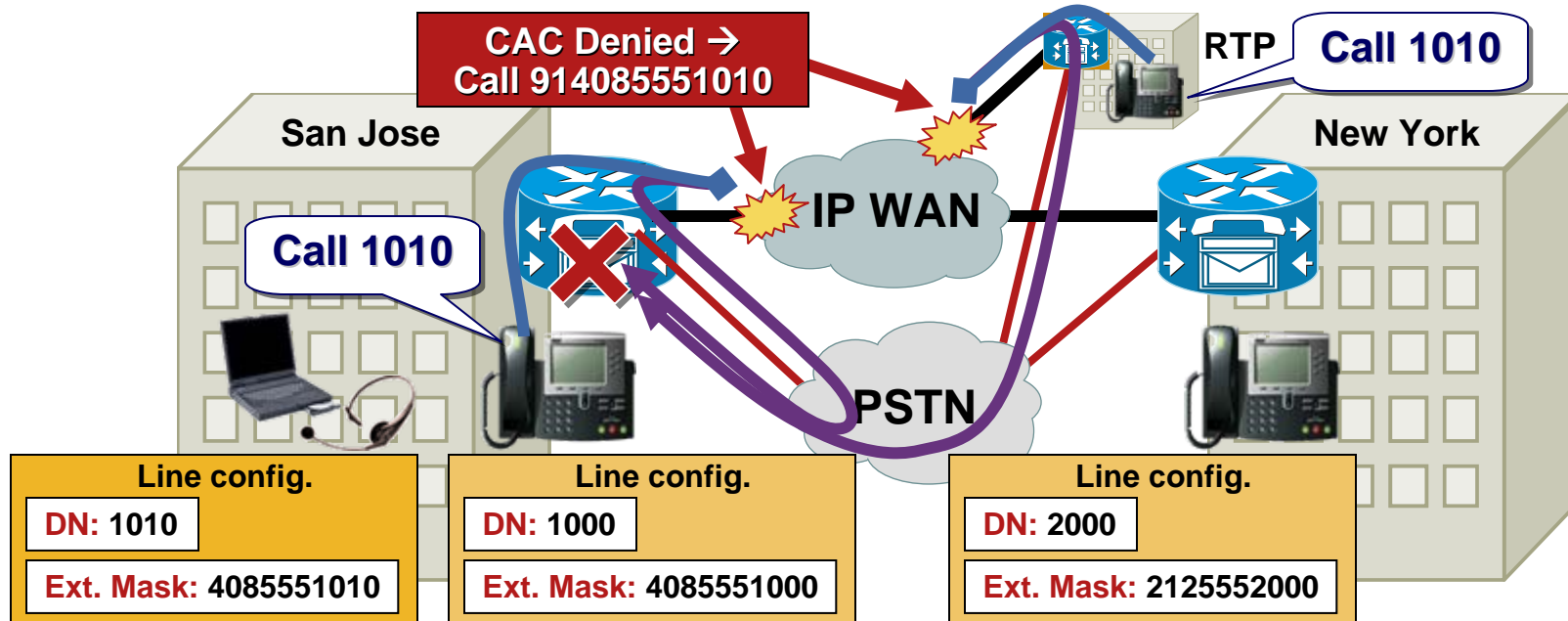
Calls from SJ IP phones use SJ PSTN GW

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

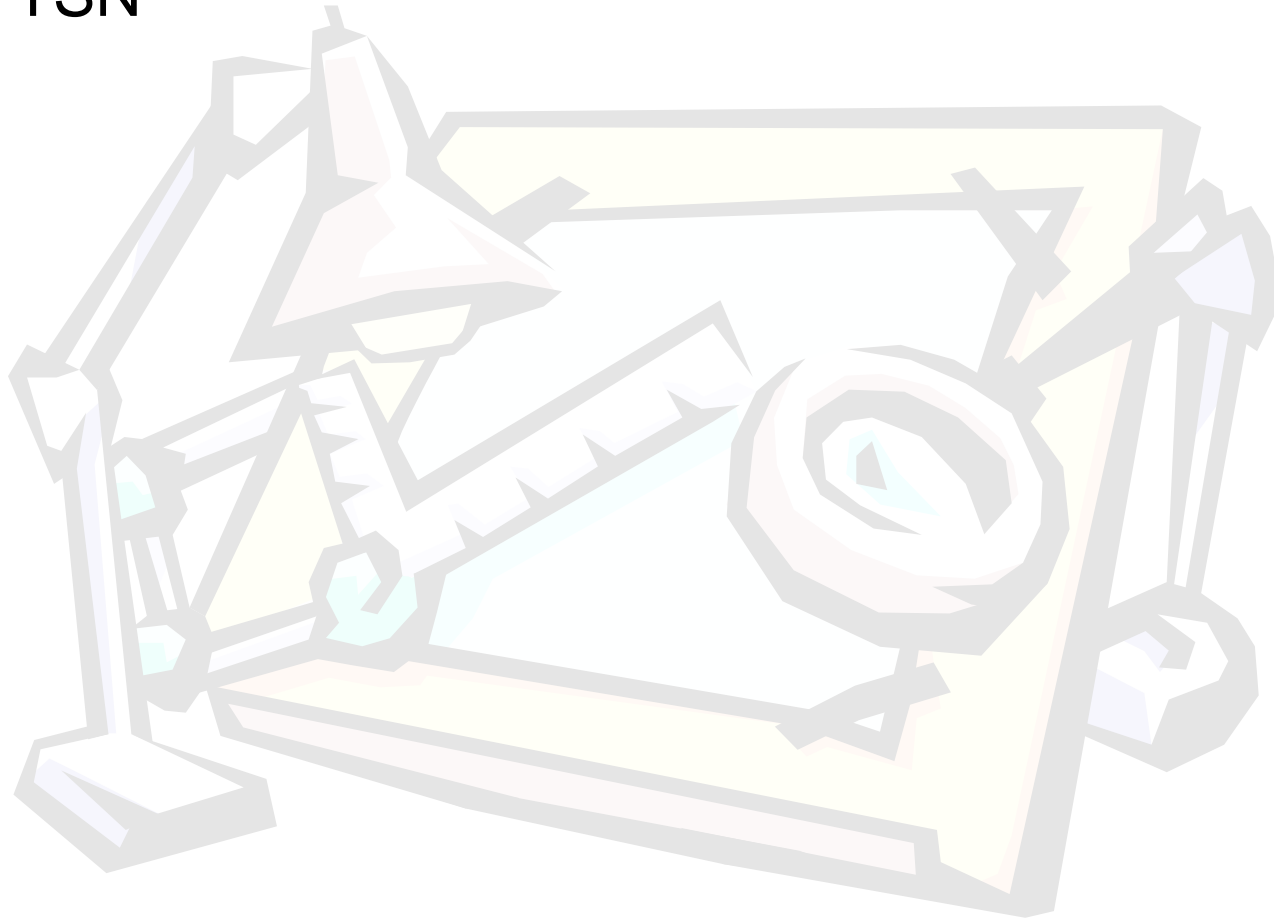
## AAR Interactions



- AAR is inherently incompatible with device mobility across sites (same as for EM 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 CSS's contain internal numbers only to prevent routing loops

# Appendix

- VoPTSN



# What Is Voice over the PSTN (VoPSTN)?

- A variation on the Centralized 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

## Global Considerations

No Streaming of Audio to Central Site, Thus No:

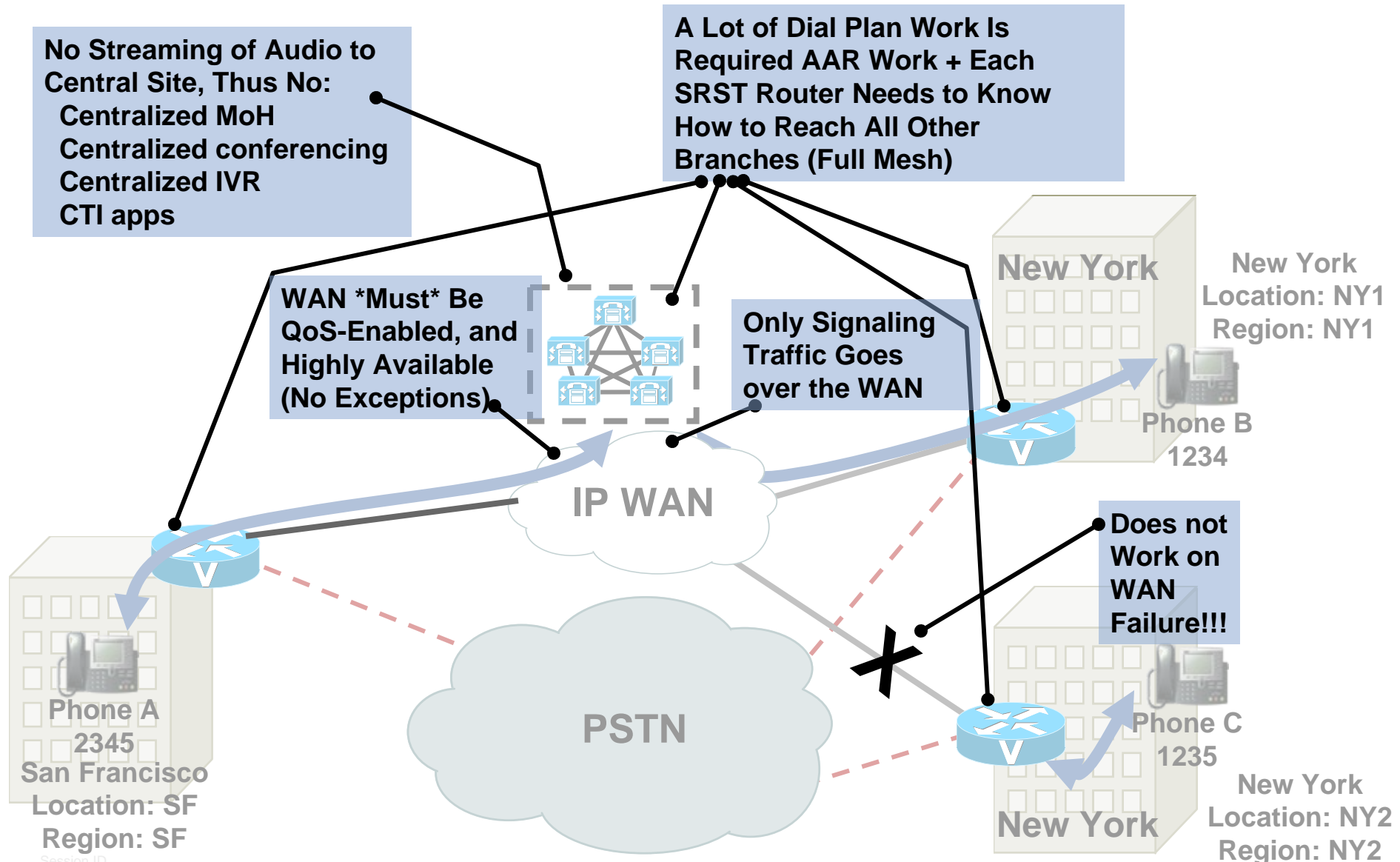
- Centralized MoH
- Centralized conferencing
- Centralized IVR
- CTI apps

A Lot of Dial Plan Work Is Required AAR Work + Each SRST Router Needs to Know How to Reach All Other Branches (Full Mesh)

WAN \*Must\* Be QoS-Enabled, and Highly Available (No Exceptions)

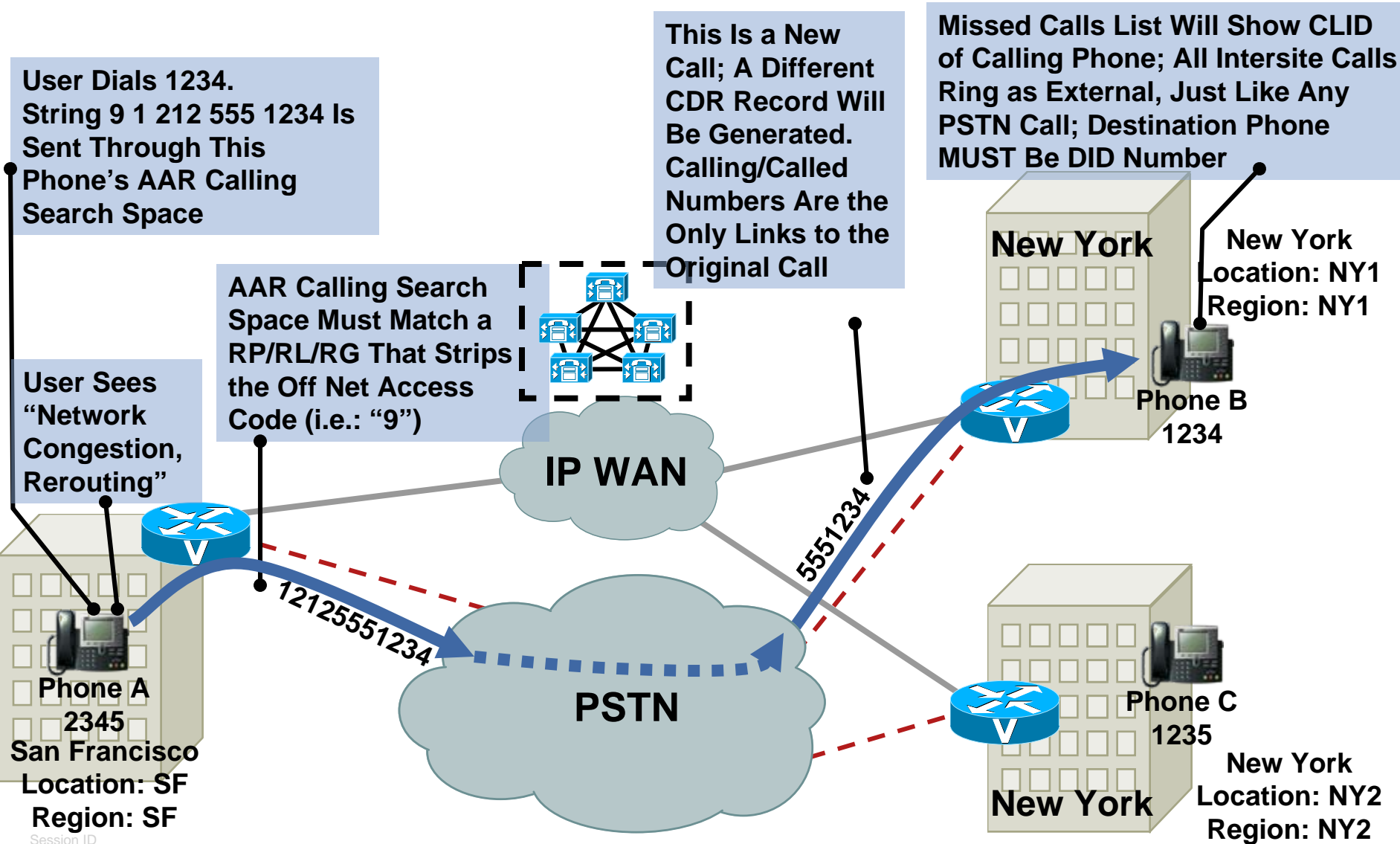
Only Signaling Traffic Goes over the WAN

Does not Work on WAN Failure!!!



# VoPSTN Using AAR

## Intersite Calls





# VoPSTN Using AAR

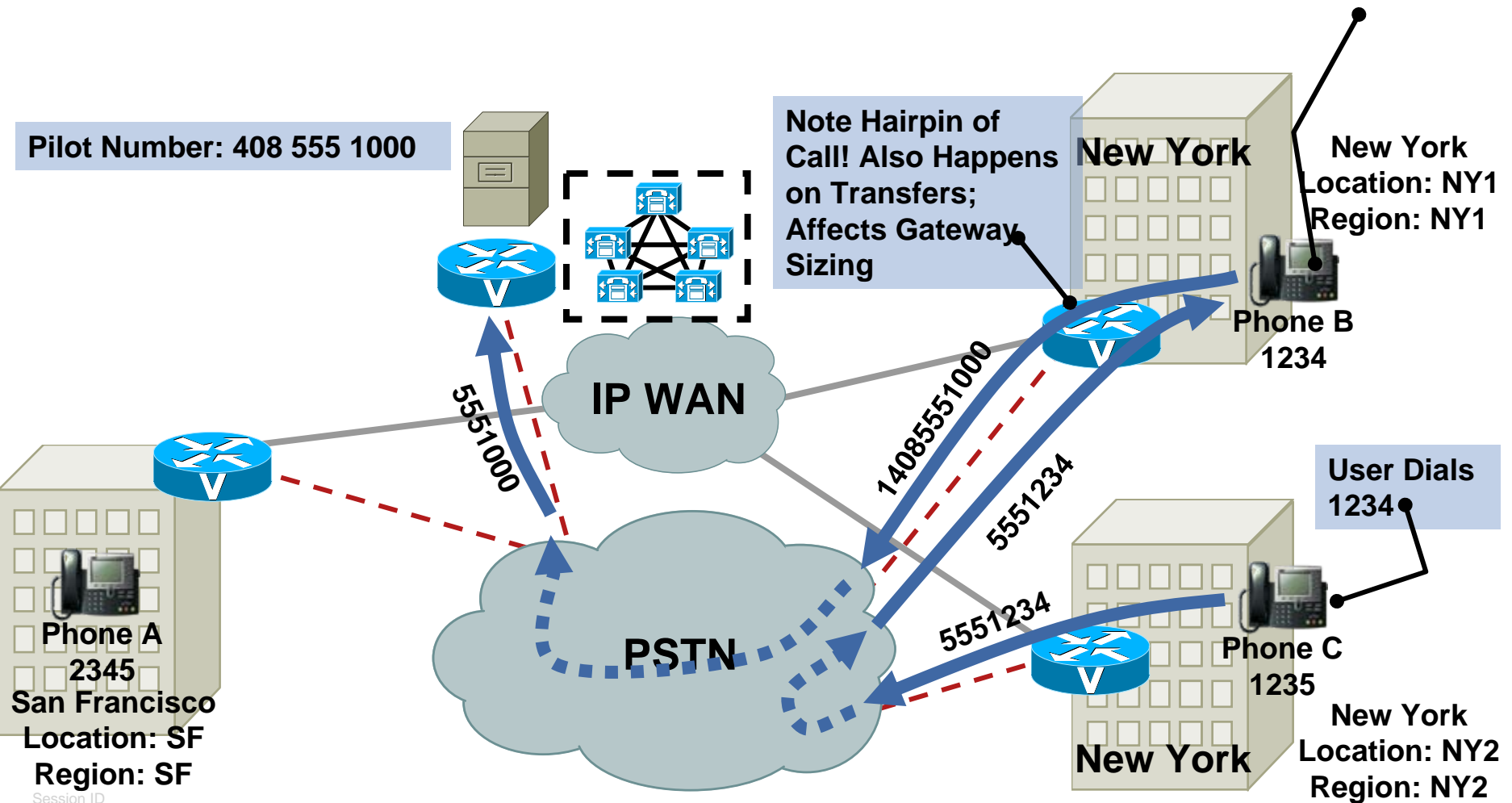
## Centralized Voicemail

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

## Shared Lines Considerations

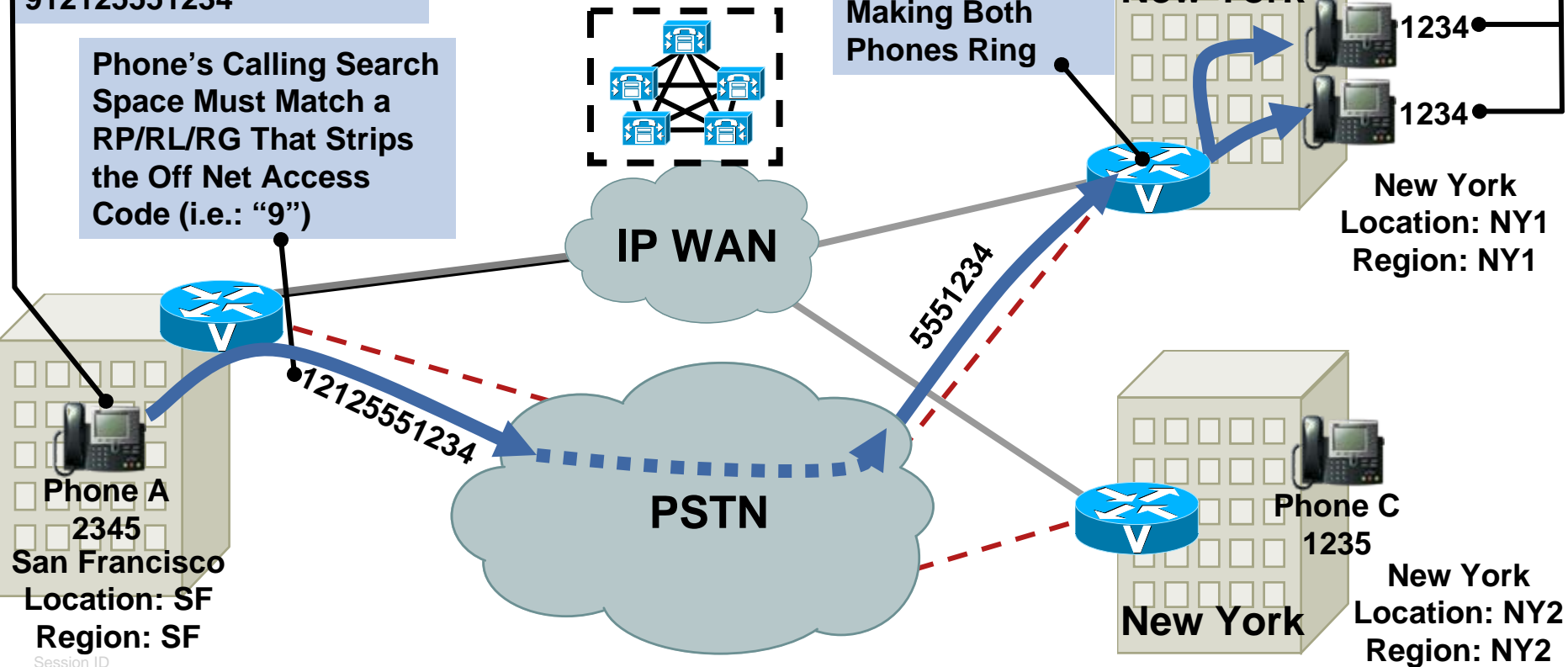
User Dials 1234;  
1234 Matches a TP in the  
Phone's Dialing Plan,  
Expanding to  
912125551234

Phone's Calling Search  
Space Must Match a  
RP/RL/RG That Strips  
the Off Net Access  
Code (i.e.: "9")

**AAR Should Not Be  
Used to Reach  
Remote Shared  
Lines, as It Would  
Launch Multiple  
Parallel PSTN Calls**

GW's CSS Must  
Include Partition  
Containing the  
Shared Line,  
Making Both  
Phones Ring

DN 1234 Must Be in a  
Site-Specific Partition,  
Not Included in Off-Site  
Calling Search Spaces



# VoPSTN Using AAR

## Summary

- Only accommodates SCCP destinations
- RDNIS required for centralized 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 centralized MoH
- No centralized conferencing
- All transferred calls are hairpinned
- All calls forwarded to outside locations are hairpinned
- If you tailor the WAN for signaling 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

- DN's at each site are placed in different partitions
- Relies on PSTN route patterns to call other sites
- For Cisco UC Manager, all calls are **external** calls
- No "on-net" features across sites (e.g. CallBack)
- No easy migration to fullblown VoIP
- **Note:** Abbreviated dialing possible with translation rules on branch GW's

