Advanced Dial Plan Design

BRKUCT-3012

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Cisco Networkers 2007

HOUSEKEEPING

- We value your feedback, don't forget to complete your online session evaluations after each session and complete the Overall Conference Evaluation which will be available online from Friday.
- Visit the World of Solutions on Level -01!
- Please remember this is a 'No Smoking' venue!
- Please switch off your mobile phones!
- Please remember to wear your badge at all times including the Party!
- Do you have a question? Feel free to ask them during the Q&A section or write your question on the Question form given to you and hand it to the Room Monitor when you see them holding up the Q&A sign.

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)

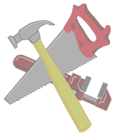
- To explore the design and implementation possibilities of Cisco's IP telephony system
  Design based on Cisco CallManager 4.X and 5.0

- Aspects we will cover:
  - Dial plan elements
    (Call routing logic, partitions and calling search spaces...)
  - Design guidelines
    (Classes of service, multisite deployments, extension mobility...)

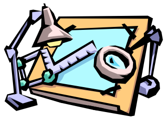
Overall Agenda

- Planning Considerations

- Dial Plan Elements



- 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” for an outside line, or five digits to reach colleagues)

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)

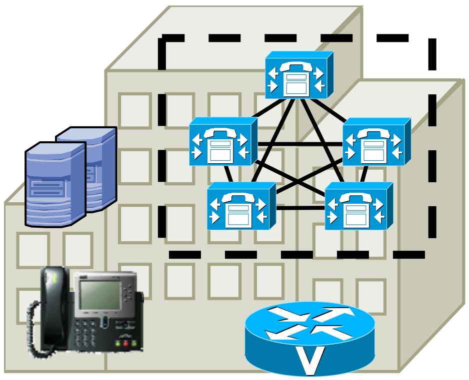
- Remember: the best tool to start with is this:

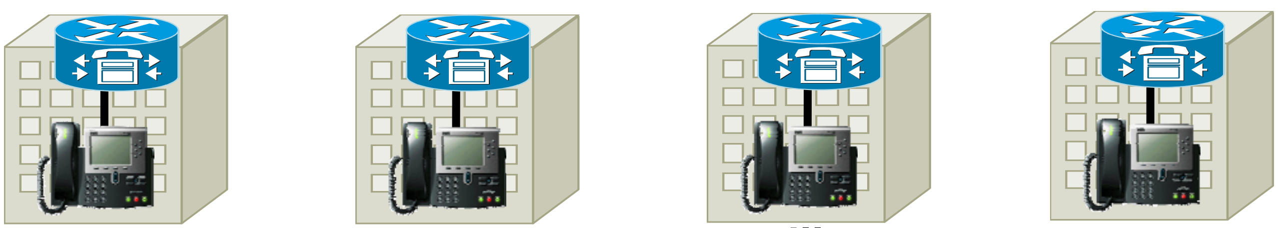
Planning Considerations
Uniform Dial Plans Are Simple

Q: Could this system use a *uniform* 3 digit dial plan?
A: No! Marseille and Brest DID ranges overlap in the last 3 digits.

Q: Ok, how about 4 digit uniform dial plan?
A: No! overlaps again!

Because each time you call extensions 1120 through 1129 in Brest, you get the emergency service (by calling 112)

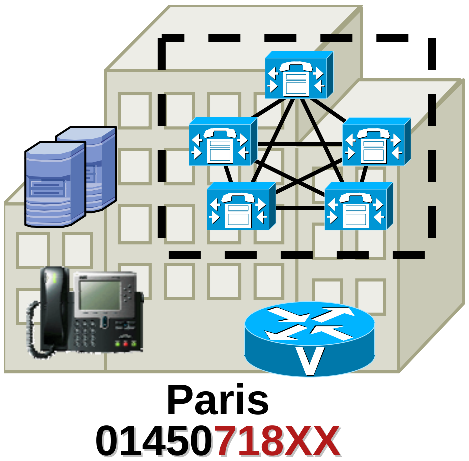




Planning Considerations
Uniform Dial Plans Are Simple (2)

Q: Fine! How about a 5 digit uniform dial plan?
A: Currently, yes! No overlap in the current ranges of DID numbers assigned.

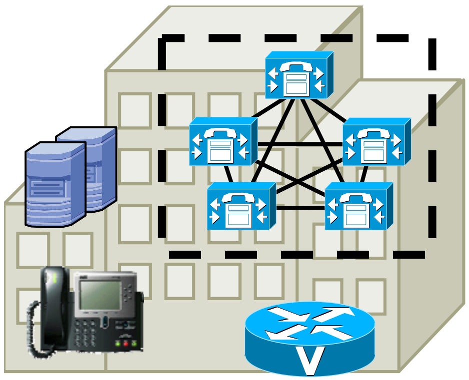
Q: Great! How about that new office we want to get in Nice? Room for it in our dial plan?
A: Sure. Well, maybe: it cannot use a DID range where the fourth digit after the prefix is 0, and cannot overlap with 575XX, 291XX, 754XX, 311XX, or 718XX…



Planning Considerations
Uniform Dial Plans Are Simple (3)

Q: If all I could get from the Telco in Nice is a DID range of 04935754XX, could I not dial 6 digits to reach a Nice phone, and 5 digits anywhere else? That way, I avoid the overlap between Nice and Lille.

A: No! Because calls to Lyon (e.g.: 57540) will sometimes overlap with calls to Nice's phones (e.g.: 575403), forcing the inter-digit timeout to occur before the call is routed.



Q: What do I do now? Go to 6 digits?

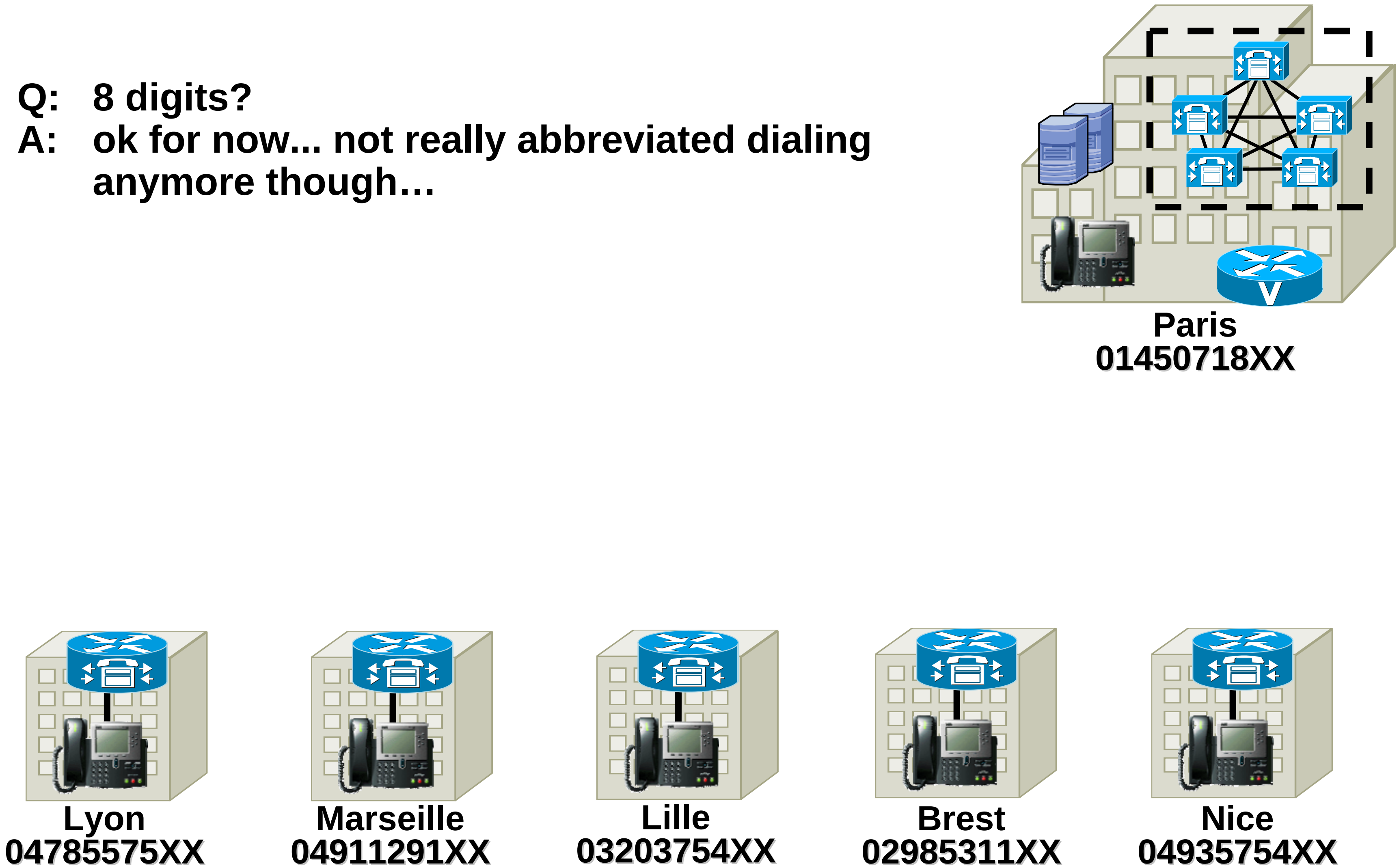
A: No: the Paris site has a 0 in the 6^{th} position. Overlaps with the PSTN access code…

Q: 7 digits?

A: No: Marseille starts with 112!

Planning Considerations
Uniform Dial Plans Are Simple (or so we hoped)

Q: 8 digits?
A: ok for now... not really abbreviated dialing anymore though…

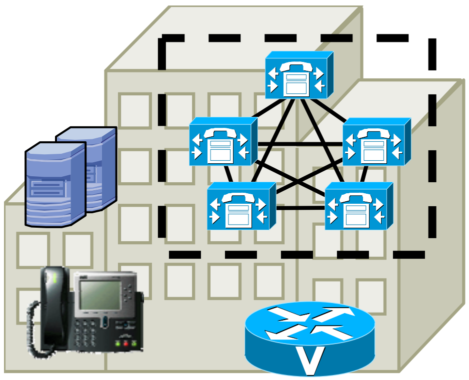


Planning Considerations

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 8 + 5 digits for on-net, inter-site calls?

A: No!
You essentially have the following to play with: 2XXXX, 3XXXX, 4XXXX, 5XXXX, 6XXXX, 7XXXX, 9XXXX
250 DID ranges, the need for more than a whole 5 digit range for a single site, and dividing the rest into 250 un-equal parts. Future planning, numbering plan changes, etc…

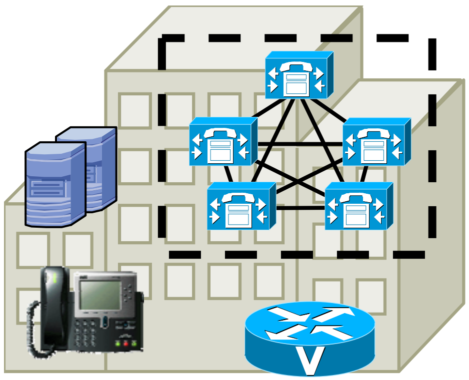


Planning Considerations

What if I have many, many more sites? More users? (2)

Q: What to do?

A: Site codes are a good idea.
- 0 = outside line, all combinations
- 8 + site code (3 digits would work up to 1000 sites), followed by a 4 digit extension
- [2-79]XXX: on-net, intra-site dialing



Paris
014455XXXX
014507XXXX
Site code 123
(and 124)

Toulouse
056155XXXX
Site code 012

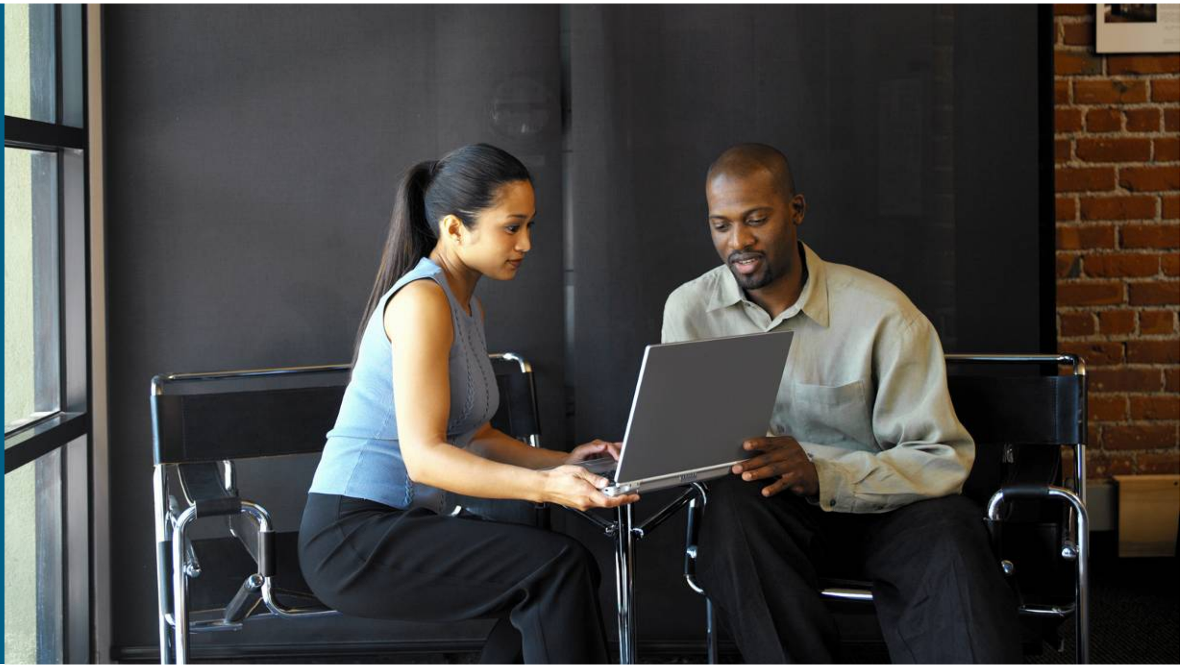
Clermont-Ferrand
04736651XX
Site code 345

Strasbourg
0388775XXX
Site code 256

Metz
0387441XXX
Site code 390

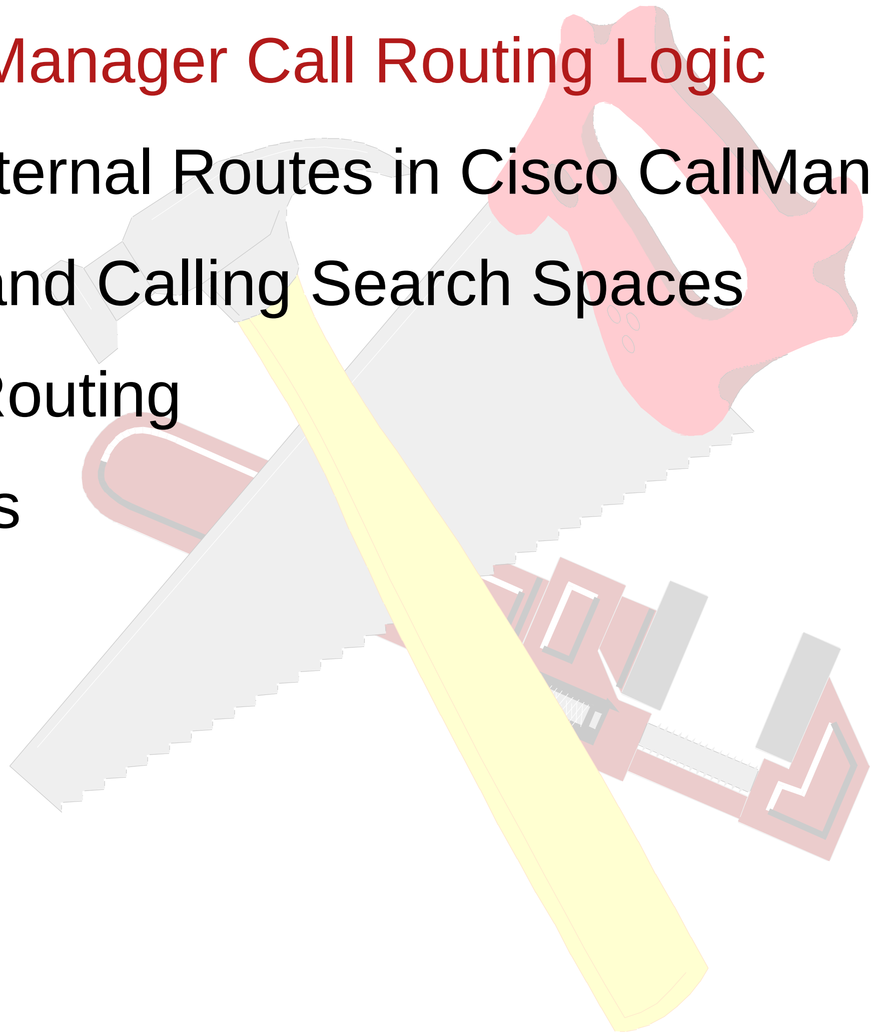
Bordeaux
05569954XX
Site code 822

Dial Plan Elements

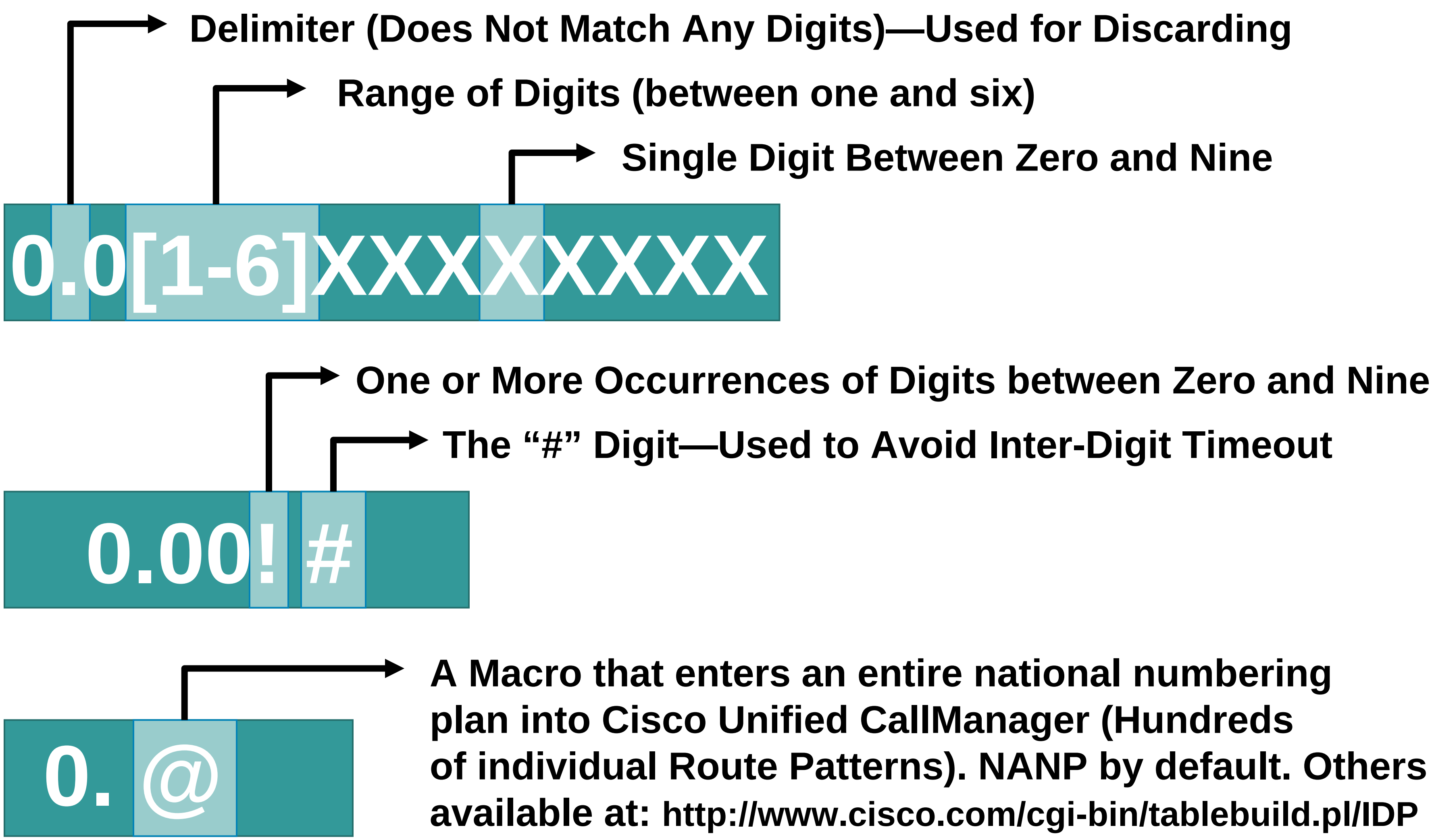


Dial Plan Elements Agenda

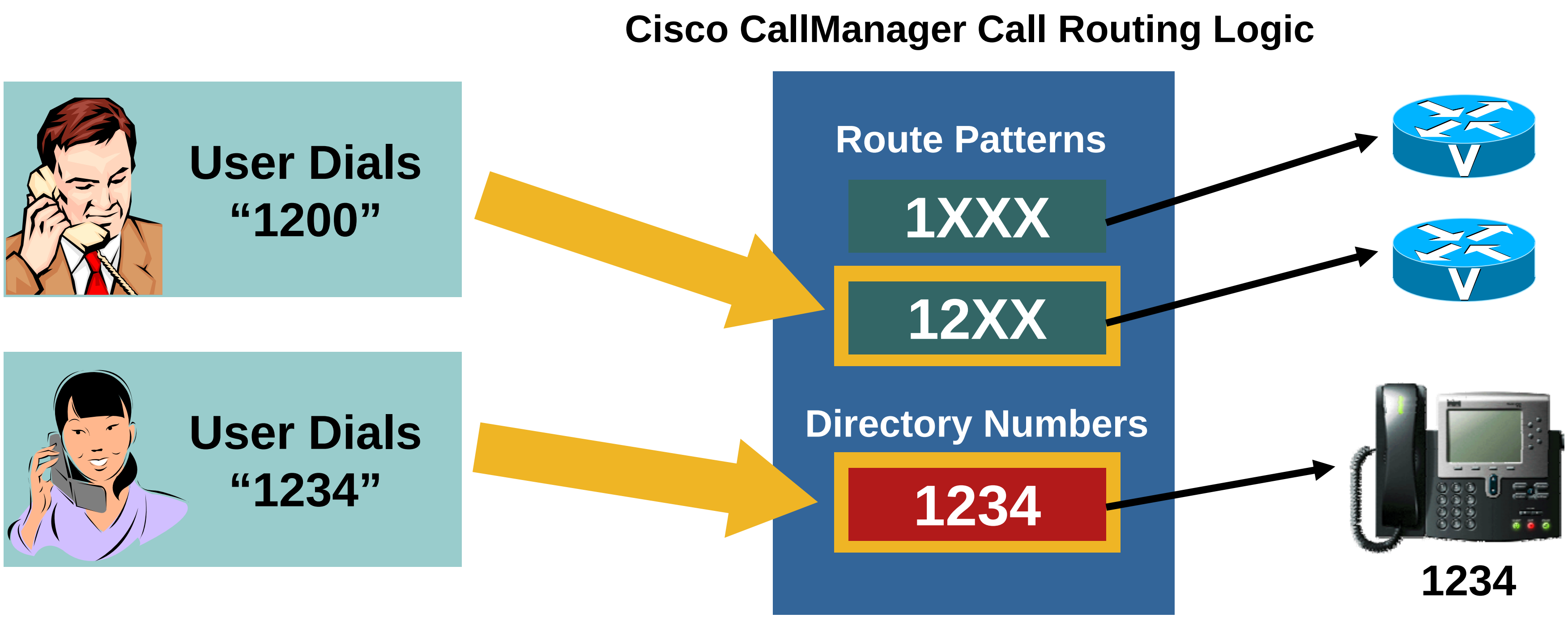
- Cisco CallManager Call Routing Logic
- External/Internal Routes in Cisco CallManager
- Partitions and Calling Search Spaces
- Alternate Routing
- Other Tools



Defining External Routes
Commonly Used Route Pattern Wildcards



Cisco CallManager Call Routing Logic
Basic Principle



- Cisco CallManager matches the most specific pattern (longest-match logic)
- An IP phone directory number is a special case of route pattern that matches a single number

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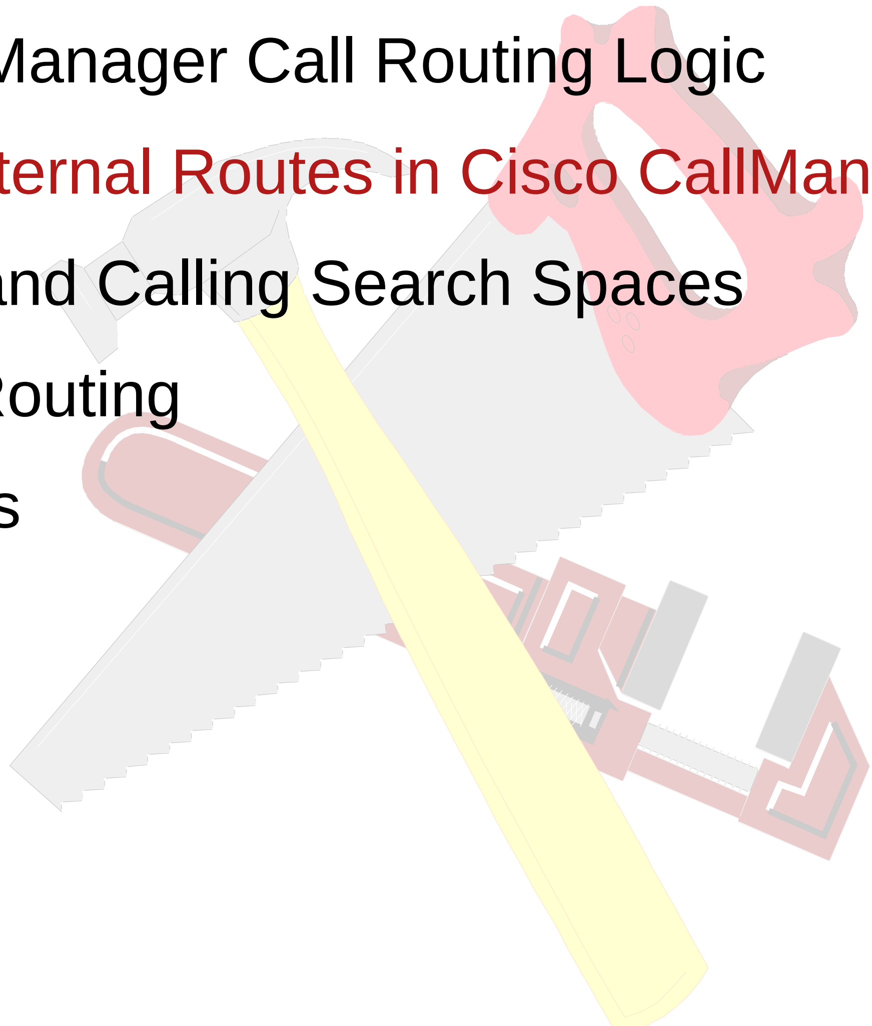
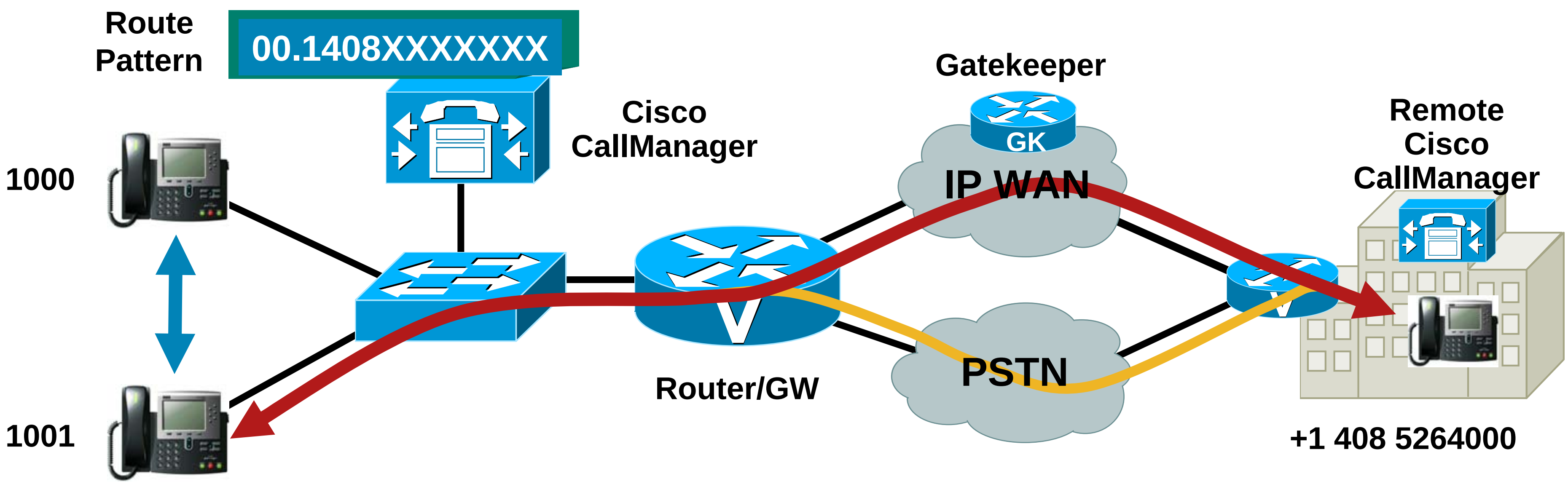


Dial Plan: The “IP Routing” of IP Telephony



Cisco CallManager Routes Two Basic Call Types:

On-Cluster Calls: Destination Directory Number (DN) is registered with Cisco CallManager. DNs are considered “internal” routes.

Off-Cluster Calls: External Route Patterns Must Be Configured on Cisco CallManager

Alternate routes: Allow On-Cluster and Off-Cluster calls to attempt alternate paths to destination (e.g.: IP WAN not available, go through PSTN)

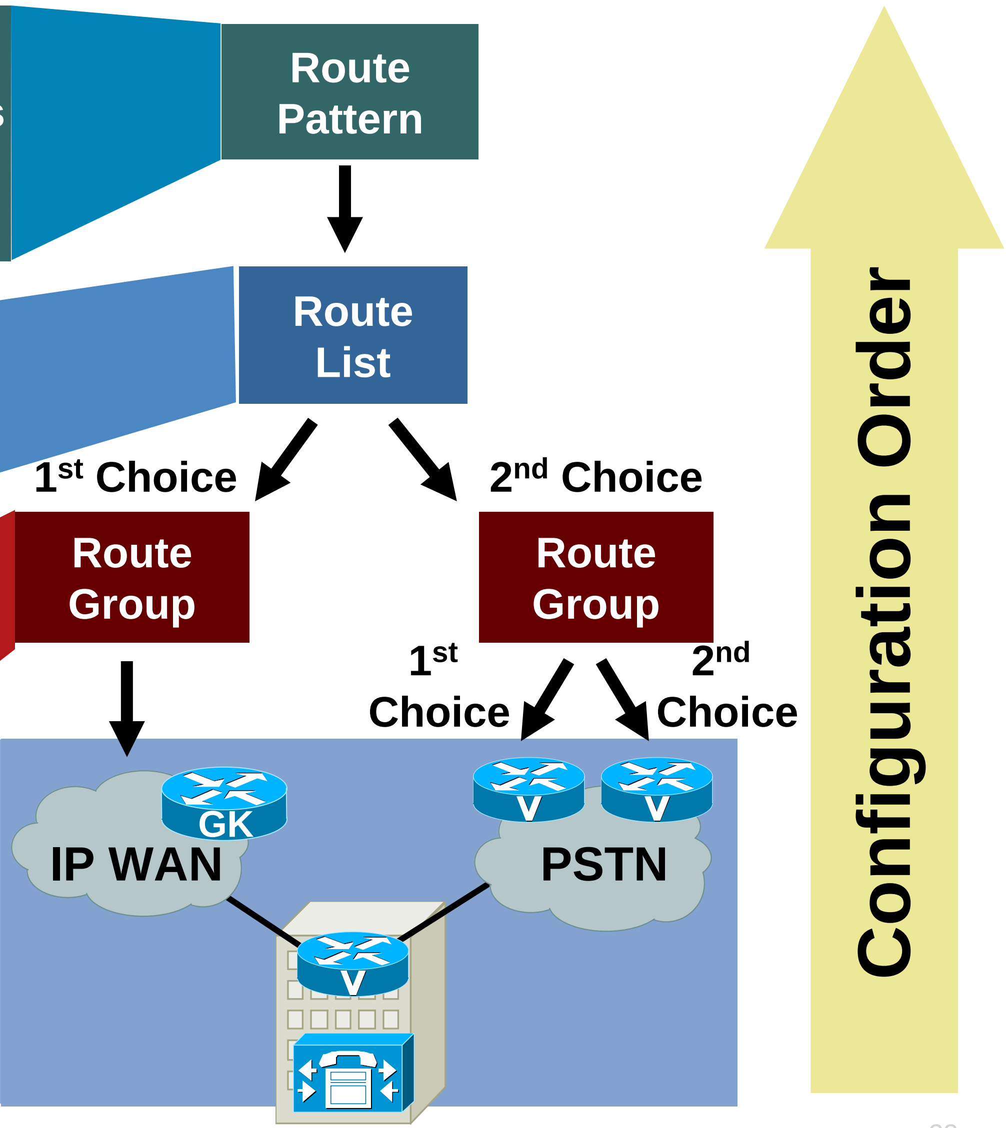
External Routes in Cisco CallManager
Overall Structure

Route Pattern
- Matches dialed number for external calls
- Performs digit manipulation (optional)
- Points to a route list for routing

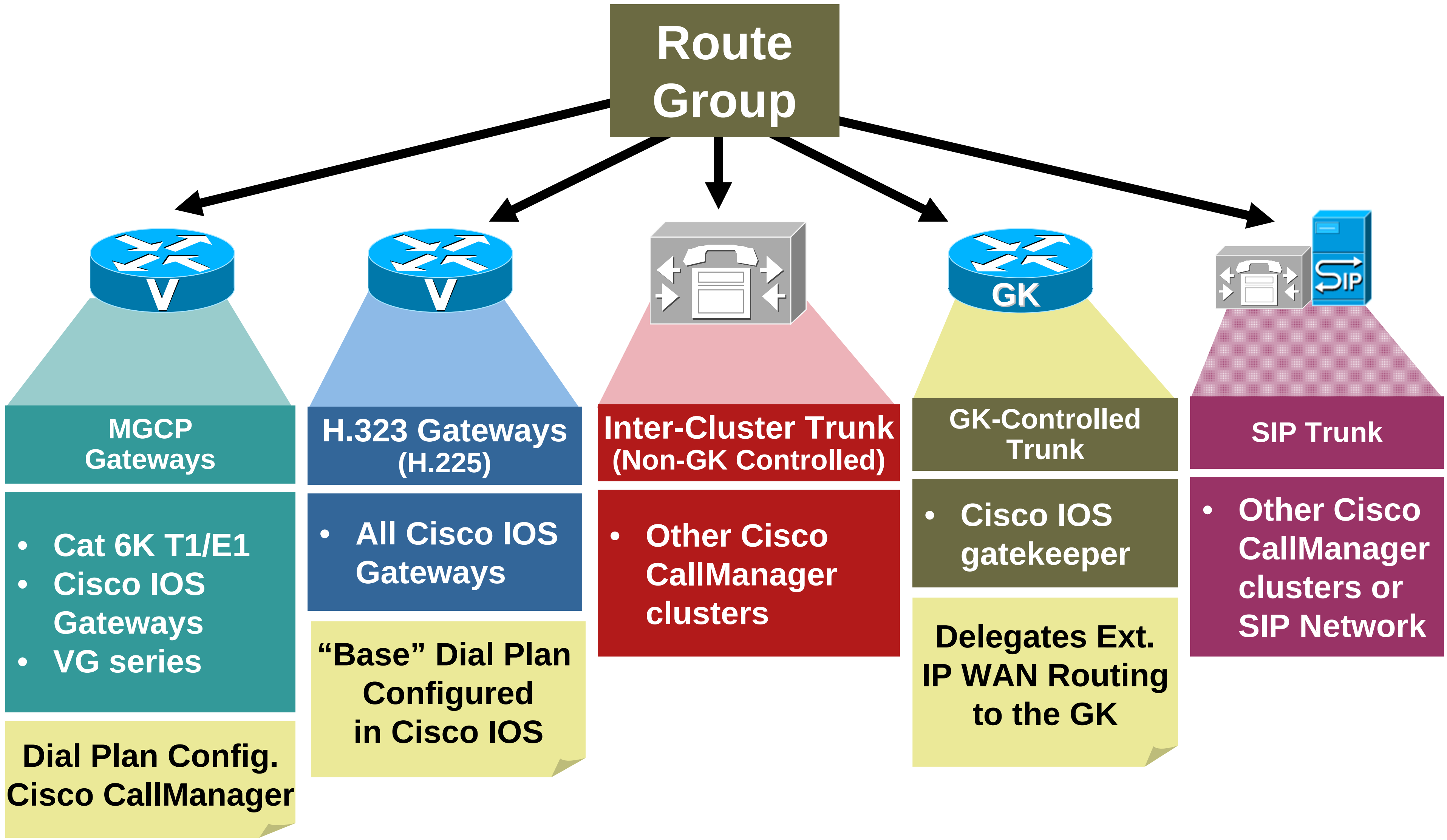
Route List
- Chooses path for call routing
- Points to prioritized route groups

Route Group
- Performs digit manipulation
- Points to the actual devices

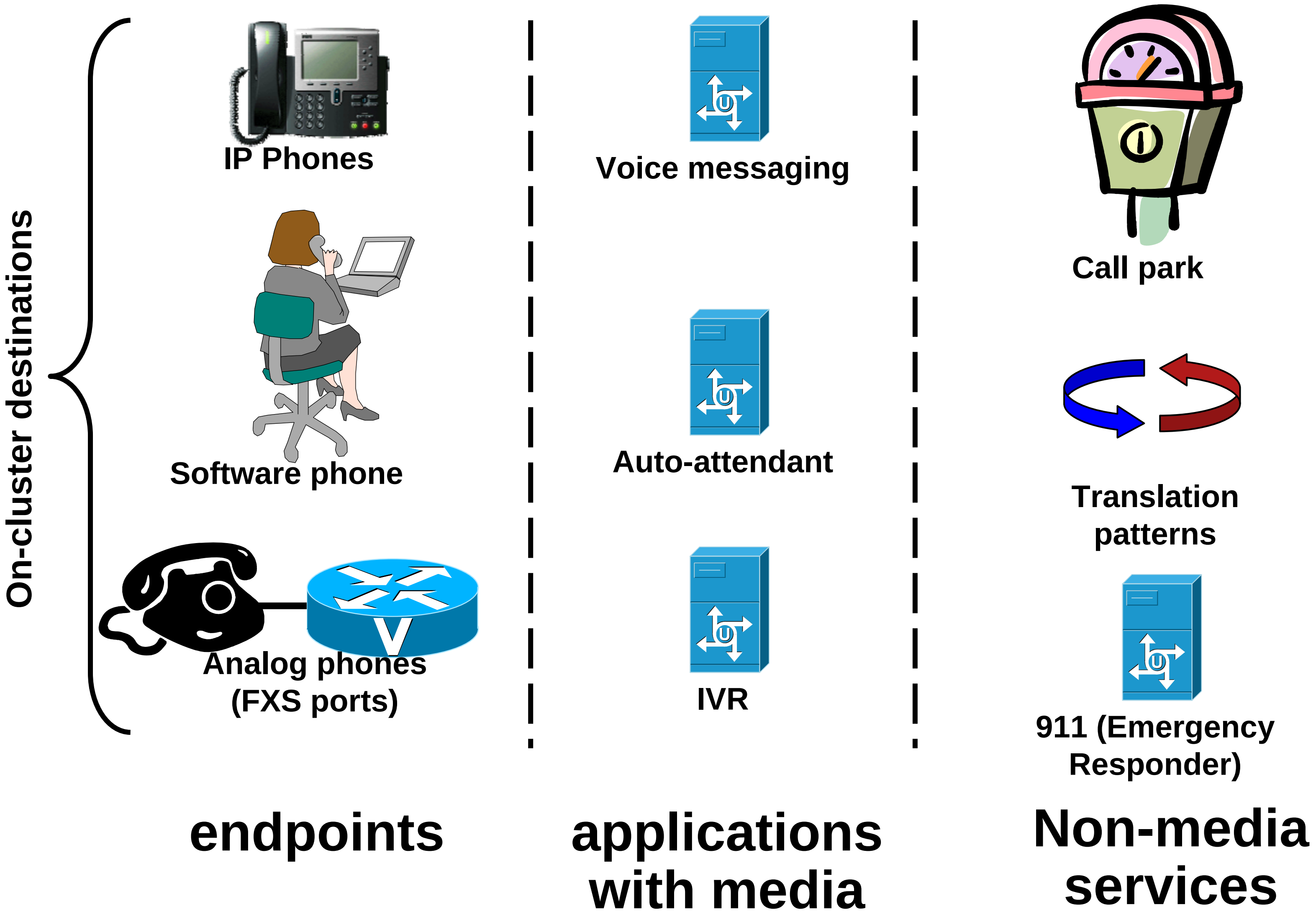
Devices
- Gateways (H.323, MGCP)
- Gatekeeper
- Trunk (H.225, ICT, SIP)



Defining External Routes
Route Group Device Types



Internal Routes in CallManager



Dial Plan Elements Agenda

- Cisco CallManager Call Routing Logic
- External/Internal Routes in Cisco CallManager
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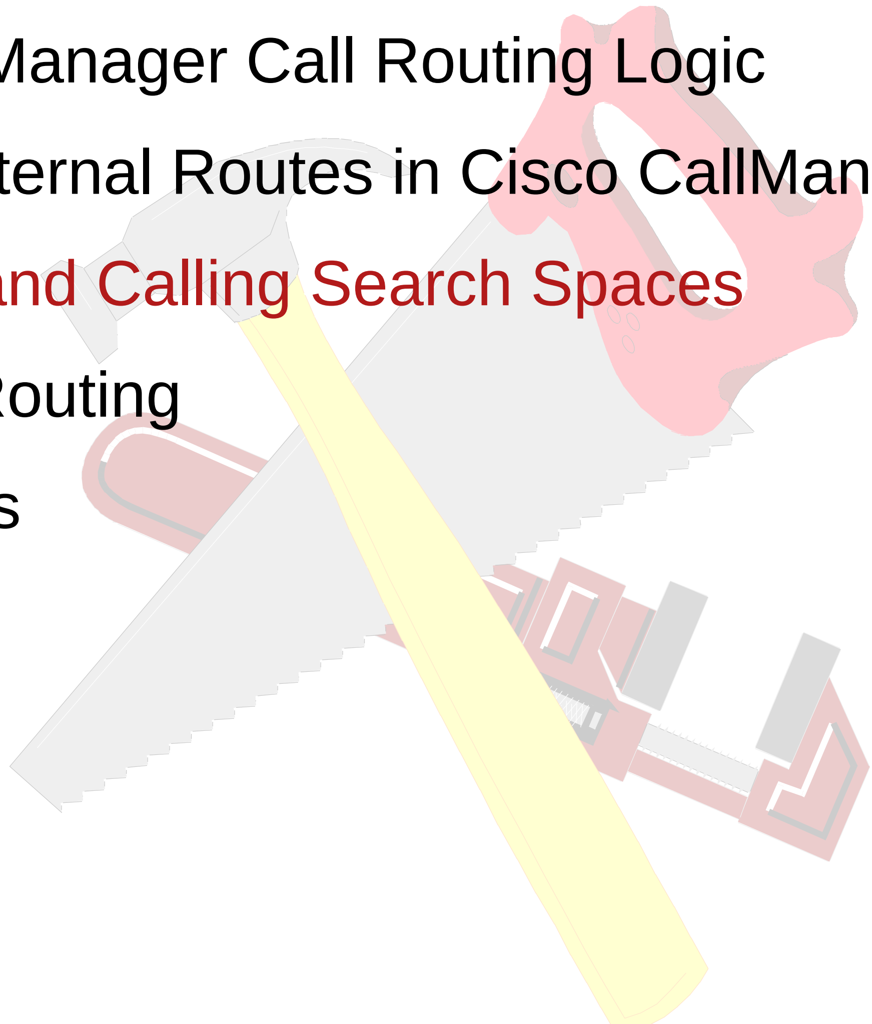
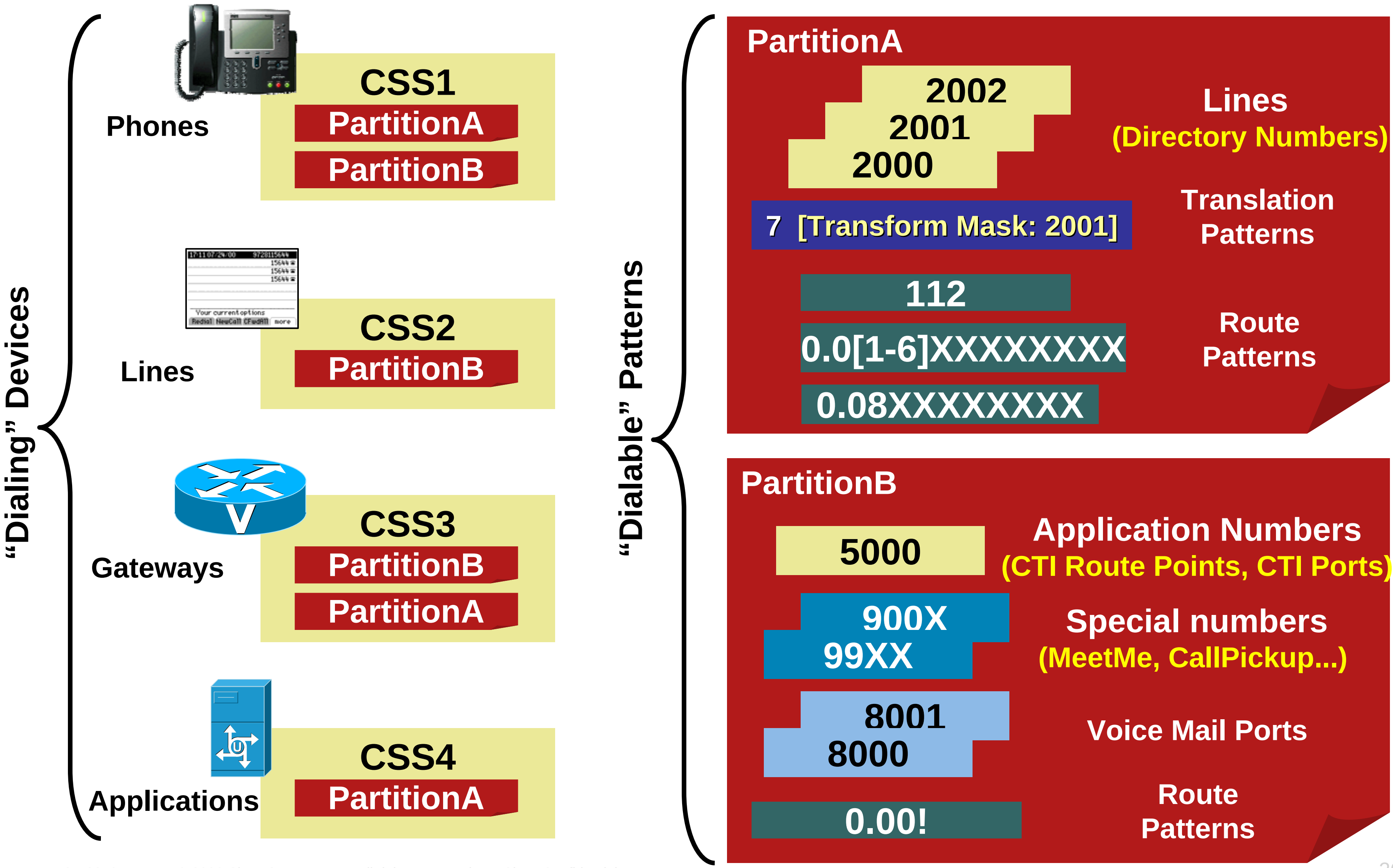


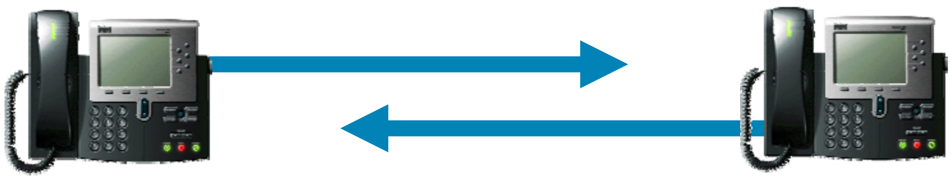
Building Classes of Service
Partitions and Calling Search Spaces



Partitions and Calling Search Spaces
Q3: Quick Quiz Question

What Is Needed for Phone A to Be Able to Call Phone B and Vice Versa?

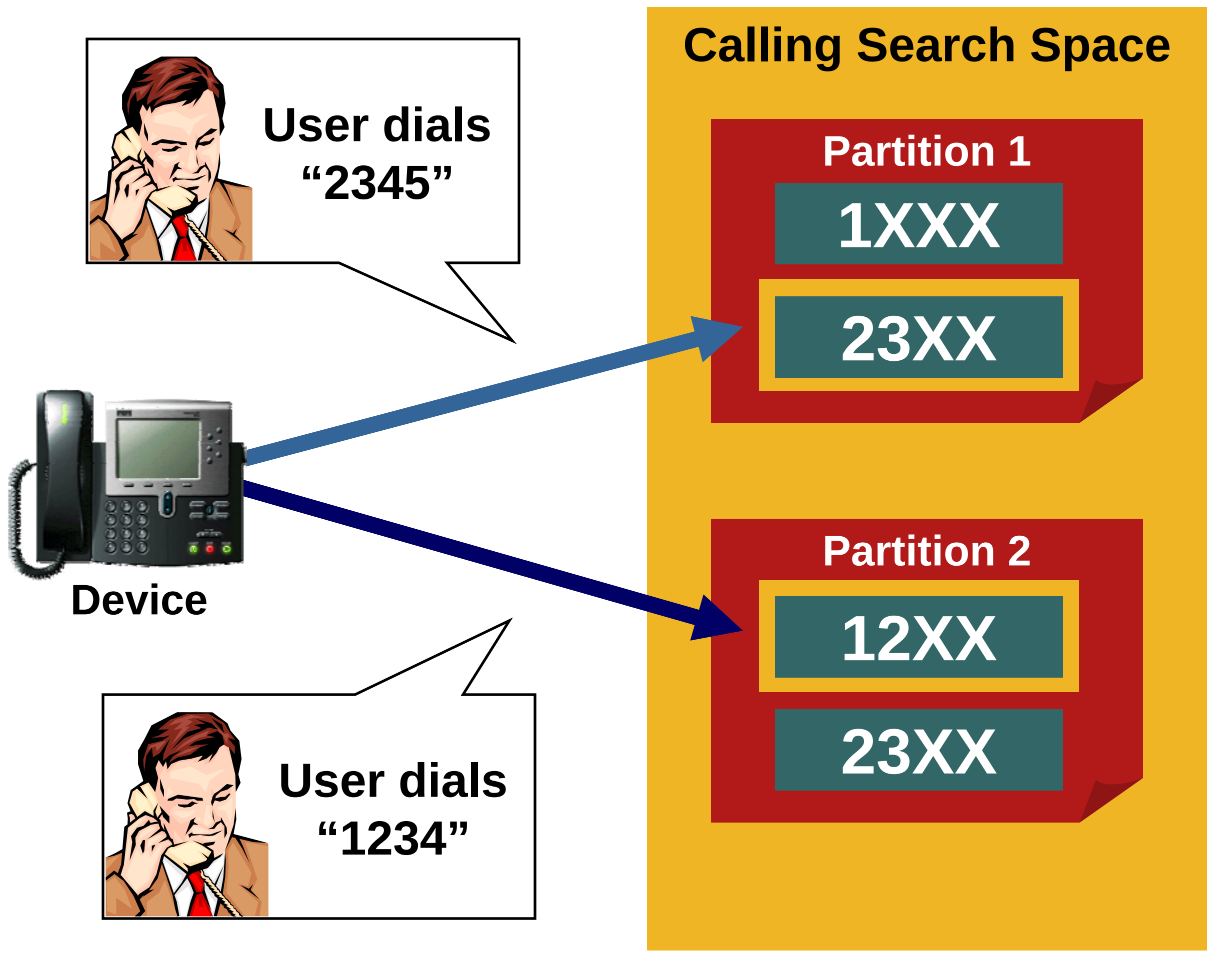
Phone A
Line 1000



Phone B
Line 2000

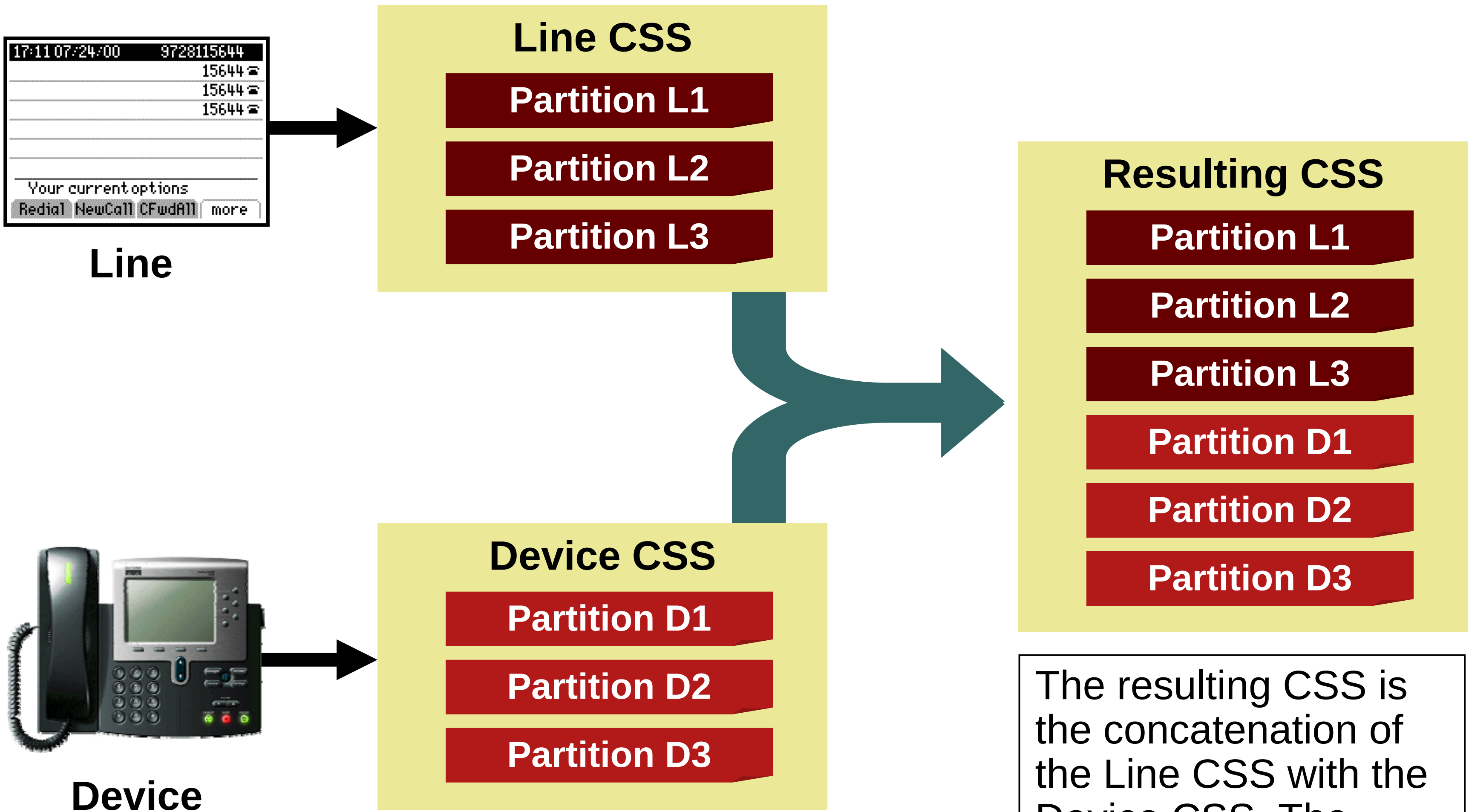
- ☐ Line 1000 and Line 2000 Must Be in the Same Partition
- ☐ Phone A and Phone B Must have same Calling Search Space
- ☐ All of the above
- ☐ None of the above

Partitions and Calling Search Spaces
Impact of Partition Order in a CSS



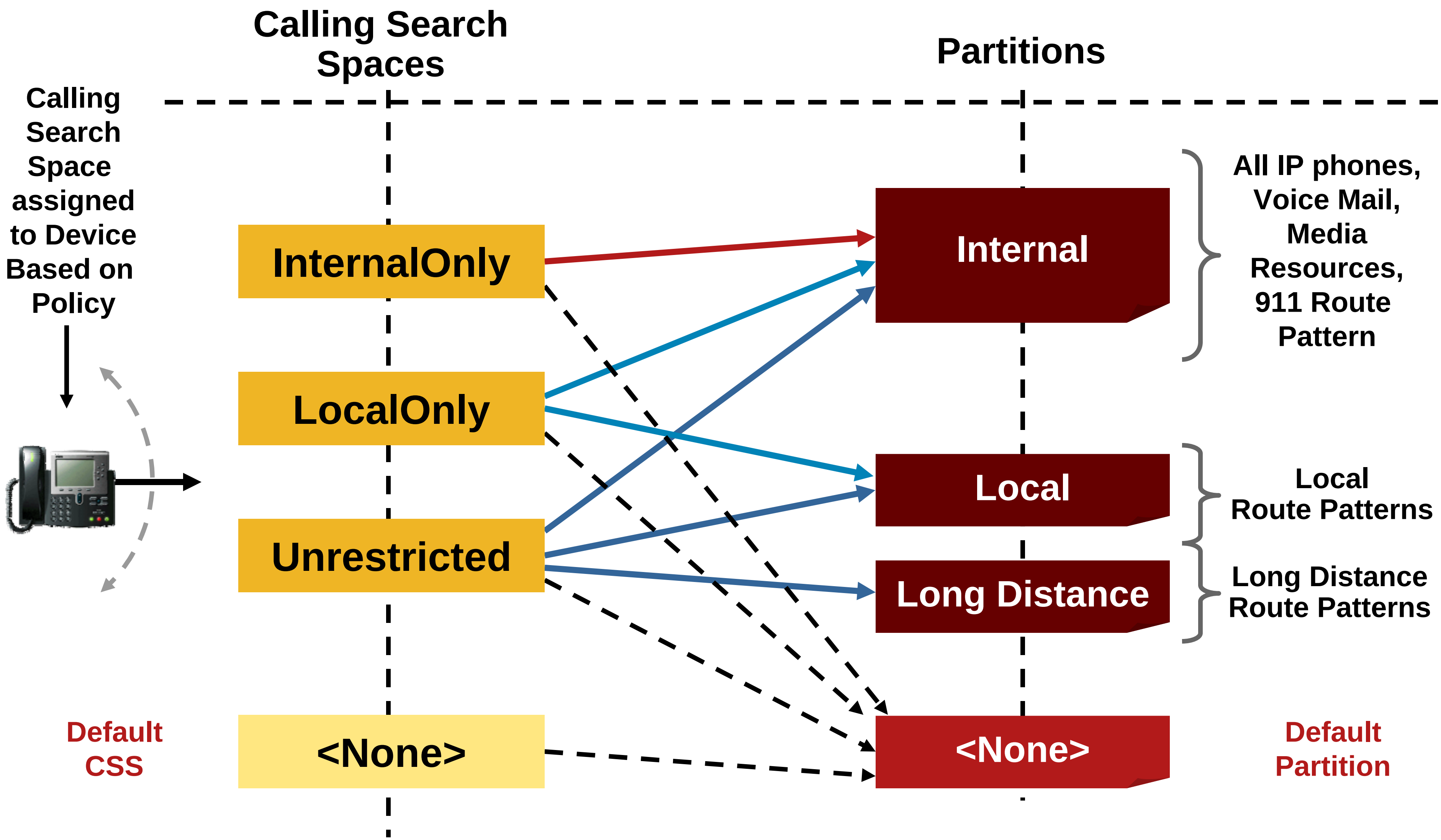
- Most specific patterns are chosen irrespective of partition order
- Partition order is only used as a tie-breaker in case of equal matches

Partitions and Calling Search Spaces
Device CSS-Line CSS Interaction

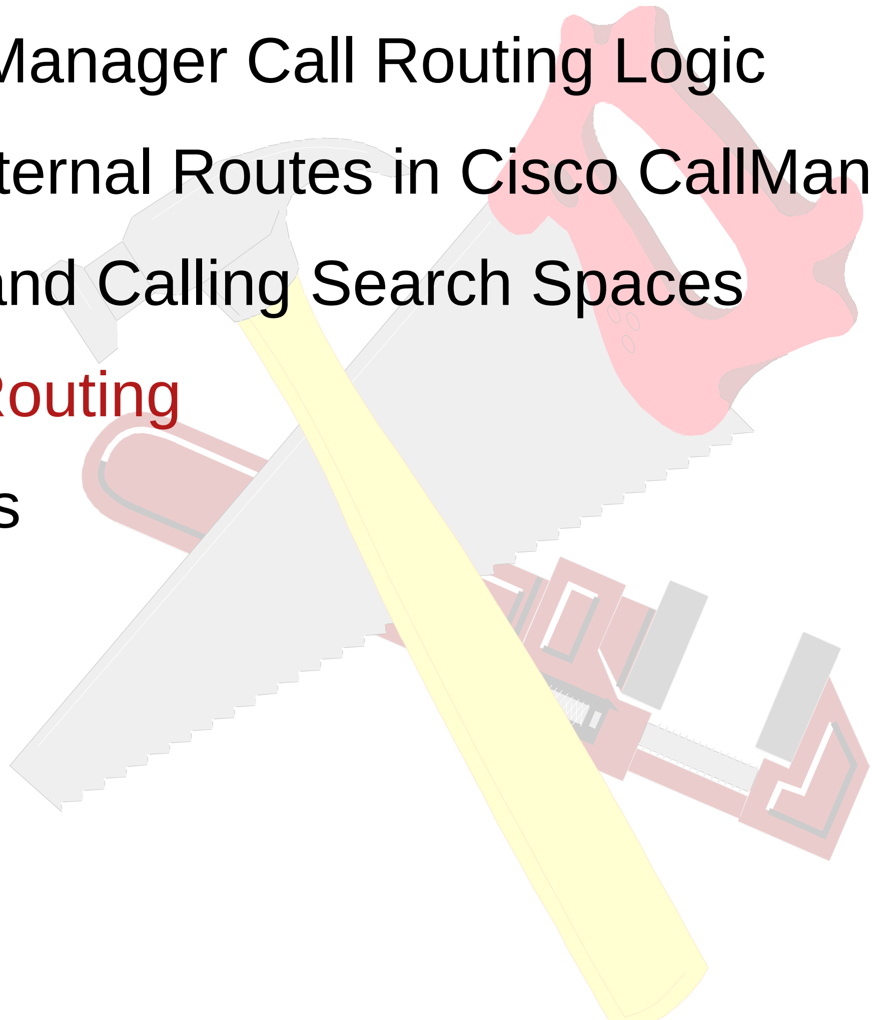


The resulting CSS is the concatenation of the Line CSS with the Device CSS. The <none> CSS is always implied at the end.

Partitions and Calling Search Spaces
Typical Use and Default Values



Dial Plan Elements Agenda



- Cisco CallManager Call Routing Logic
- External/Internal Routes in Cisco CallManager
- Partitions and Calling Search Spaces
- Alternate Routing
- Other Tools

Alternate Routing

- Multiple mechanisms to allow CUCM to route a call through an alternate path if the preferred path is not available

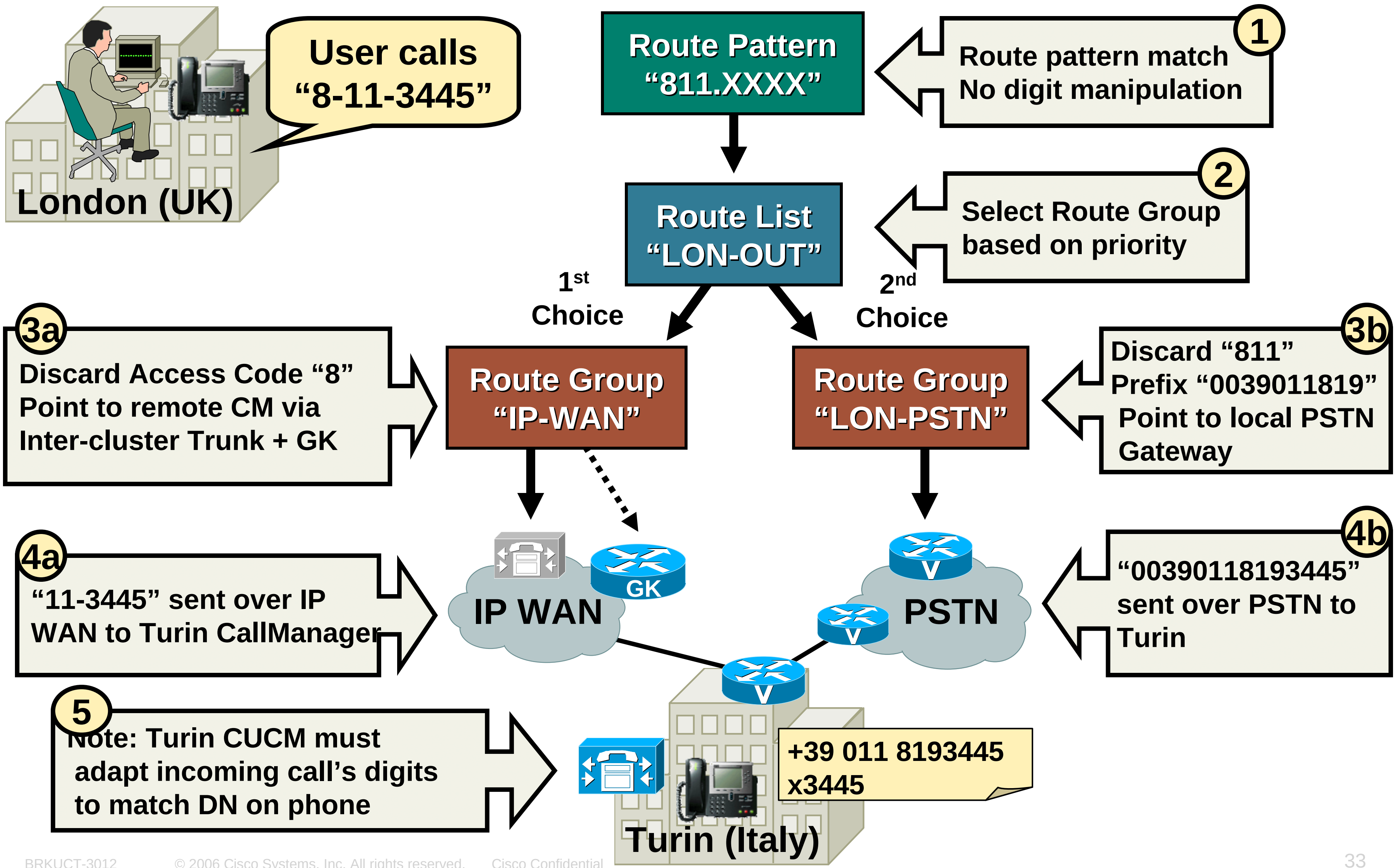
  e.g.: IP path not usable, then overflow the call through the PSTN

- External routes can use Route Lists / Route Groups
- Internal routes can use:

  Automated Alternate Routing for calls to on-net IP endpoints when there is not enough bandwidth

  Call Forward Un-Registered (CFUR) for calls to IP endpoints when the destination is unreachable (e.g.: a remote site in SRST)

Alternate Routing for External routes
The route list/route group construct

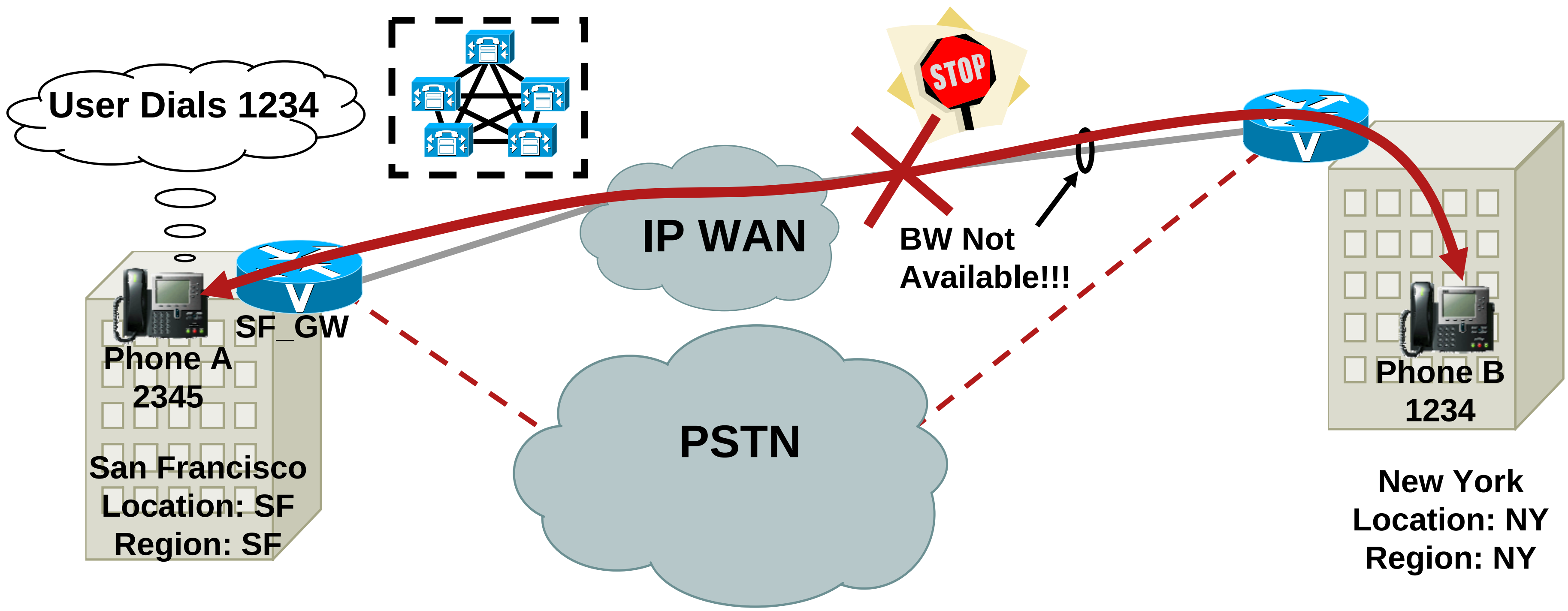


Alternate Routing for internal routes
CAC denial without AAR

Request: Call from
2345 to 1234?

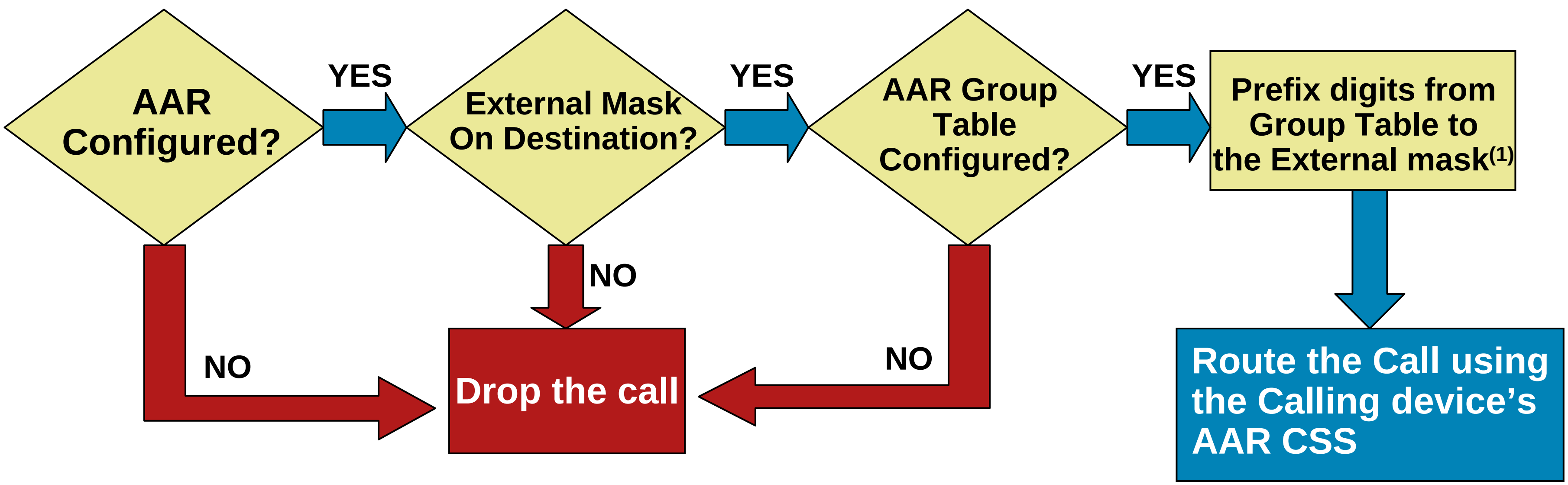
DO NOT PLACE
THAT CALL!!!

2345: Region SF; Location SF
1234: Region NY; Location NY
Codec SF-NY: G.729, ergo 24k
Location SF: 24k: OK!
Location NY: 1k Left: NO!!!



Alternate Routing for internal routes
AAR Situation with CallManager 4.0, 4.1 & 5.X

- Call is automatically re-routed using number configured in External Phone Number Mask when bandwidth is not sufficient (call admission control denial)
- AAR decision tree in CallManager 4.0, 4.1 & 5.X:



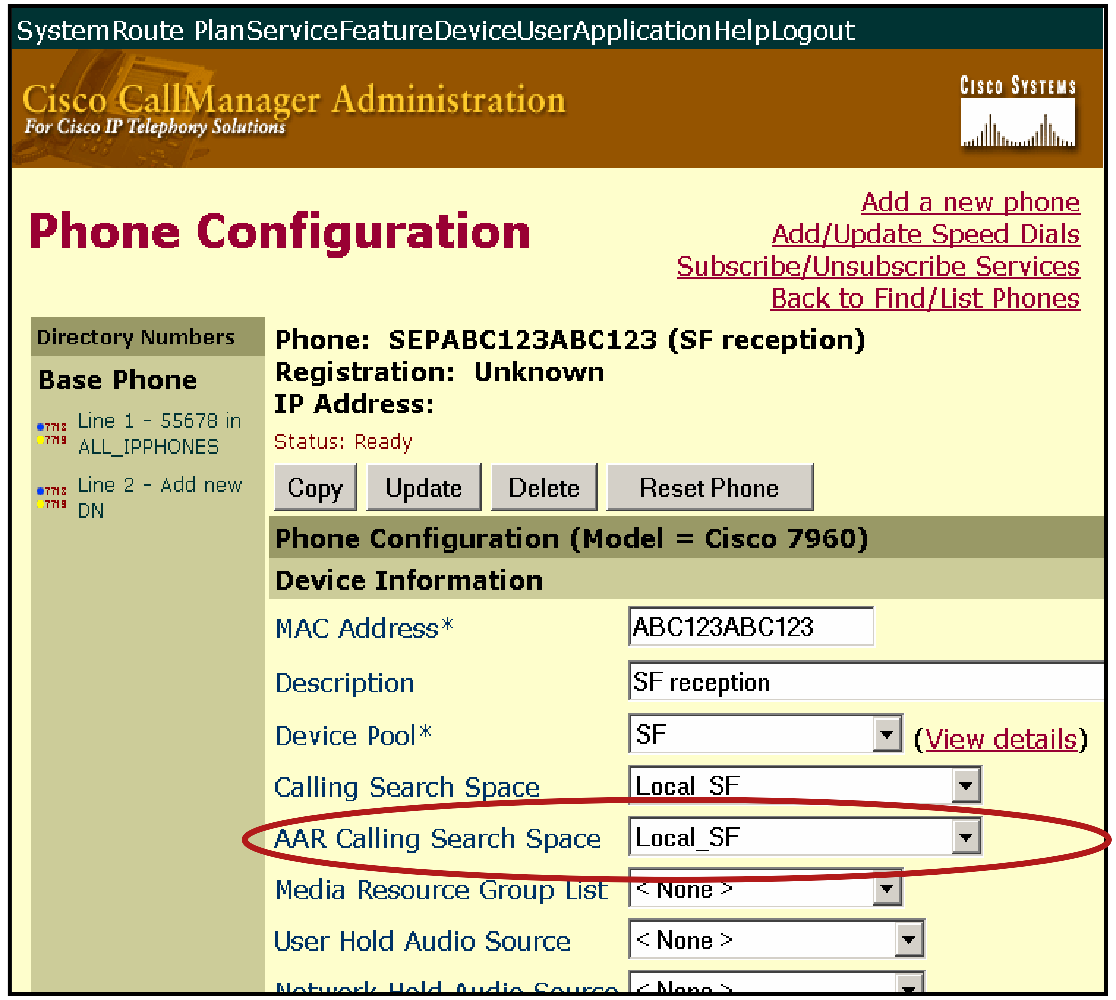
(1) Mask is combined with the digits dialed originally

Alternate Routing for internal routes
AAR Group Assigned to DN

- DNs are assigned to an AAR group
- But, the CSS used for AAR calls is on the device (see next slide)

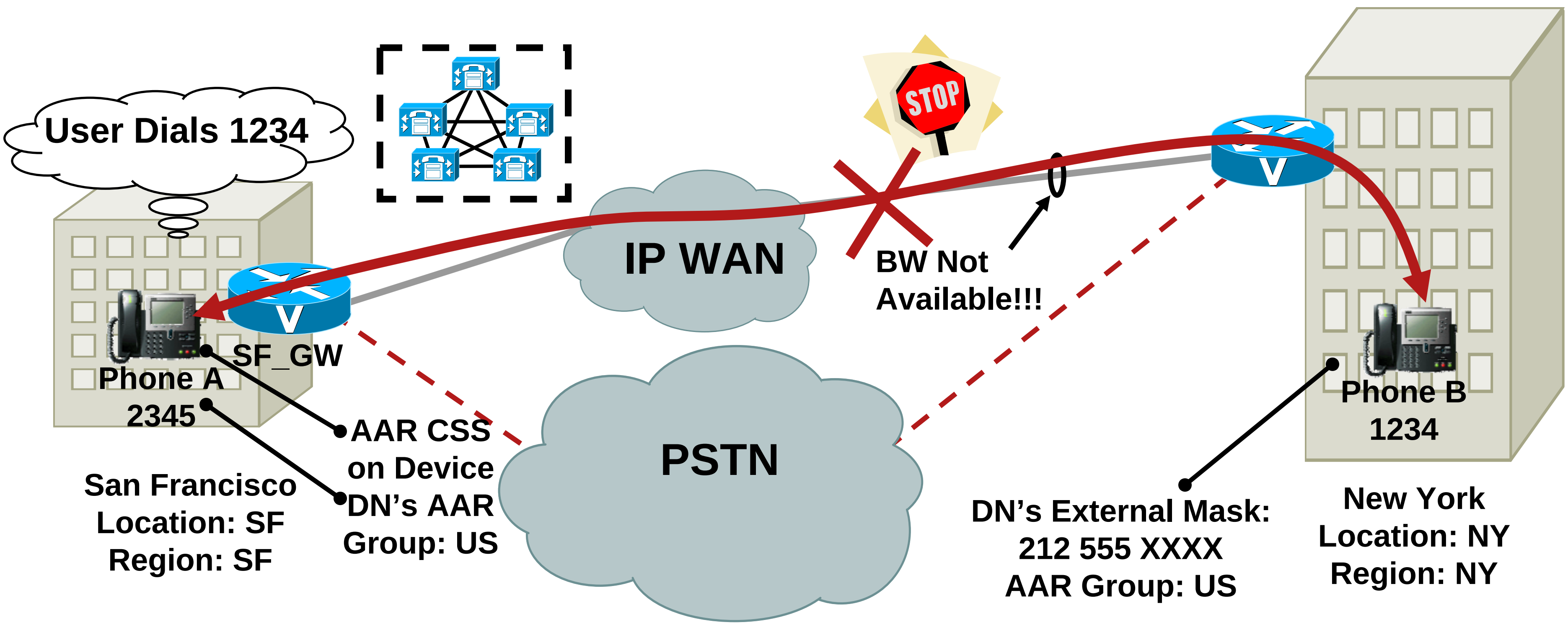
Alternate Routing for internal routes
AAR Calling Search Space

- Be mindful of this for extension mobility
- ***<u>This is how an AAR-specific route can be chosen</u>***
- GW typically needs to be co-located (since un-availability of WAN bandwidth is what triggers the AAR mechanism)



Alternate Routing for internal routes
AAR configuration Details

Called DN's External Party Phone Number Mask: 212555XXXX
AAR Groups Tell Me to Prefix 91, So New Destination Is: 912125551234
AAR CSS of Originating Device Contains R.P. 91[2-9]XX[2-9]XX XXXX Pointing to SF_GW
Let's Request a Call from 2345 to SF_GW

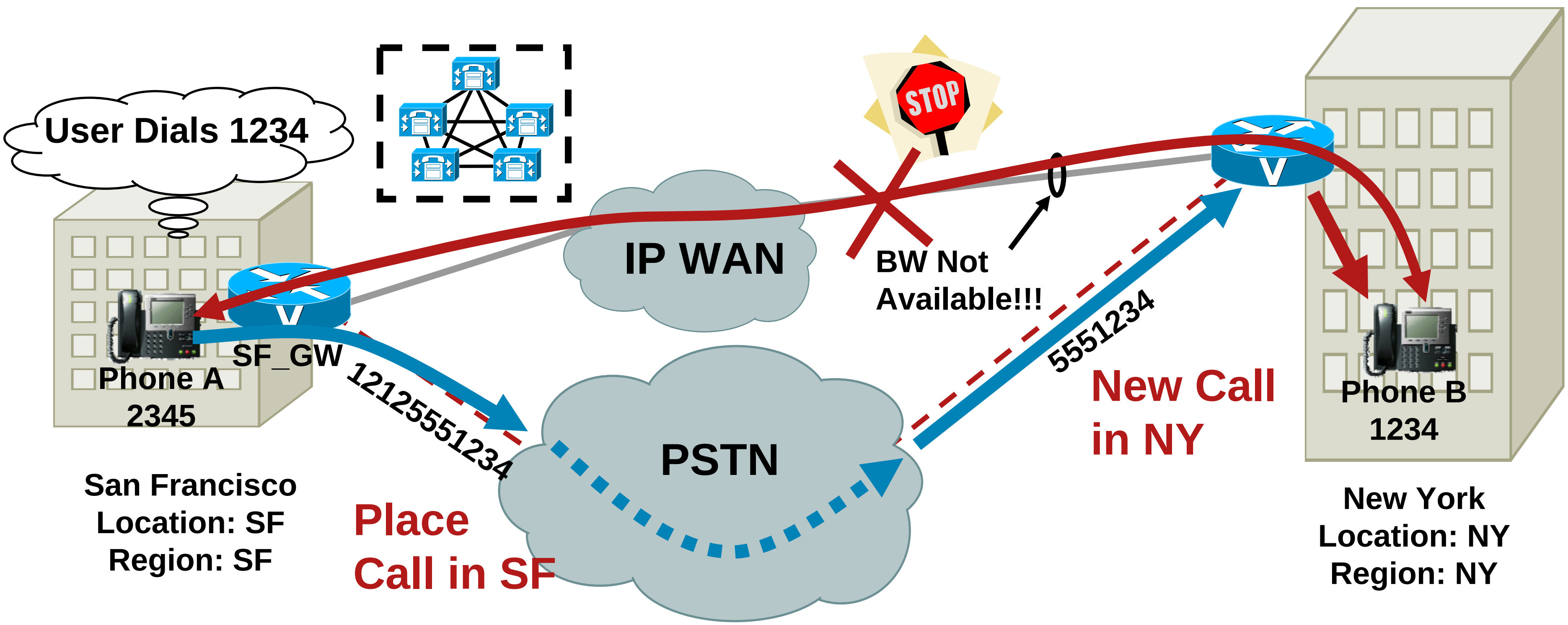


Alternate Routing for internal routes
AAR Rerouting the Call

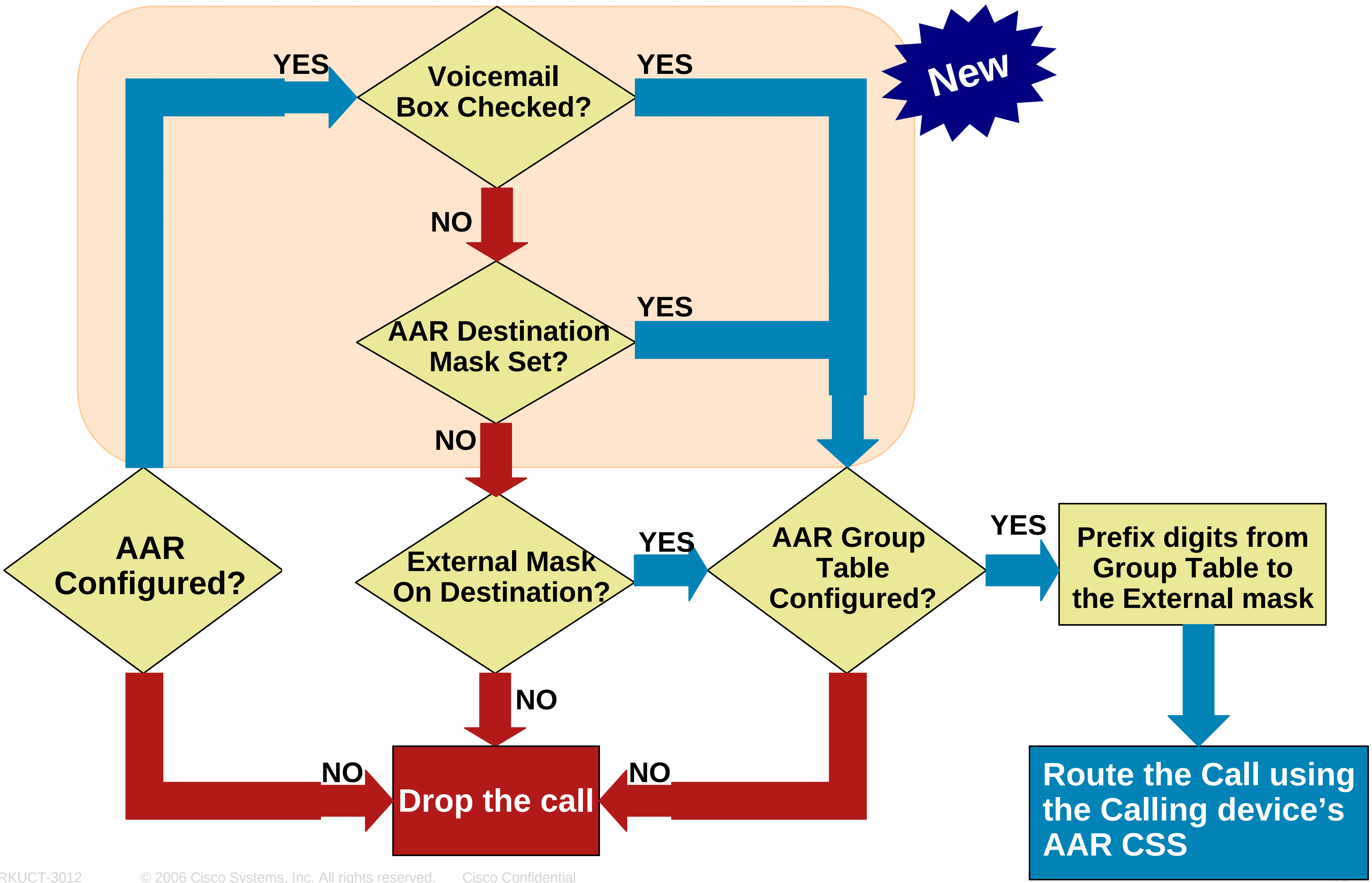
Request: Call from 2345 to SF_GW?

Go Ahead!!!

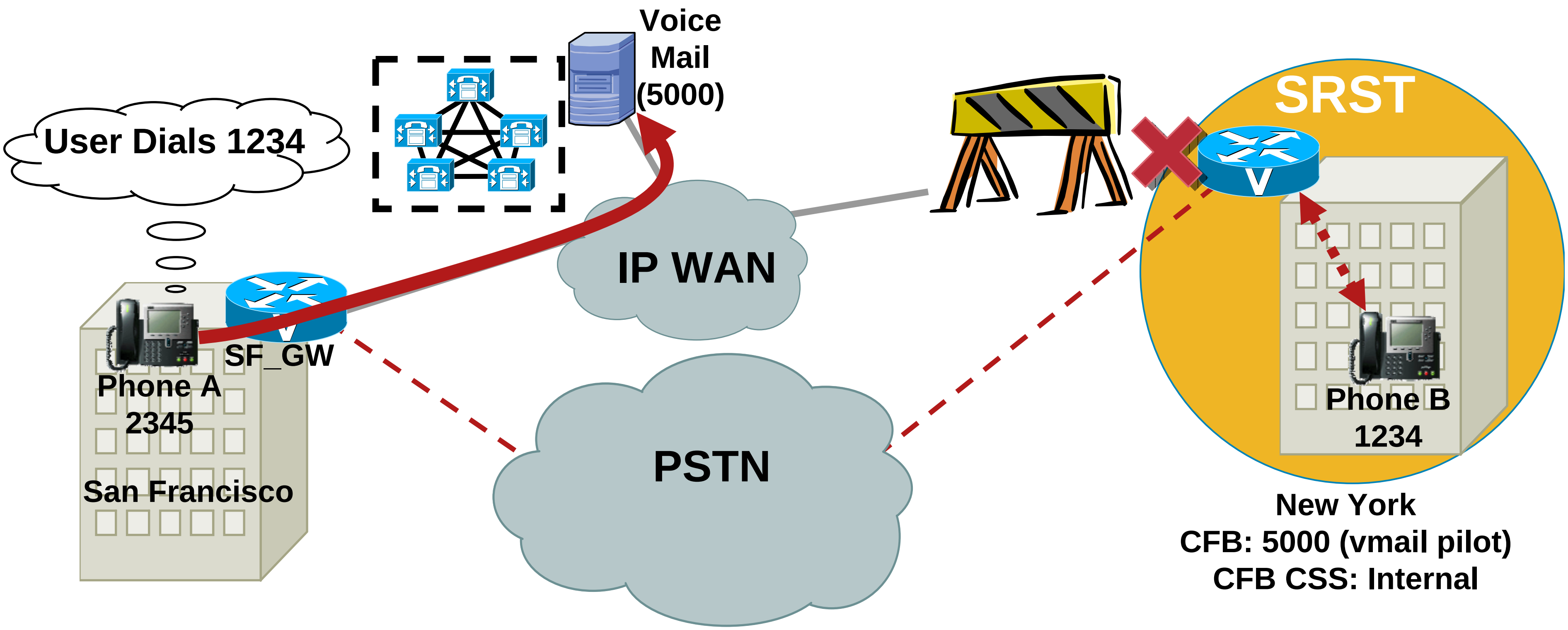
2345: Region SF; Location SF
SF_GW: Region SF; Location SF
Codec SF-SF: G.711, ergo 80k
Same Location: CAC OK!
GO!



Alternate Routing for internal routes
AAR decision Tree with CallManager 4.2

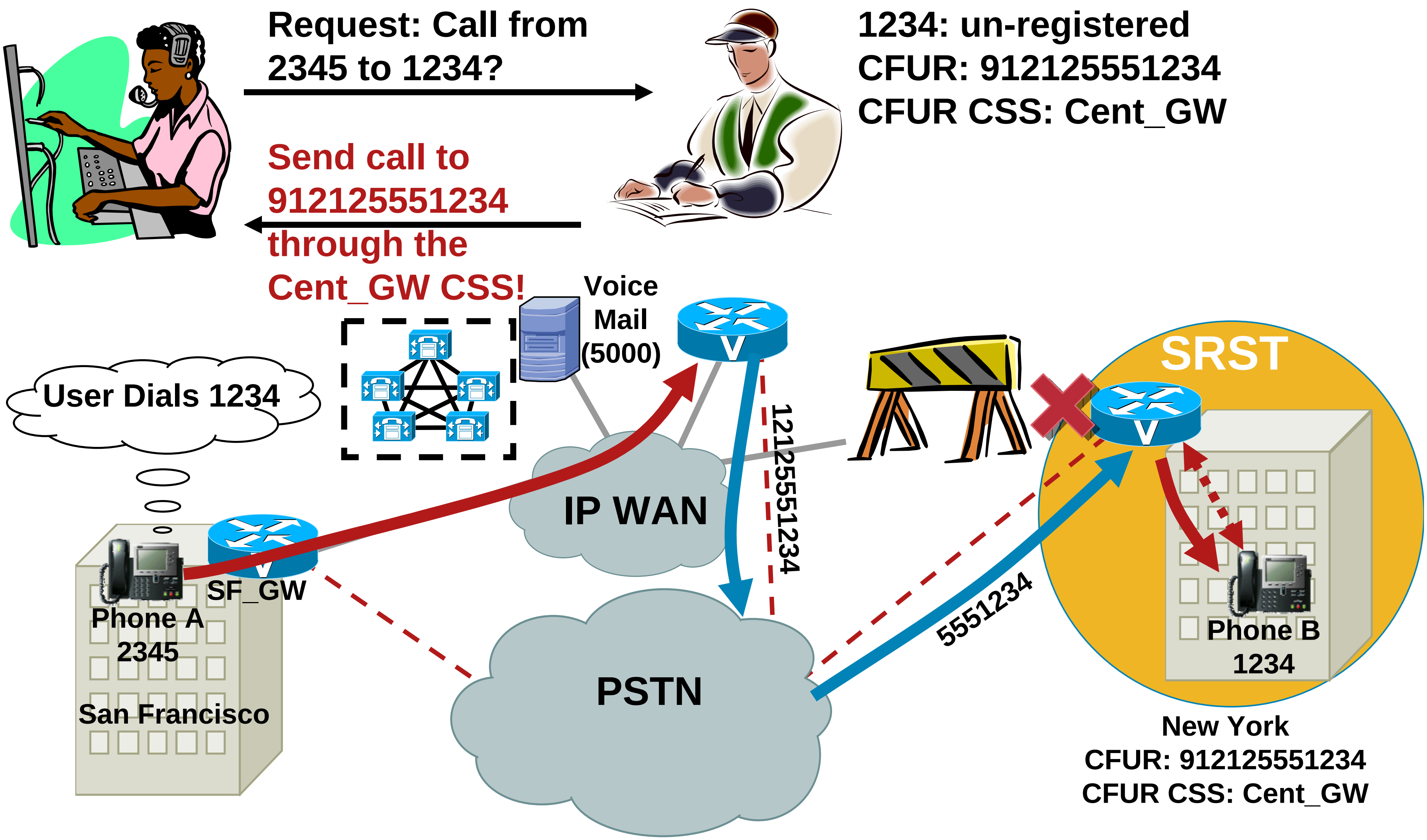


Alternate Routing for internal routes
Without Call Forward Unregistered (CFUR)



Alternate Routing for internal routes

With Call Forward Unregistered (CFUR)



Alternate Routing for internal routes
With Call Forward Unregistered (CFUR)

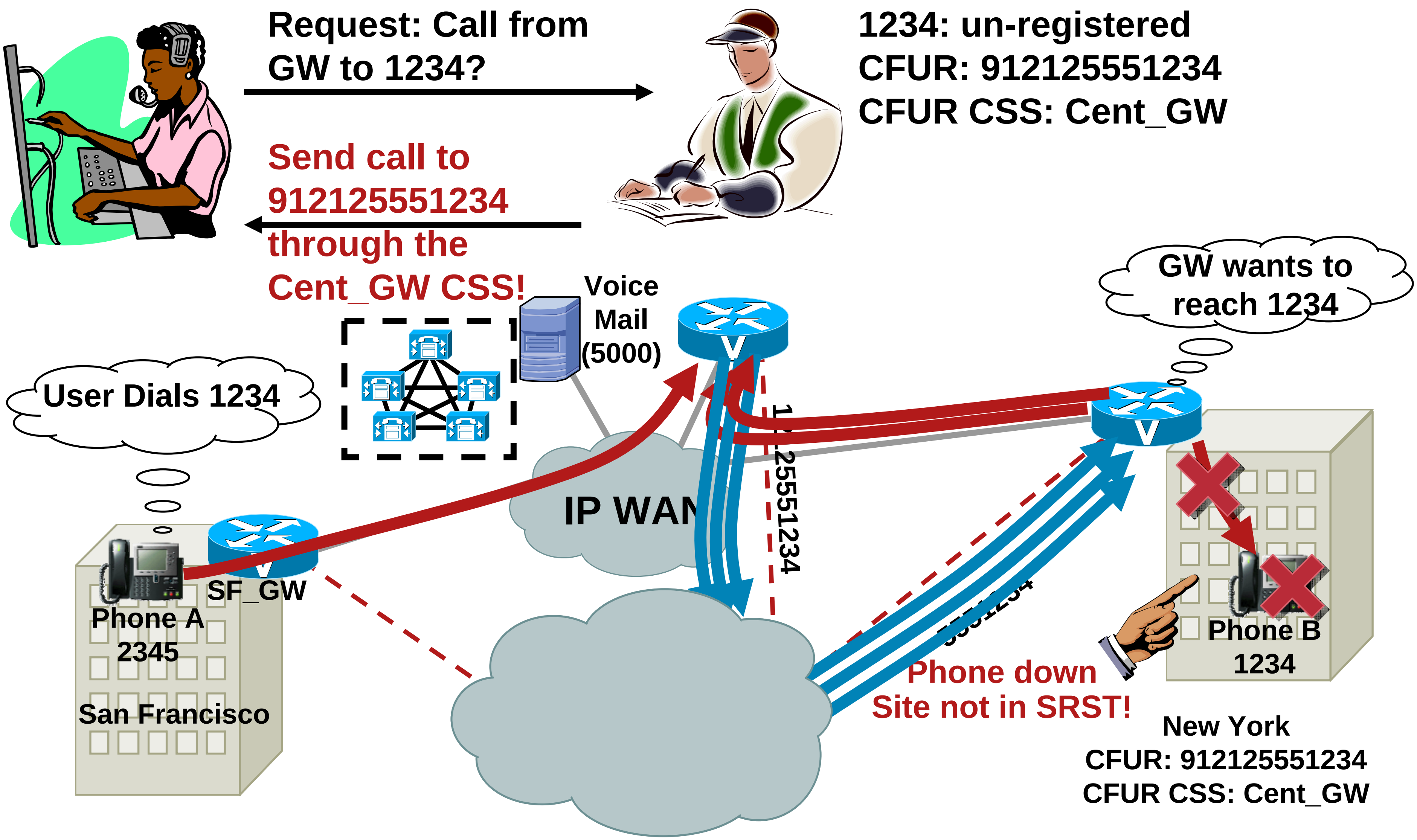
- Reroutes calls to unregistered DN's using number specified in "Call Forward Unregistered" (CFUR) field
- Destination number same irrespective of calling phone's PSTN dialing requirements: previous example a problem for say, a site in Europe where the dialed number should be 0 00 1 212 555 1234
- CFUR CSS same irrespective of calling phone's dial plan: not able to use different GW based on calling site

    If CFUR CSS is left to <none>, calling phone's CSS is used. ***NOT A PROTECTED FEATURE!!!!***

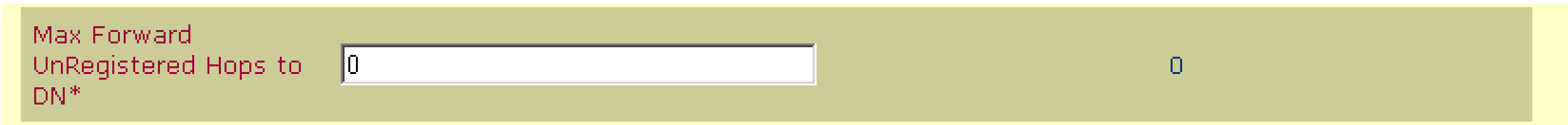
    Calling phone's class of service must allow call

- Number in CFUR field needs to include PSTN access codes
- What happens if phone is "merely" un-registered?
- Beware of loops: GWs should not be allowed to place calls to number ranges that deliver calls to the GW itself. Next page has illustration: we will be looking at what happens after the first CFUR attempt

Alternate Routing for internal routes
With Call Forward Unregistered (CFUR)



Alternate Routing for internal routes
With Call Forward Unregistered (CFUR)



- CFUR CSS cannot be expected to be able to avoid loops in this situation.
- CFUR is invoked whenever DN is unregistered, including when EM is logged out or the phone is unplugged
- Set service parameter to 1 (or 2) to limit loops *(value may need to be higher if forwarding “chains” are used for voicemail or other applications)*
- When looping call is dropped, caller hears fast-busy

Dial Plan Elements Agenda

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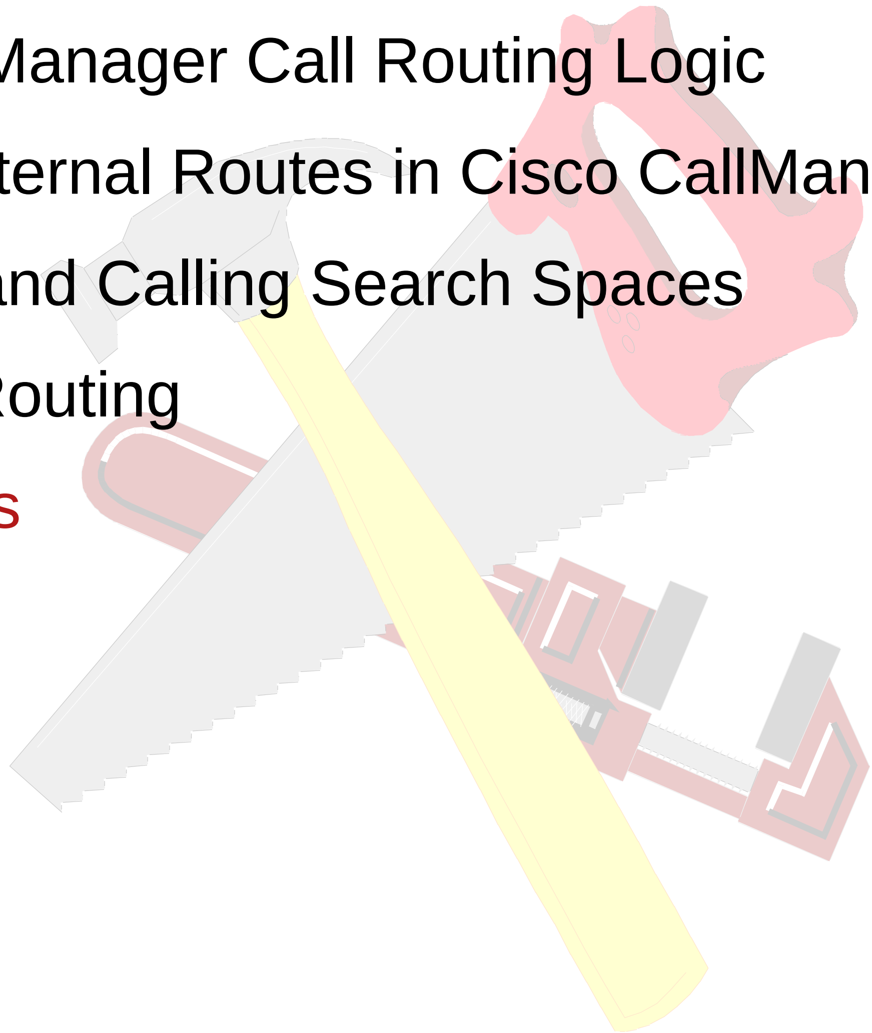
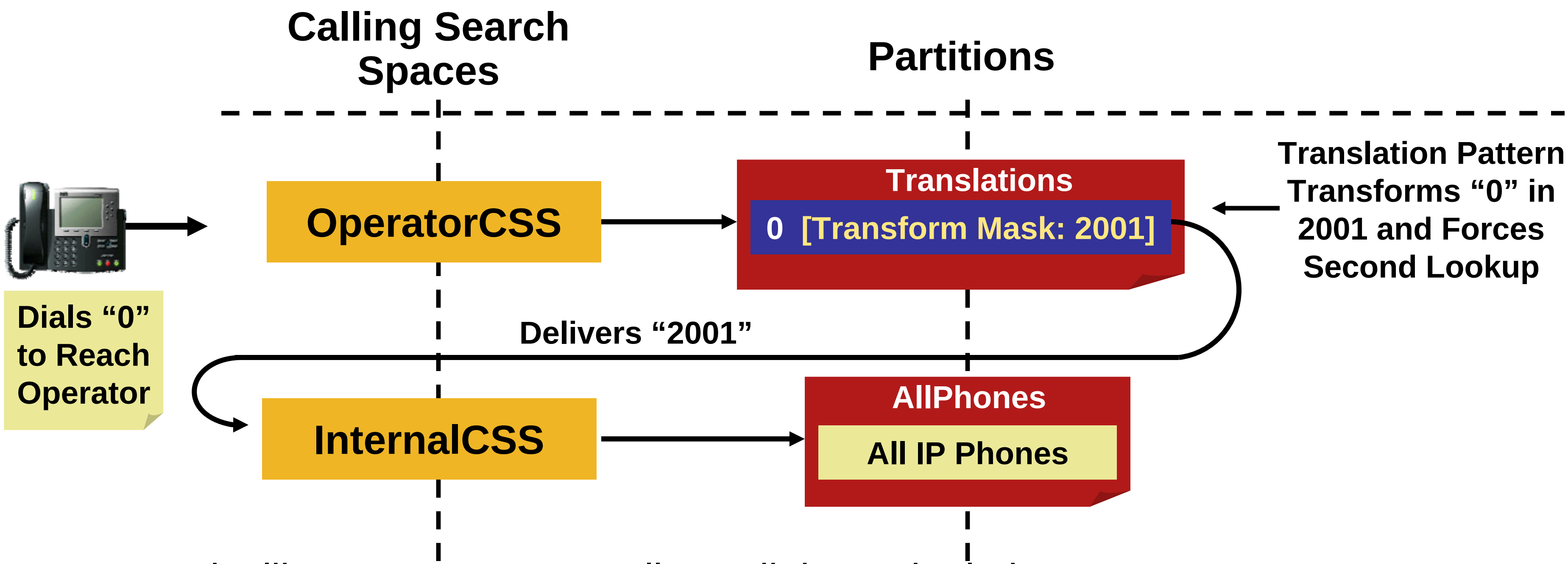
- External/Internal Routes in Cisco CallManager
- Partitions and Calling Search Spaces
- Alternate Routing
- Other Tools



Other Tools
Translation Patterns: The Basics



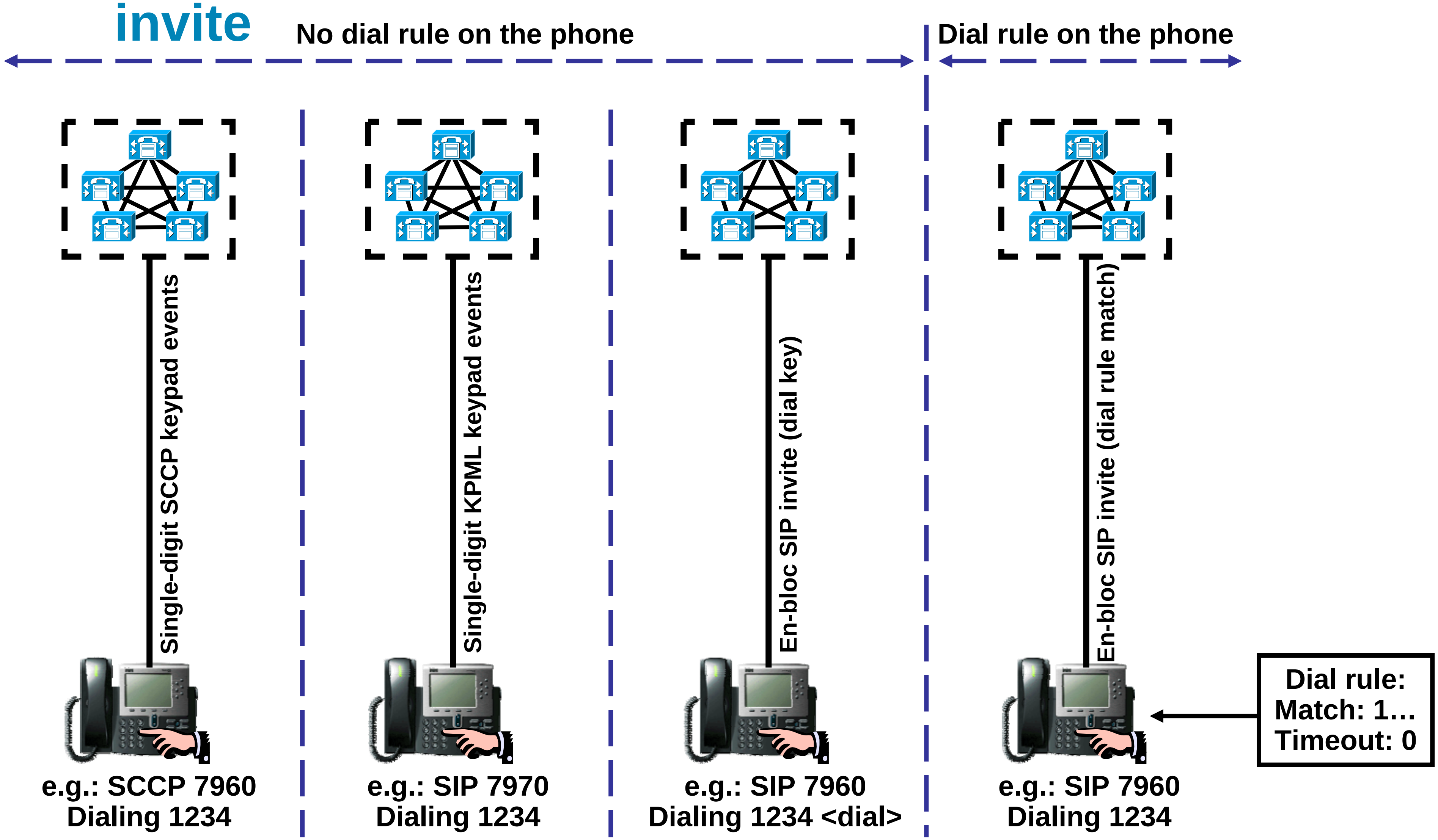
- Looks like a route pattern, allows digit manipulation
- Instead of sending calls outside via a route list, forces second lookup in Cisco CallManager, using a (possibly different) calling search space
- Translation Patterns are “Urgent Priority” by nature: as soon as they match, the inter-digit timer is aborted, and the best match pattern is selected to route the call.

Other Tools
SIP Dial Rules (Cisco Call Manager 5.X)

- SIP phones, used with SIP Dial Rules, can place the function of “pattern recognition” in the phone
- Dial Rules perform “local matching” of dialed digits; sends digit “enbloc” to Call Manager
- Applicable only for SIP Phones
- SIP phones can be configured with, or without SIP dial rules
- Basic patterns: Digits, Period (Any digit), Comma (Secondary Dial tone)

Other Tools

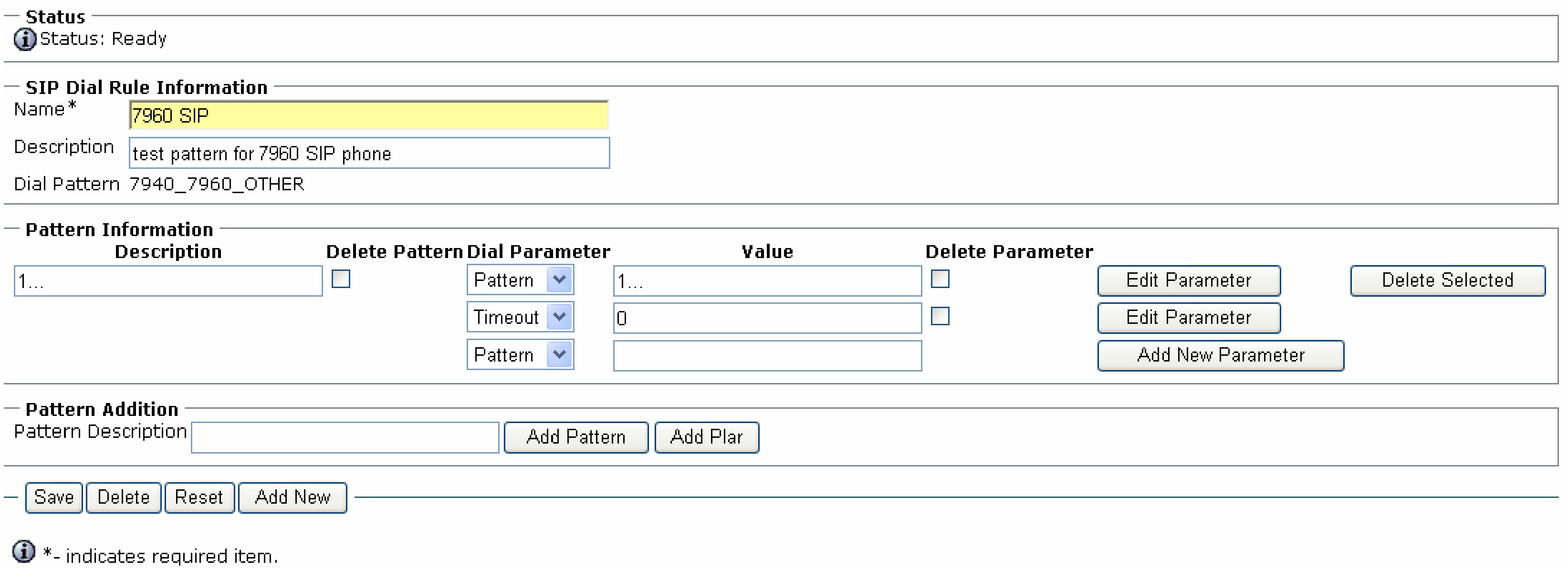
SIP dial rules – pattern recognition triggering SIP invite



Other Tools

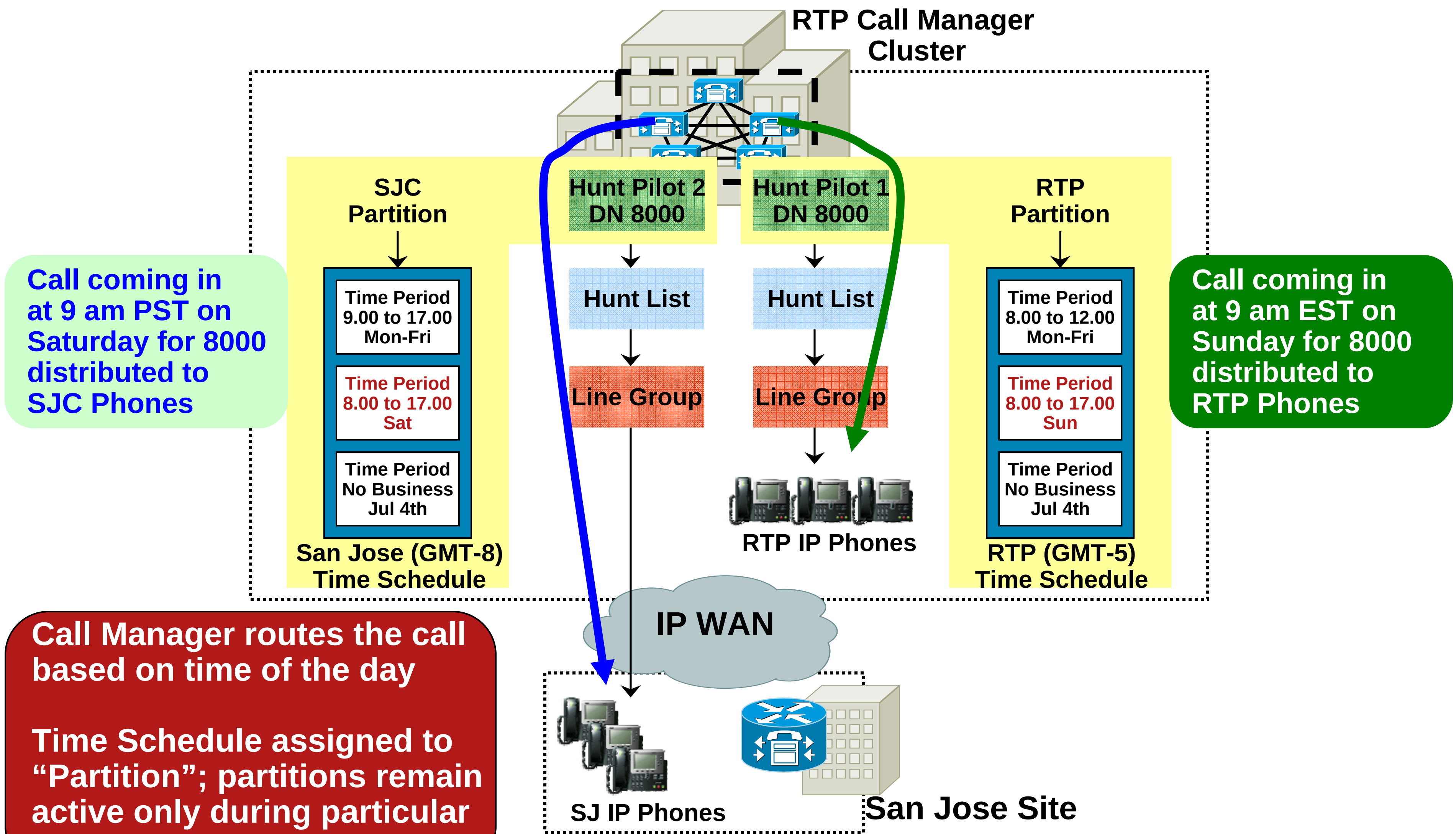
SIP dial rules – pattern recognition triggering SIP invite

Sample Dial rule: “match 1… immediately”



More information in the dial plan chapter of the Cisco Unified Communications SRND Based on Cisco Unified CallManager 5.0. www.cisco.com/go/srnd

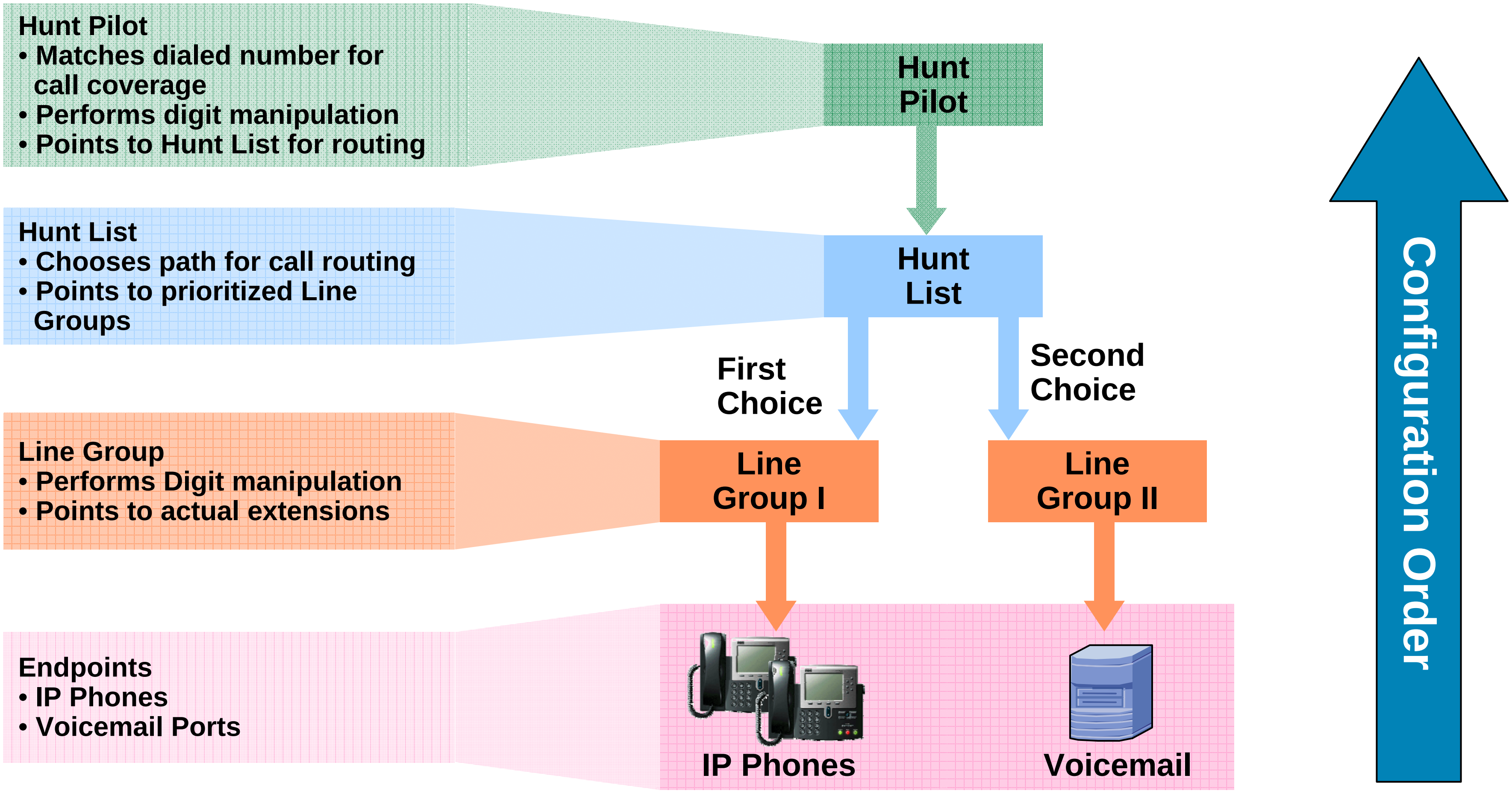
Other Tools
Time of the day Routing



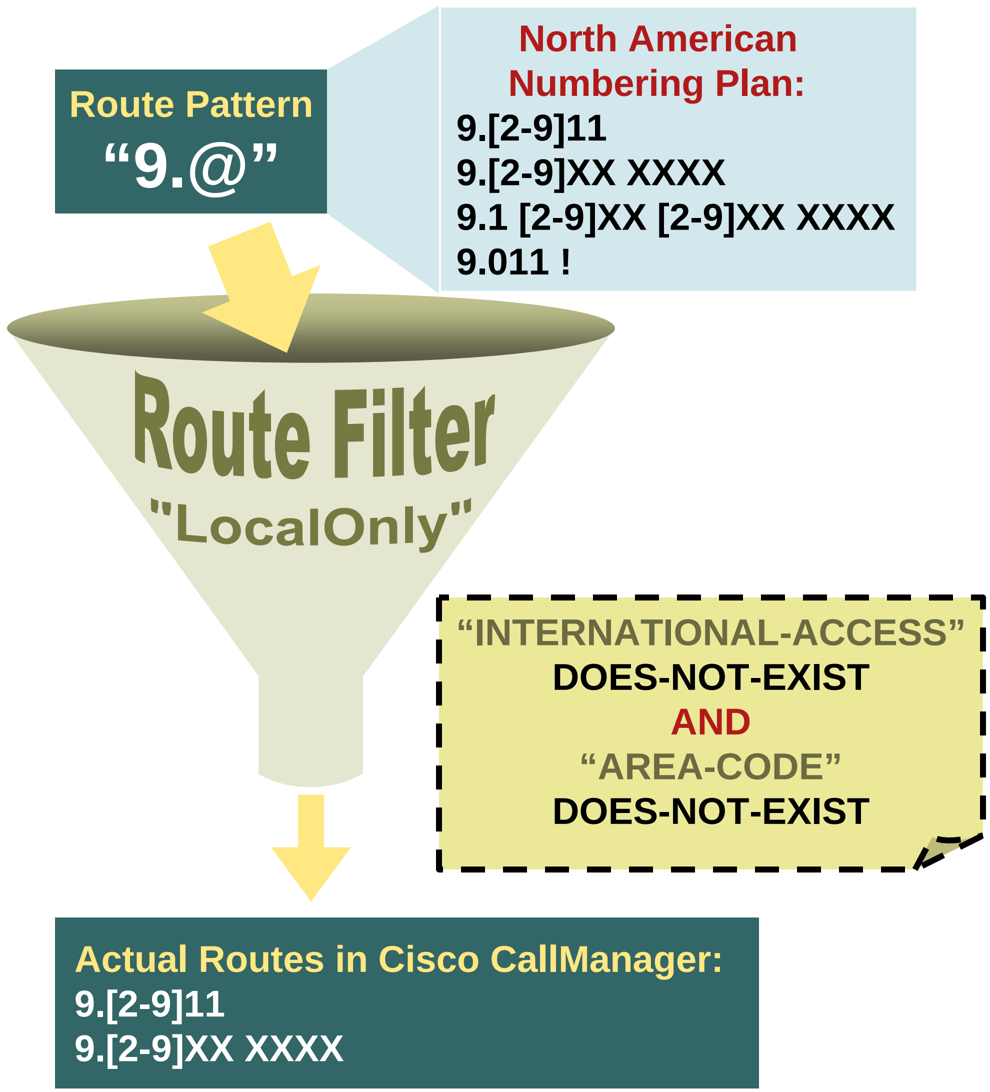
Call Manager routes the call based on time of the day

Time Schedule assigned to “Partition”; partitions remain active only during particular time configured

Other Tools
Hunt Options



Other Tools
Route Filters: The Basics



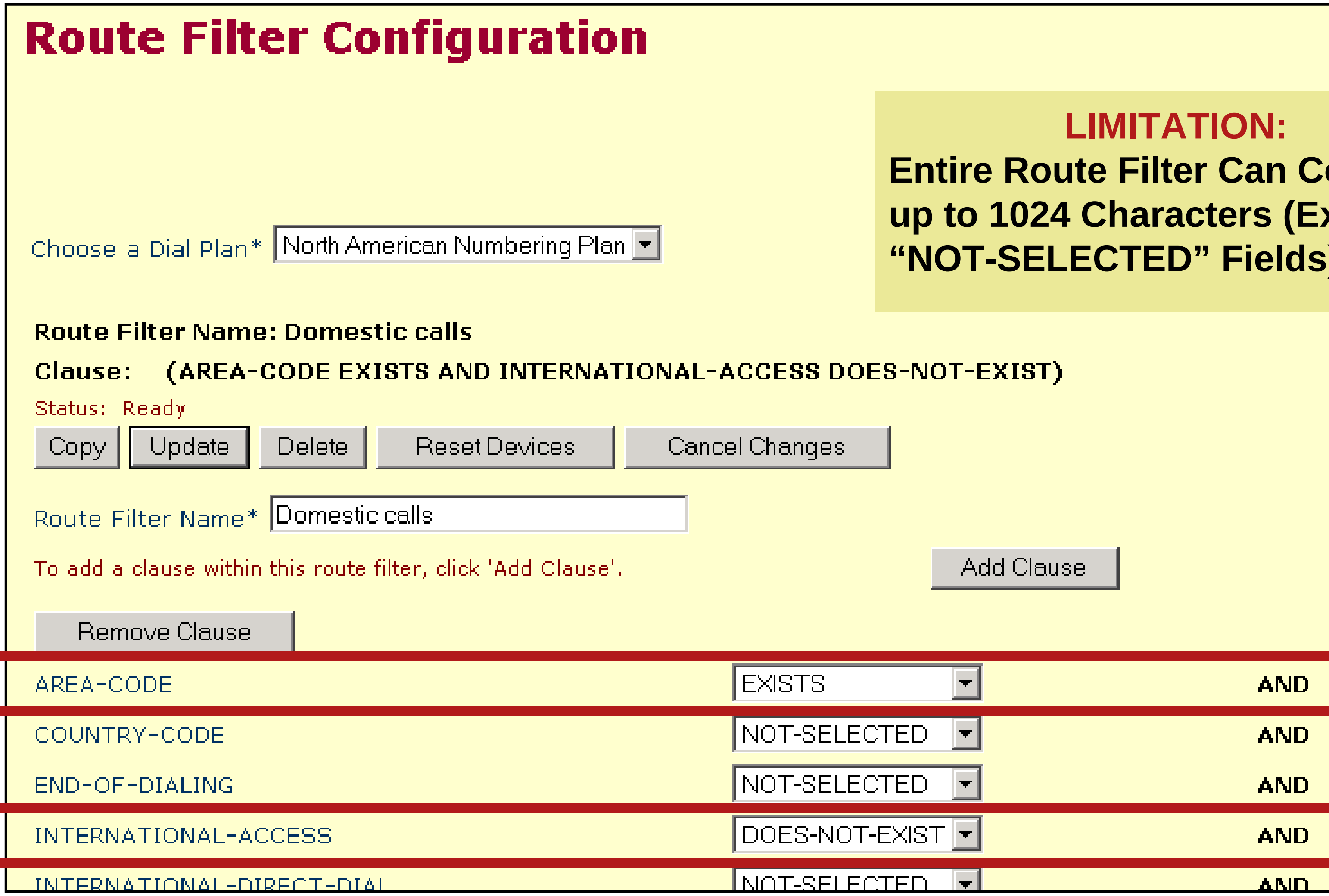
- The “@” wildcard represents all the routes defined in the national numbering plan
- Cisco CallManager identifies tags in each number:

  INTERNATIONAL-ACCESS

  AREA-CODE

  OFFICE-NUMBER ...
- Route filters are logical expressions that operate on these tags
- Useful for blocking 900, pay-per-call, international...

Other Tools
Route Filters: Configuration



LIMITATION:
Entire Route Filter Can Contain up to 1024 Characters (Excludes "NOT-SELECTED" Fields)

Other Tools
DNA and IDP

- Dialed Number Analyzer Tool

  Dial plan troubleshooting tool: simulate calls from specific IP phones/gateways/trunks or from a certain CSS and observe routing behavior

  (Ships as a plugin with CCM 3.3(4), 4.0(1) and later)

- International Dial Plan downloads

  Allows to create country-specific numbering plans and import them into CCM to enable use of the "@" macro

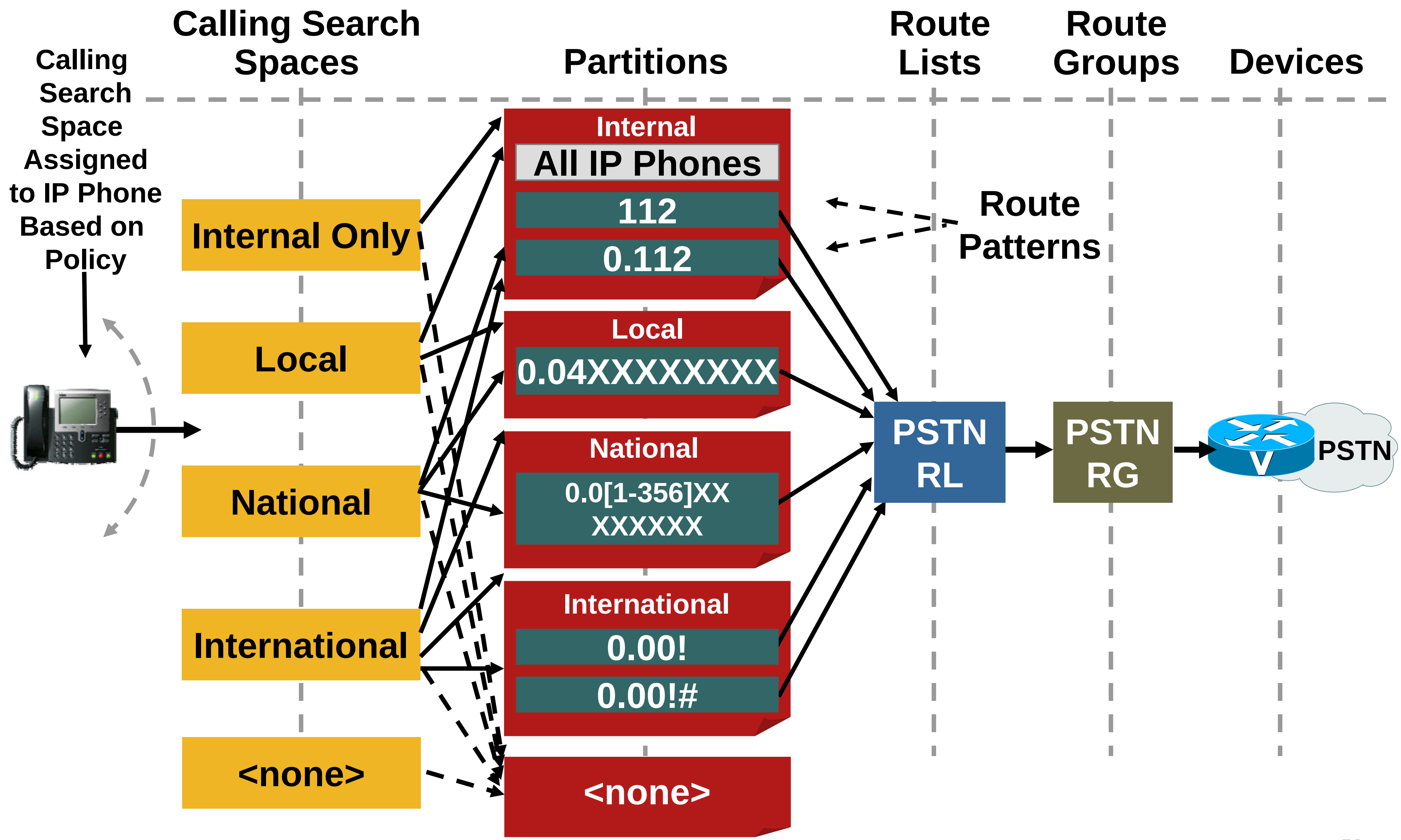
  http://www.cisco.com/cgi-bin/tablebuild.pl/IDP

Design Guidelines

Design Best Practices Agenda

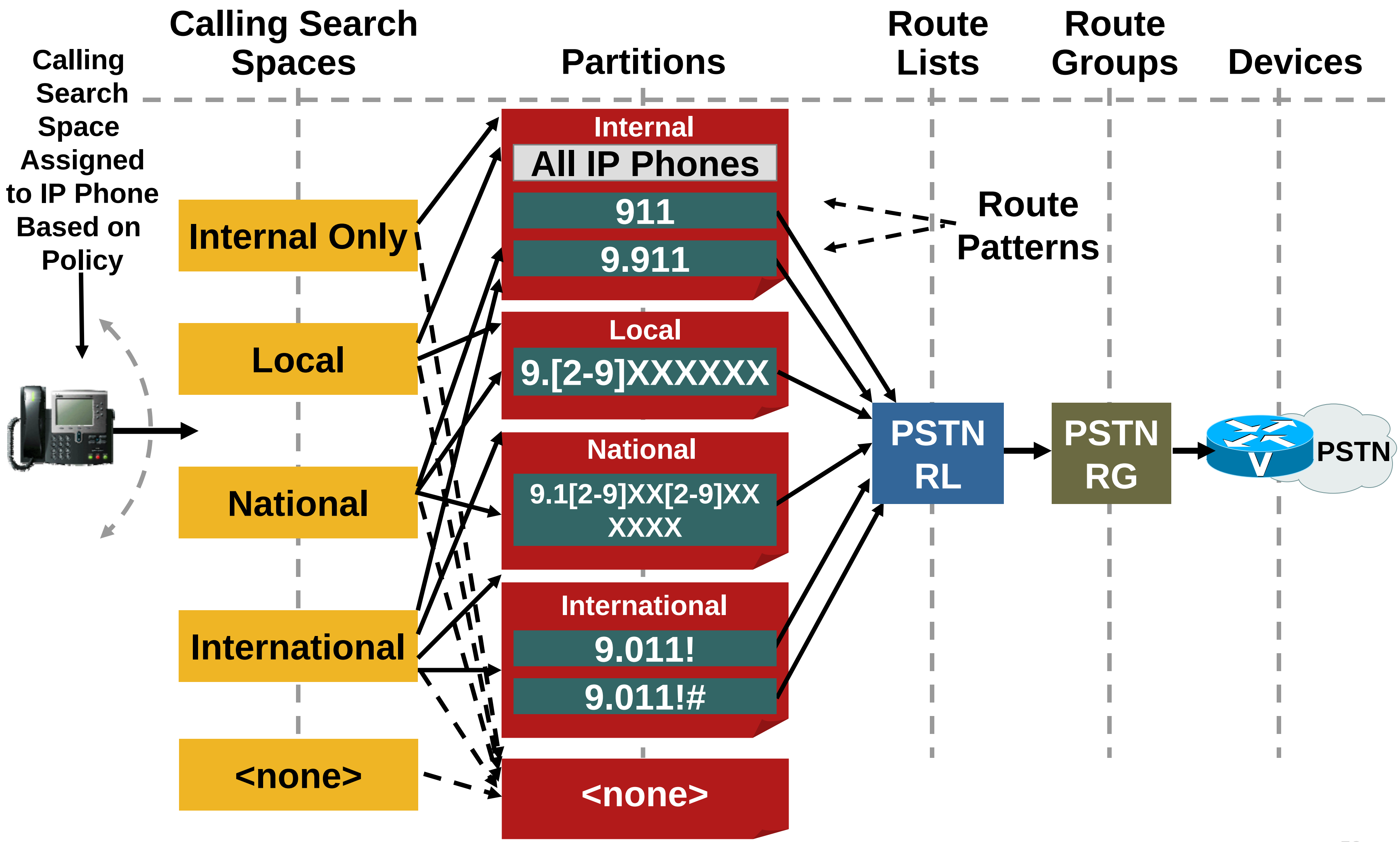
- Building Classes of Service
  - Traditional CSS Approach
  - Line/Device CSS Approach
- Multisite Deployments
- Mobility Considerations

Traditional CSS Approach
Example of Composite View - France



Traditional CSS Approach

Example of Composite View – North America

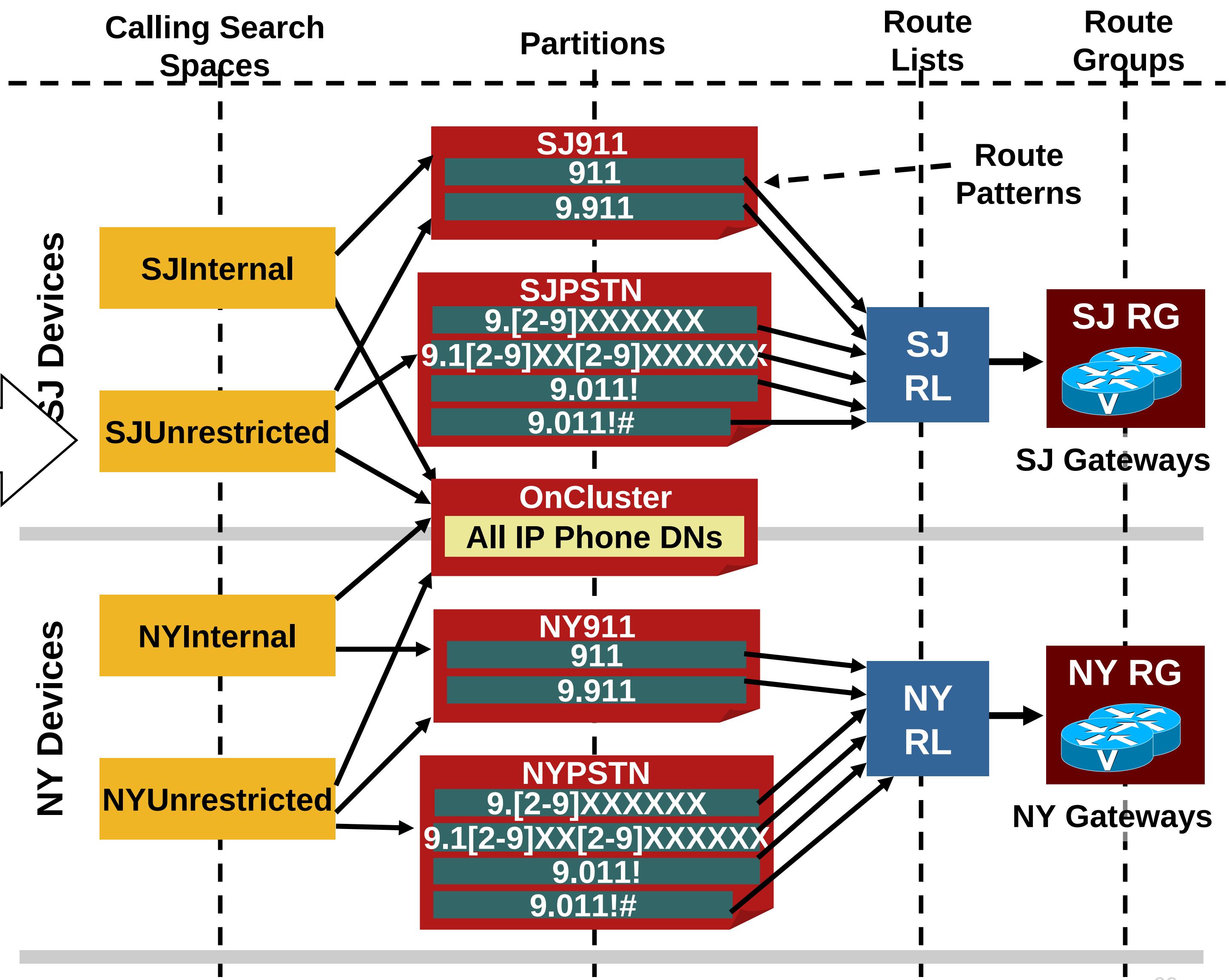


Traditional CSS Approach
Scalability for Centralized Deployments

Device CSS Dictates:
- Class of Service
- Path Selection

 # CSS = N x C

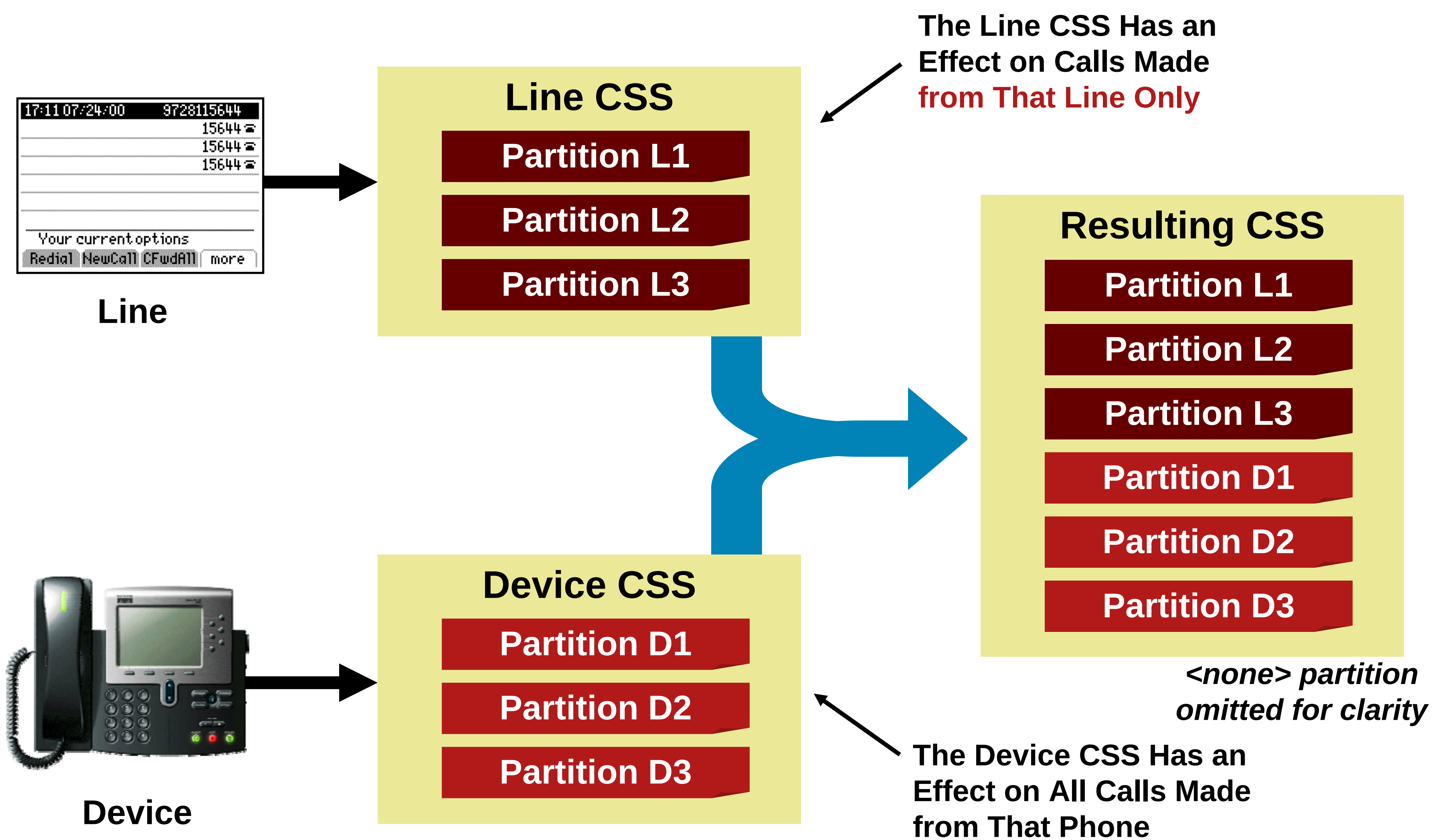
N = # of Sites
C = # of Classes of Service



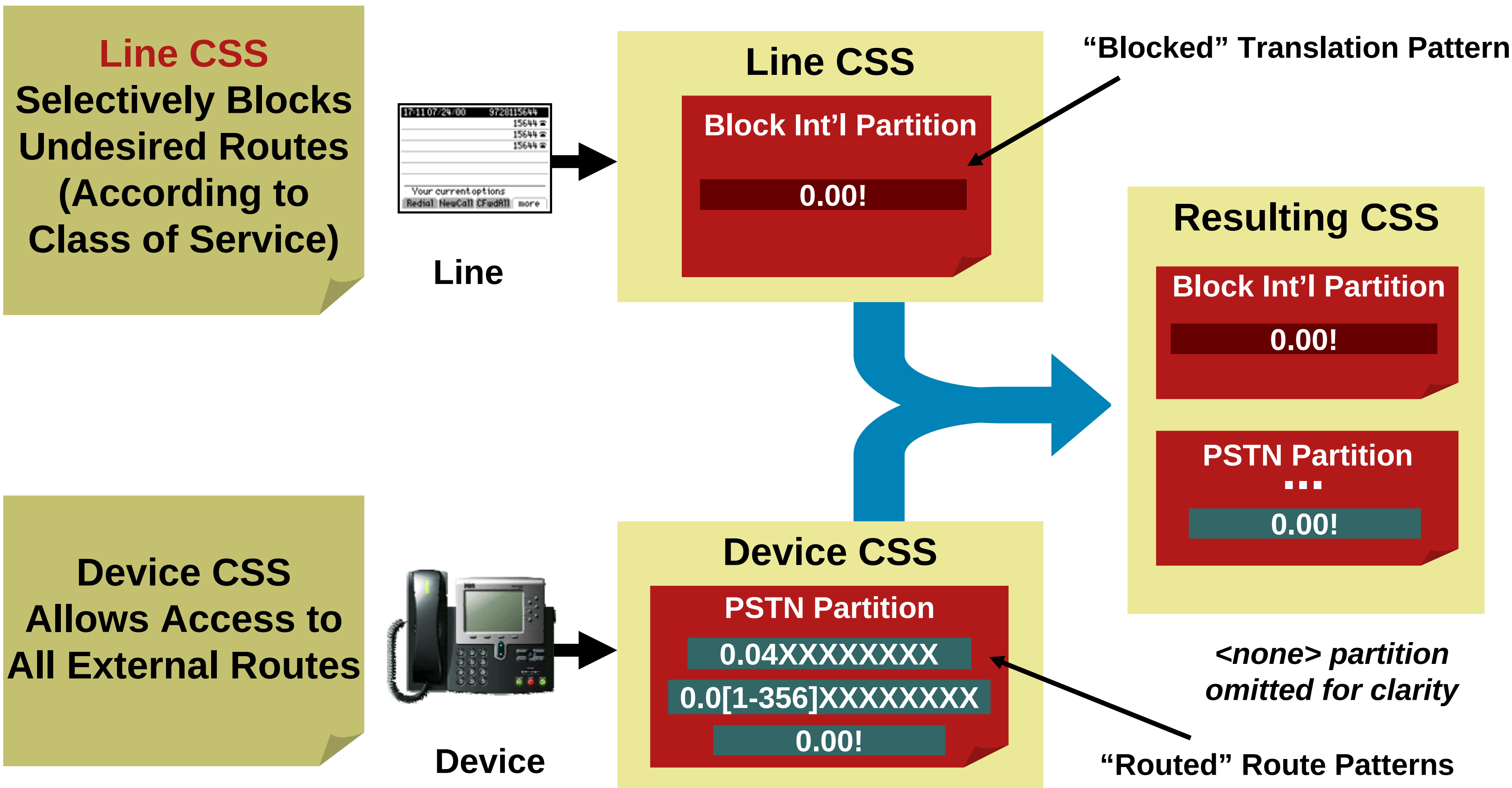
Design Best Practices Agenda

- 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



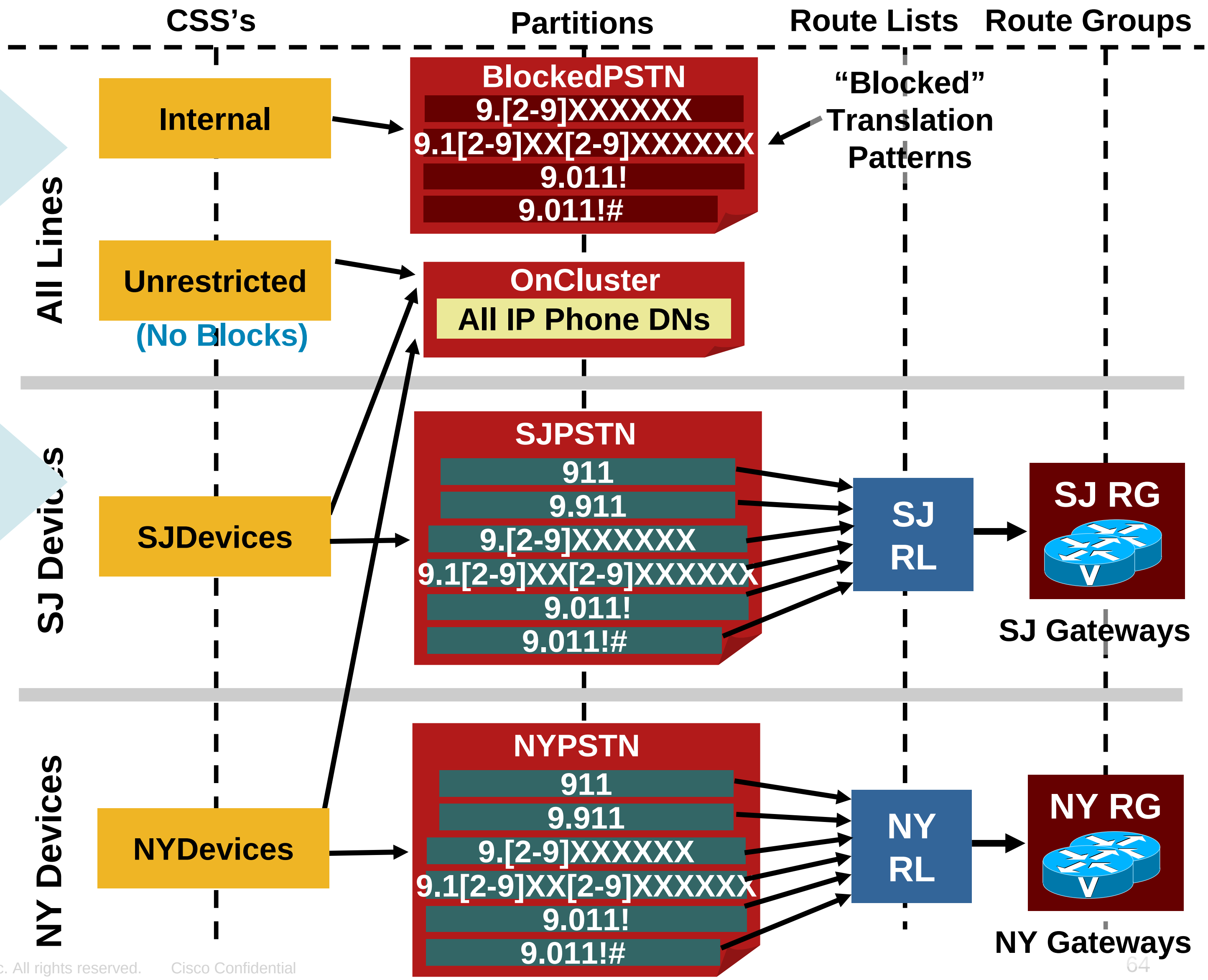
The Line/Device CSS Approach
Scalability for Centralized Deployments

Line CSS Dictates:
- Class of Service

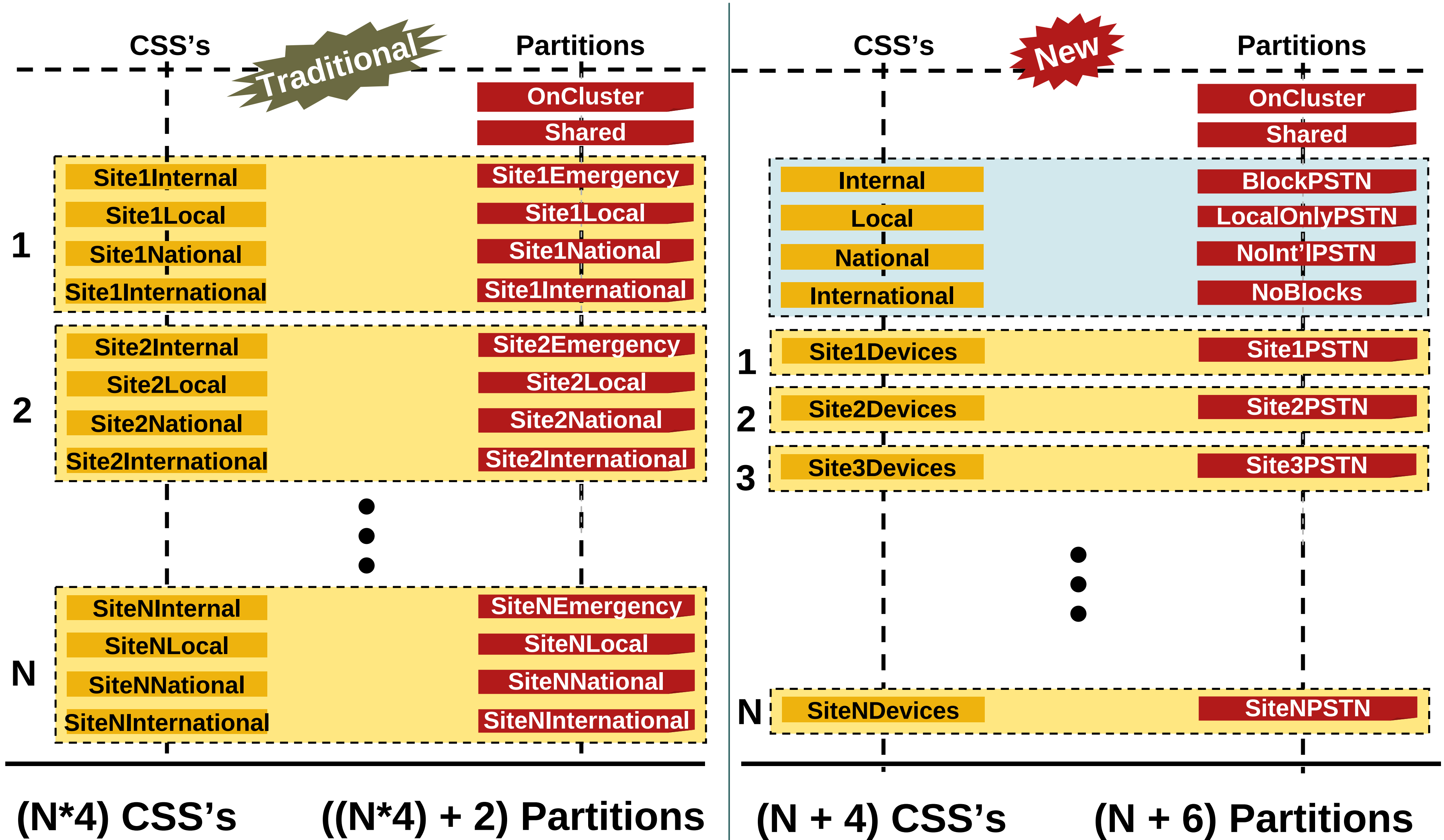
Device CSS Dictates:
- Path Selection

 # CSS = N + C

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



The Line/Device CSS Approach
Comparison of the Two Methods



The Line/Device CSS Approach
CallForward Caveats

- 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
- If forwarded calls must have some intermediate restriction (e.g., no international calls), this approach may loose efficiency, as additional site-specific CSS's will be needed

New

- In CUCM 5.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
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

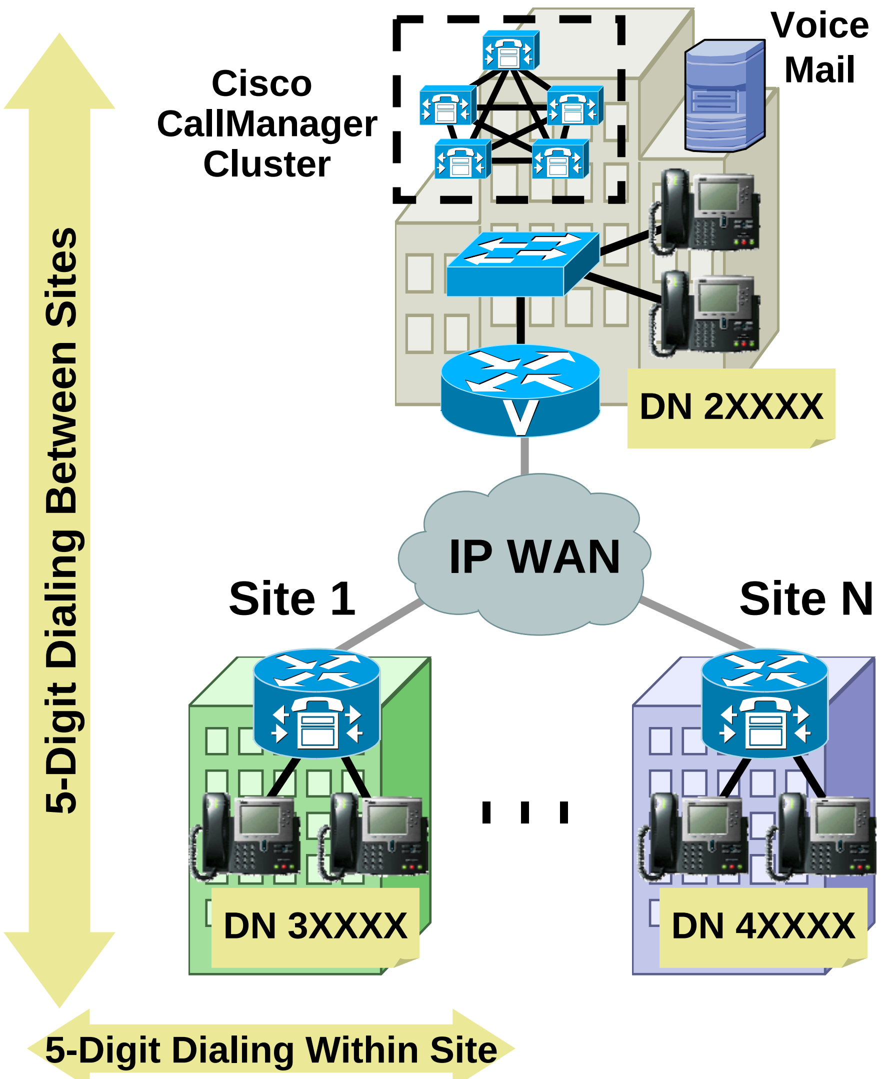
- Priority order between line and device is reversed for CTI route points and CTI ports; therefore, the Line/Device CSS approach cannot be *directly* applied to CTI devices, such as Softphone (not Communicator)

  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

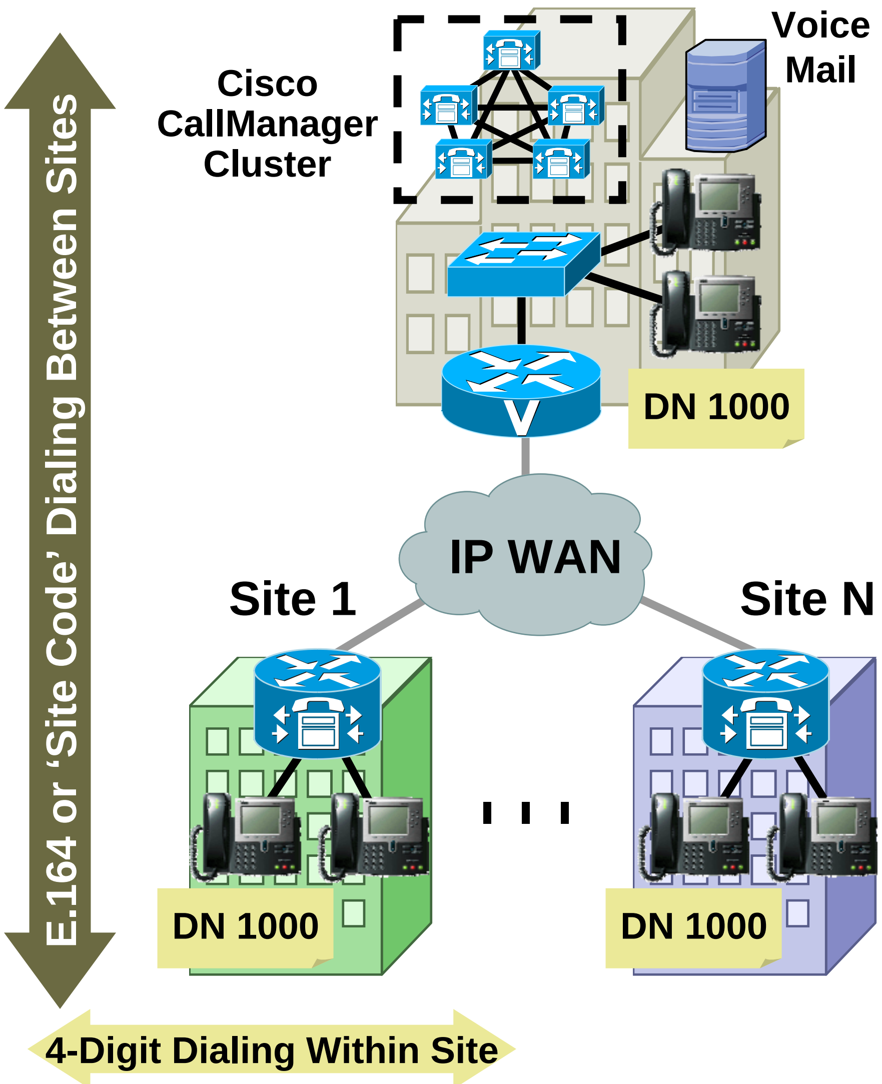
- Building Classes of Service
- Multisite Deployments
  - Choosing a Dial Plan Approach
  - Uniform On-Net Dialing
  - Variable-Length On-Net Dialing with Partitioned Addressing
  - Variable-Length On-Net Dialing with Flat Addressing
- Mobility Considerations

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)

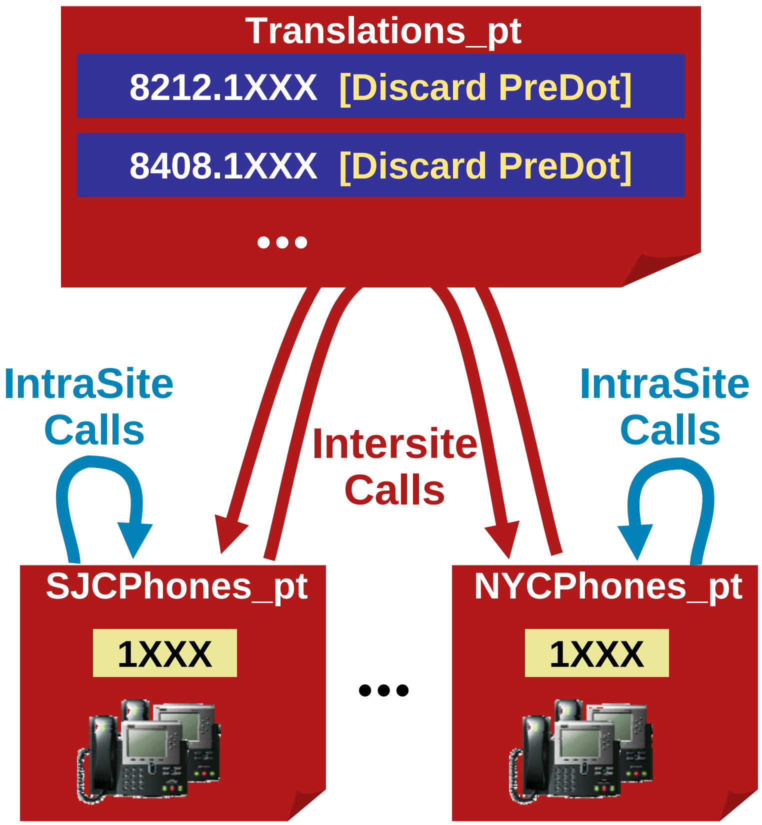
Choosing a Dial Plan Approach
Variable-Length On-Net Dialing (VLOD)



- 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

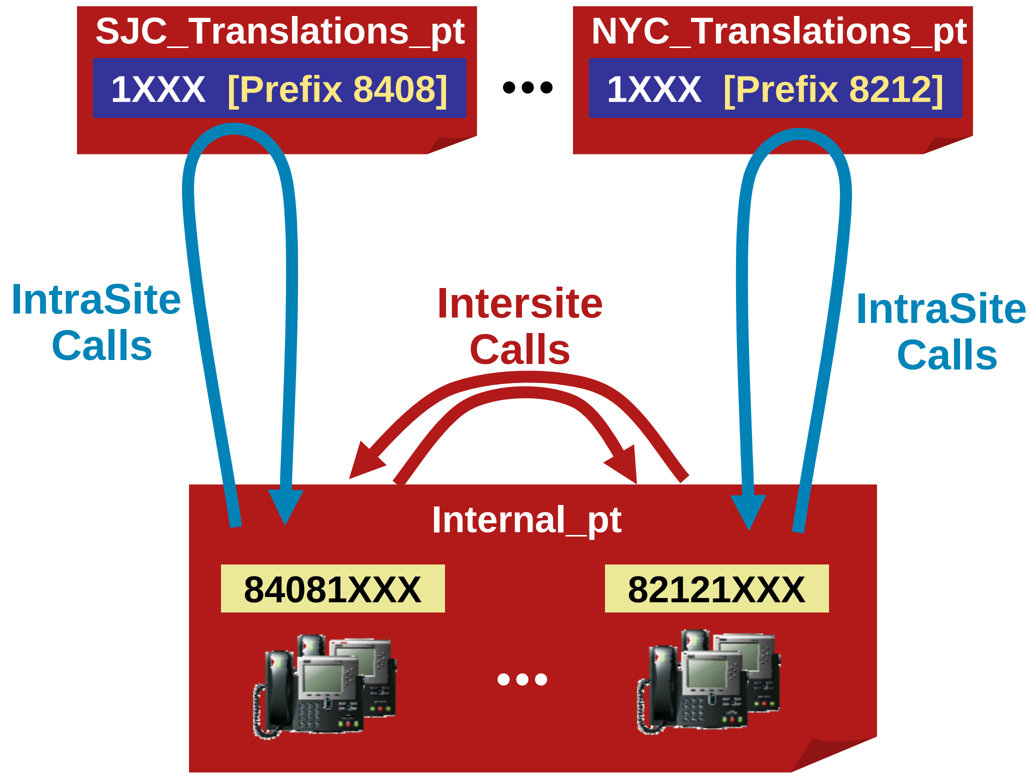
Choosing a Dial Plan Approach
Addressing Methods for VLOD

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)?

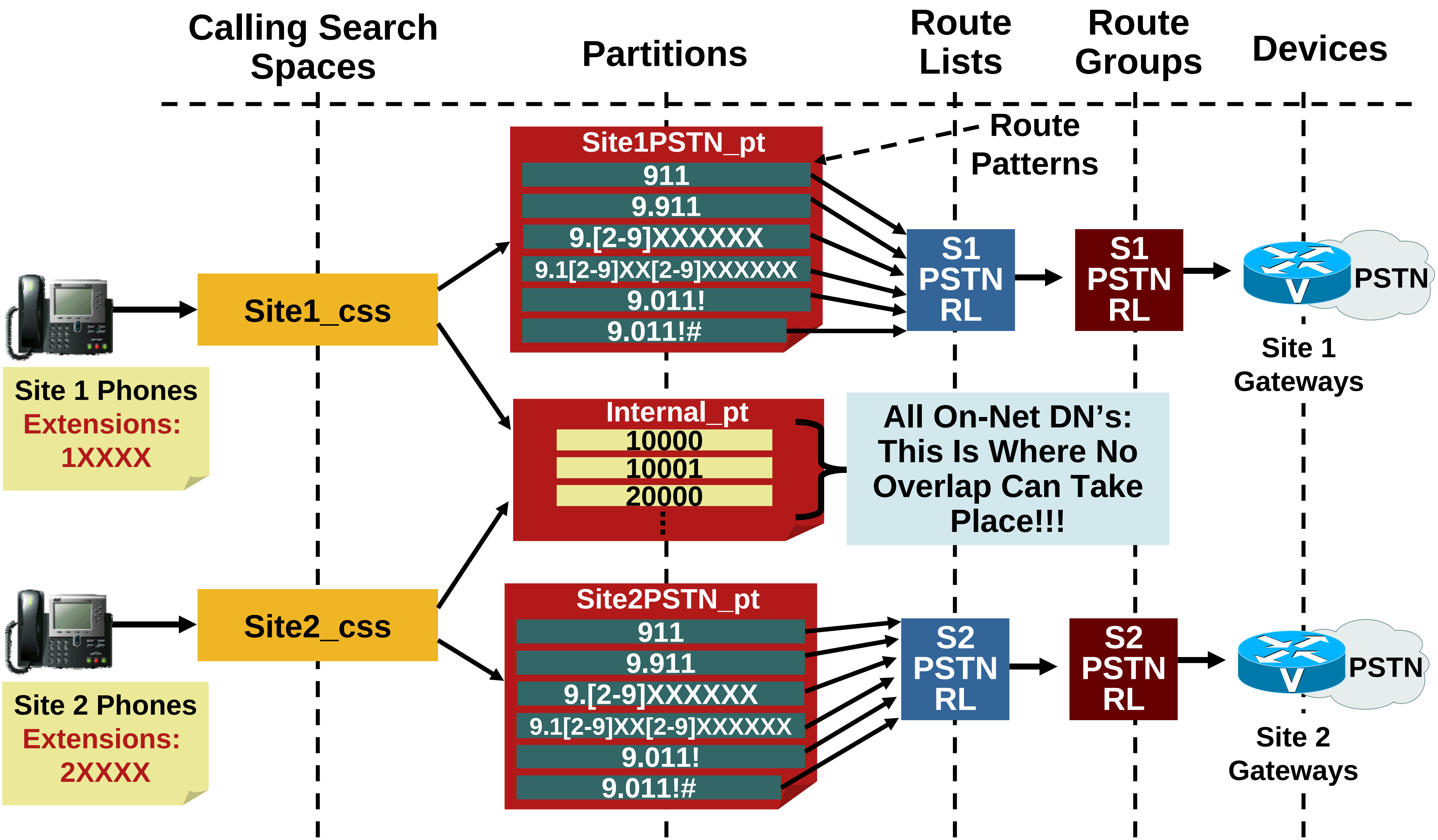
Design Best Practices Agenda

- Building Classes of Service
- MultiSite Deployments
  - Choosing a Dial Plan Approach
  - Uniform On-Net Dialing
  - Variable-Length On-Net Dialing with Partitioned Addressing
- 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 relatively small
- Number of sites is not expected to grow significantly in the future

Uniform On-Net Dialing
Composite View



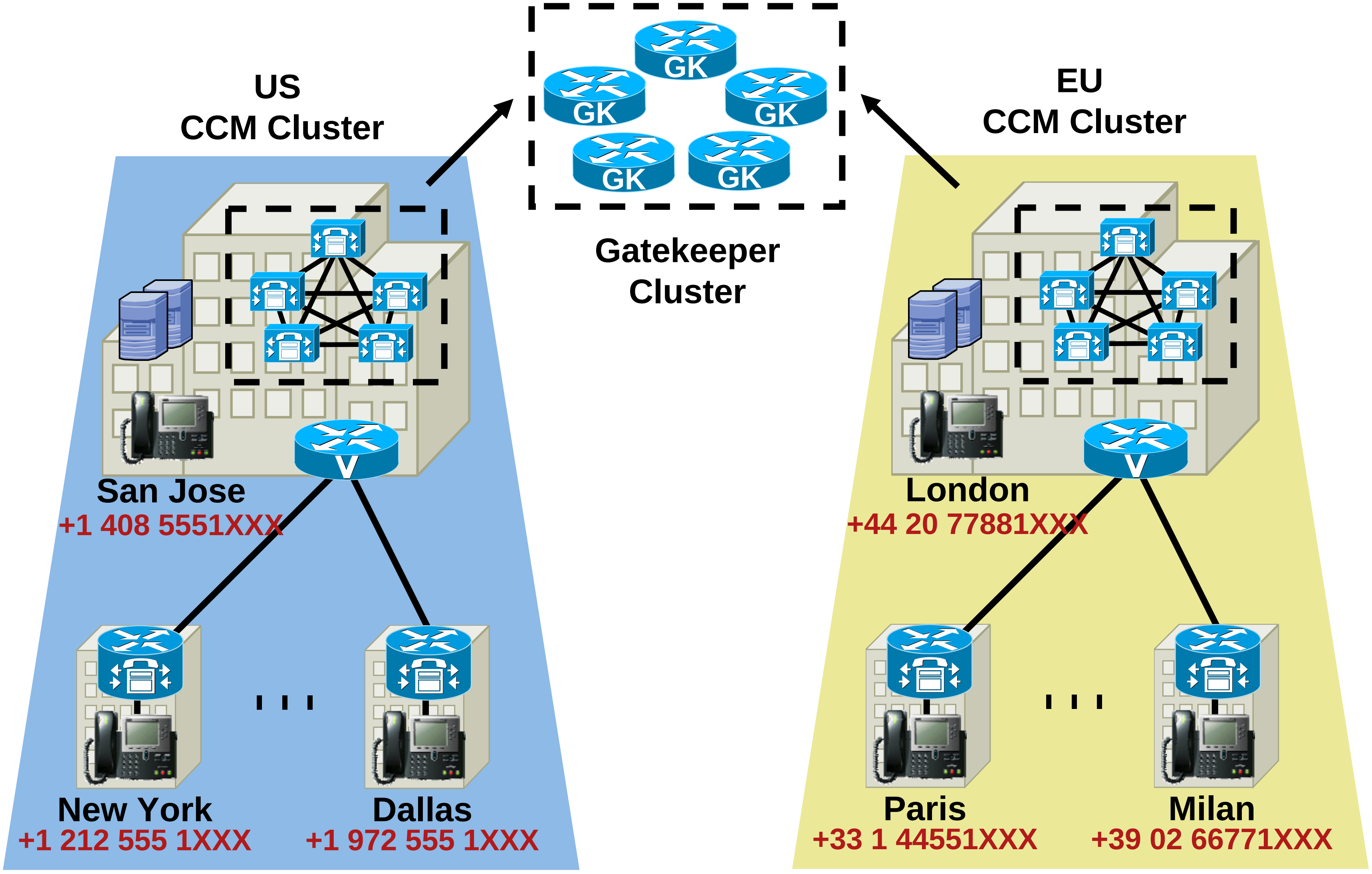
Design Best Practices Agenda

- 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
- Mobility Considerations

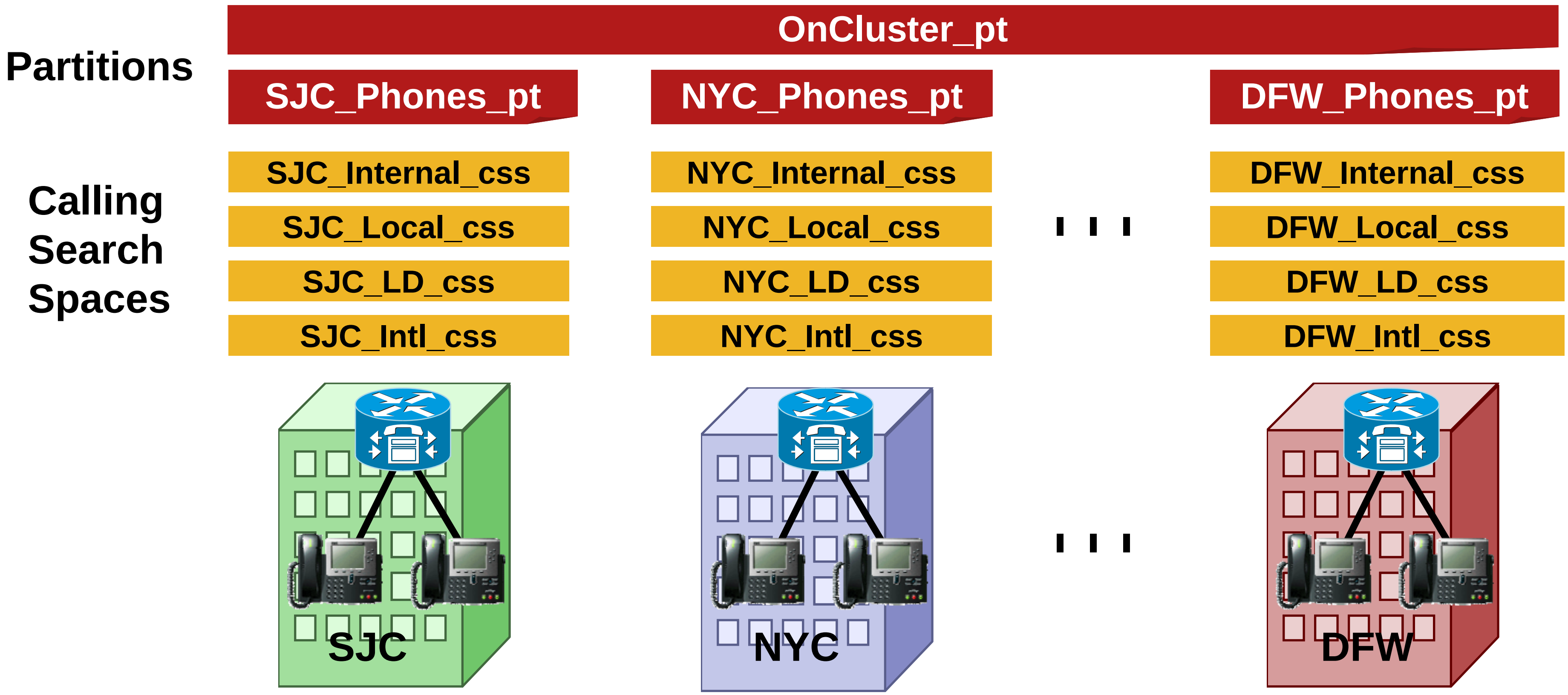
VLOD with Partitioned Addressing
Use this Model if...

- 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

VLOD with Partitioned Addressing
Hypothetical Customer Example

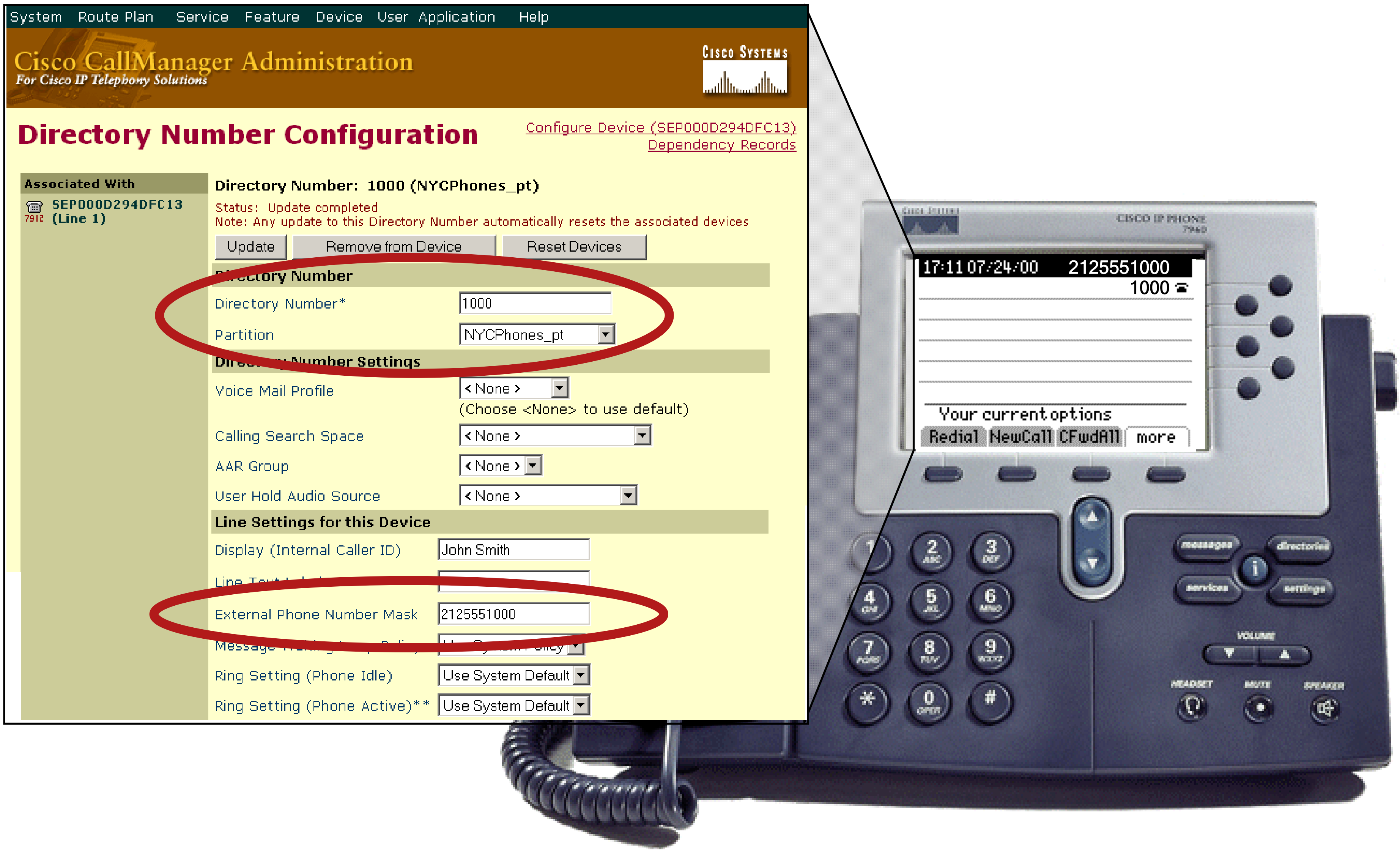


VLOD with Partitioned Addressing
Partitions and Calling Search Spaces

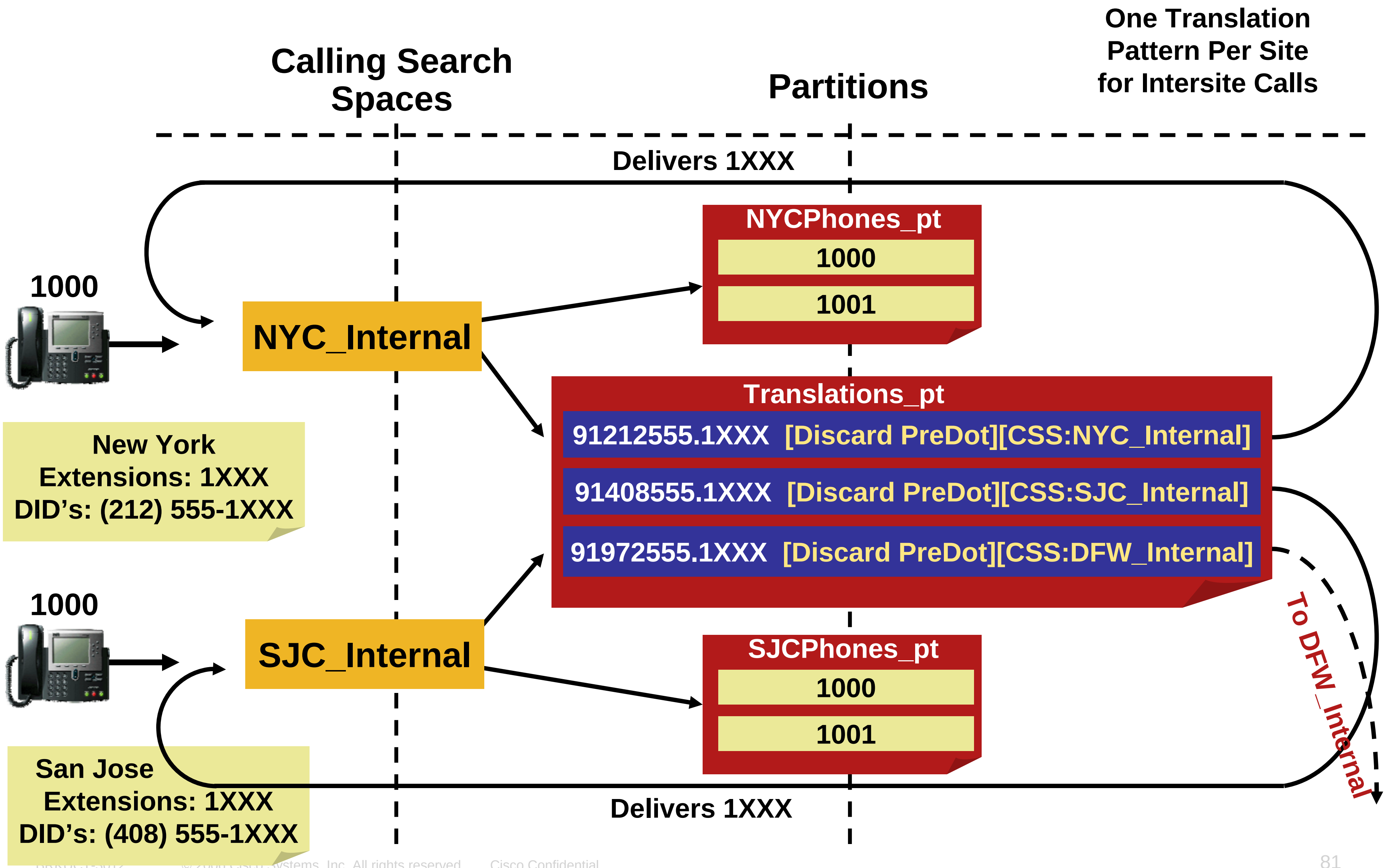


* **Note: If Using the Line/Device CSS Approach, the Number of CSS's Can Be Reduced**

VLOD with Partitioned Addressing
Line Configuration



VLOD with Partitioned Addressing
Intersite Calls Within a Cluster



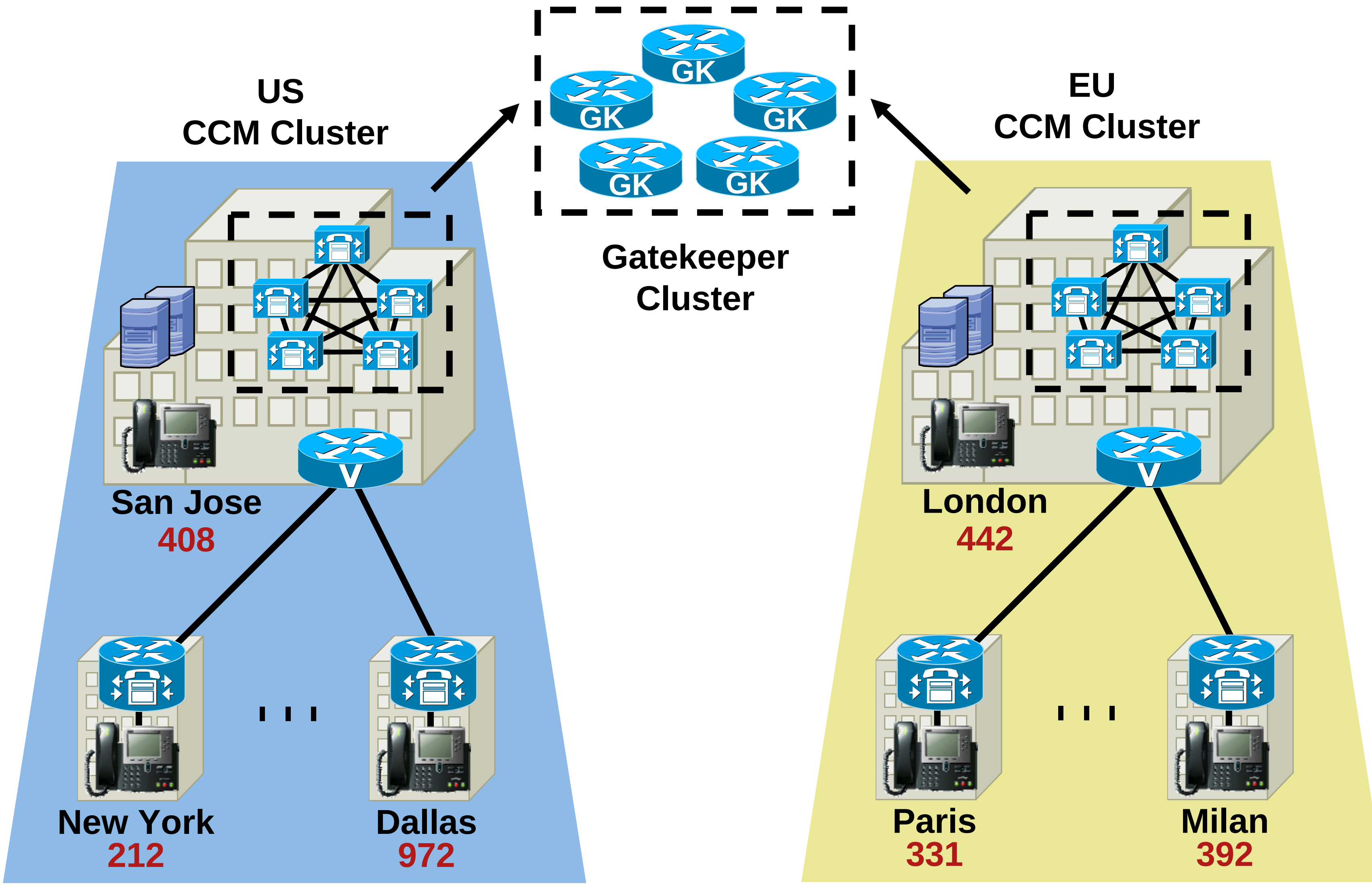
Design Best Practices Agenda

- 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
- Mobility Considerations

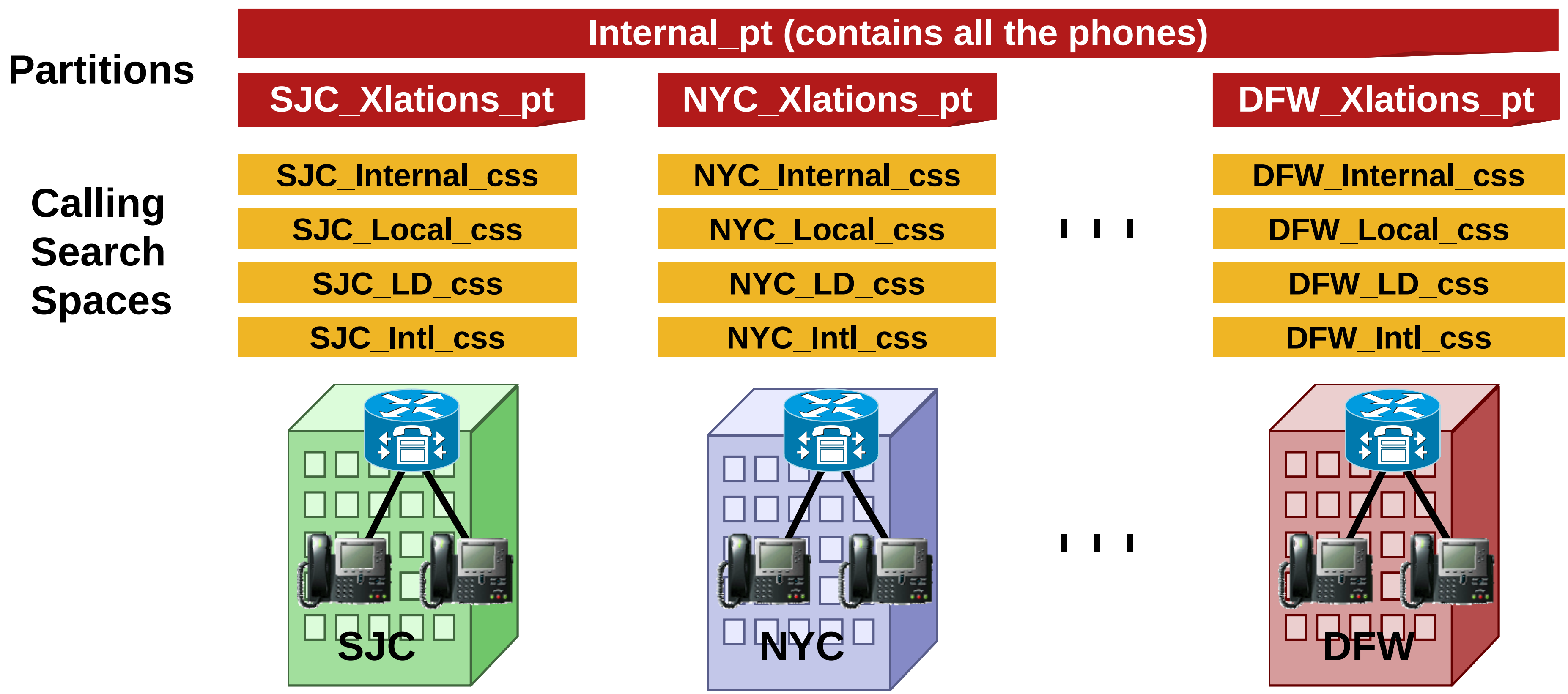
VLOD with Flat Addressing
Use this Model if...

- 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

VLOD with Flat Addressing
Site Code Assignment

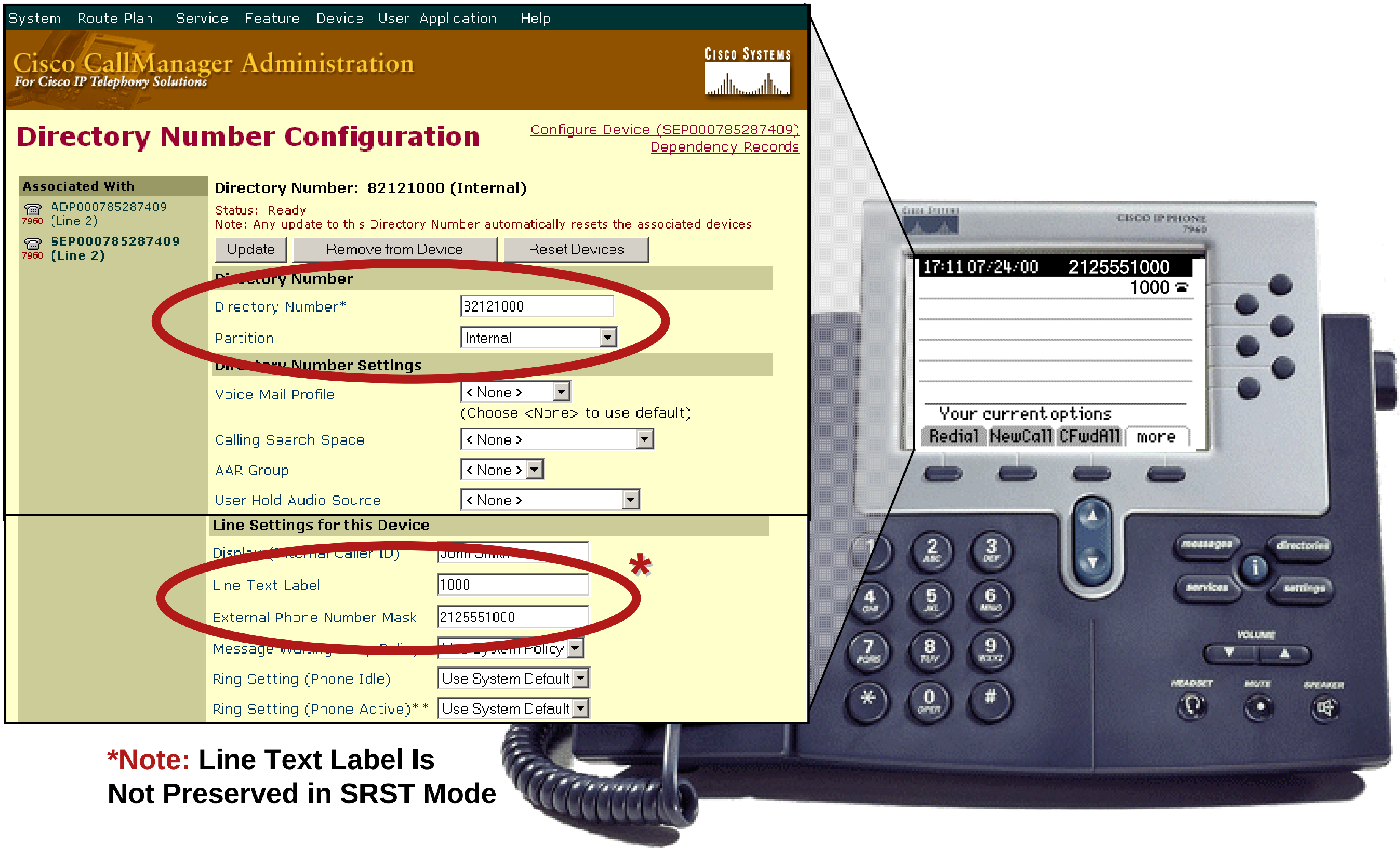


VLOD with Flat Addressing
Partitions and Calling Search Spaces



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

VLOD with Flat Addressing
Line Configuration

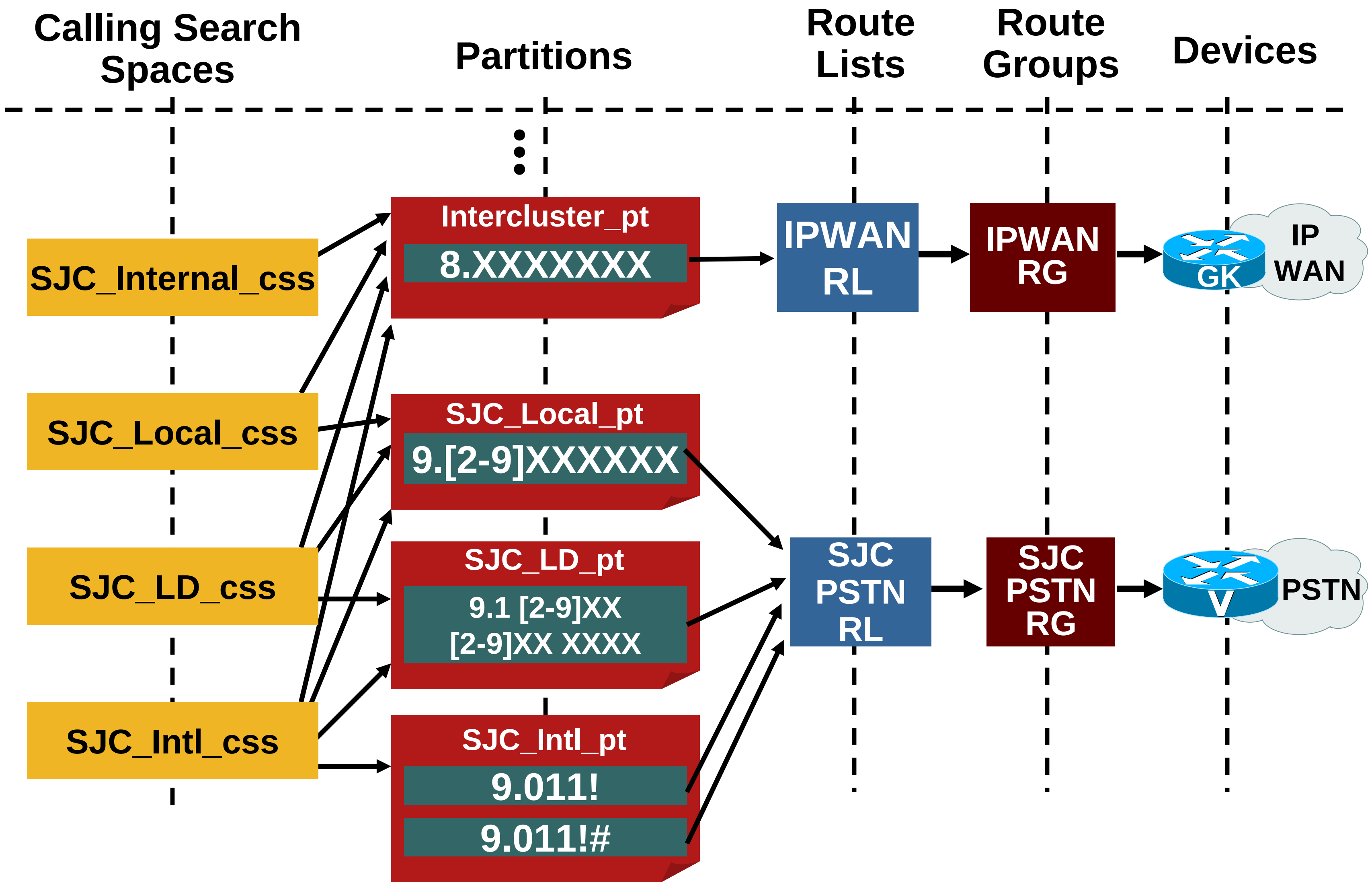


***Note:** **Line Text Label Is Not Preserved in SRST Mode**

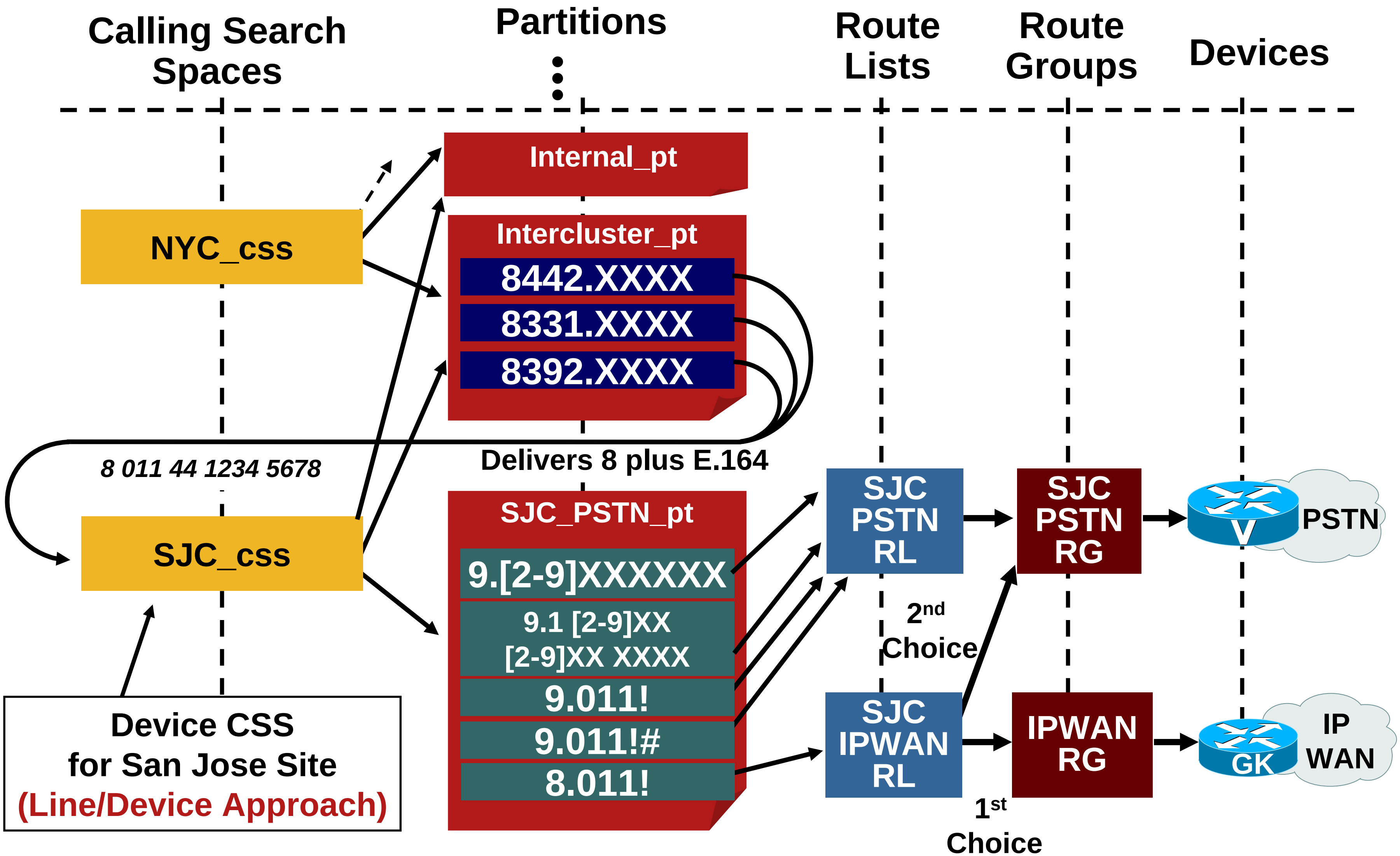
VLOD with Flat Addressing
Outgoing inter-cluster WAN/PSTN Calls

- 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

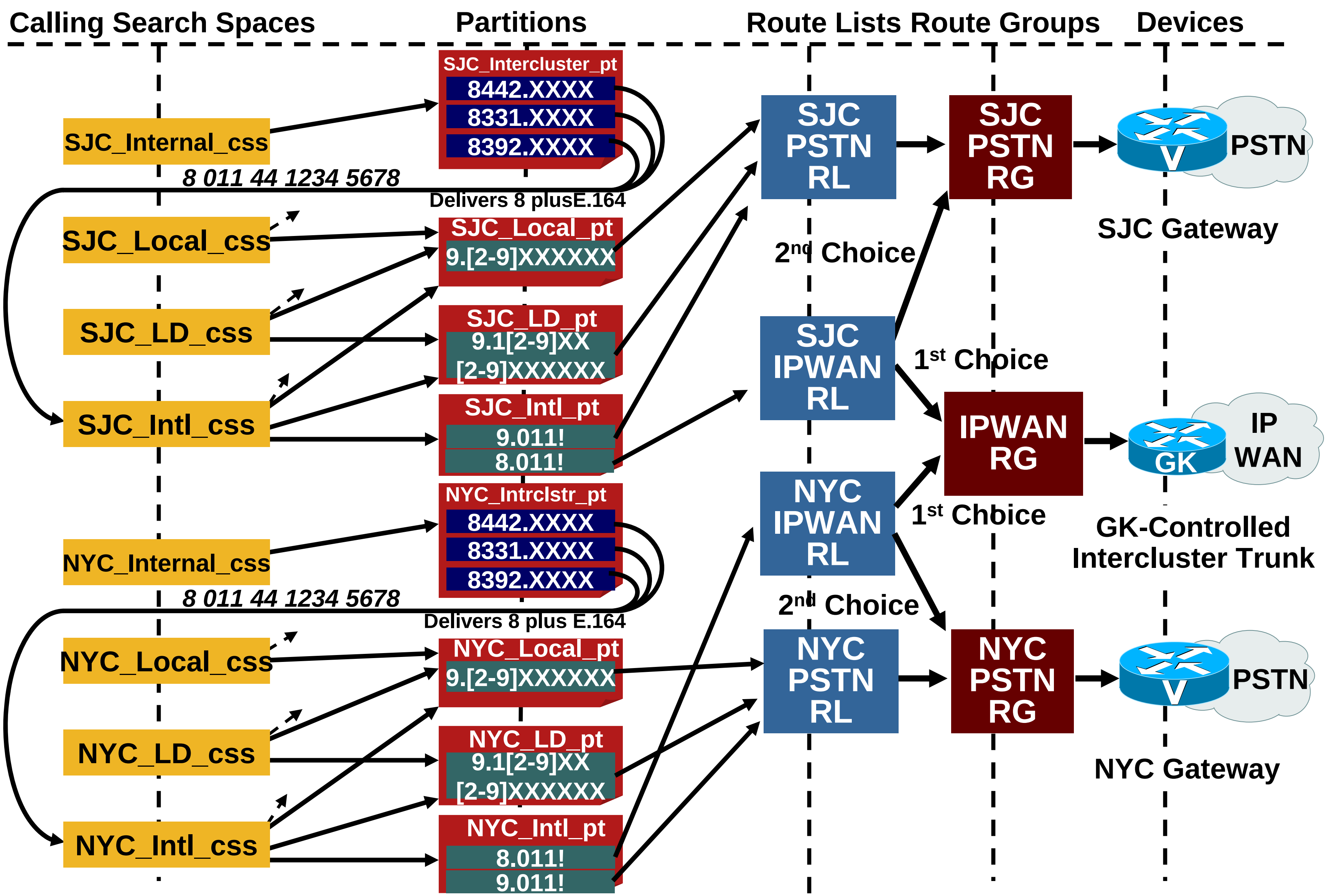
VLOD with Flat Addressing
Outgoing PSTN/IP WAN Calls: Option 1



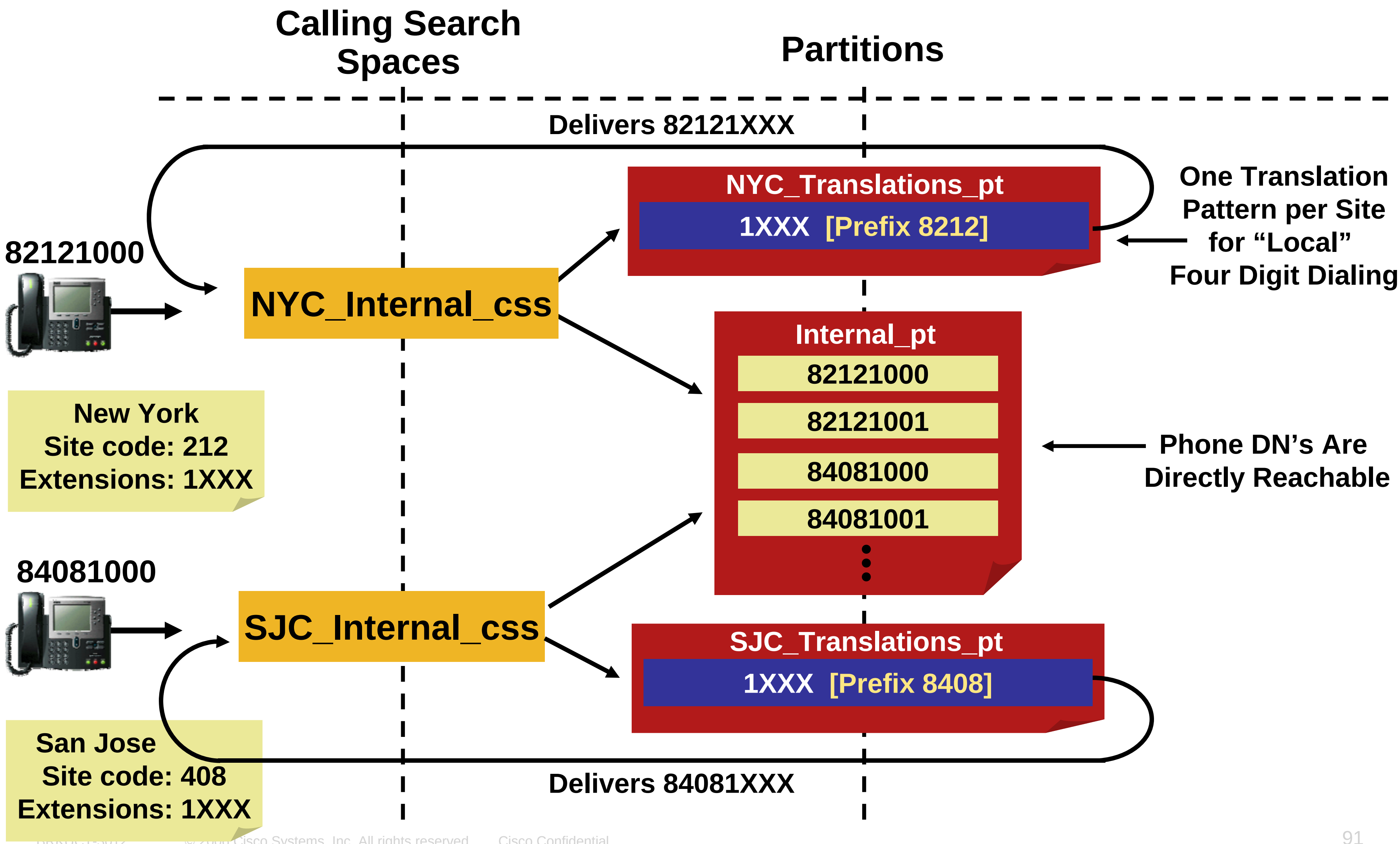
VLOD with Flat Addressing
Outgoing PSTN/IP WAN Calls: Option 2



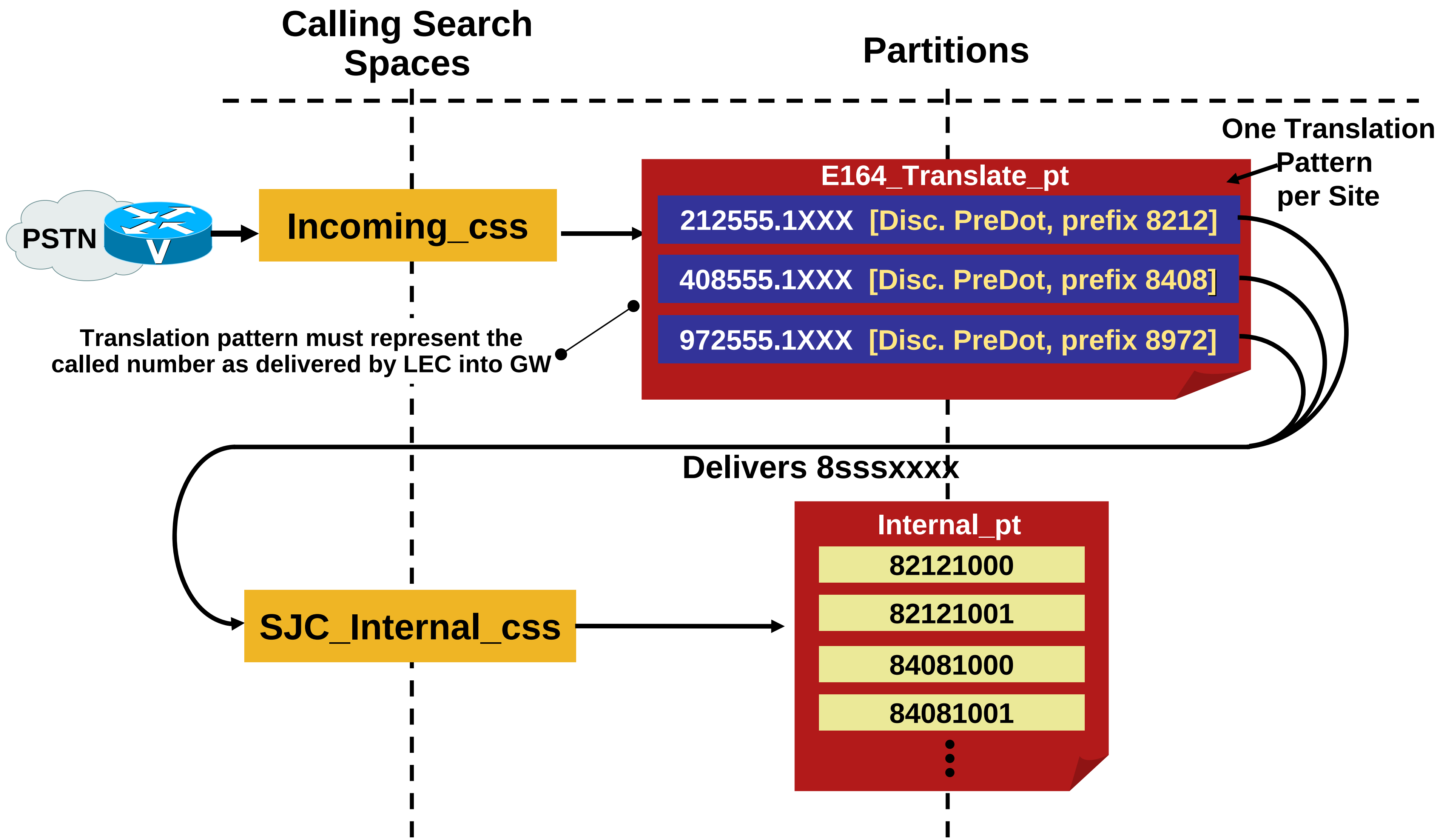
VLOD with Flat Addressing
Outgoing PSTN/IP WAN Calls: Option 3



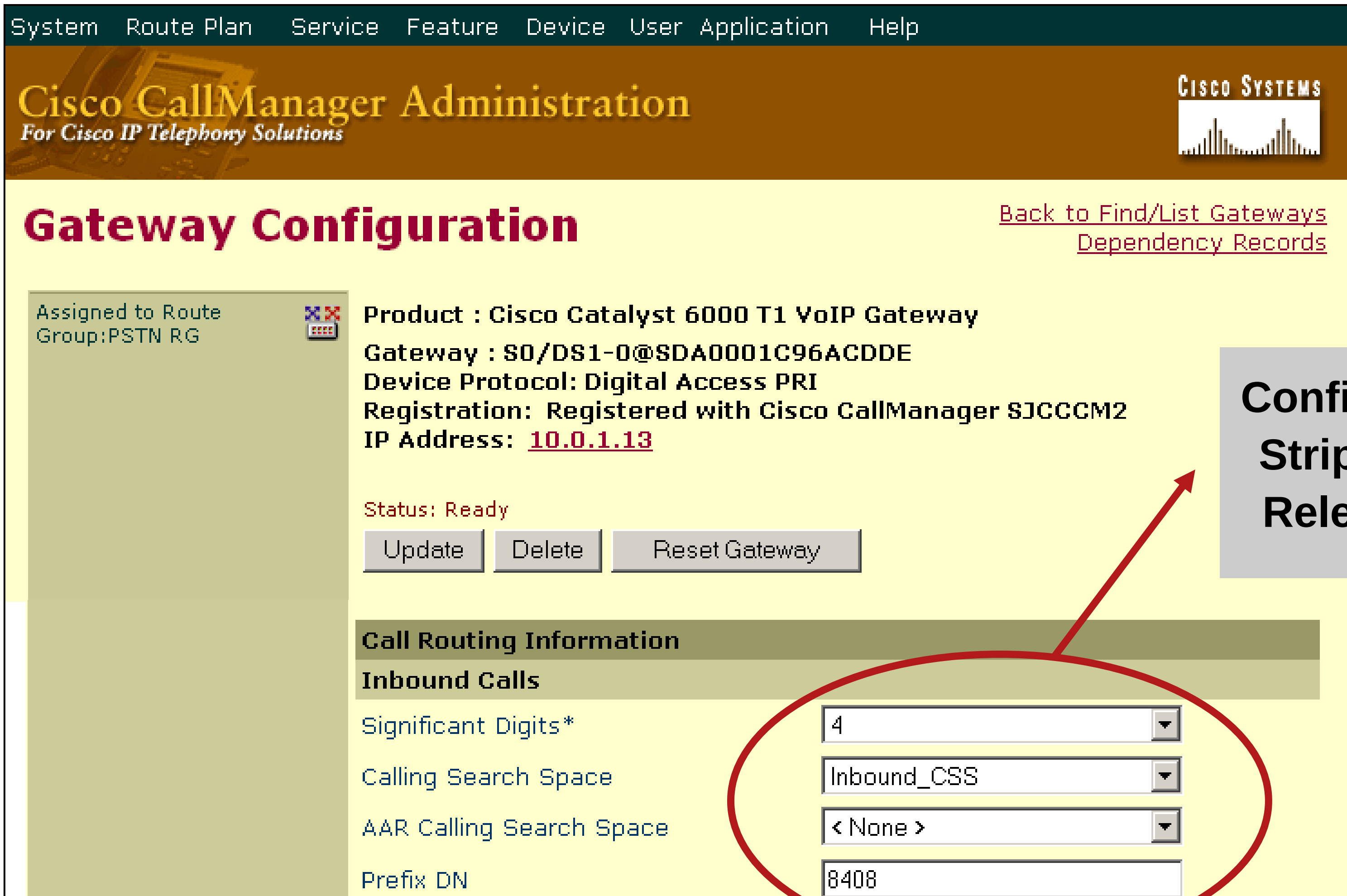
VLOD with Flat Addressing
Intra/Inter-site Calls Within a Cluster



VLOD with Flat Addressing
Incoming PSTN/IP WAN Calls



VLOD with Flat Addressing
Incoming PSTN/ IP WAN Calls (Alternative)



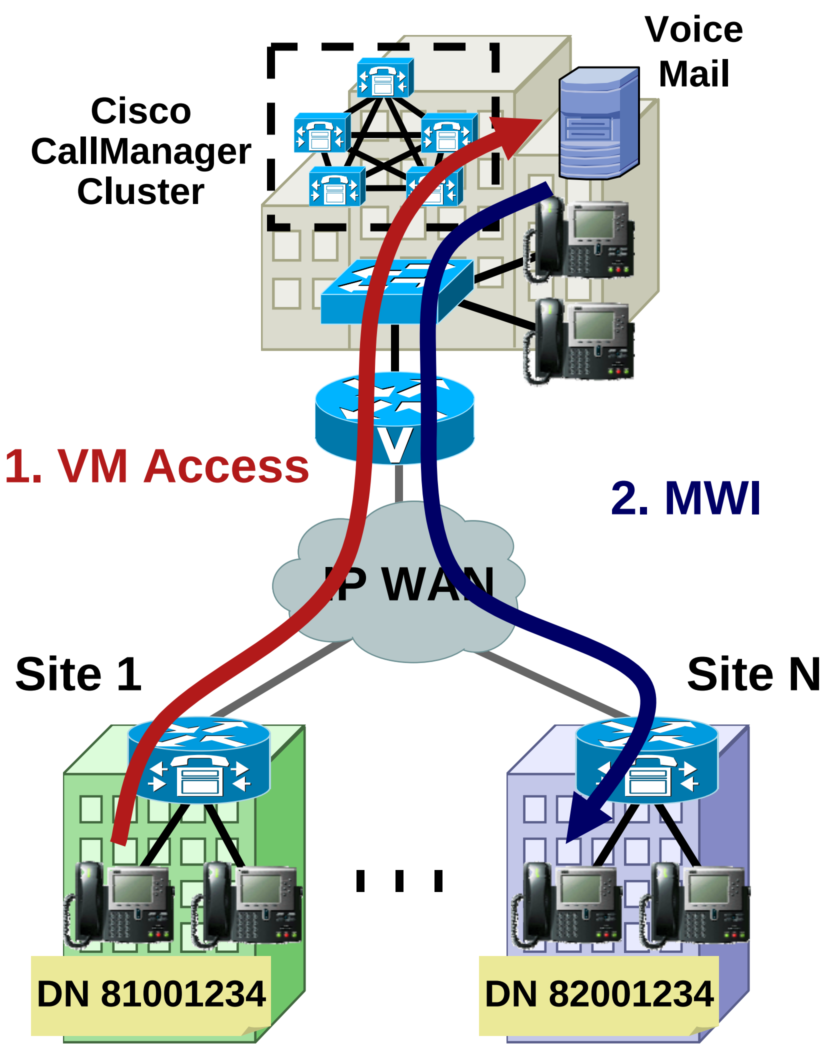
Configure GW to Strip and Prefix Relevant Digits

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...
!
```

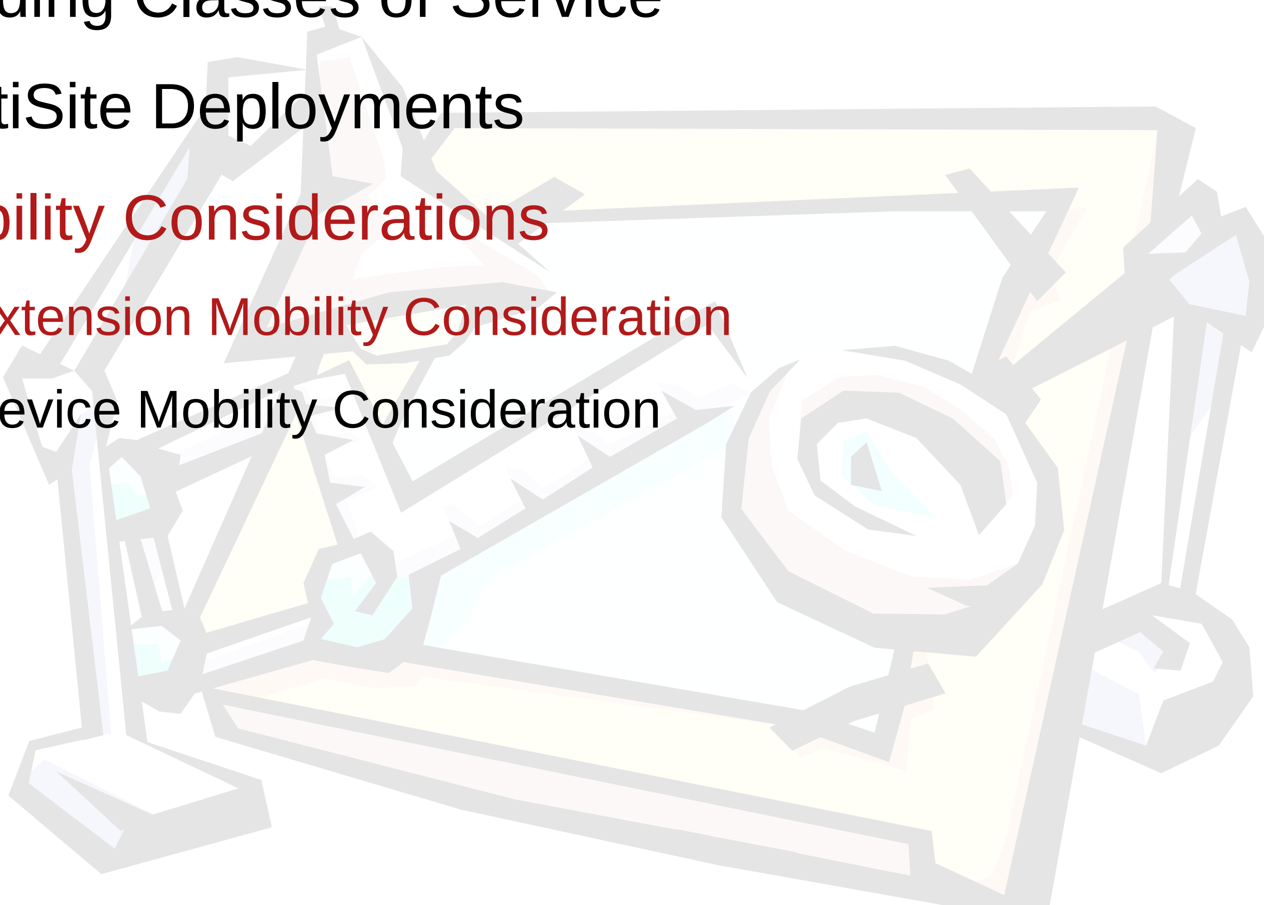
VLOD with Flat Addressing
Voice Mail Integration



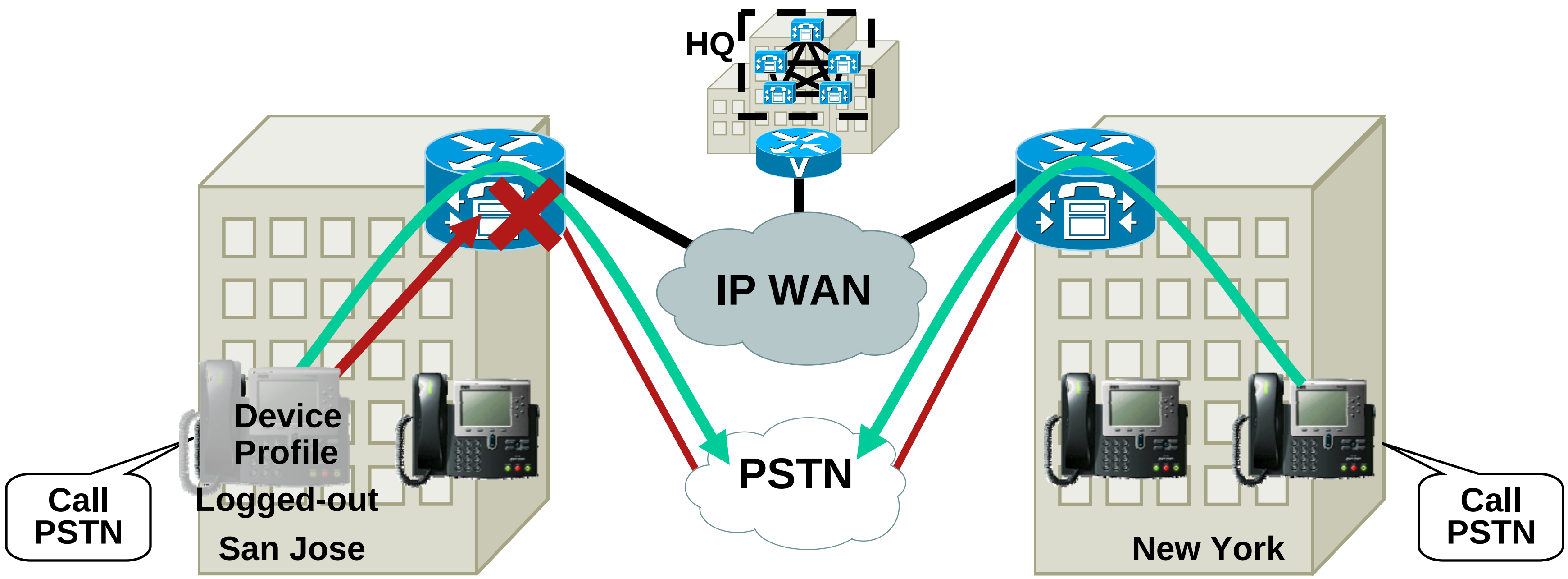
- Each eight digit extension is unique ➔ it can be used to identify a voicemail box
- No need to use masks in voicemail profile
- No translations necessary for MWI

Design Best Practices Agenda

- Building Classes of Service
- MultiSite Deployments
- Mobility Considerations
    - Extension Mobility Consideration
    - Device Mobility Consideration

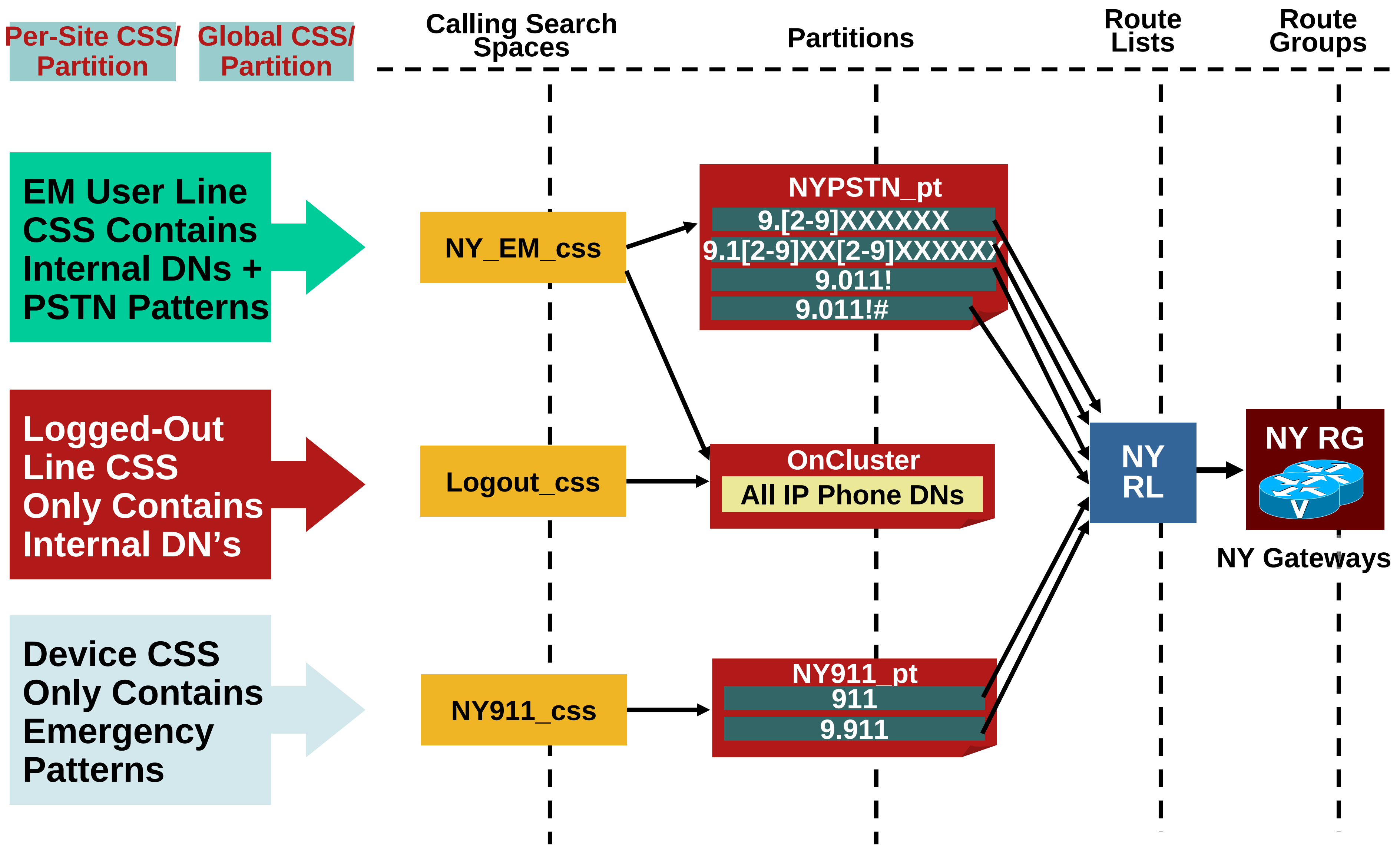


Extension Mobility Considerations
Requirements

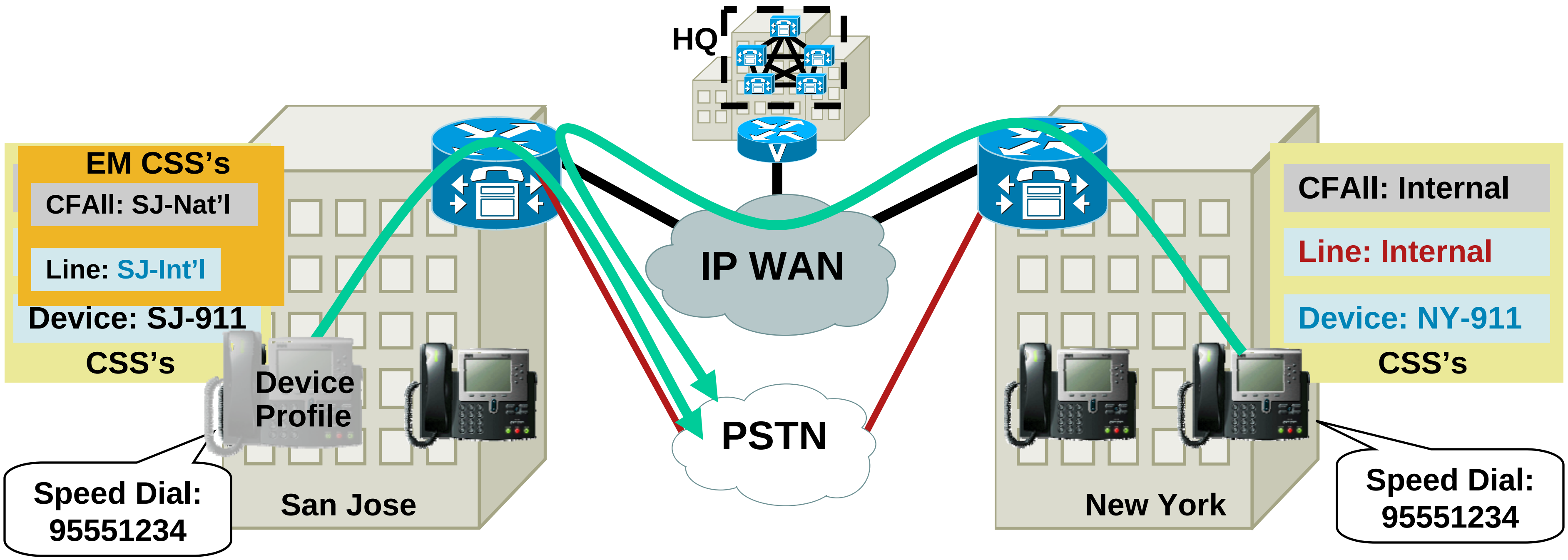


- Allow users to log in at different sites with a single device profile
- Restrict PSTN calls when logged out
- Always route emergency calls via local gateway
- Optional: route all PSTN calls via local gateway

Extension Mobility Considerations
Traditional Dial Plan Approach

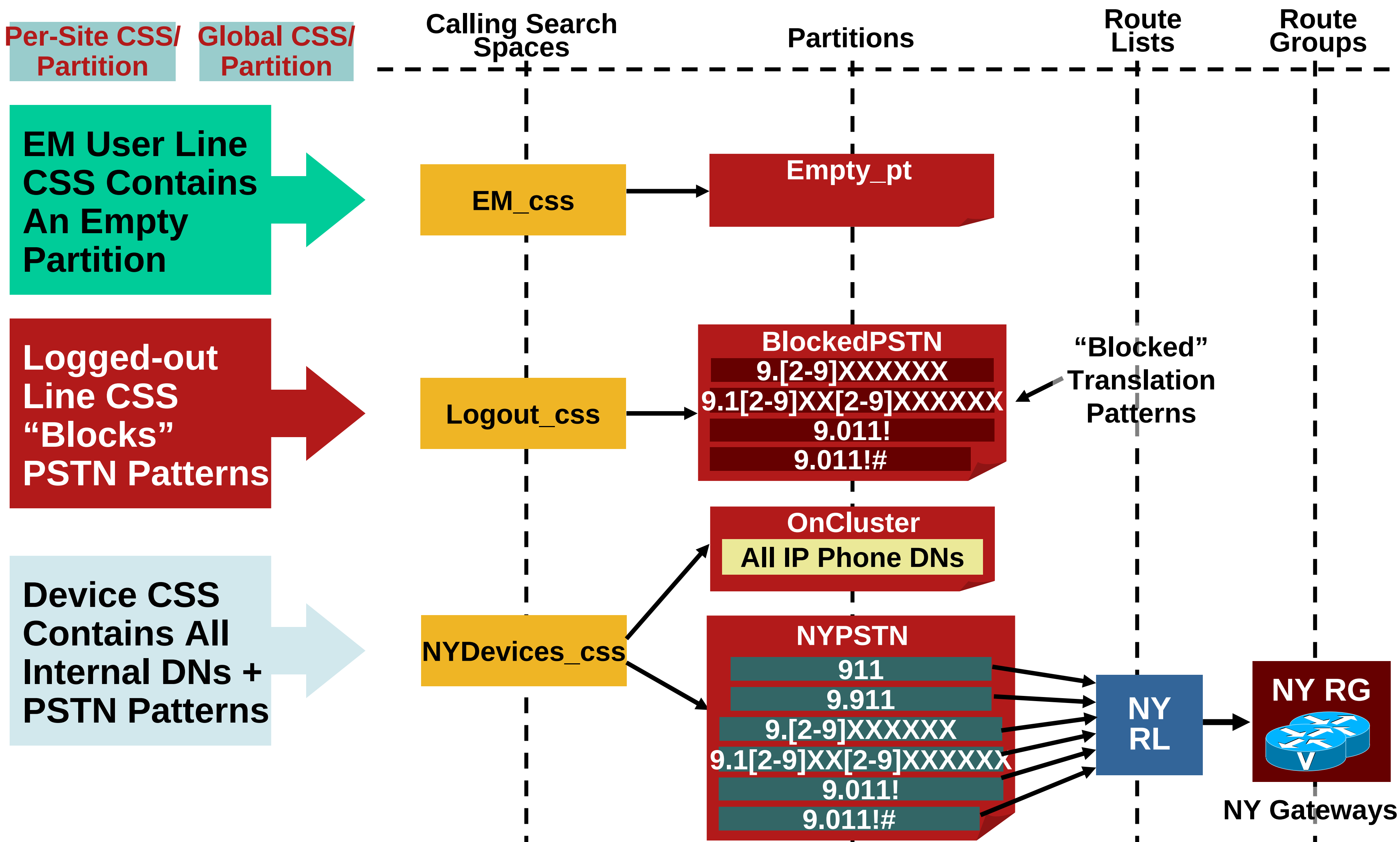


Extension Mobility Considerations
Traditional Dial Plan Approach: Behavior

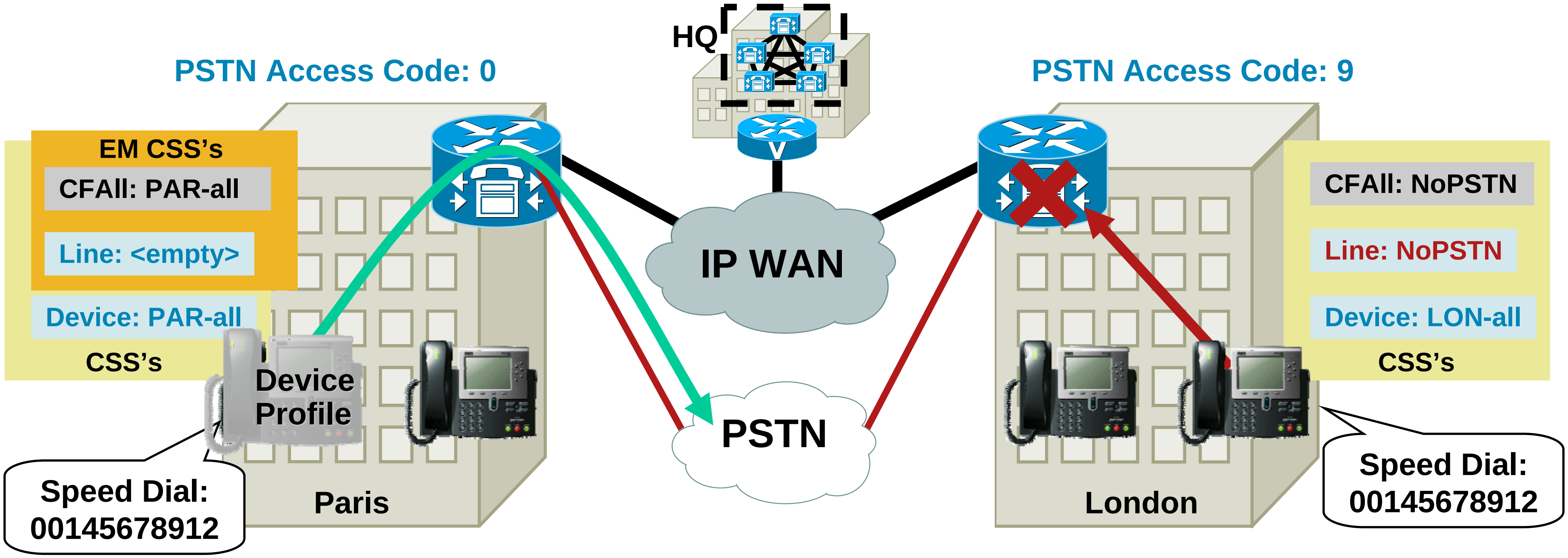


- Emergency calls routed via local gateway
- Other PSTN calls routed via "home" gateway
- User dialing habits and speed dials are automatically preserved

Extension Mobility Considerations
Line/Device Dial Plan Approach

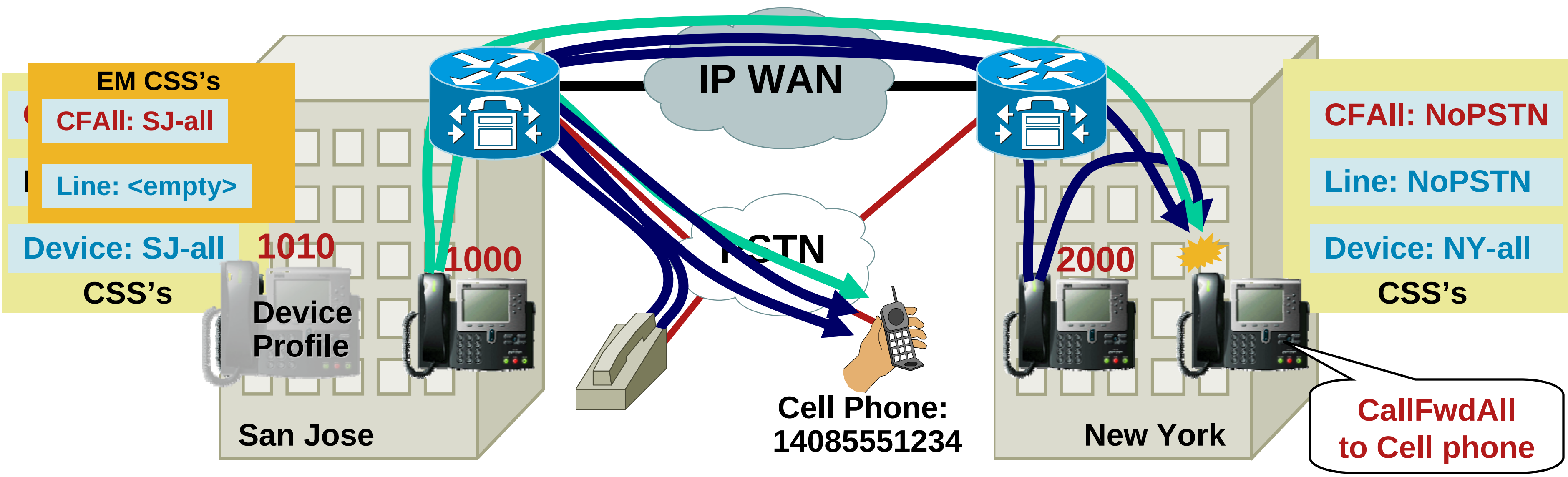


Extension Mobility Considerations
Line/Device Dial Plan Approach: Behavior



- 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

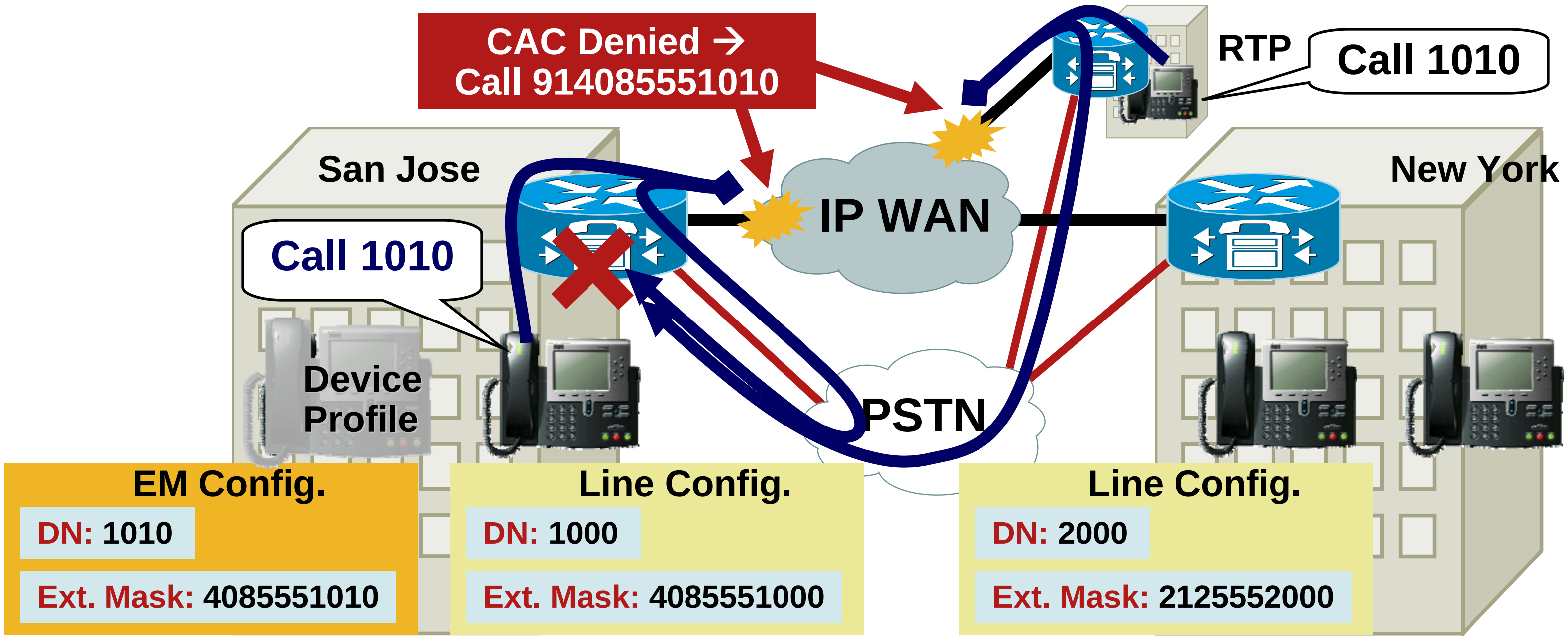
Extension Mobility Considerations
Line/Device Dial Plan Approach: Forwarded Calls



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

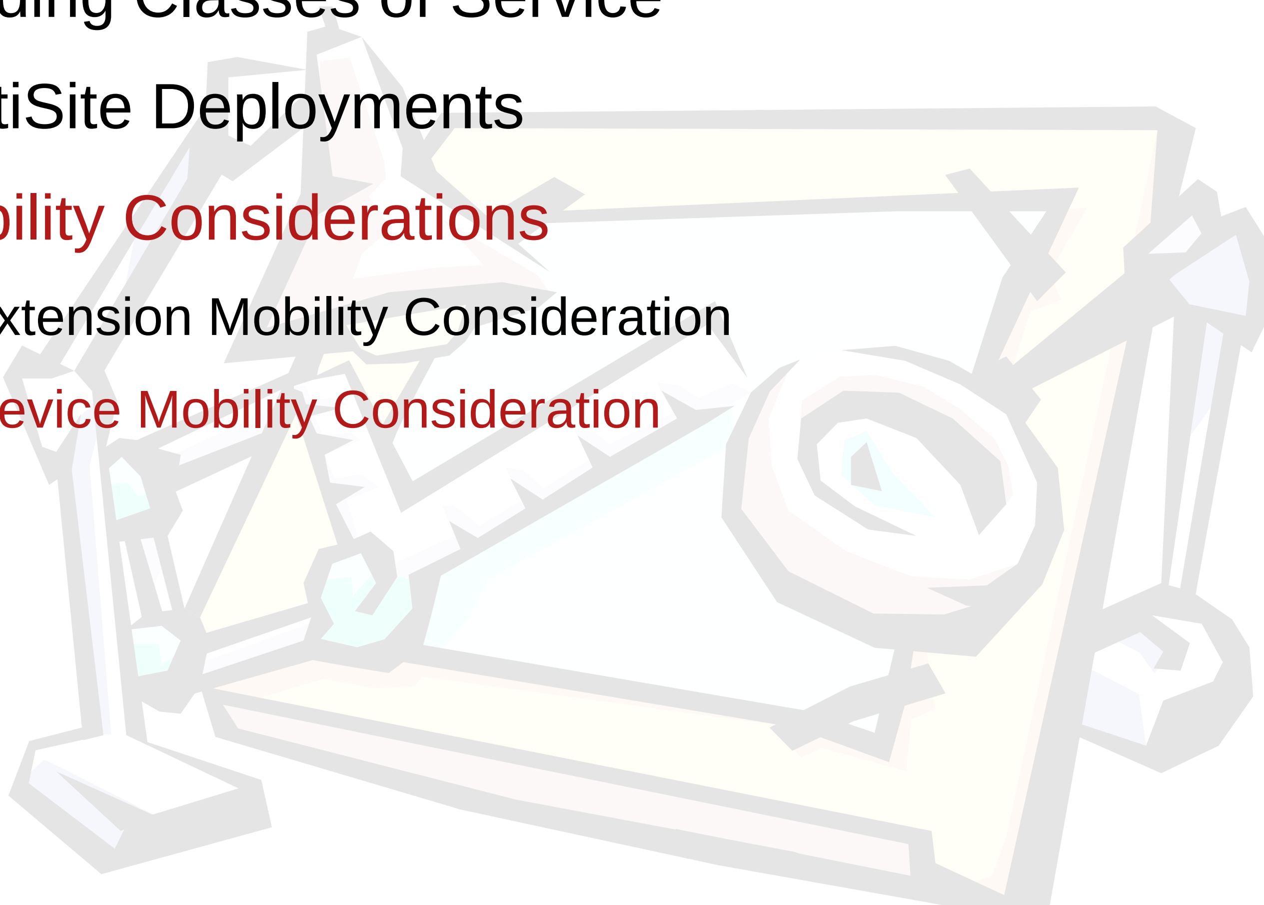
Extension Mobility Considerations
AAR Interaction



- 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 Consideration
    - Device Mobility Consideration



Device Mobility Considerations
High-level Behavior - *CallManager 4.2 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 CallManager:

    Device in Home Location

    Device in Roaming Location

Device Mobility
Device Pool Changes

Device Pool

CallManager Group
Auto-reg CSS

Roaming Sensitive Settings

Date/Time Group
Region
MRGL
Network Locale
SRST Reference
Location
Physical Location
Device Mobility Group

Impacts CAC, Media Resource & SRST

Device Mobility Related Information

Device CSS
AAR Group
AAR CSS

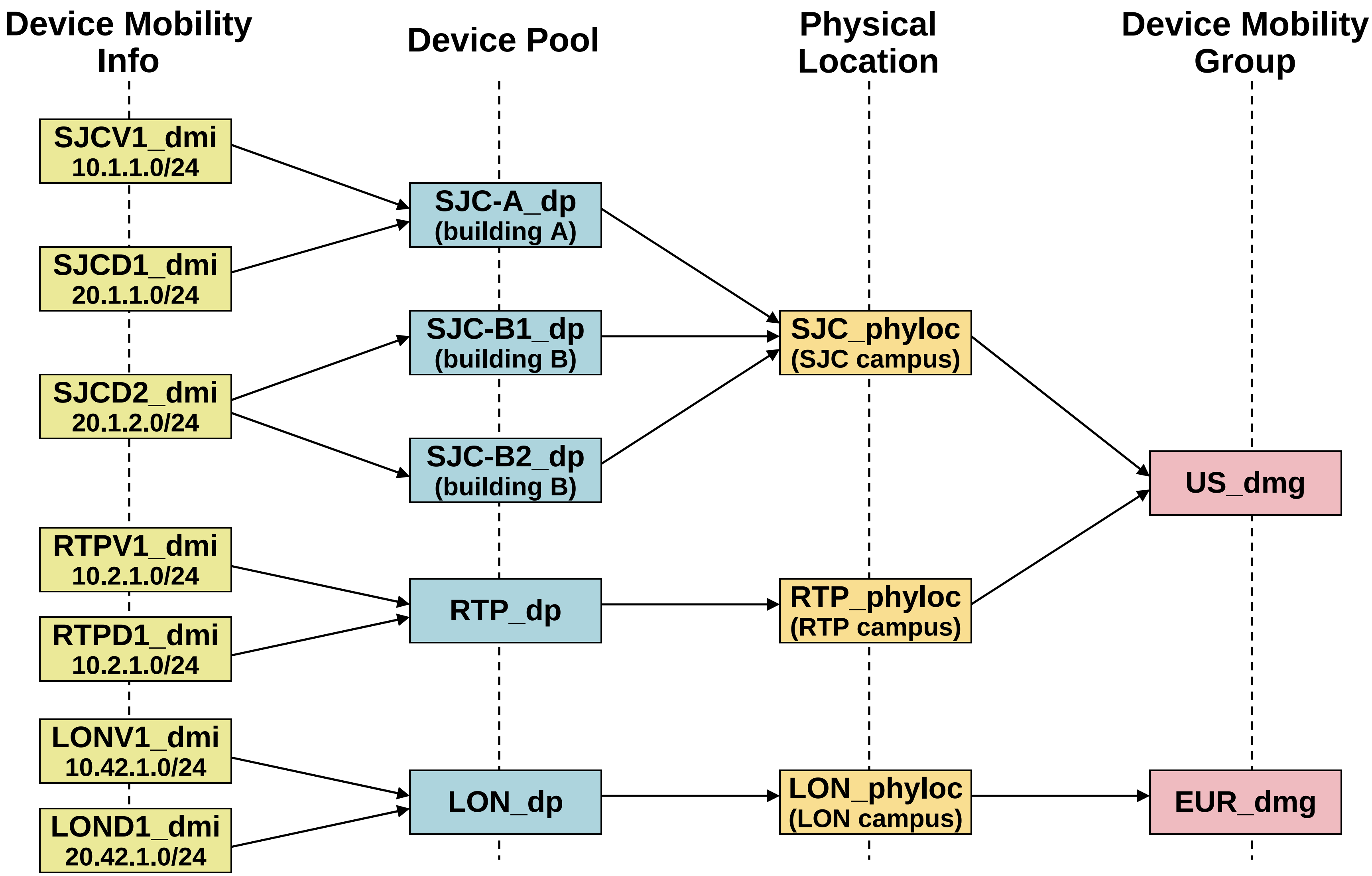
Impacts Dial Plan

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



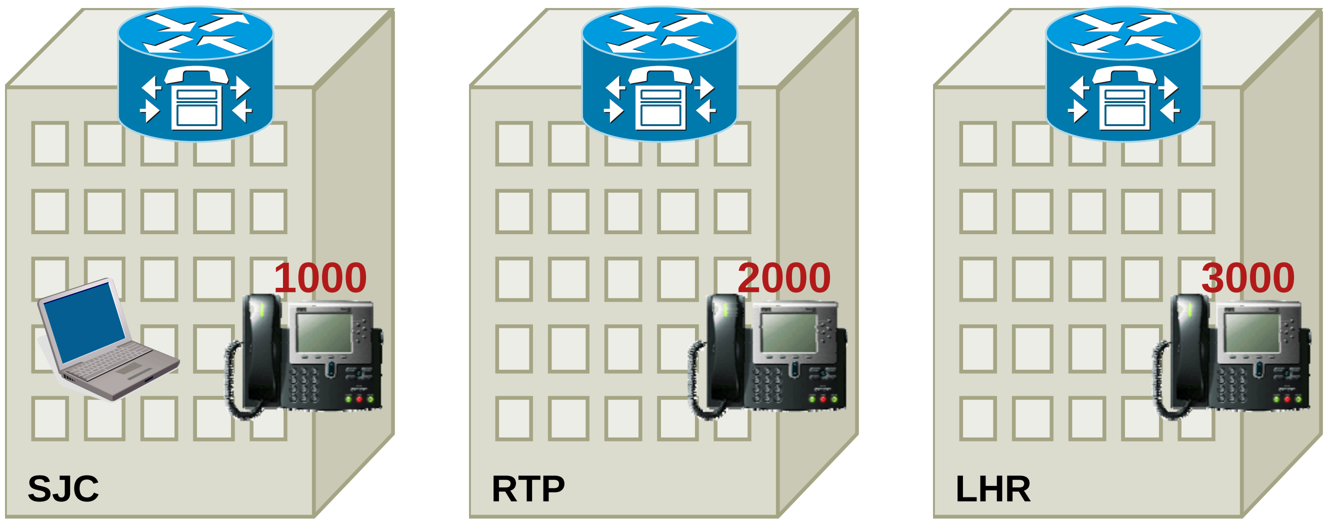
Device Mobility Considerations

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



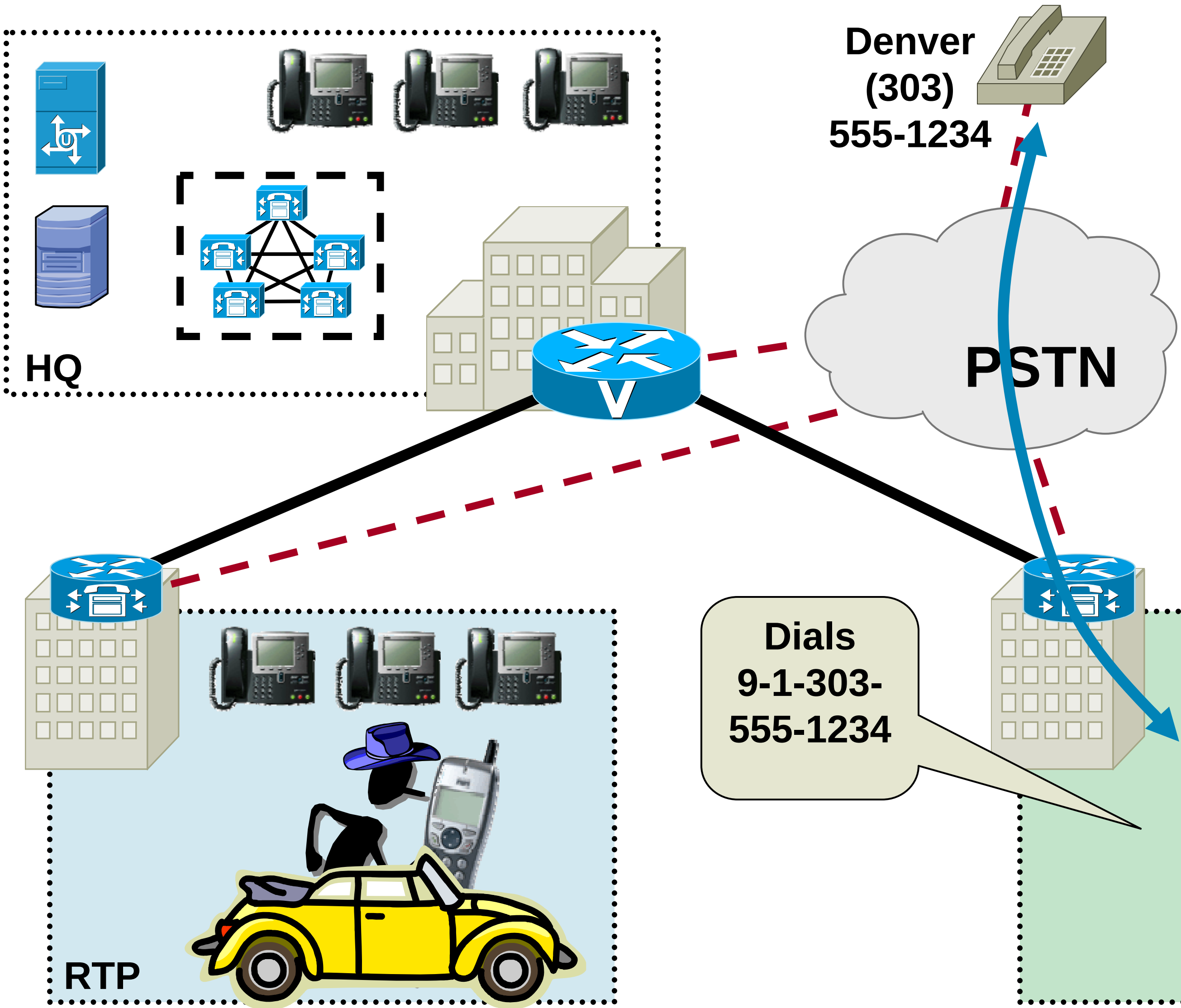
Note: When roaming from SJC to LHR, we are crossing DMGs Dial Plan-related information does not change.

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.

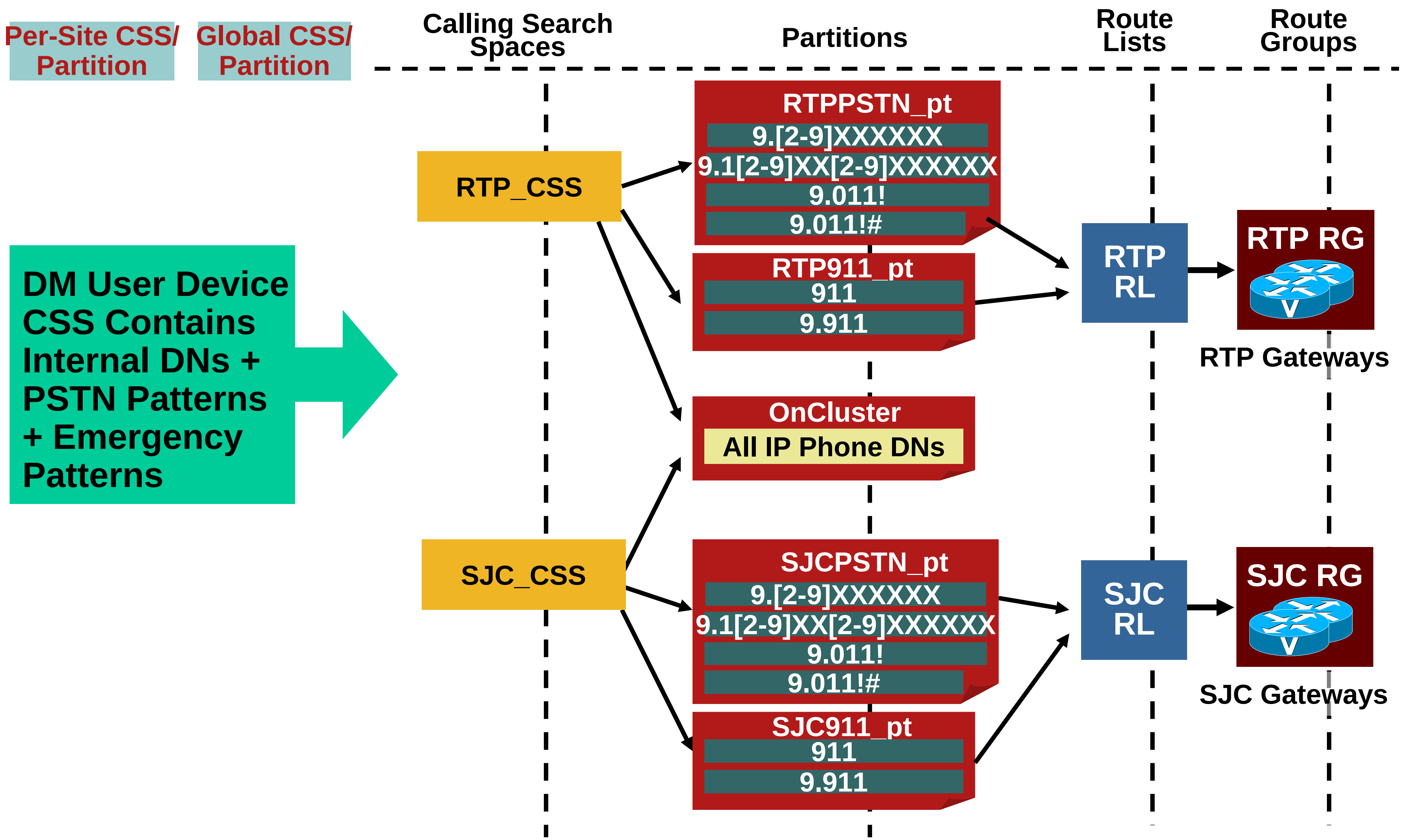
Device Mobility Related Information
Changes only when roaming <u>within the same DMG.</u>

Device Mobility Considerations
Requirements (Call Manager 4.2)



- 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
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

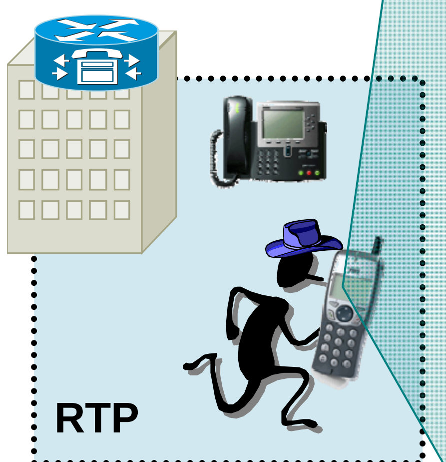
Copy | Update | Delete | Reset Phone

Phone Configuration (Model = Cisco 7960)

Device Information

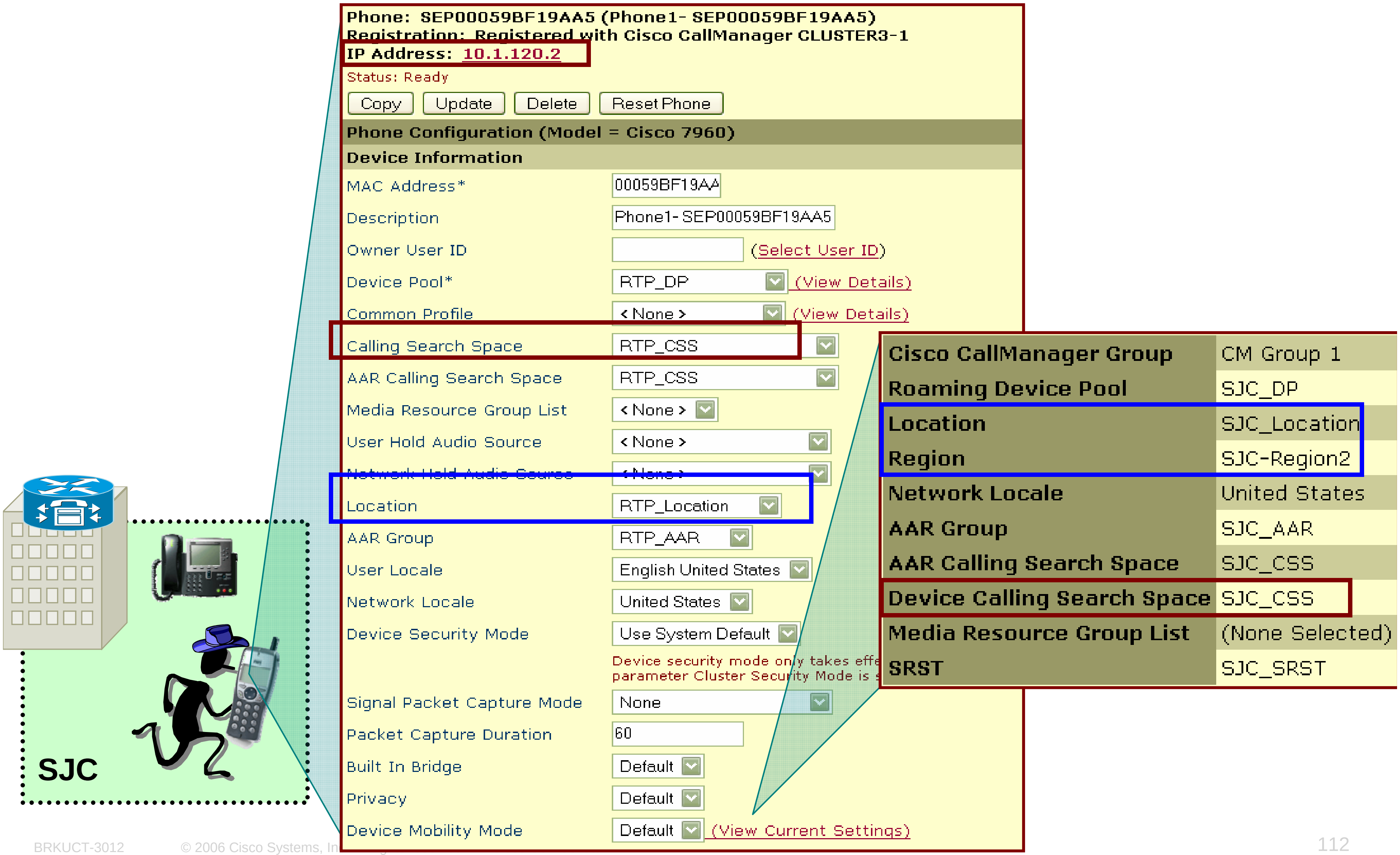
MAC Address*	00059BF19AA
Description	Phone1- SEP00059BF19AA5
Owner User ID	(Select User ID)
Device Pool*	RTP_DP (View Details)
Common Profile	< None > (View Details)
Calling Search Space	RTP_CSS
AAR Calling Search Space	RTP_CSS
Media Resource Group List	< None >
User Hold Audio Source	< None >
Network 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 Device security mode only takes effe parameter Cluster Security Mode is s
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

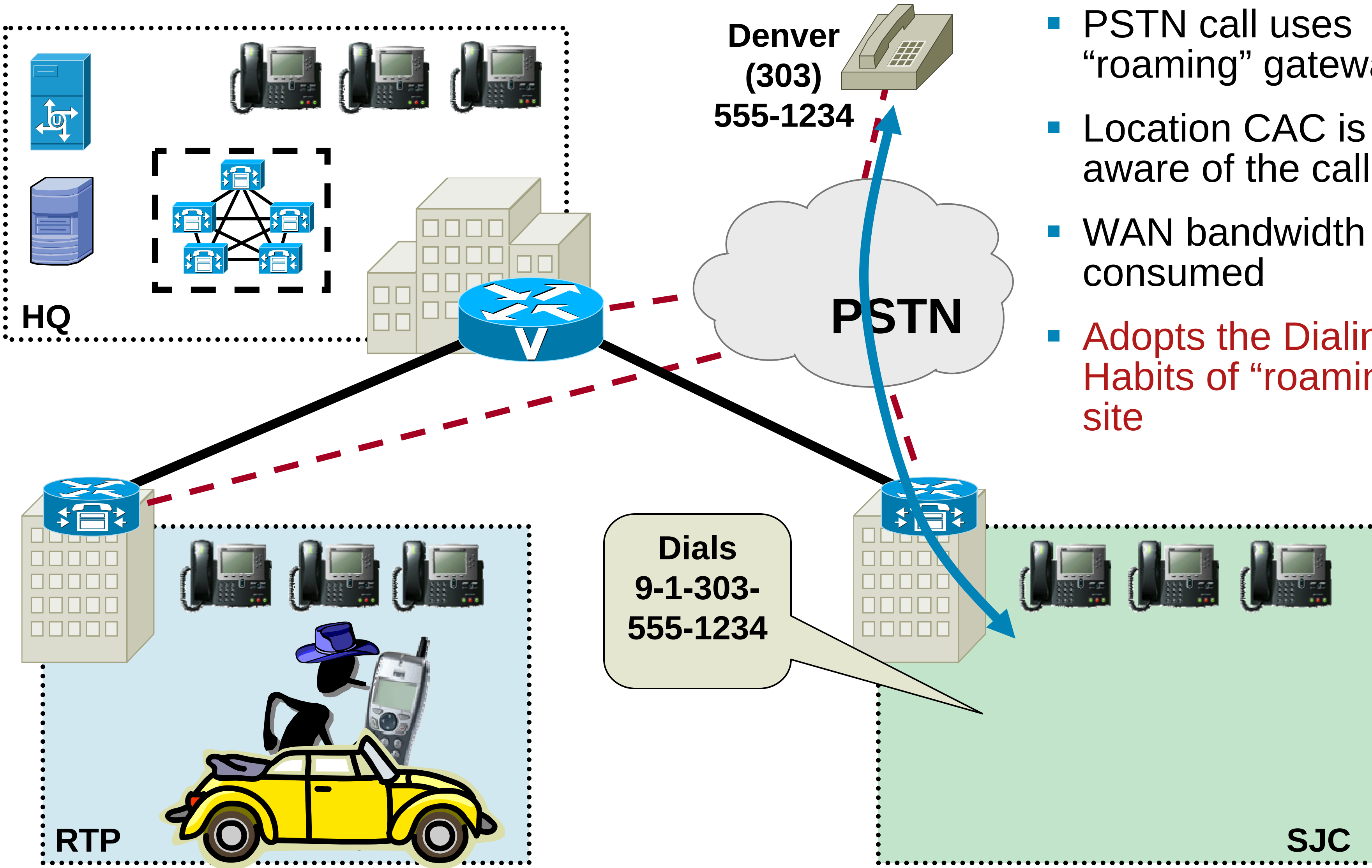
Phone Configuration (Model = Cisco 7960)

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
	Device security mode only takes effe… parameter Cluster Security Mode is …
Signal Packet Capture Mode	None
Packet Capture Duration	60
Built In Bridge	Default
Privacy	Default
Device Mobility Mode	Default (View Current Settings)

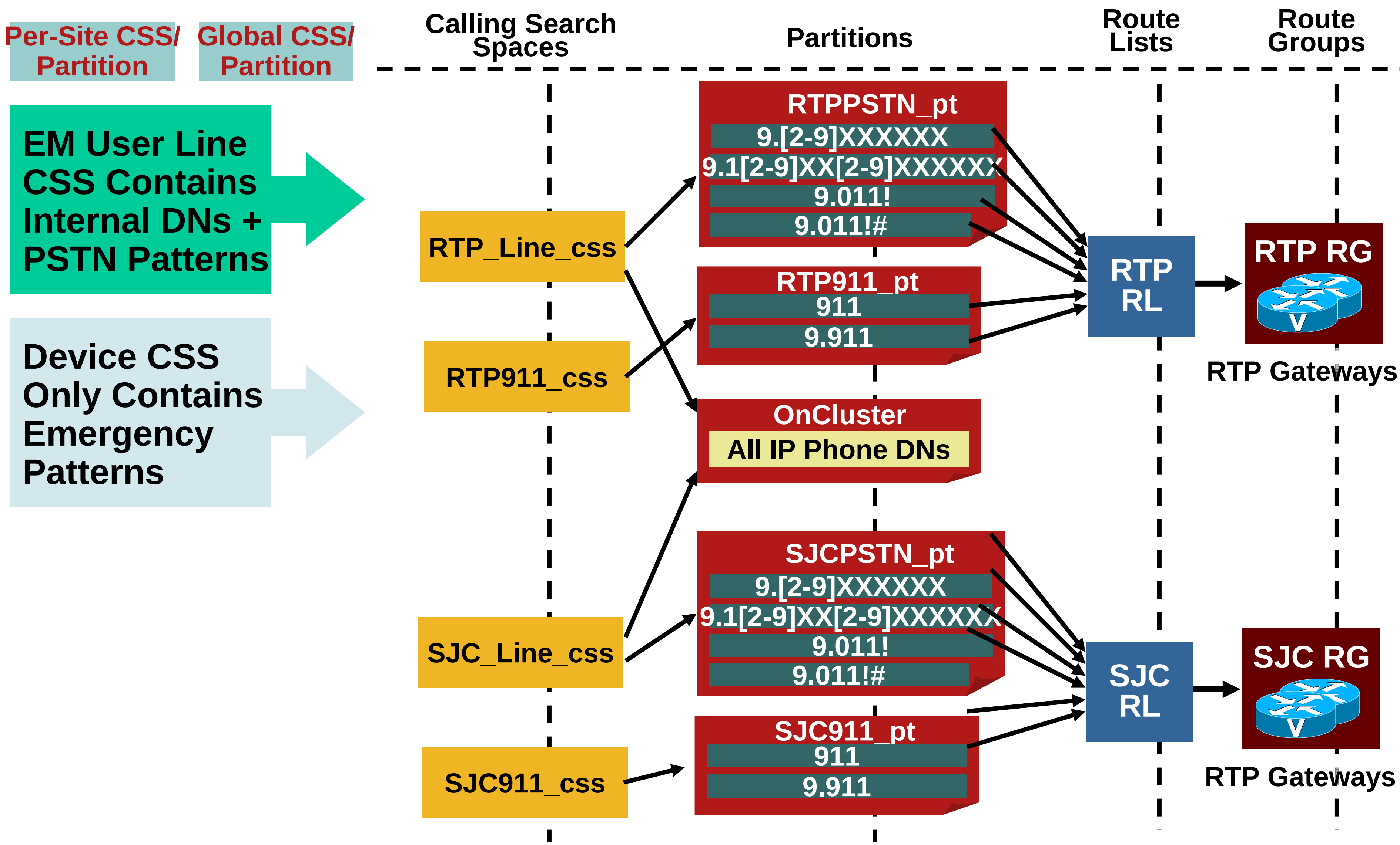
Cisco CallManager Group	CM Group 1
Roaming Device Pool	SJC_DP
Location	SJC_Location
Region	SJC-Region2
Network Locale	United States
AAR Group	SJC_AAR
AAR Calling Search Space	SJC_CSS
Device Calling Search Space	SJC_CSS
Media Resource Group List	(None Selected)
SRST	SJC_SRST

Device Mobility Considerations
Traditional Dial Plan Approach: Behavior

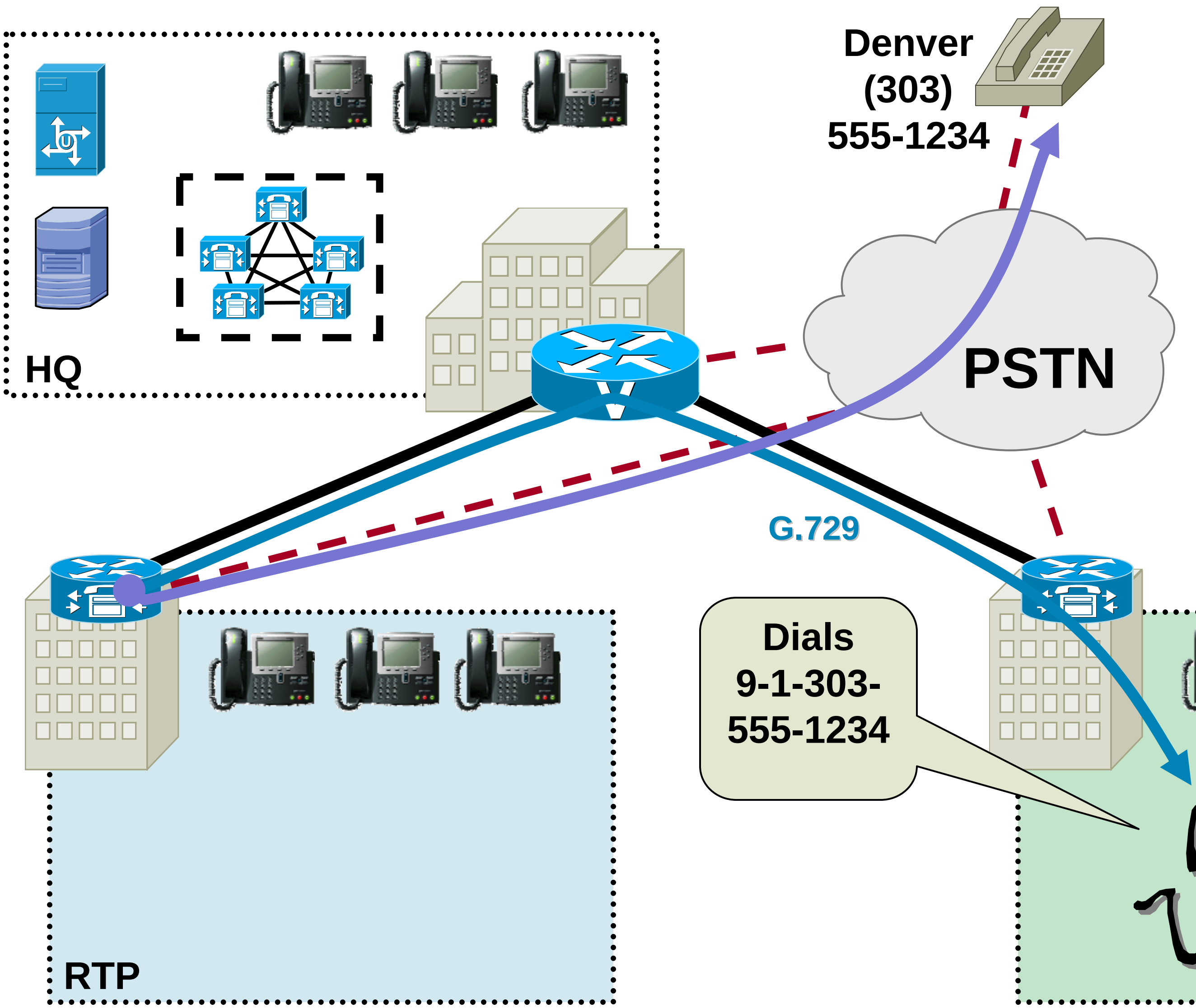


- PSTN call uses "roaming" gateway
- Location CAC is aware of the call
- WAN bandwidth not consumed
- Adopts the Dialing Habits of "roaming" site

Device Mobility Considerations
Traditional Dial Plan Approach (EM Approach)

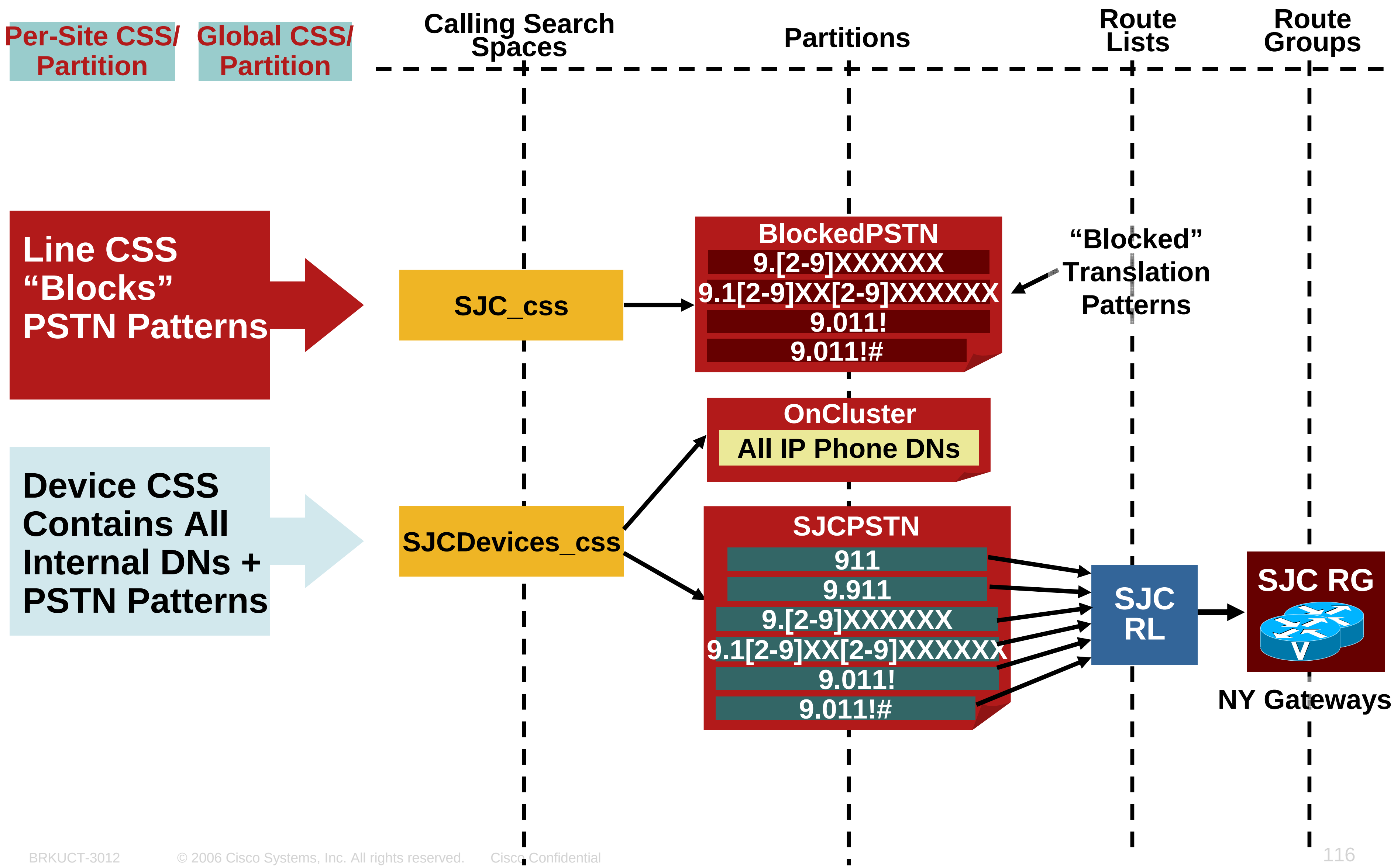


Device Mobility Considerations
Traditional Dial Plan (EM Approach): Behavior

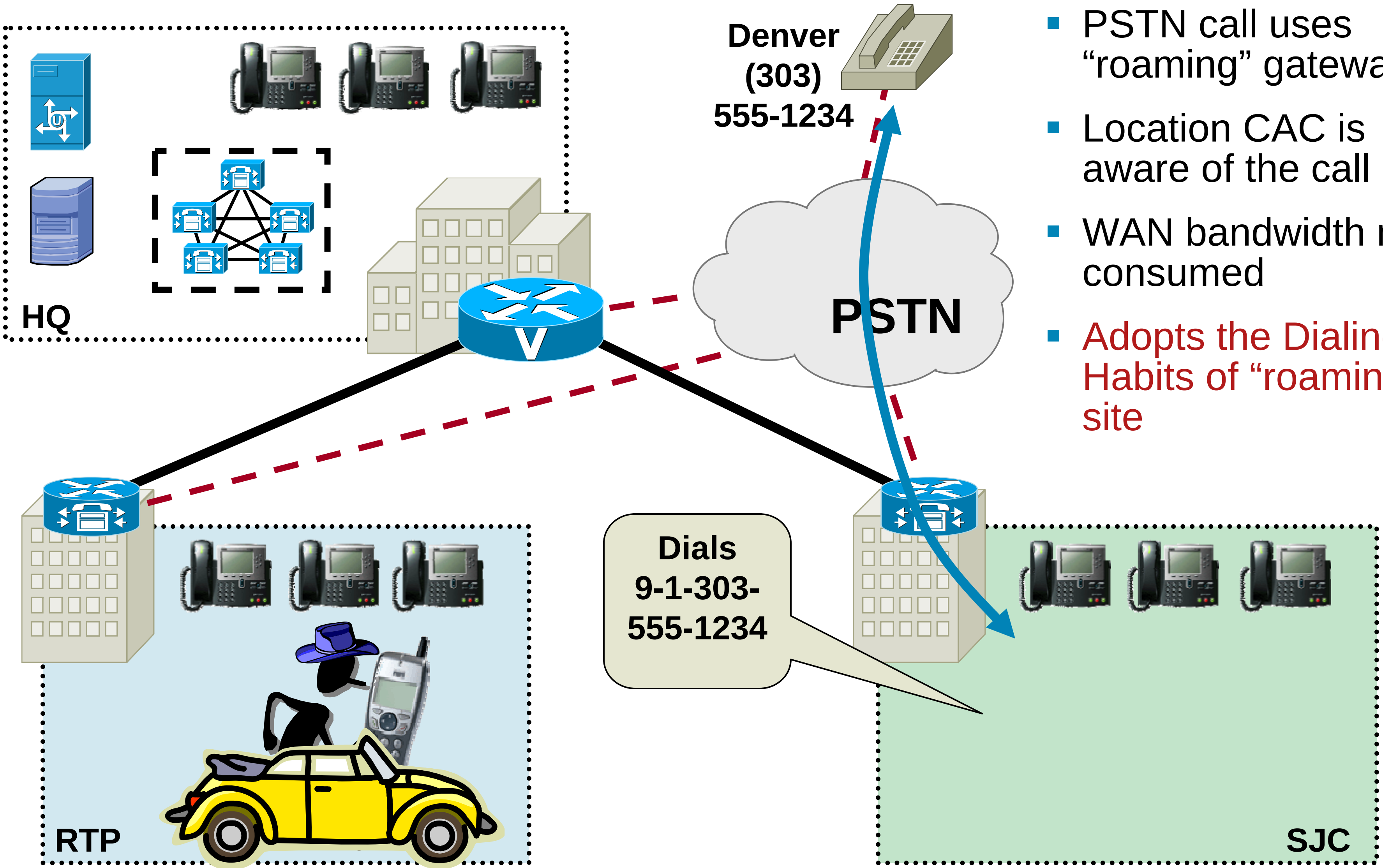


- PSTN call uses "home" gateway
- Location CAC is aware of the call
- G.729 CODEC uses more WAN bandwidth
- Preserves the dialing habits

Device Mobility Considerations
Line/Device Dial Plan Approach

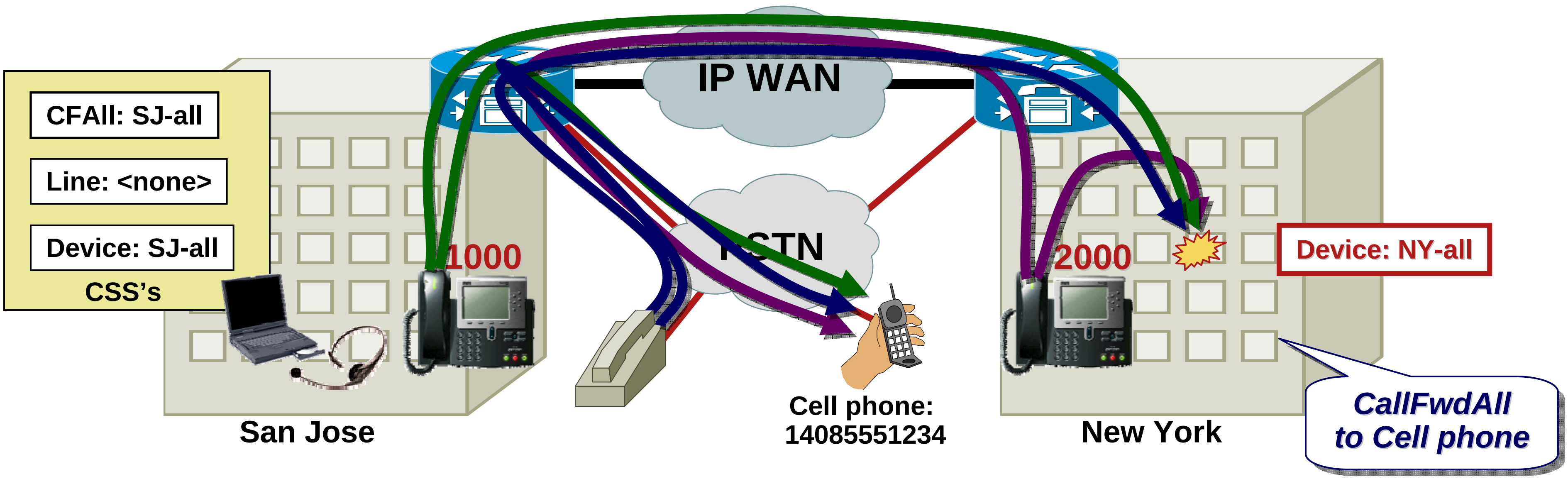


Device Mobility Considerations
Line/Device Dial Plan Approach: Behavior



- PSTN call uses "roaming" gateway
- Location CAC is aware of the call
- WAN bandwidth not consumed
- Adopts the Dialing Habits of "roaming" site

Device Mobility Consideration
Line/Device Dial Plan Approach: Forwarded Calls



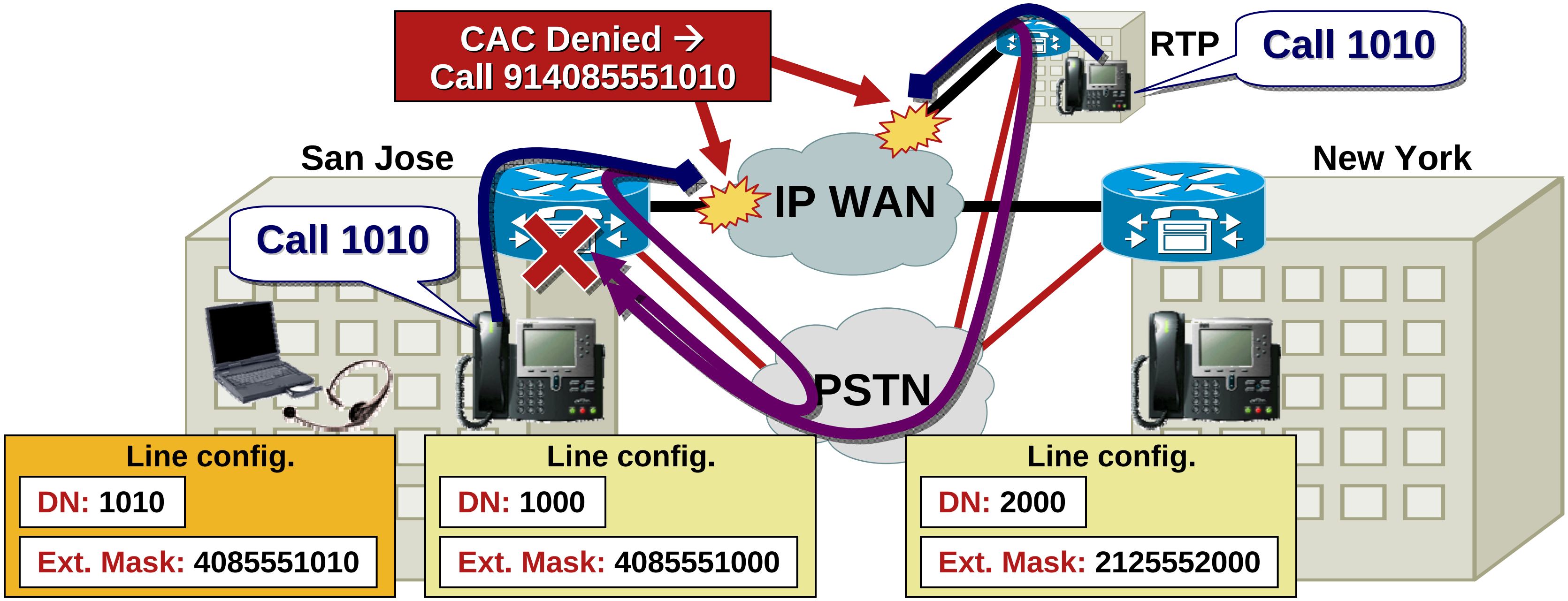
- When a SJ user moves to NY site and forwards his phone to a PSTN number:

  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

Conclusions

Conclusions
General Recommendations

- KEEP IT SIMPLE!
- Plan for future growth
- Use Gatekeeper-controlled Intercluster Trunks when more than two Cisco CallManager clusters are present
- Normalize DNs to the full E.164 when using Gatekeeper for dial plan resolution

Conclusions
Summary: What Did We Cover?

- Planning an enterprise IP telephony dial plan—uniform vs. variable-length dialing
- Enterprise IP telephony dial plan elements—the tools and how to use them
- Design recommendations in different areas of dial plan:
  - Classes of service
  - Dialing architectures
  - Addressing methods

For More Information

Dial Plan

The dial plan is one of the key elements of an IP Telephony system, and an integral part of all call processing agents. Generally, the dial plan is responsible for instructing the call processing agent on how to route calls. Specifically, the dial plan performs the following main functions:

- Endpoint addressing

  Reachability of internal destinations is provided by assigning directory numbers (DNs) to all endpoints (such as IP phones, fax machines, and analog phones) and applications (such as voicemail systems, auto attendants, and conferencing systems)

- Path selection

  Depending on the calling device, different paths can be selected to reach the same destination. Moreover, a secondary path can be used when the primary path is not available (for example, a call can be transparently rerouted over the PSTN during an IP WAN failure).

- Calling privileges

  Different groups of devices can be assigned to different classes of service, by granting or denying access to certain destinations. For example, lobby phones might be allowed to reach only internal and local PSTN destinations, while executive phones could have unrestricted PSTN access.

- Digit manipulation

  In some cases, it is necessary to manipulate the dialed string before routing the call; for example, when rerouting over the PSTN a call originally dialed using the on-net access code, or when expanding an abbreviated code (such as 0 for the operator) to an extension.

- Call coverage

  Special groups of devices can be created to handle incoming calls for a certain service according to different rules (top-down, circular hunt, longest idle, or broadcast).

This chapter examines the following main aspects of dial plan:

- Planning Considerations, page 10-2

  This section analyzes the thought process involved in planning an IP Telephony dial plan, ranging from the number of digits used for internal extensions to the overall architecture of a company's internal dial plan. (Prerequisite: Some familiarity with dial plans in general.)

- Dial Plan Elements, page 10-7

  This section provides detailed explanations of the elements of a Cisco IP Telephony dial plan. Covered topics include call routing logic, calling privileges, and digit manipulation techniques for

More Details in: Chapter 10 of the IP Telephony SRND for Cisco CallManager 4.x and 5.0, Available at: http://www.cisco.com/go/srnd/

Meet the Experts
Unified Communications Technologies

- Janet Byron
  Technical Leader

- Jan-Willem Ruys
  Consulting Engineer

- Luc Bouchard
  Technical Marketing Engineer

- Mariano O'Kon
  Consulting Systems Engineer

- Paul Tindall
  Consulting System Engineer

- Richard Dodsworth
  Consulting Systems Engineer

Meet the Experts
Unified Communications Technologies

- TJ Schuler
  Technical Marketing Engineer
- Tobias Neumann
  Consulting Systems Engineer
- Tony Mulchrone
  Technical Mktg Eng
- Yves Torjman
  Consulting System Engineer
- Zorela Sora
  Consulting Engineer

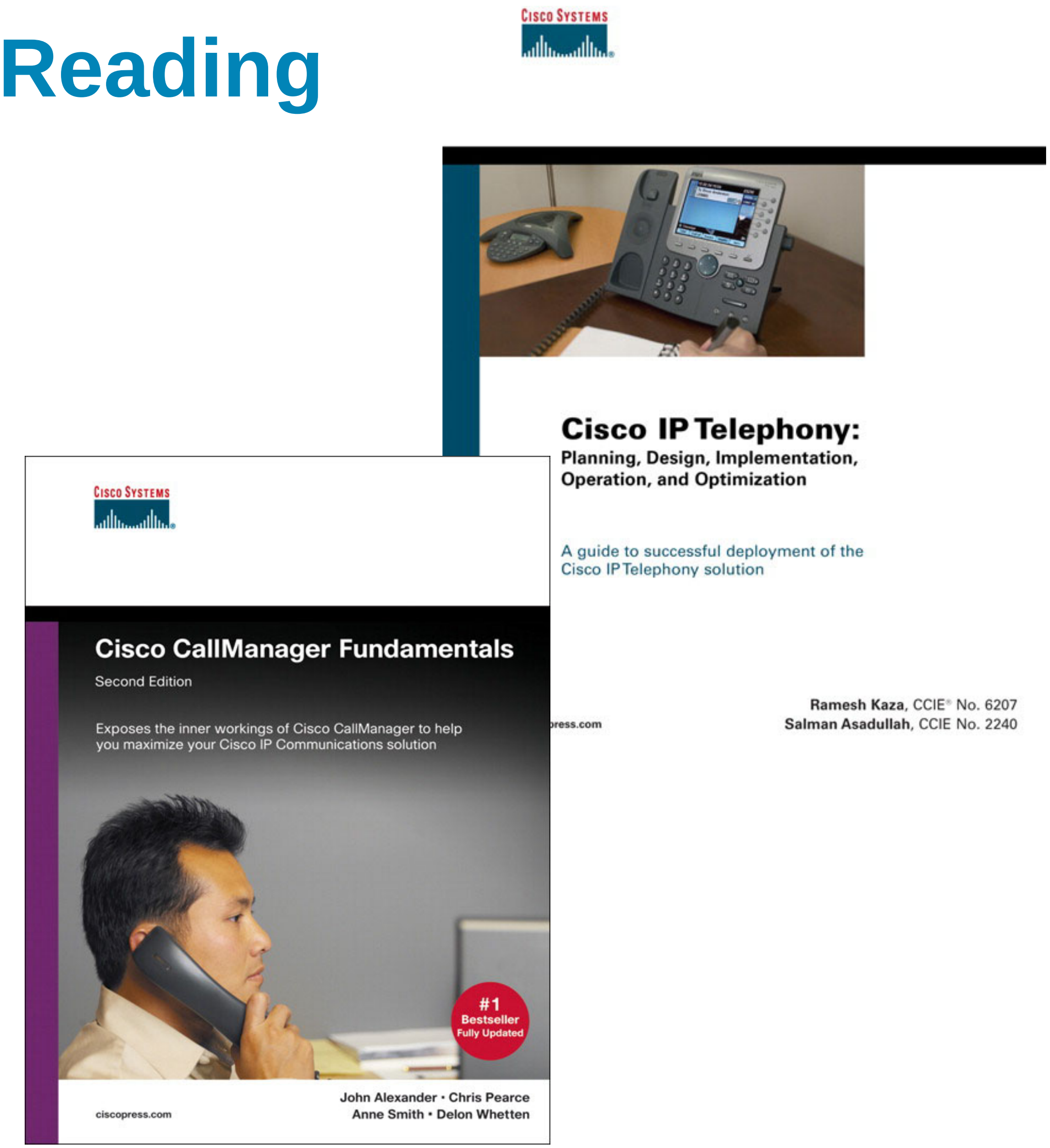




Recommended Reading

BRKUCT - 3012

- Cisco CallManager Fundamentals
- Cisco IP Telephony: Planning, Design, Implementation, Operation, and Optimization

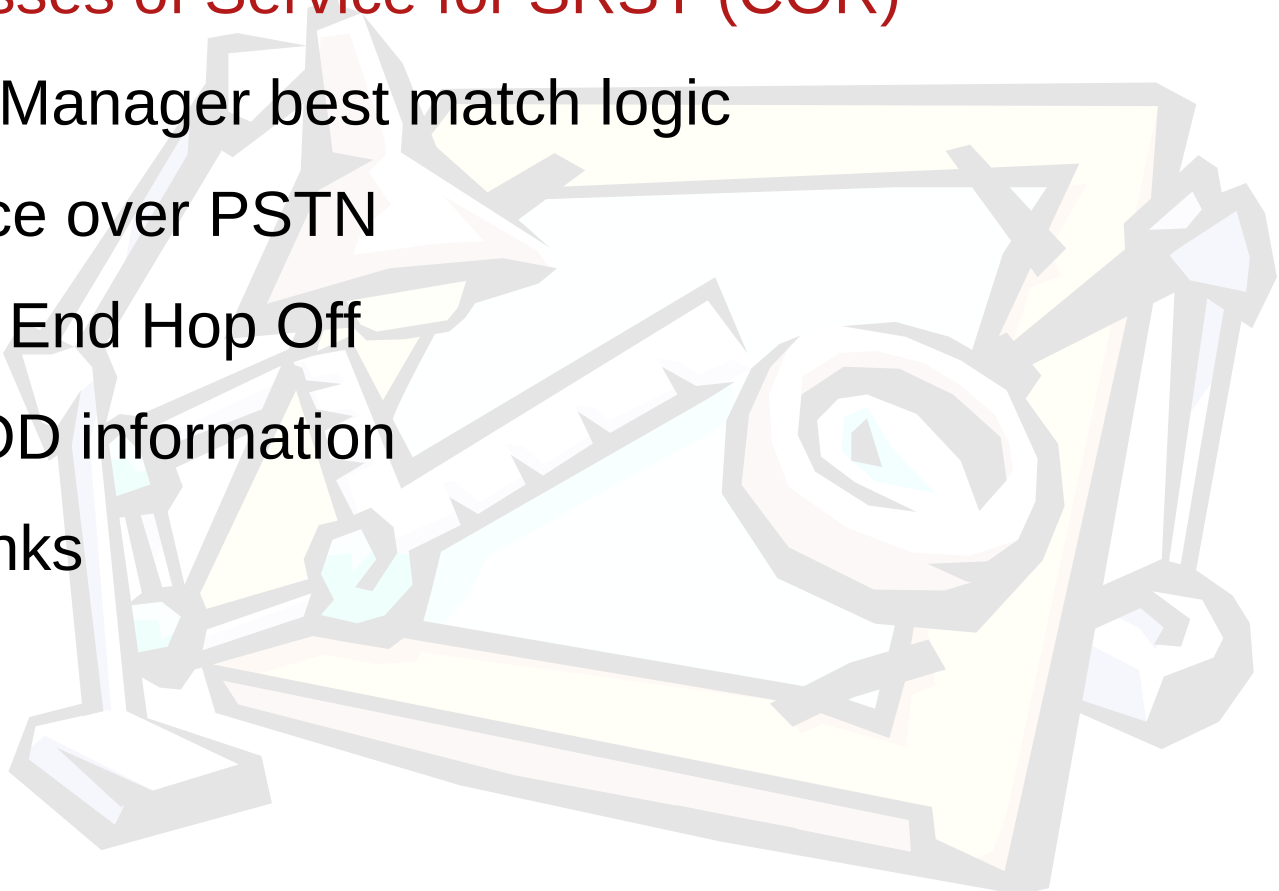


Available in the Cisco Company Store

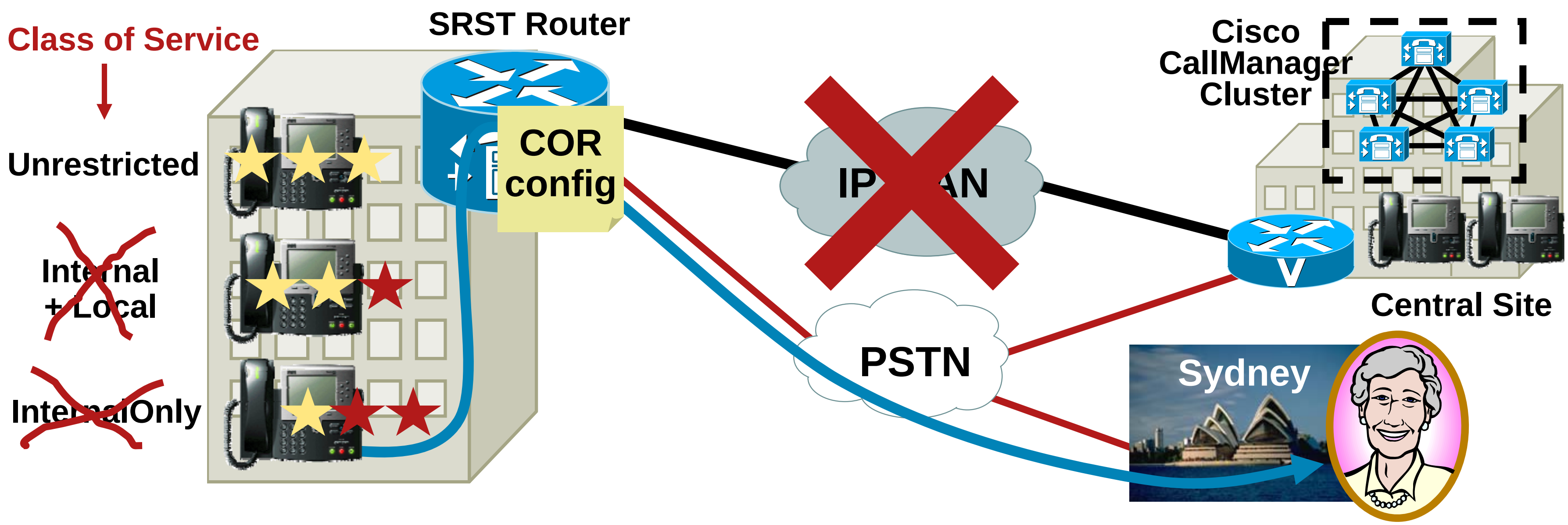
Appendix
Reference material follows

Appendix

- Classes of Service for SRST (COR)
- CallManager best match logic
- Voice over PSTN
- Tail End Hop Off
- VLOD information
- Trunks

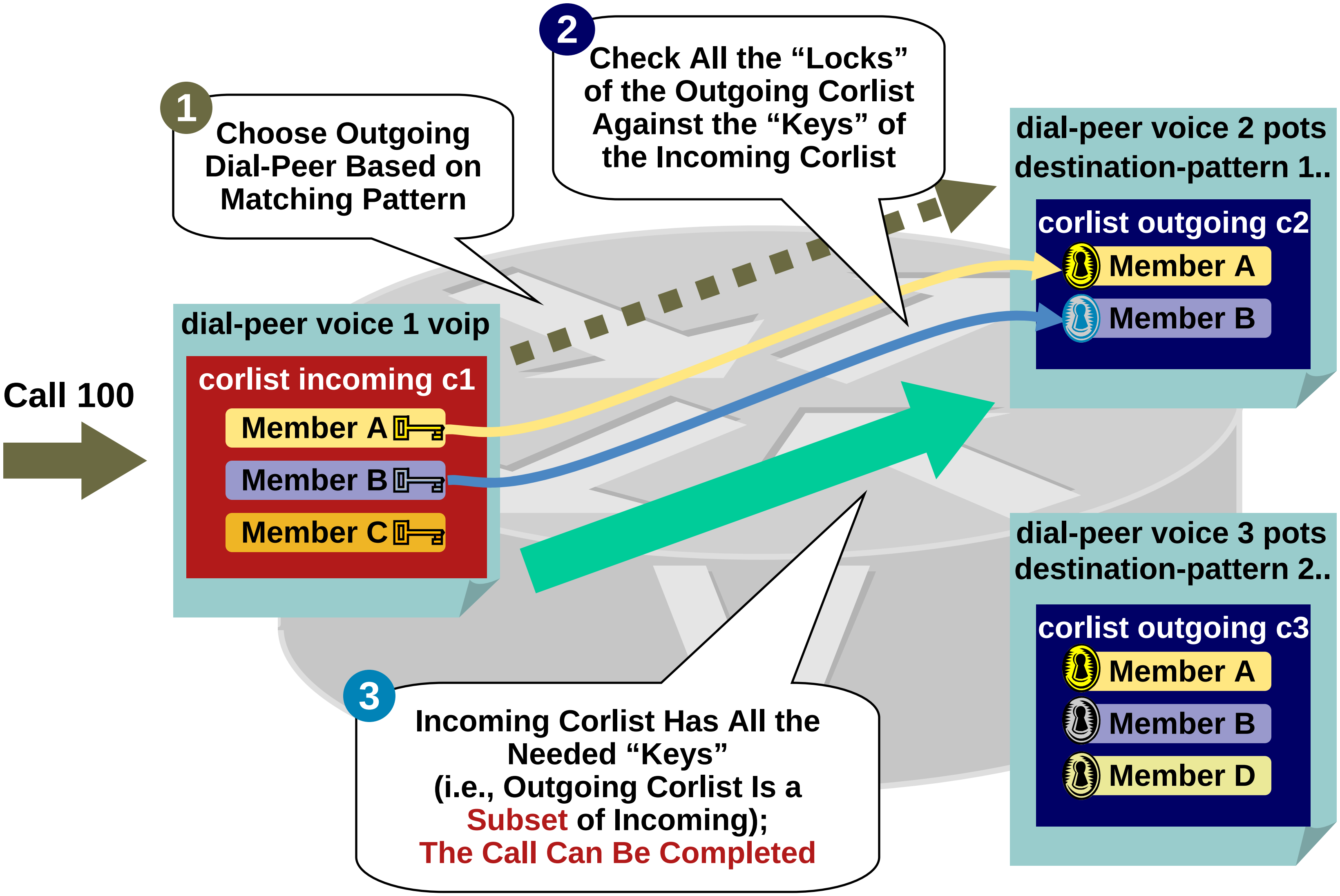


Classes of Service for SRST (COR)
Rationale

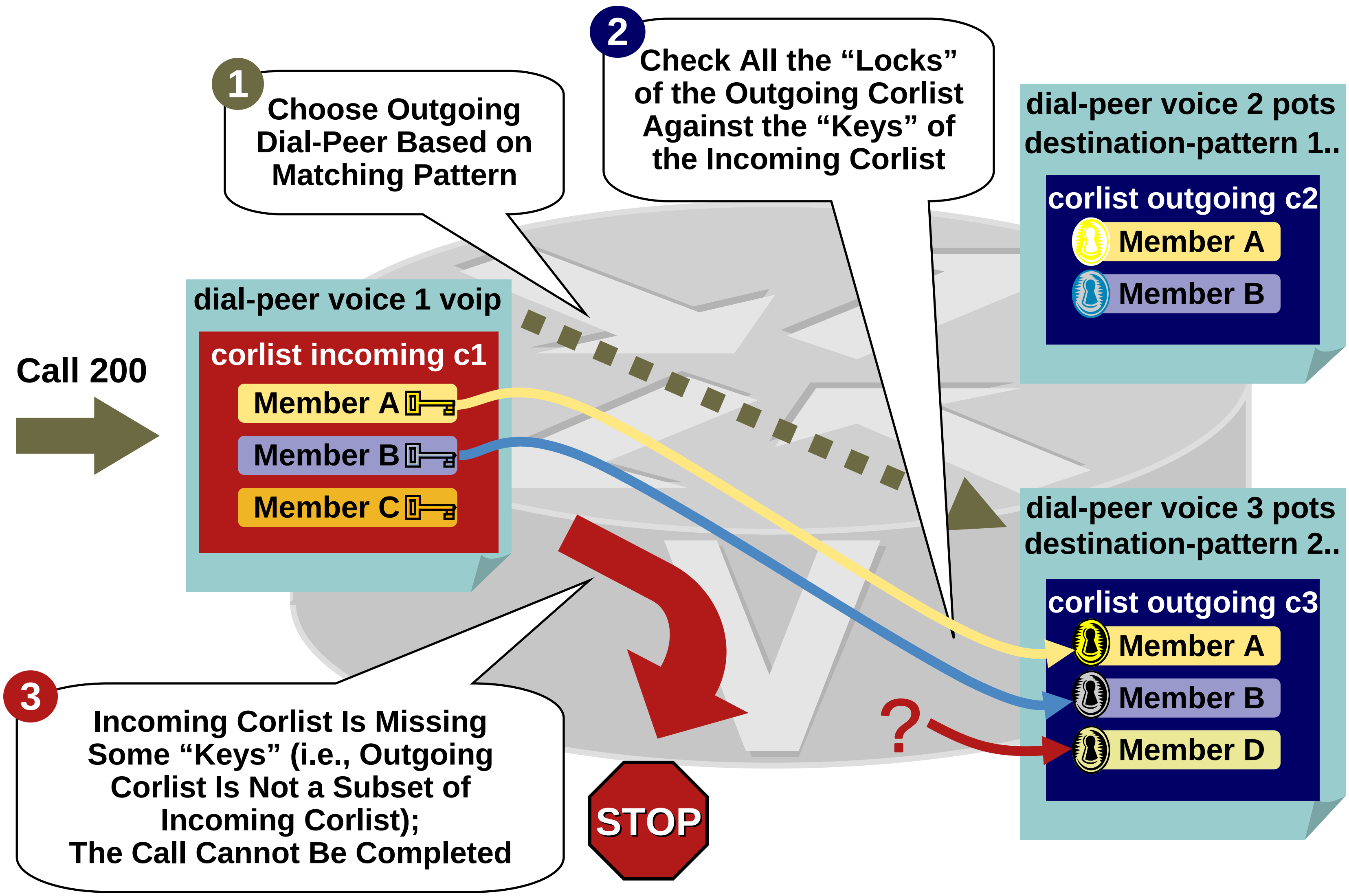


- When WAN connection is lost, Cisco CallManager classes of service are also lost ➔ All remote phones gain unrestricted PSTN access
- COR configuration on branch router allows preservation of classes of service in SRST mode

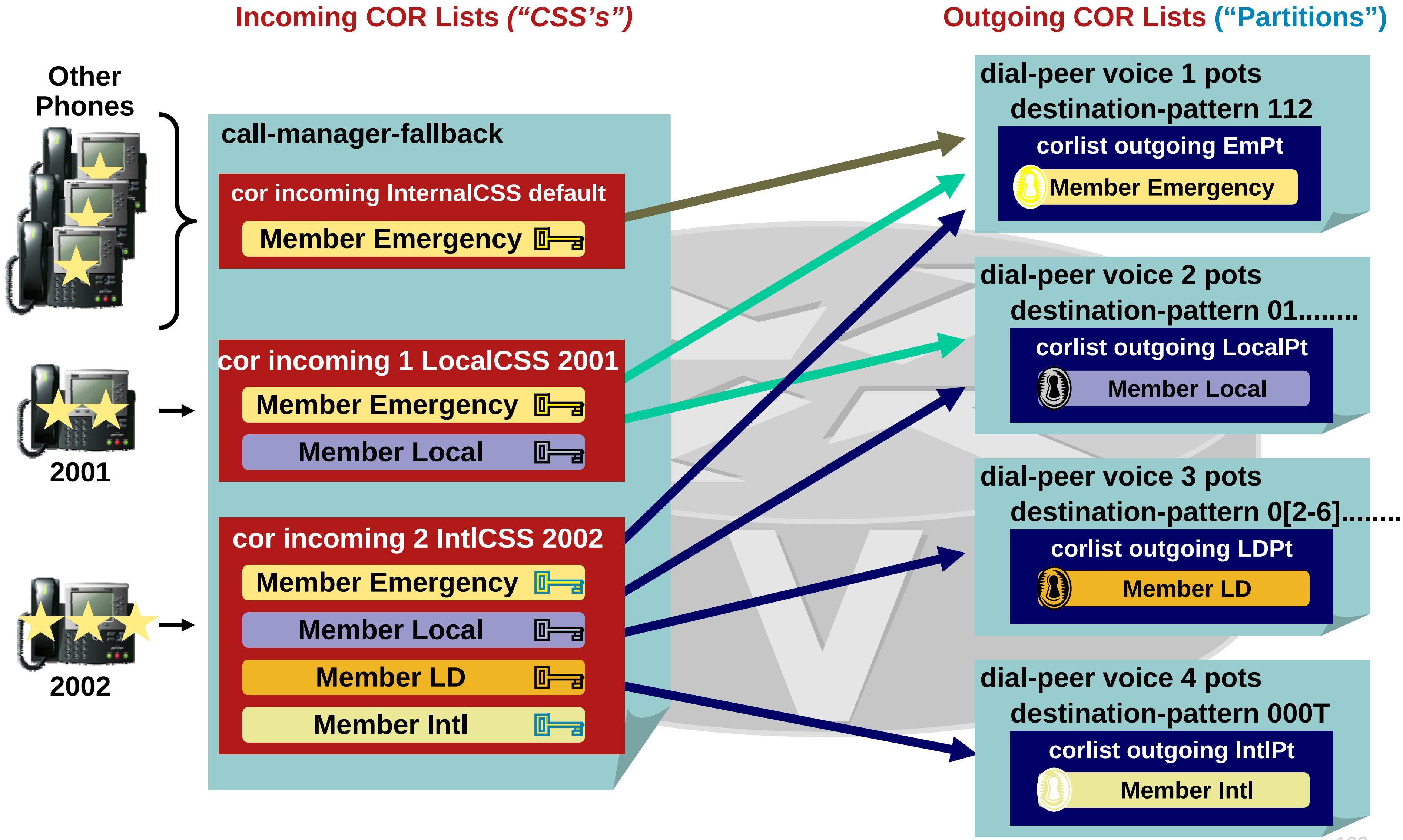
Classes of Service for SRST (COR)
COR Logic (1)



Classes of Service for SRST (COR)
COR Logic (2)



Classes of Service for SRST (COR)
How to Recreate "Partitions" and "CSS's"



Classes of Service for SRST (COR)
Step-by-Step Guidelines

- Define meaningful tags (Emergency, VMail, Local, LD, Intl)
- Define “simple” COR lists (with only one tag as a member) to be used as “partitions”
- Assign the “partitions” as outgoing COR lists to the appropriate POTS dial peers
- Define COR lists to be used as “CSS” (containing a subset of the tags as members)
- Assign the “CSS” as incoming COR lists to the different phone numbers under the SRST commands

Classes of Service for SRST (COR)
COR: Cisco IOS Configuration Basics

STEP 1

```
dial-peer cor custom
  name A
  name B
  name C
  name D
```

Define "Tags" for COR List Members

STEP 2

```
dial-peer cor list c1
  member A
  member B
  member C

dial-peer cor list c2
  member A
  member B

dial-peer cor list c3
  member A
  member B
  member D
```

Create COR Lists with Various Combinations of Tags

STEP 3

```
dial-peer voice 1 voip
  corlist incoming c1
  session target ipv4:1.1.1.1
  dtmf-relay h245-alpha

call-manager-fallback
  cor incoming c2 default
  cor incoming c3 1 2001
  cor incoming c3 2 2004-2007

dial-peer voice 2 pots
  corlist outgoing c3
  destination-pattern 1..
  port 1/0:23
```

Associate Incoming and Outgoing COR Lists with Voip/Pots Dial-Peers and Cisco CallManager-Fallback

Classes of Service for SRST (COR)
SRST COR Limitations

- Maximum number of “cor incoming” statements under call-manager-fallback is 5 (plus default) in SRST 2.1 (Cisco IOS 12.2(13)T14)
- Maximum number of “cor incoming” statements under call-manager-fallback is 20 (plus default) in SRST 3.0 (Cisco IOS 12.2(15)ZJ3)
- If “manager” phone DN’s are not consecutive and the SRST site is relatively large, this may become an obstacle to establishing appropriate classes of service
- If a device/DN is has NO corlist assignment, it is essentially unrestricted

Appendix

- Classes of Service for SRST (COR)
- CallManager best match logic
- Voice over PSTN
- Tail End Hop Off
- VLOD information
- Trunks

Cisco CallManager Call Routing Logic
Example (1)

Configured Route Patterns

User's Dial String:

Cisco CallManager Actions:

- 1111
- 1211
- 1[23]XX
- 131
- 1[0-4]XX
- 13!

Cisco CallManager Call Routing Logic
Example (2)

Configured Route Patterns

User's Dial String:

<Off Hook>

Cisco CallManager Actions:

Provide Dial Tone
Wait

1111	Might Match
1211	Might Match
1[23]XX	Might Match
131	Might Match
1[0-4]XX	Might Match
13!	Might Match

```
Digit analysis: match(fqcn="9195555644", cn="15644",
    pss="PA:Line1:Cisco:Local:Long Distance:International", dd=”")
Digit analysis: potentialMatches=PotentialMatchesExist
```

Cisco CallManager Call Routing Logic
Example (3)

Configured Route Patterns

1111	Might Match
1211	Might Match
1[23]XX	Might Match
131	Might Match
1[0-4]XX	Might Match
13!	Might Match

User’s Dial String:

1

Cisco CallManager Actions:

Break Dial Tone
Wait

```
Digit analysis: match(fqcn="9195555644", cn="15644",
    pss="PA:Line1:Cisco:Local:Long Distance:International", dd=”1")
Digit analysis: potentialMatches=PotentialMatchesExist
```

Cisco CallManager Call Routing Logic
Example (4)

Configured Route Patterns

1111	Doesn’t Match
1211	Doesn’t Match
1[23]XX	Might Match
131	Might Match
1[0-4]XX	Might Match
13!	Might Match

User’s Dial String:

13

Cisco CallManager Actions:

Wait

```
Digit analysis: match(fqcn="9195555644", cn="15644",
    pss="PA:Line1:Cisco:Local:Long Distance:International", dd=”13")
Digit analysis: potentialMatches=PotentialMatchesExist
```

Cisco CallManager Call Routing Logic
Example (5)

User's Dial String:

131

Cisco CallManager Actions:

Keep Waiting; More Digits Might Cause a Different Pattern to Match

Configured Route Patterns

1111	Doesn't Match
1211	Doesn't Match
1[23]XX	Might Match
131	Match!
1[0-4]XX	Might Match
13!	Match! and Might Match

```
Digit analysis: match(fqcn="9195555644", cn="15644",
    pss="PA:Line1:Cisco:Local:Long Distance:International", dd=”131")
Digit analysis: potentialMatches=PotentialMatchesExist
```

Cisco CallManager Call Routing Logic
Example (6)

Configured Route Patterns

1111	Doesn’t Match
1211	Doesn’t Match
1[23]XX	Match!
131	Doesn’t Match
1[0-4]XX	Match!
13!	Match! and Might Match

User’s Dial String:

1311

Cisco CallManager Actions:

Keep Waiting; More Digits Might Cause a Different Pattern to Match

```
Digit analysis: match(fqcn="9195555644", cn="15644",
    pss="PA:Line1:Cisco:Local:Long Distance:International", dd=”1311")
Digit analysis: potentialMatches=PotentialMatchesExist
```

Cisco CallManager Call Routing Logic Example (7)

User's Dial String:

1311<timeout>

Configured Route Patterns

1111	Doesn't Match
1211	Doesn't Match
1[23]XX	Match!
131	Doesn't Match
1[0-4]XX	Match!
13!	Match!

Cisco CallManager Actions:

Extend Call to the Best Match

Can You Tell Which Route Pattern Is the Best Match in This Case?

Hint: We Are Being Crafty to Make Sure You Remember Forever ☺

Cisco CallManager Call Routing Logic
Example (8)

User's Dial String:

1311<Timeout>

Matches 200 Digit Strings

Matches 500 Digit Strings

Matches ∞ Digit Strings, However for the Purposes of Closest Match Routing in This Case, This Matches 100 Digit Strings Because You Only Consider the Number of Potential Strings Given the Number of Digits Dialed

Configured Route Patterns

1111	Doesn't Match
1211	Doesn't Match
1[23]XX	Match!
131	Doesn't Match
1[0-4]XX	Match!
13!	Match!

Partitions and Calling Search Spaces
Analogy

Rita Wants to Call Dave

To Do So, She Needs to Know Dave's Number

Miami Yellow Pages	
Dave	305 555 5000

Dave Lists His Number in a Directory

Rita

Dave

305 555 5000

Partitions and Calling Search Spaces
Analogy

To Look up Numbers, Rita Looks Through the Directories She Owns

If She Doesn't Have the Right Directory…

Rita's List of Directories

Dallas White Pages
Outlook Address Book
Little Black Book

…She Can't Place the Call

Miami Yellow Pages	
Dave	305 555 5000

Rita

Dave

305 555 5000

Partitions and Calling Search Spaces
Analogy

But If She Has the Directory Dave Has Listed His Number in…

Miami Yellow Pages	
Dave	305 555 5000

Rita's List of Directories

Rita's List of Directories
Dallas White Pages
Miami Yellow Pages
Little Black Book

Rita

…the Call Will Go Through

Dave

305 555 5000

Partitions and Calling Search Spaces
Analogy

The Directory in Which Dave's Number Is Listed Is His Number's **Partition**

Miami Yellow Pages	
Dave	305 555 5000

Rita's List of Directories

Dallas White Pages
Miami Yellow Pages
Little Black Book

The List of Directories in Which Rita Looks up Numbers Is Her **Calling Search Space**

Rita

Dave

305 555 5000

Appendix

- Classes of Service for SRST (COR)
- CallManager best match logic
- Voice over PSTN
- Tail End Hop Off
- VLOD information
- Trunks

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.
- We do see that it could serve as a “beach head” to win over some customers
- There are several, fundamental limitations
- Relies on AAR configuration

VoPSTN Using AAR
Global Considerations

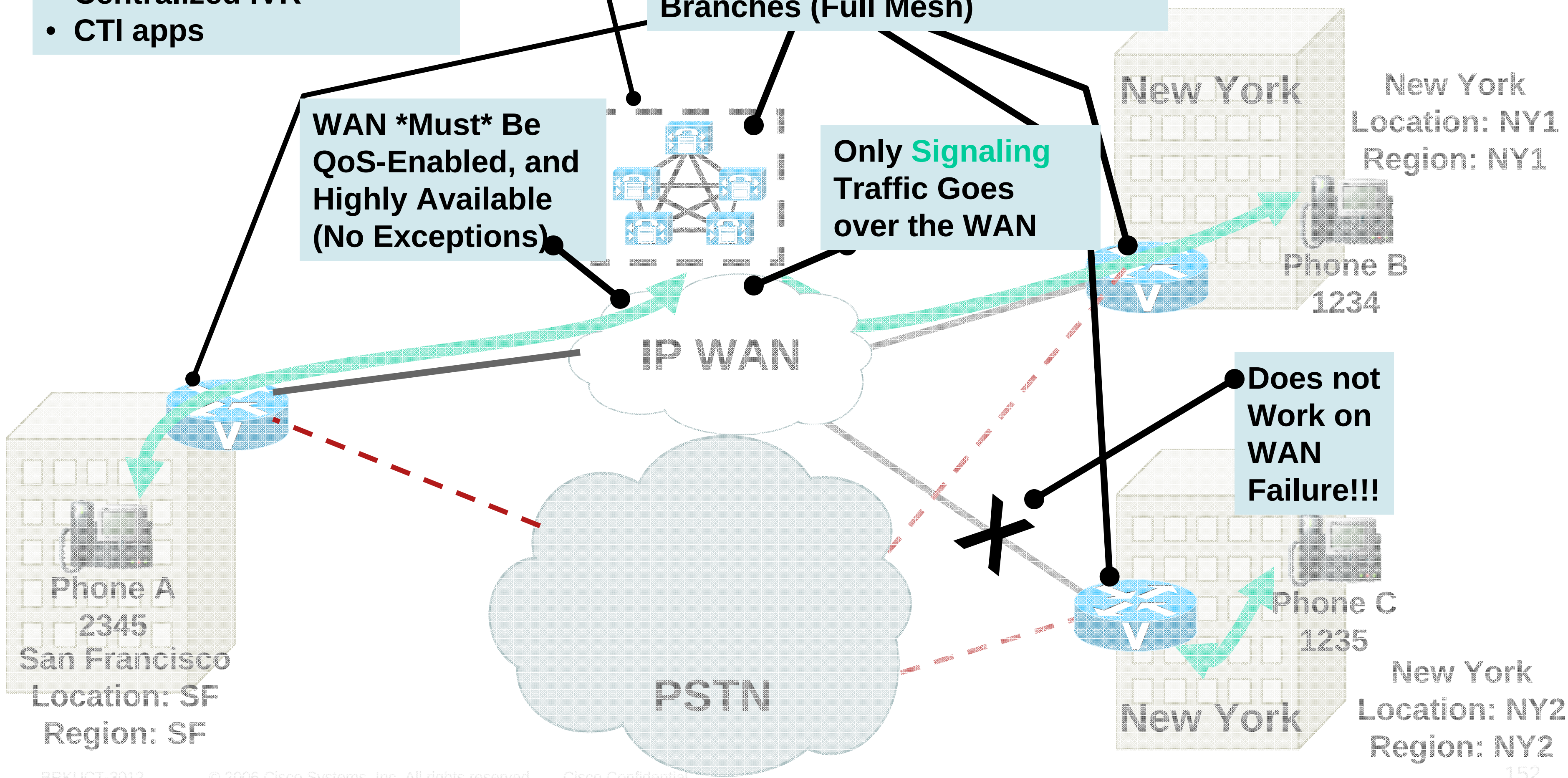
No Streaming of Audio to Central Site, Thus No:
- Centralized MoH
- Centralized conferencing
- Centralized IVR
- CTI apps

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

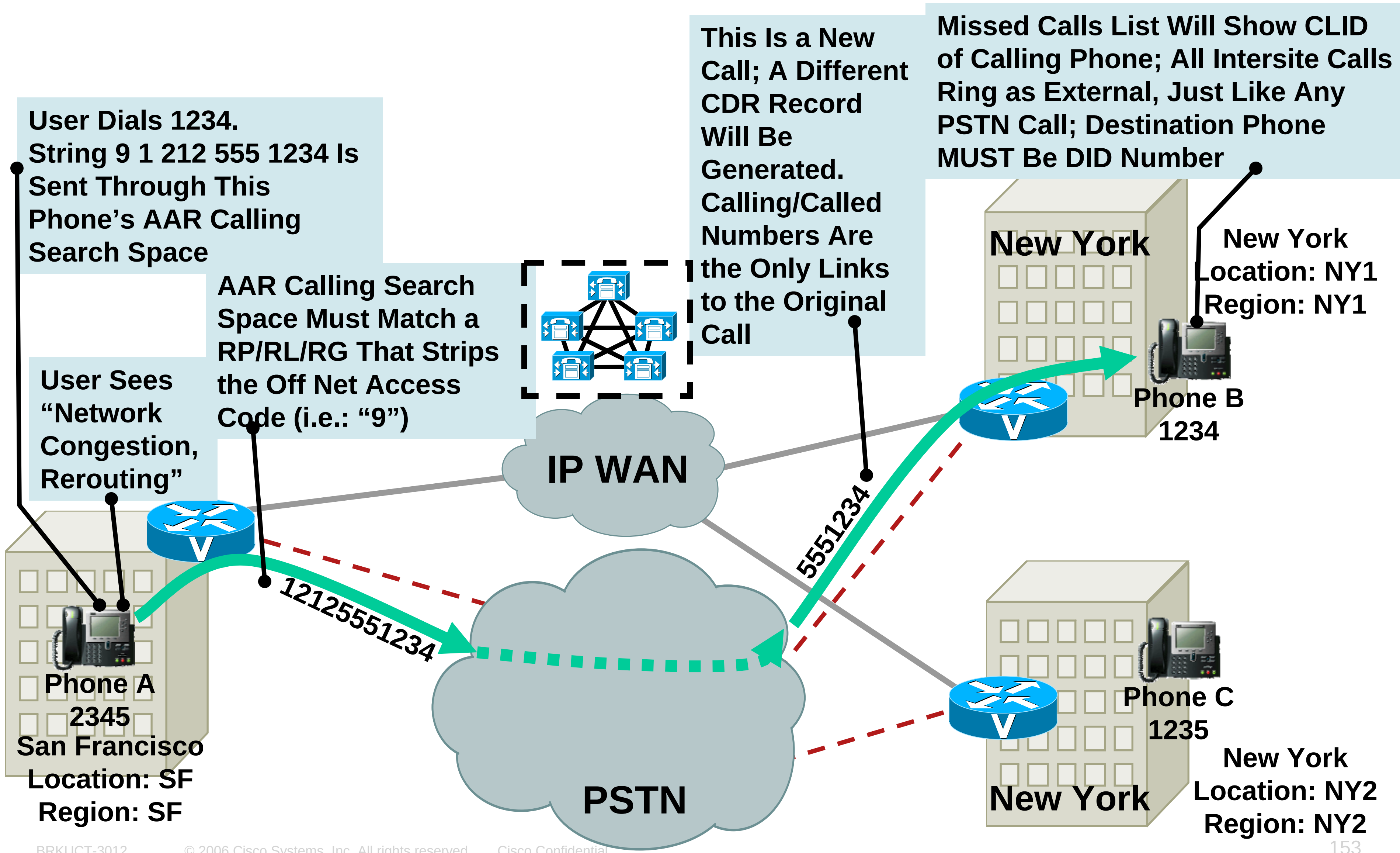
WAN *Must* Be QoS-Enabled, and Highly Available (No Exceptions)

Only Signaling Traffic Goes over the WAN

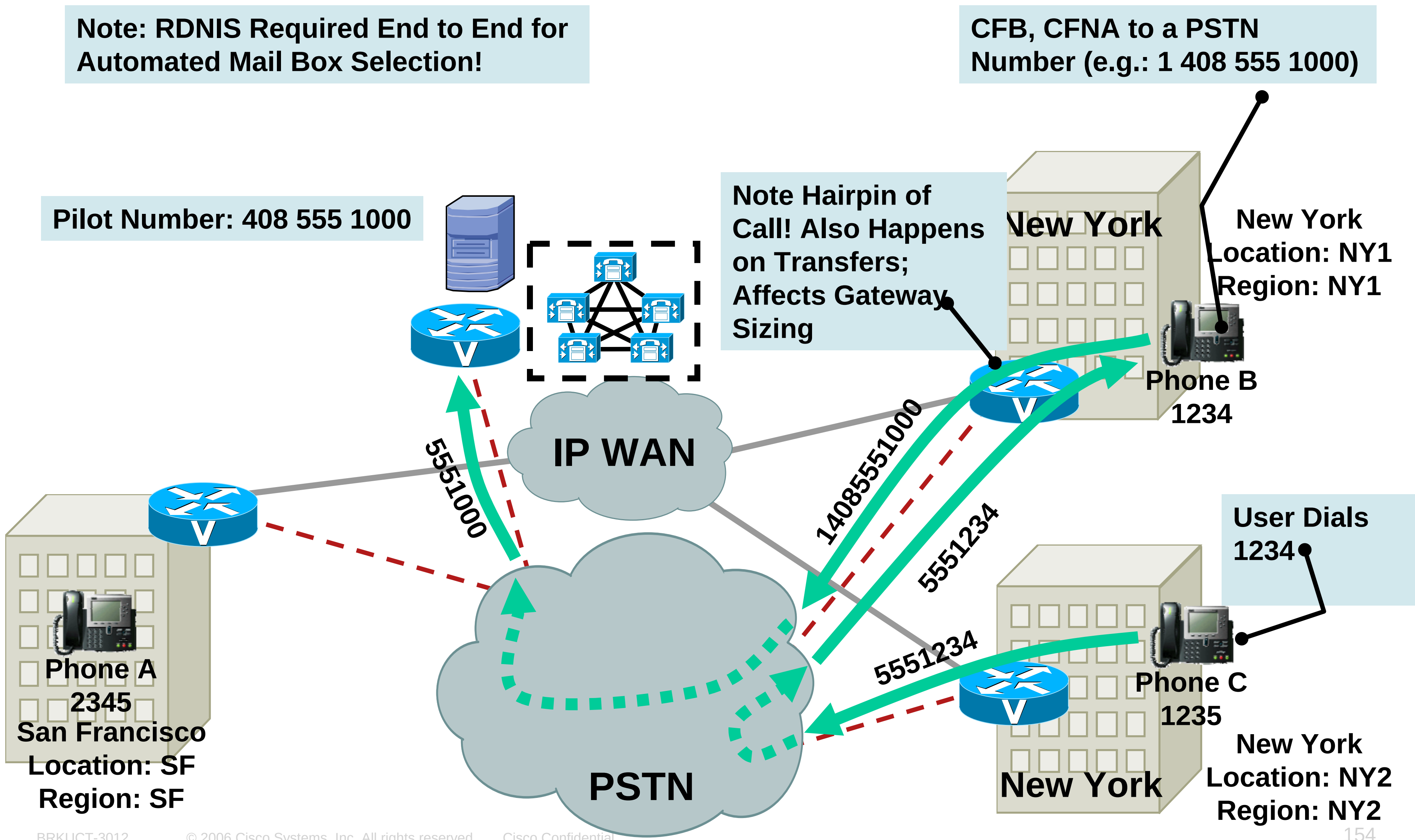
Does not Work on WAN Failure!!!



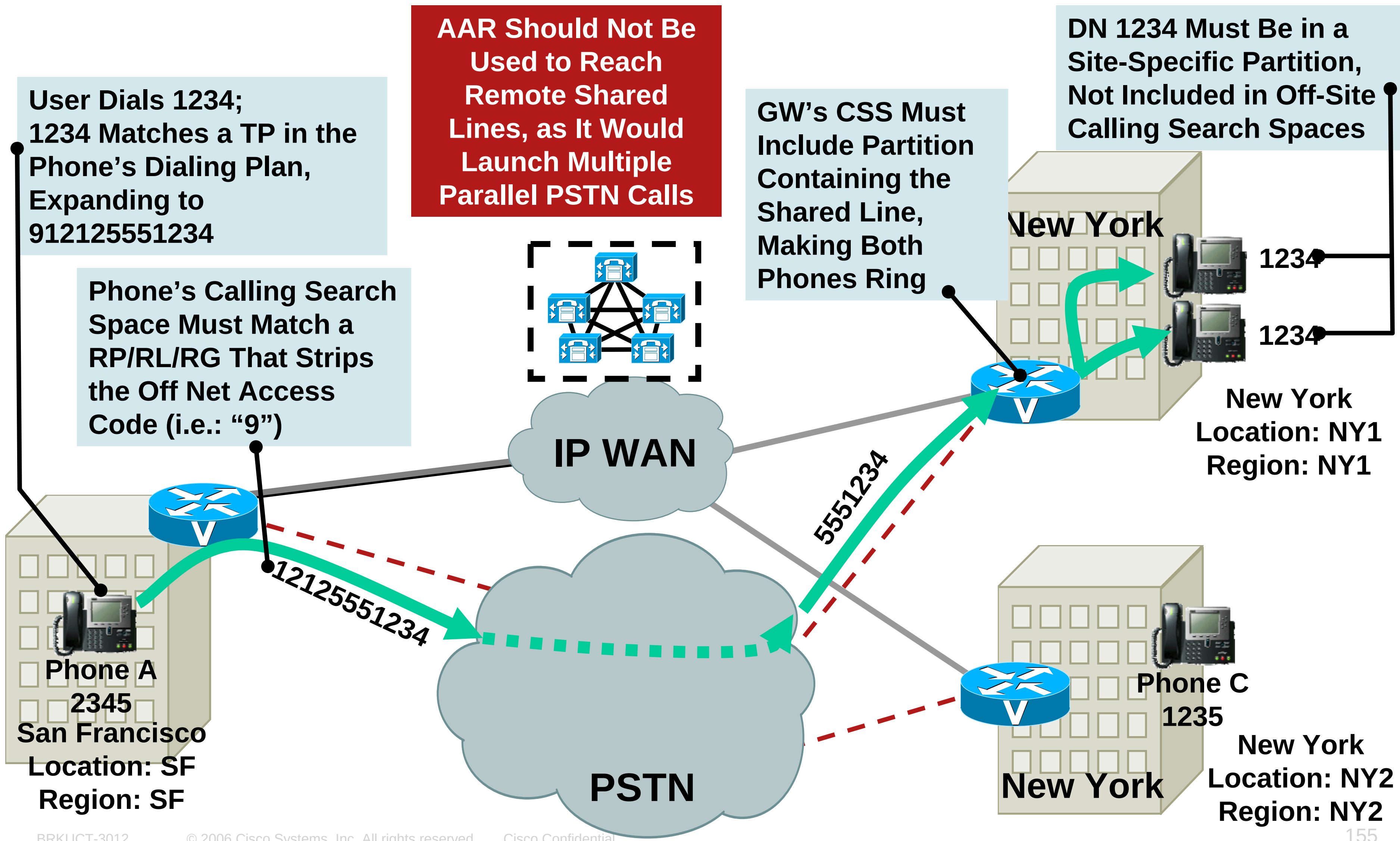
VoPSTN Using AAR
Intersite Calls



VoPSTN Using AAR
Non-Unity™ Centralized Voicemail



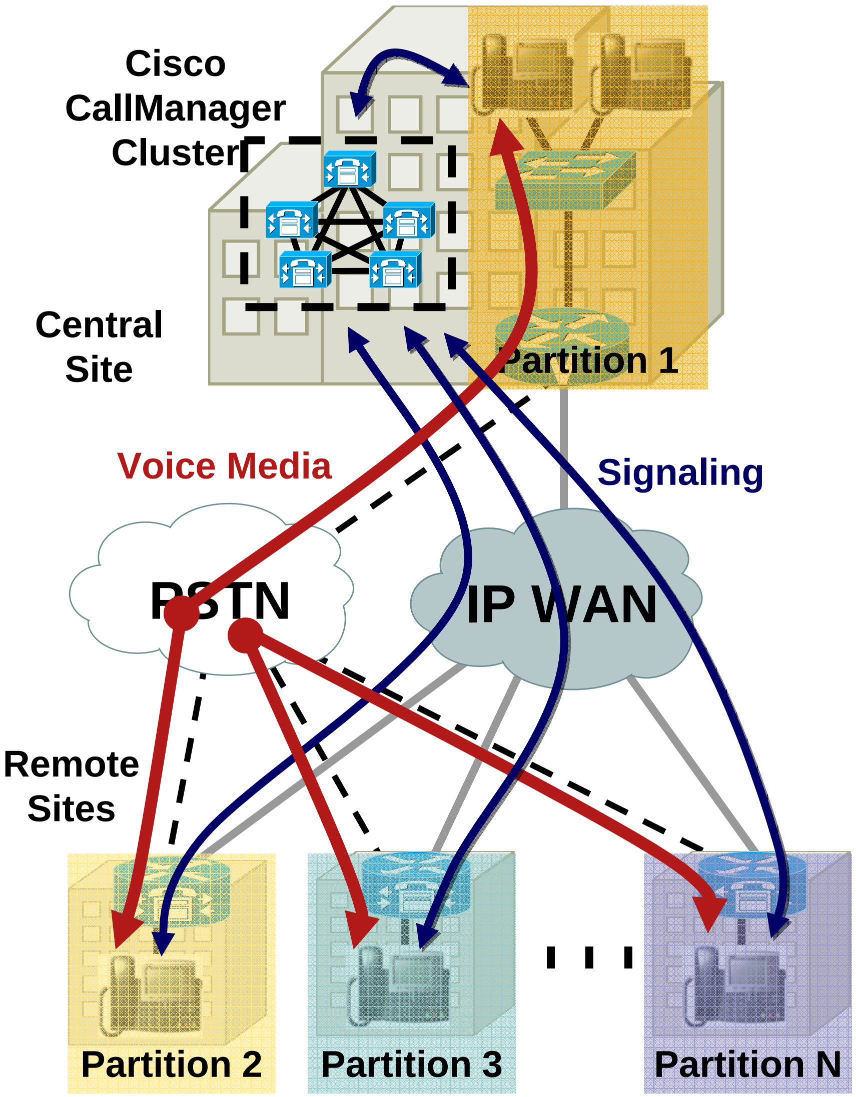
VoPSTN Using AAR
Shared Lines Considerations



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

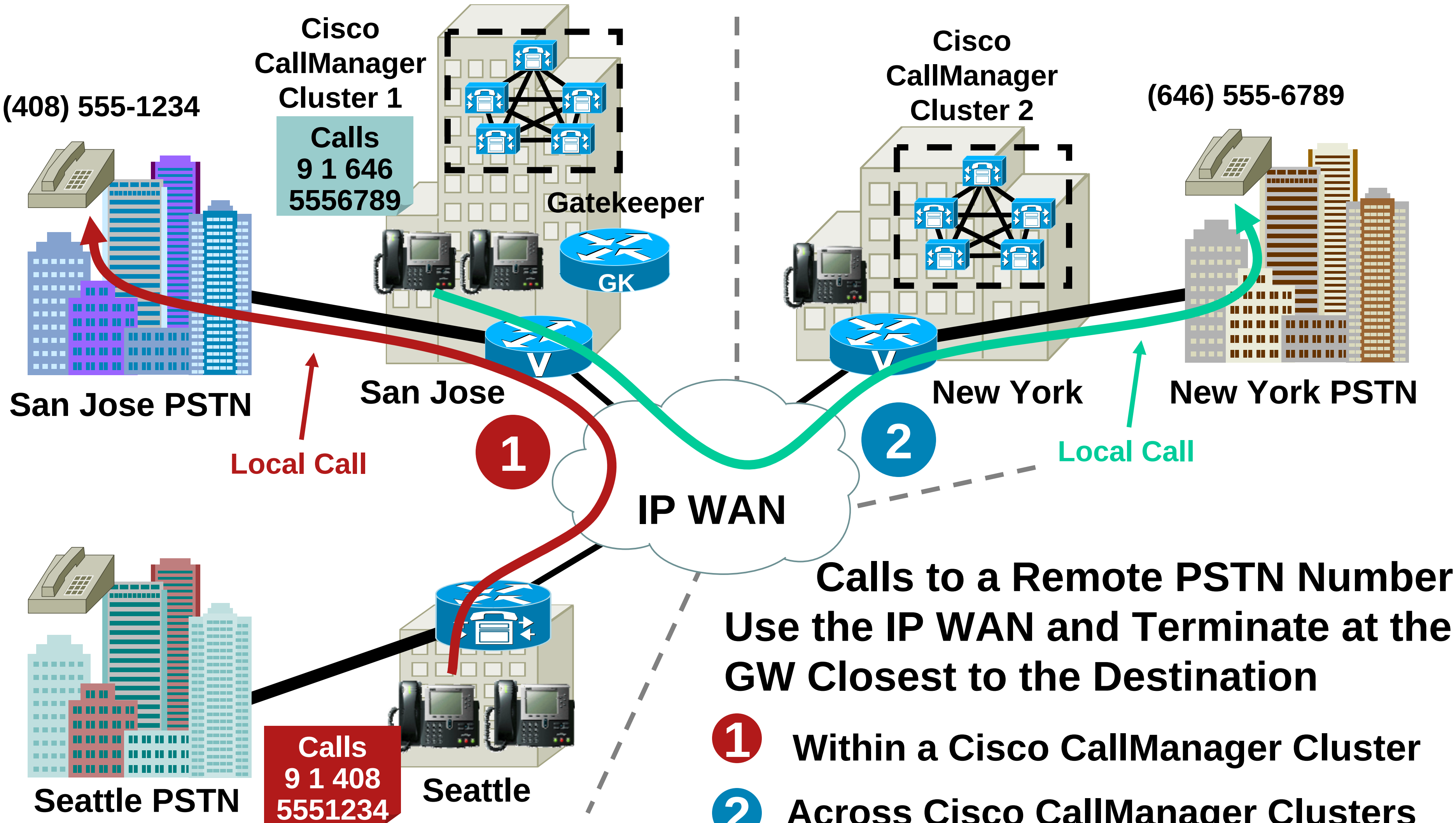


- DN’s at each site are placed in different partitions
- Relies on PSTN route patterns to call other sites
- For Cisco CallManager, all calls are external calls
- No “on-net” features across sites (e.g.: CallBack)
- No easy migration to fullblown VoIP
- NOTE: Abbreviated dialing possible with translation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

Appendix

- Classes of Service for SRST (COR)
- CallManager best match logic
- Voice over PSTN
- Tail End Hop Off
- VLOD information
- Trunks

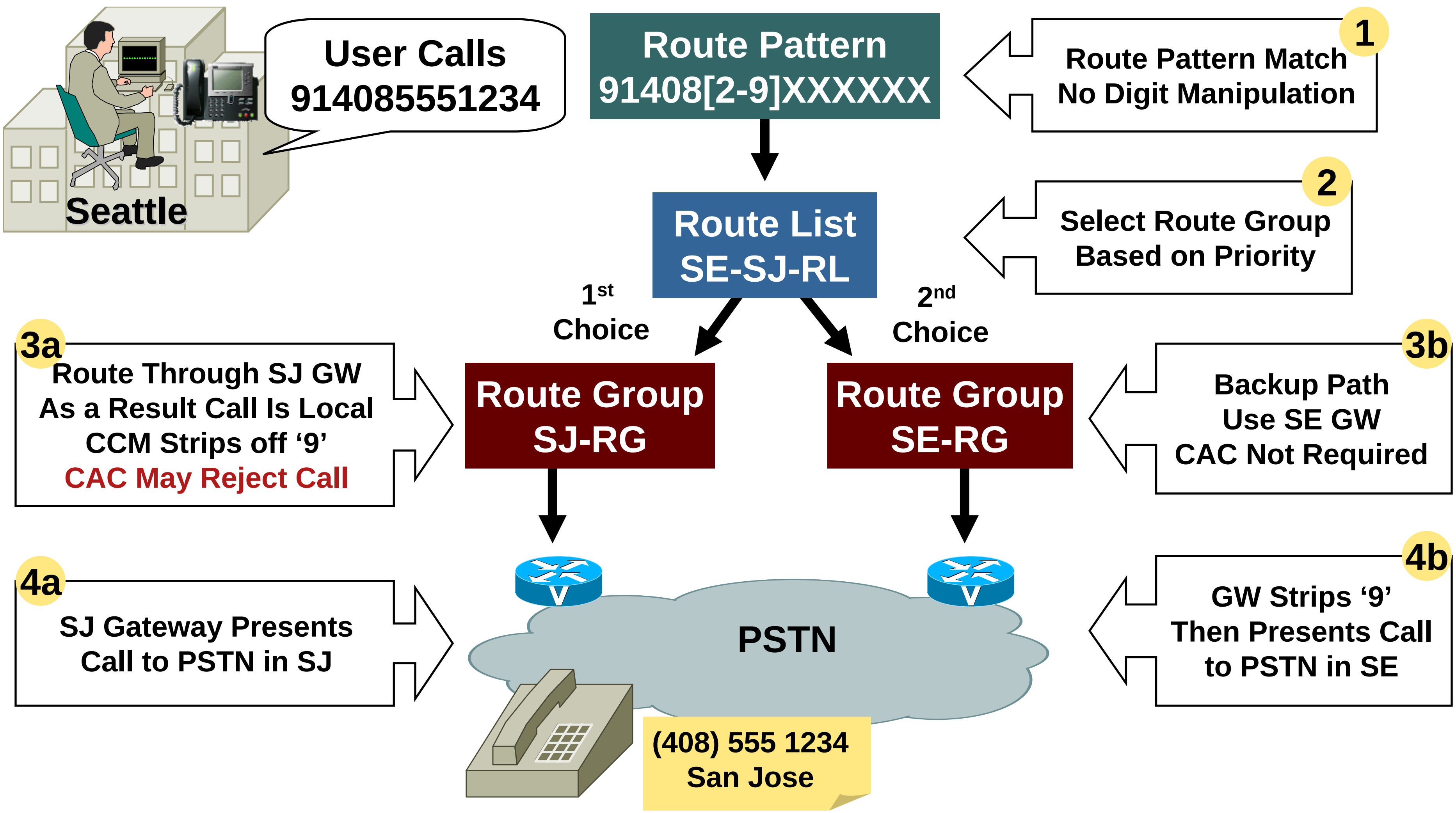
Tail-End Hop-Off (TEHO)
What Is It?



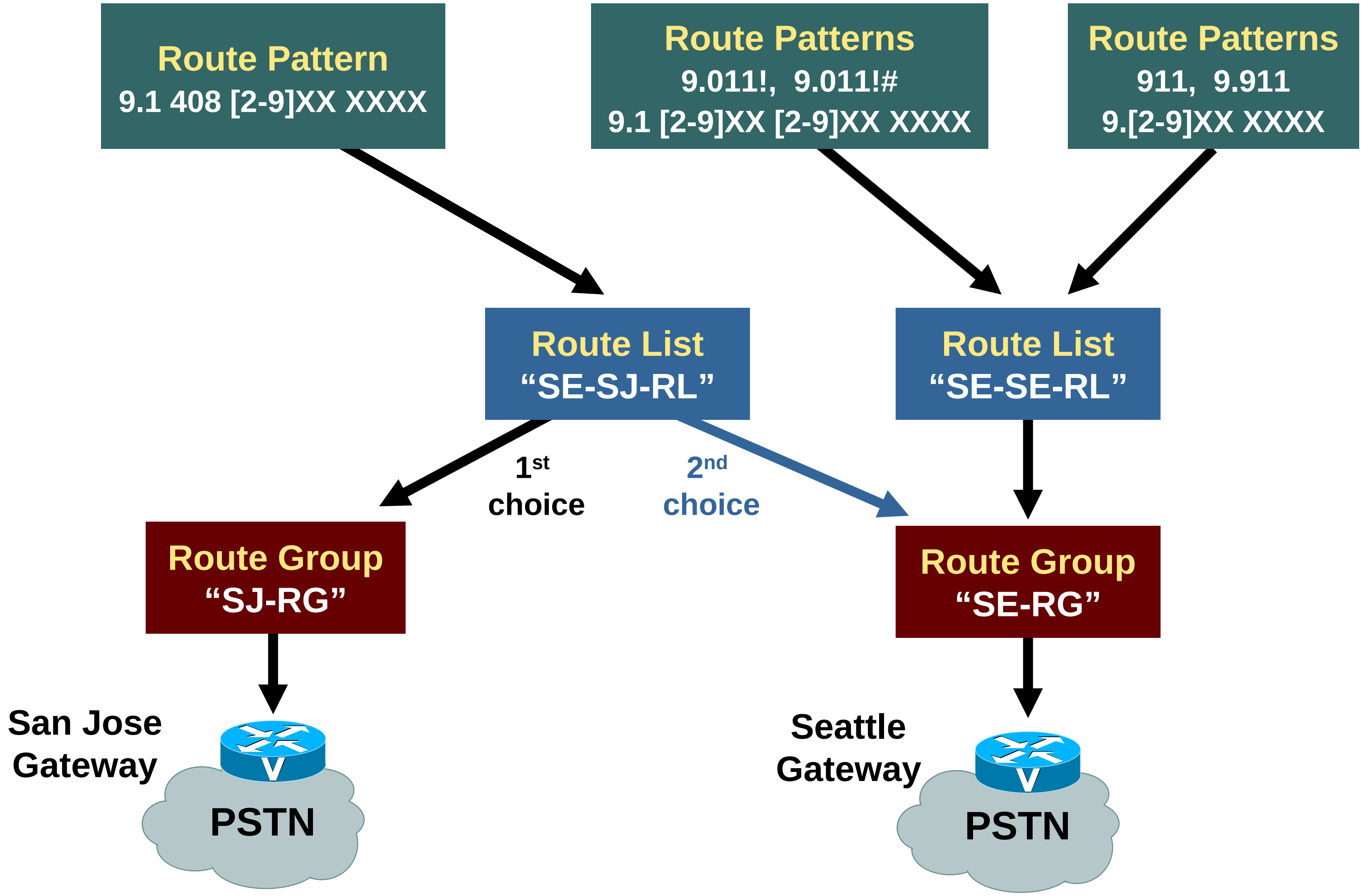
Calls to a Remote PSTN Number Use the IP WAN and Terminate at the GW Closest to the Destination

1. Within a Cisco CallManager Cluster
2. Across Cisco CallManager Clusters

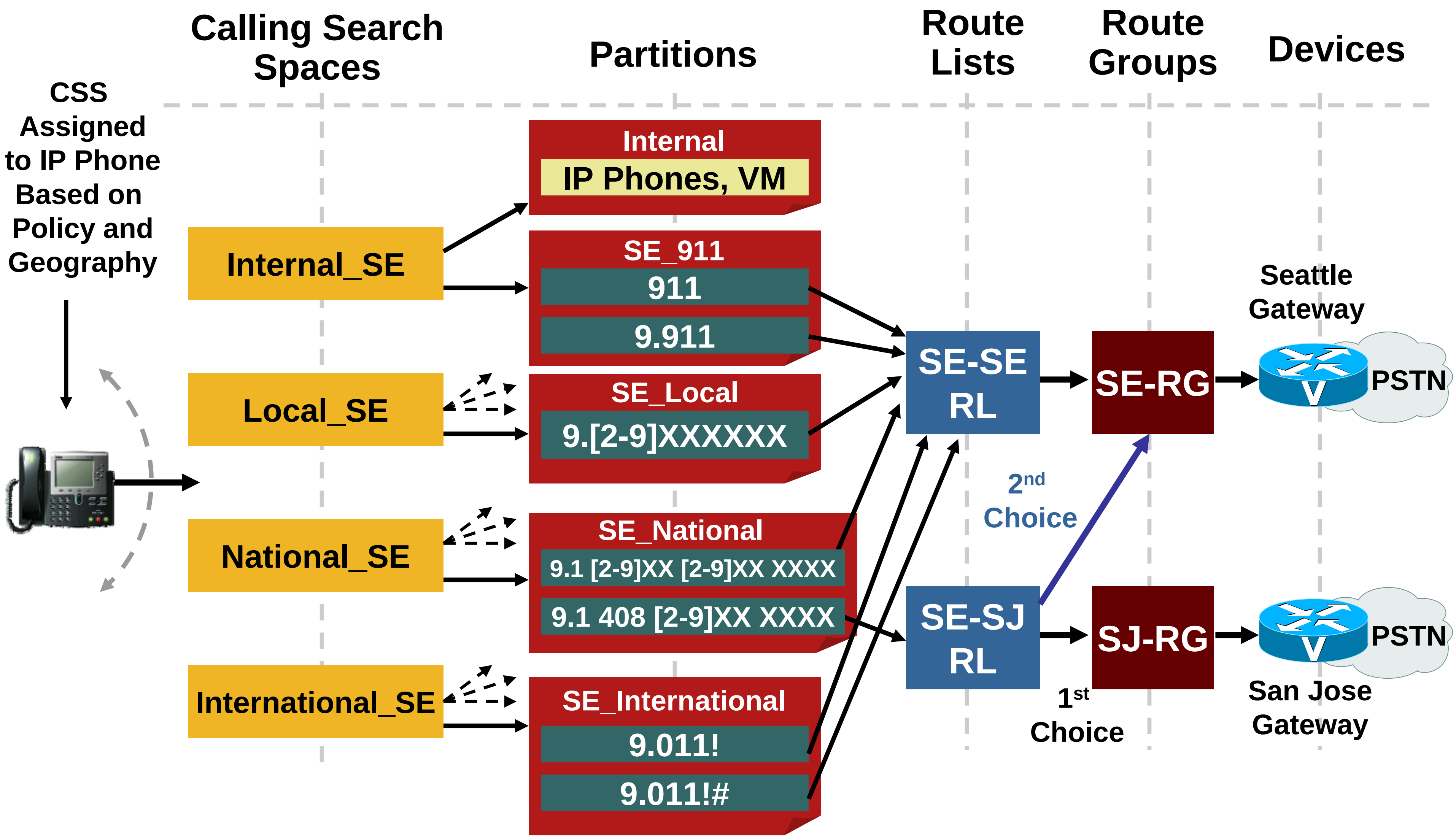
Tail-End Hop-Off (TEHO)
Intracluster: Seattle to San Jose



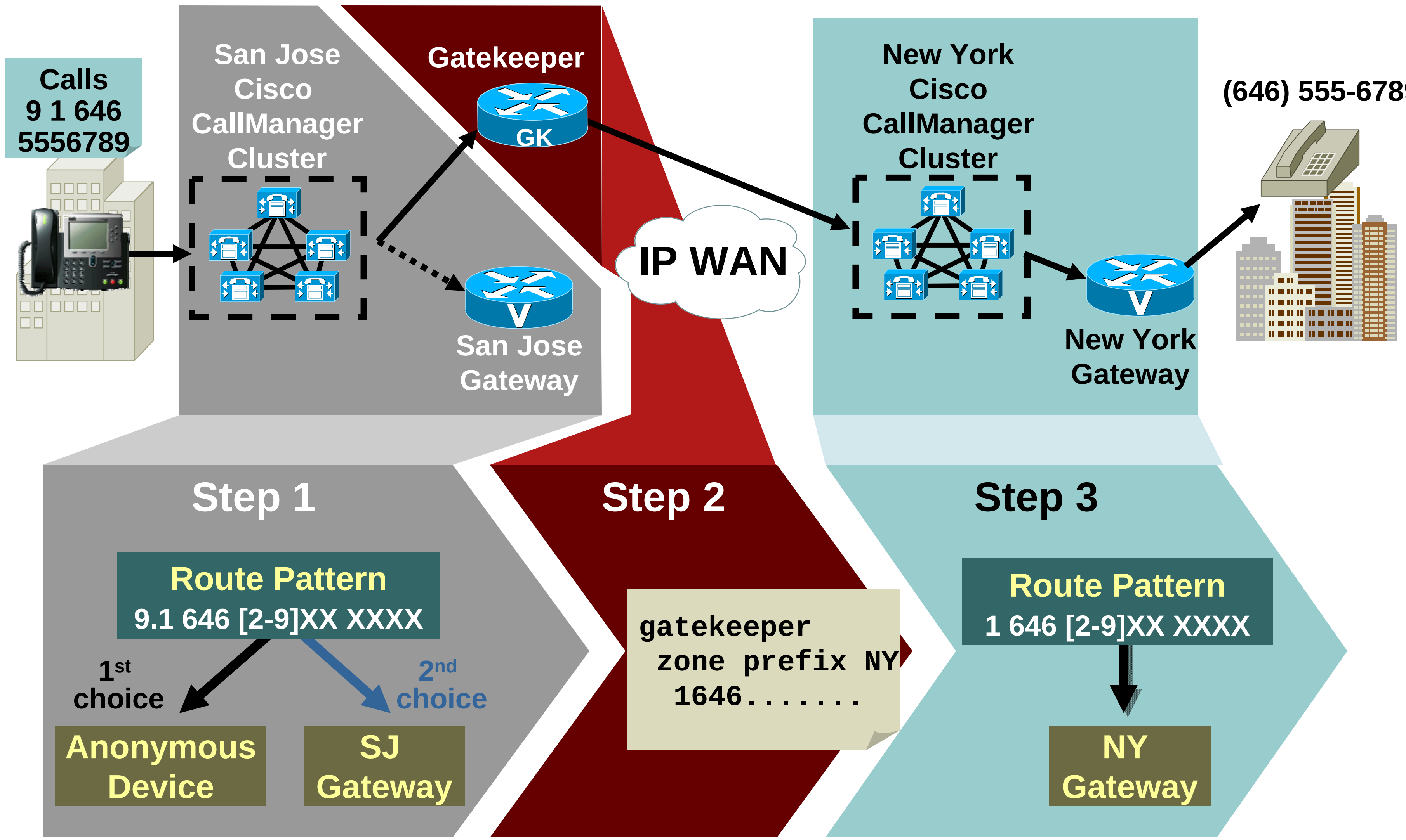
Tail-End Hop-Off (TEHO)
Intracluster: Route Patterns for Seattle



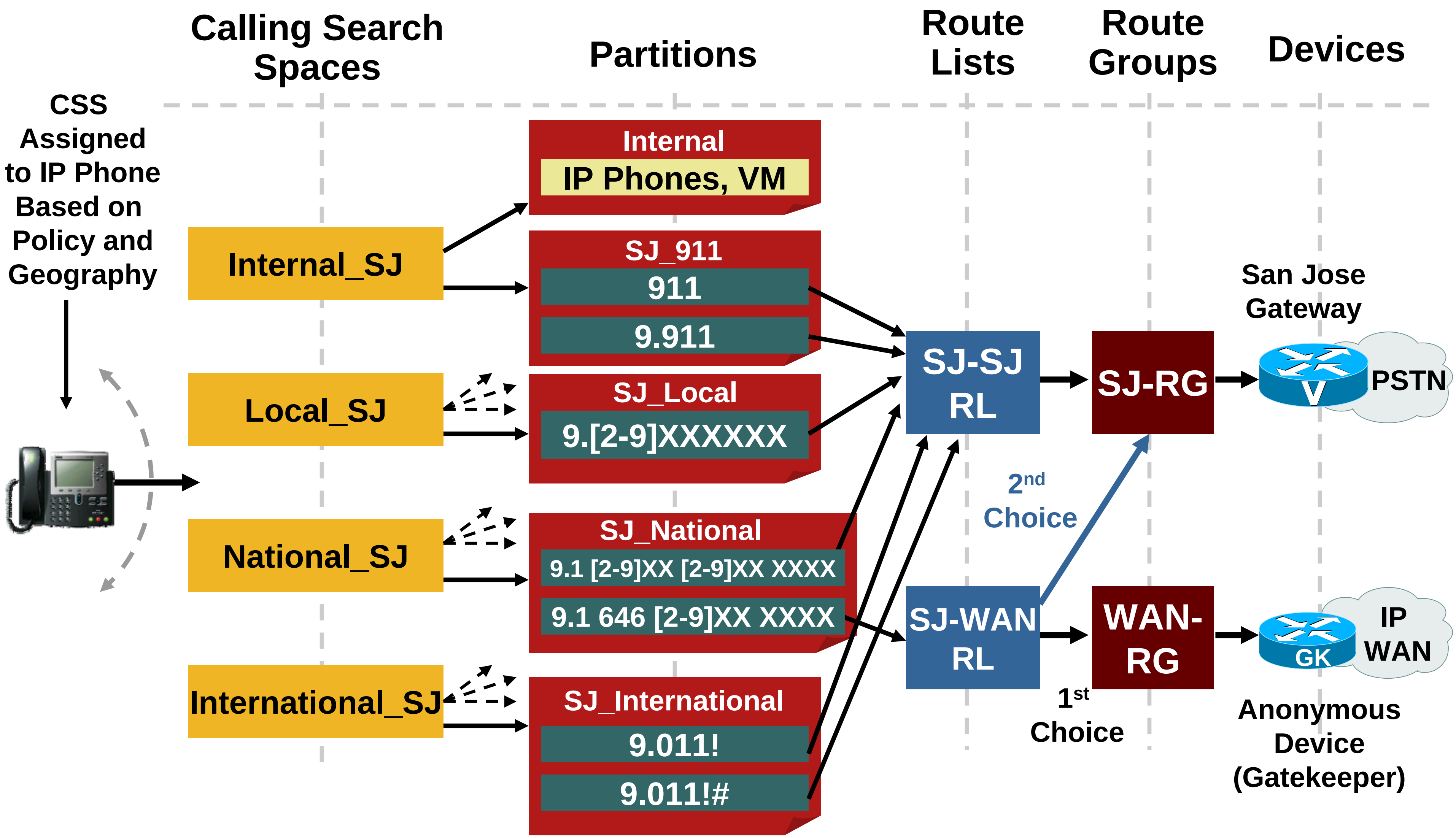
Tail-End Hop-Off (TEHO)
Intracluster: Composite Dial Plan for Seattle



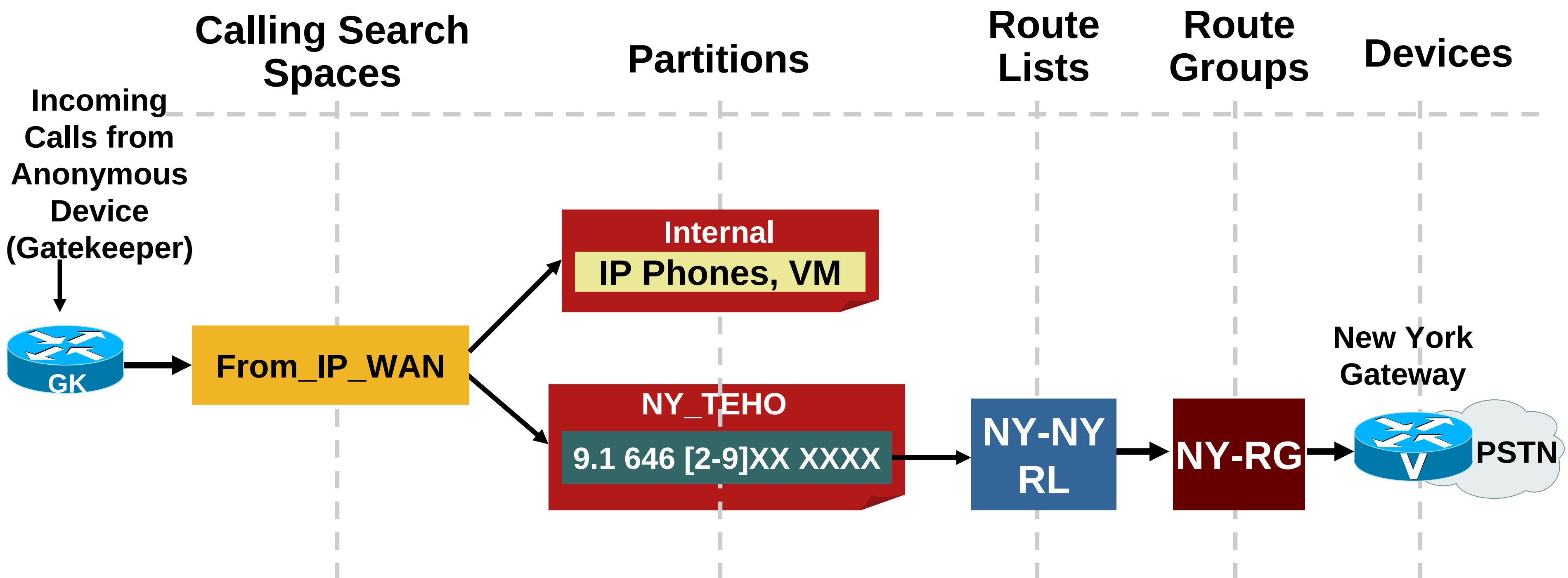
Tail-End Hop-Off (TEHO)
Intercluster: San Jose to New York



Tail-End Hop-Off (TEHO)
Intercluster: Composite Dial Plan for San Jose



Tail-End Hop-Off (TEHO)
Intercluster: Dial Plan for New York

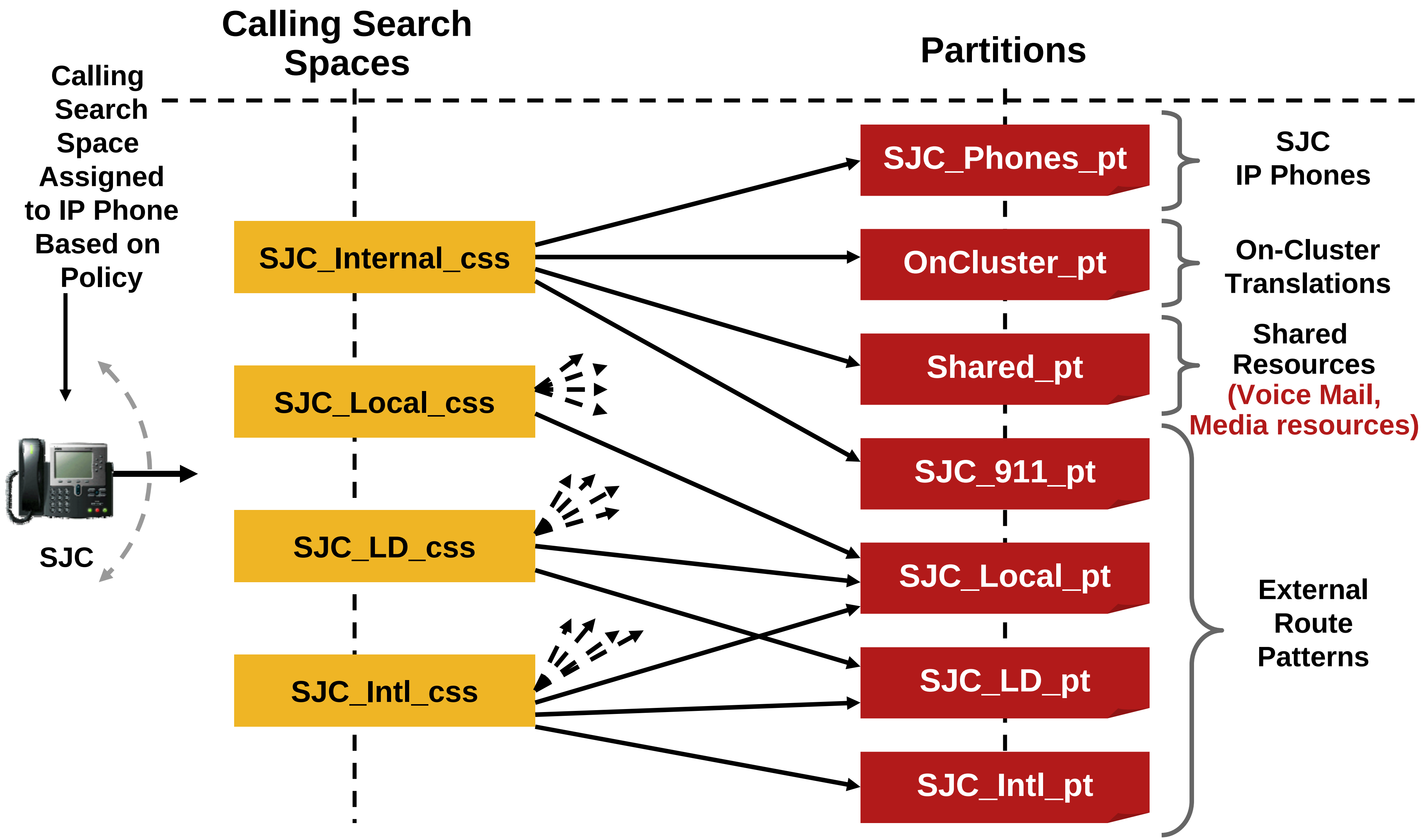


Note: To Avoid Routing Loops, Do Not Include Partitions That Contain IP WAN Routes in the "From_IP_WAN" Calling Search Space

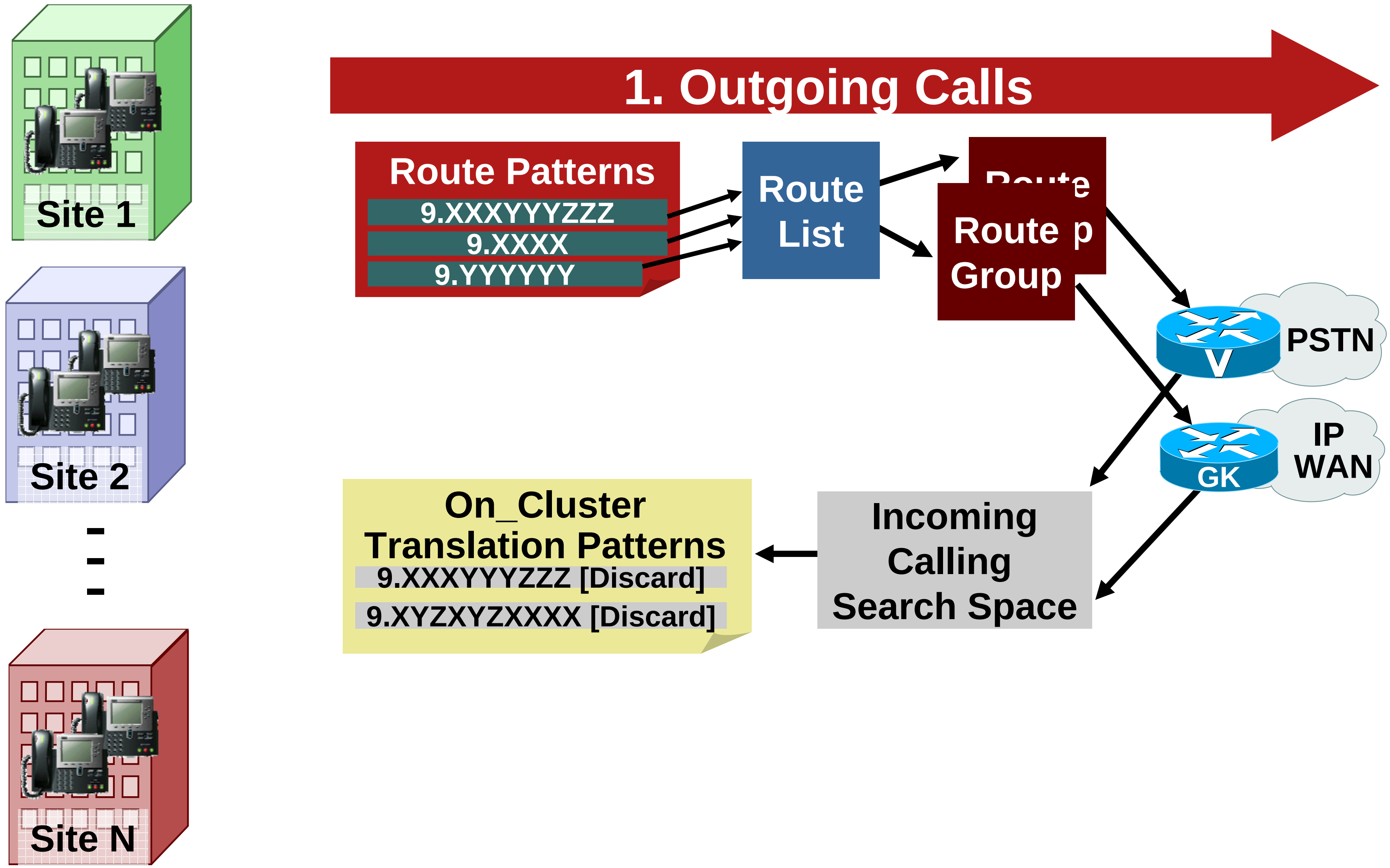
Appendix

- Classes of Service for SRST (COR)
- CallManager best match logic
- Voice over PSTN
- Tail End Hop Off
- VLOD information
- Trunks

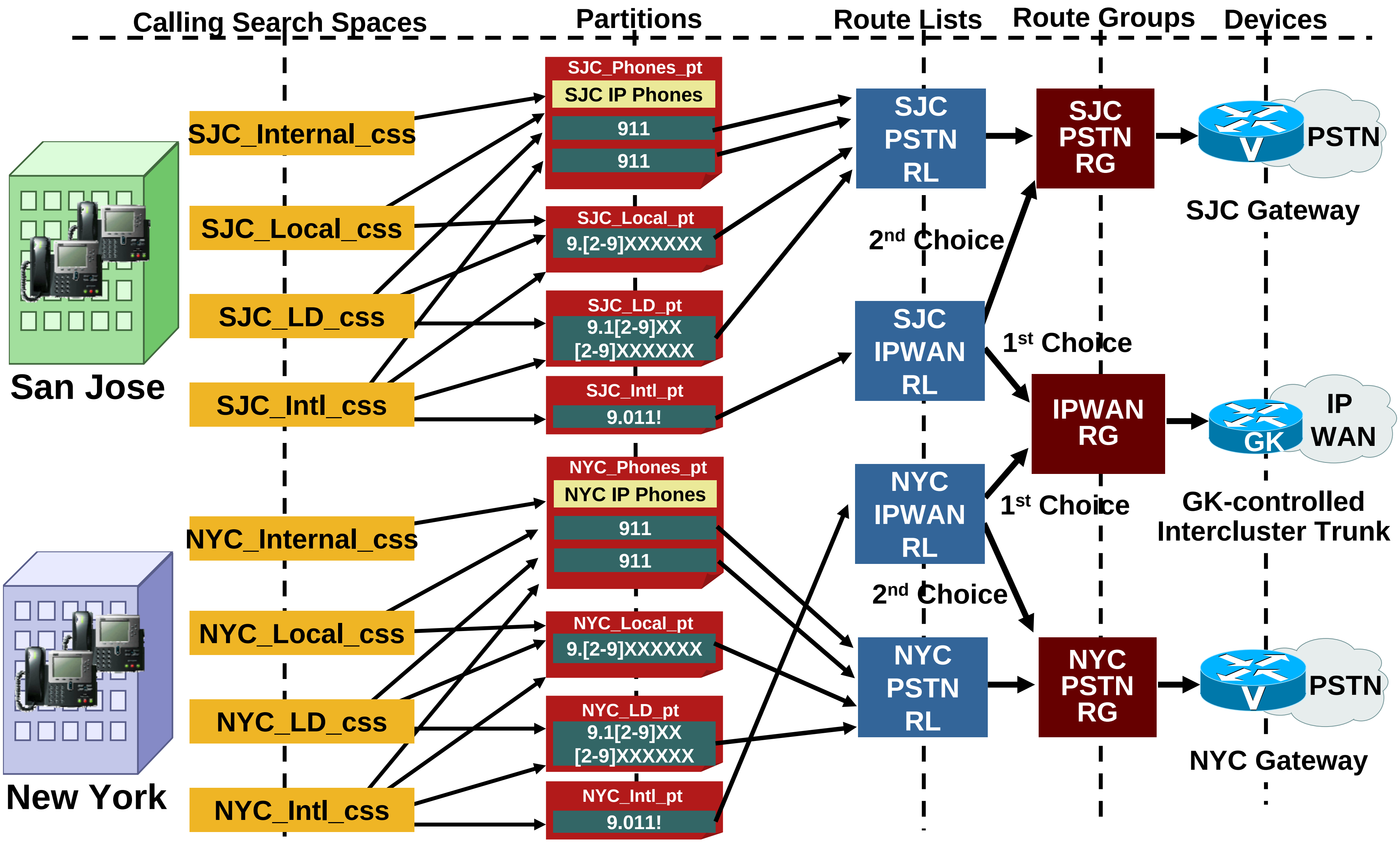
VLOD with Partitioned Addressing
View of Partitions/Calling Search Spaces



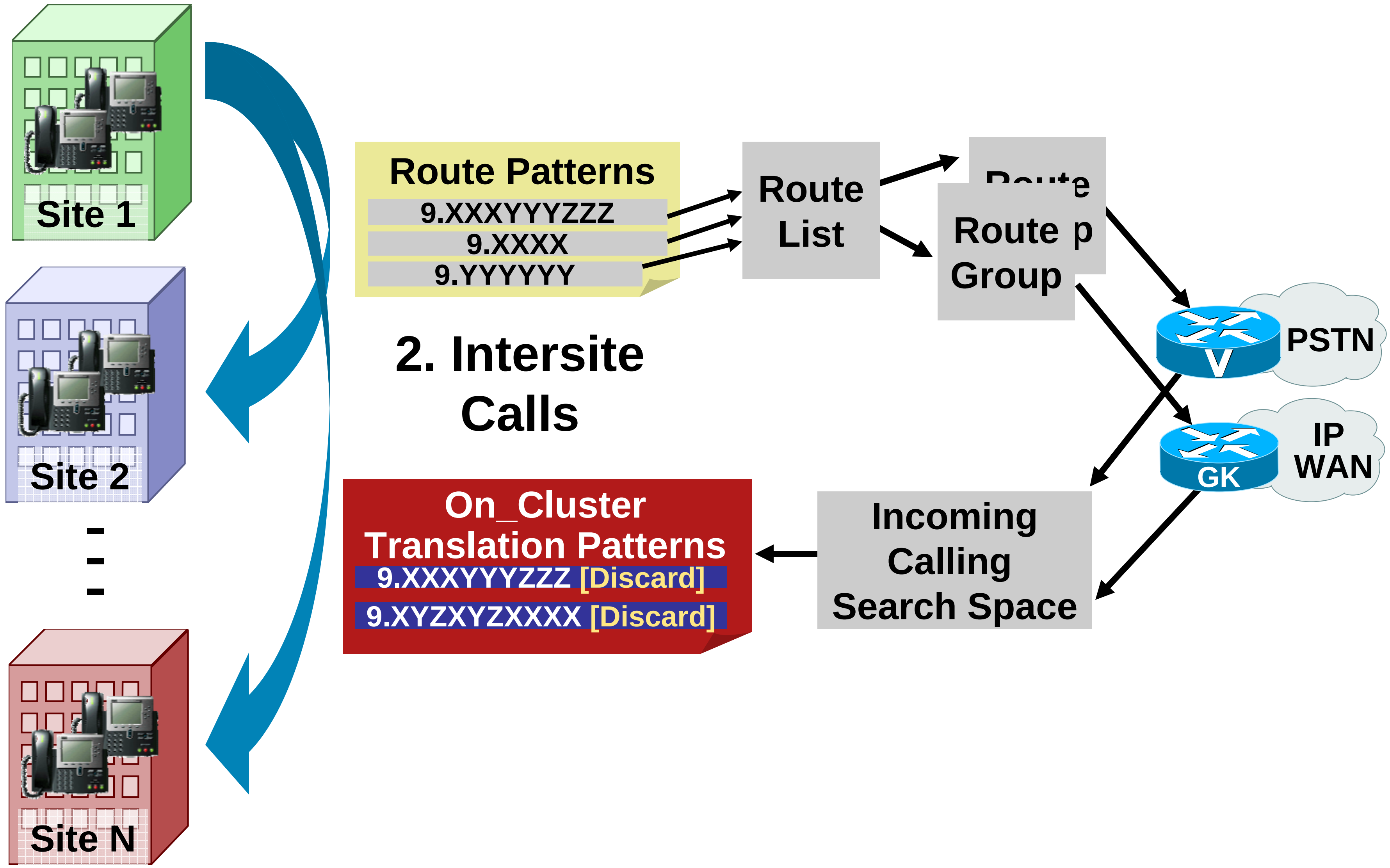
VLOD with Partitioned Addressing
Outgoing PSTN/Gatekeeper Calls



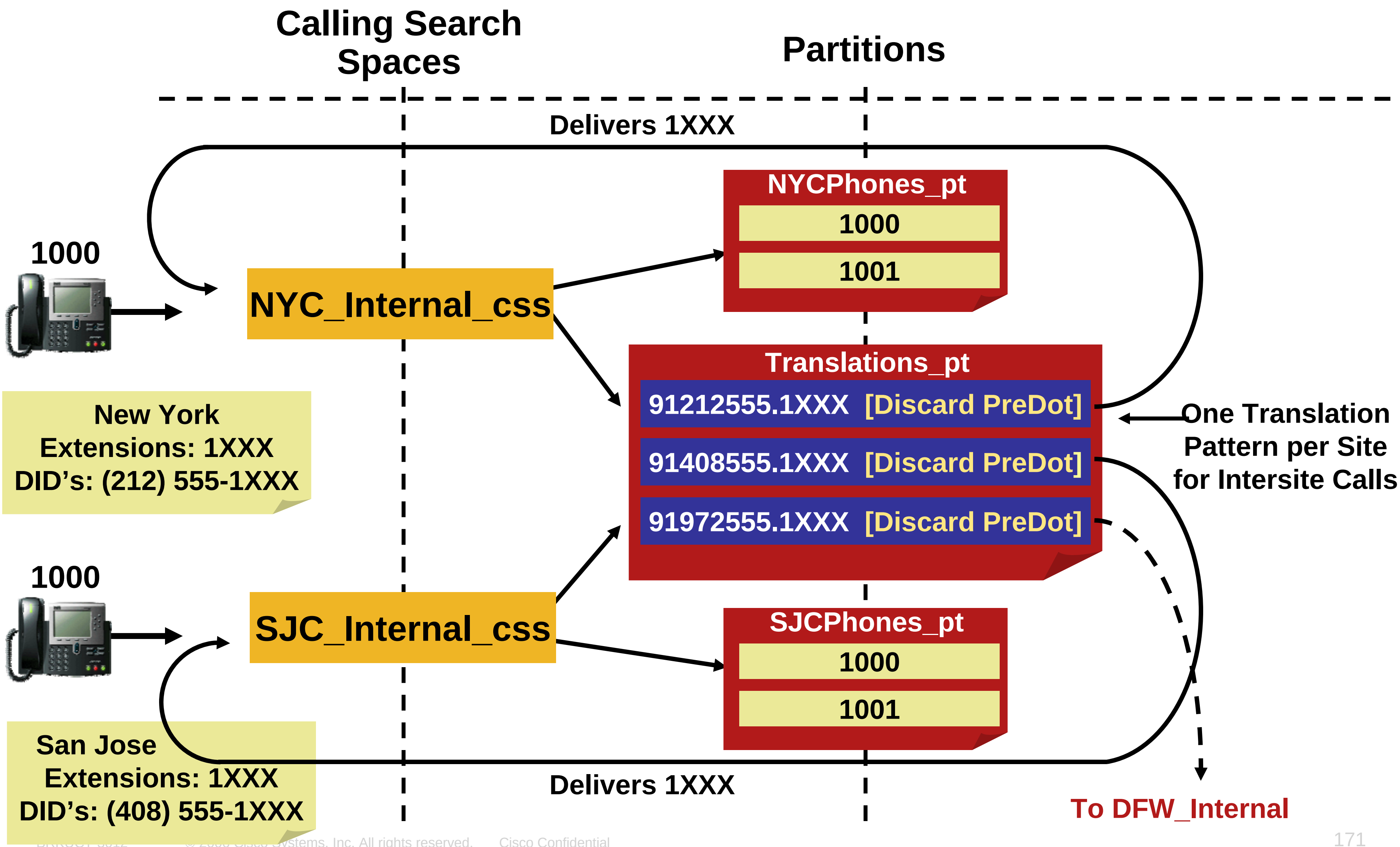
VLOD with Partitioned Addressing
Outgoing PSTN/Gatekeeper Calls



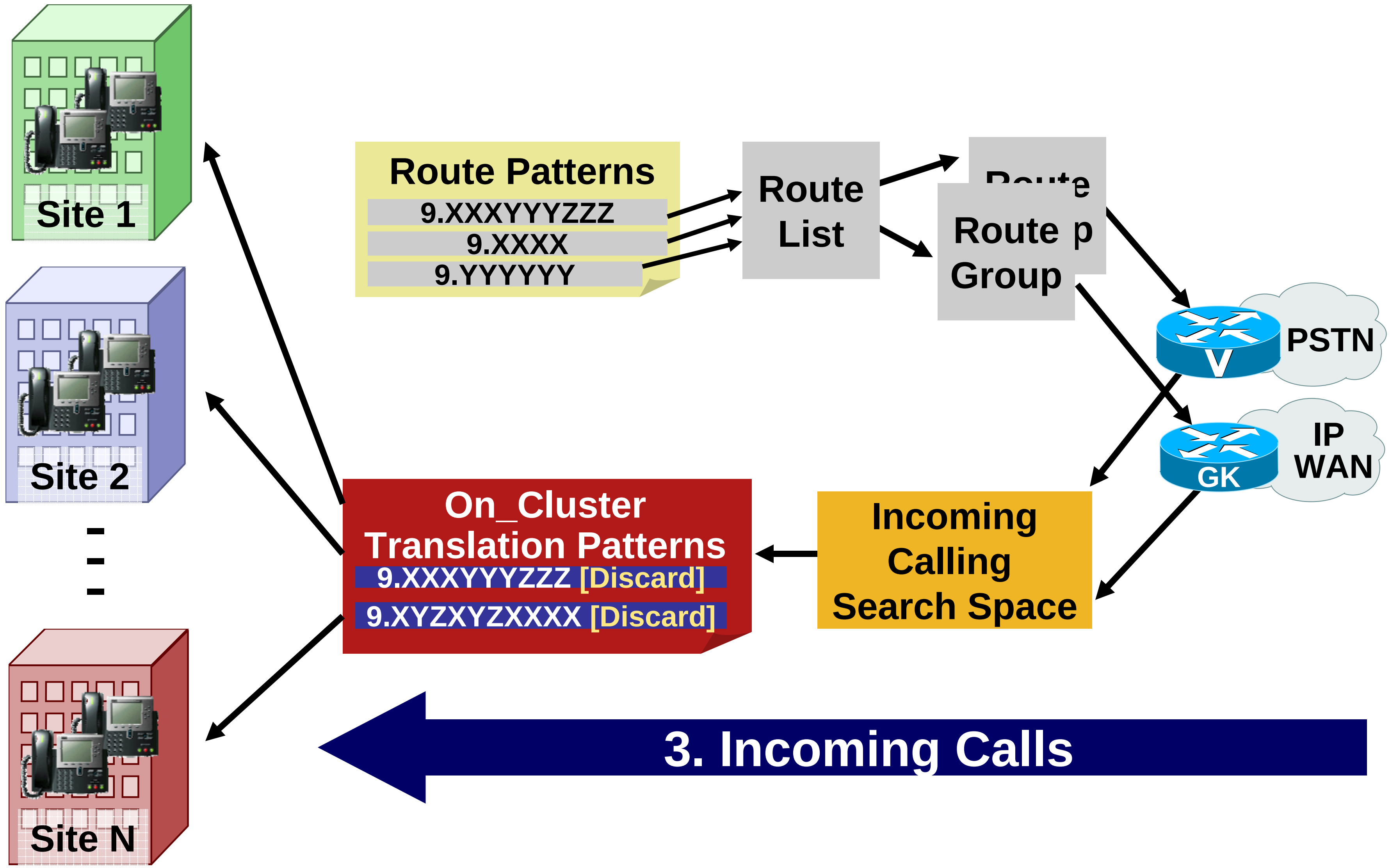
VLOD with Partitioned Addressing
Intersite Calls Within a Cluster



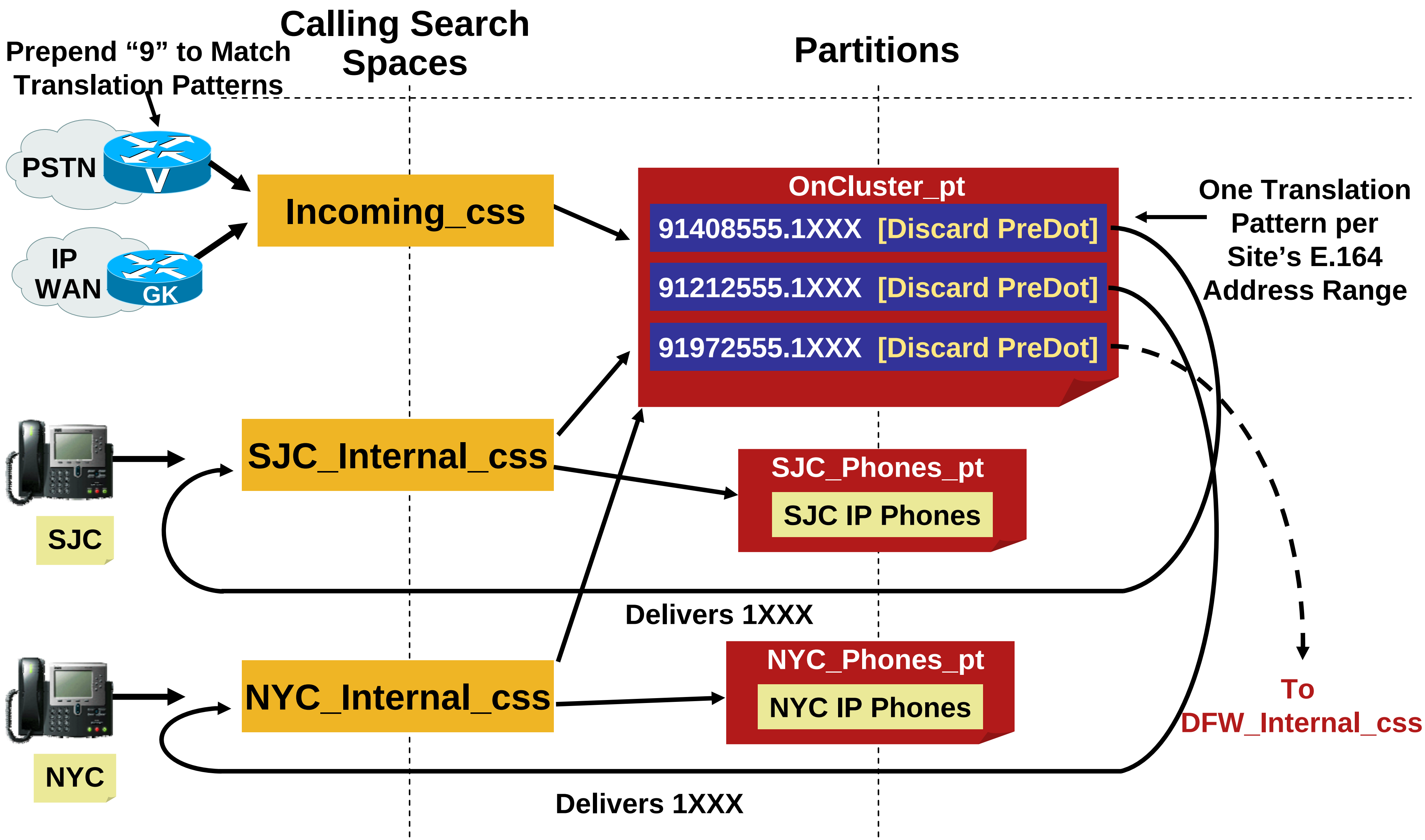
VLOD with Partitioned Addressing
Intersite Calls Within a Cluster



VLOD with Partitioned Addressing
Incoming PSTN/Gatekeeper Calls



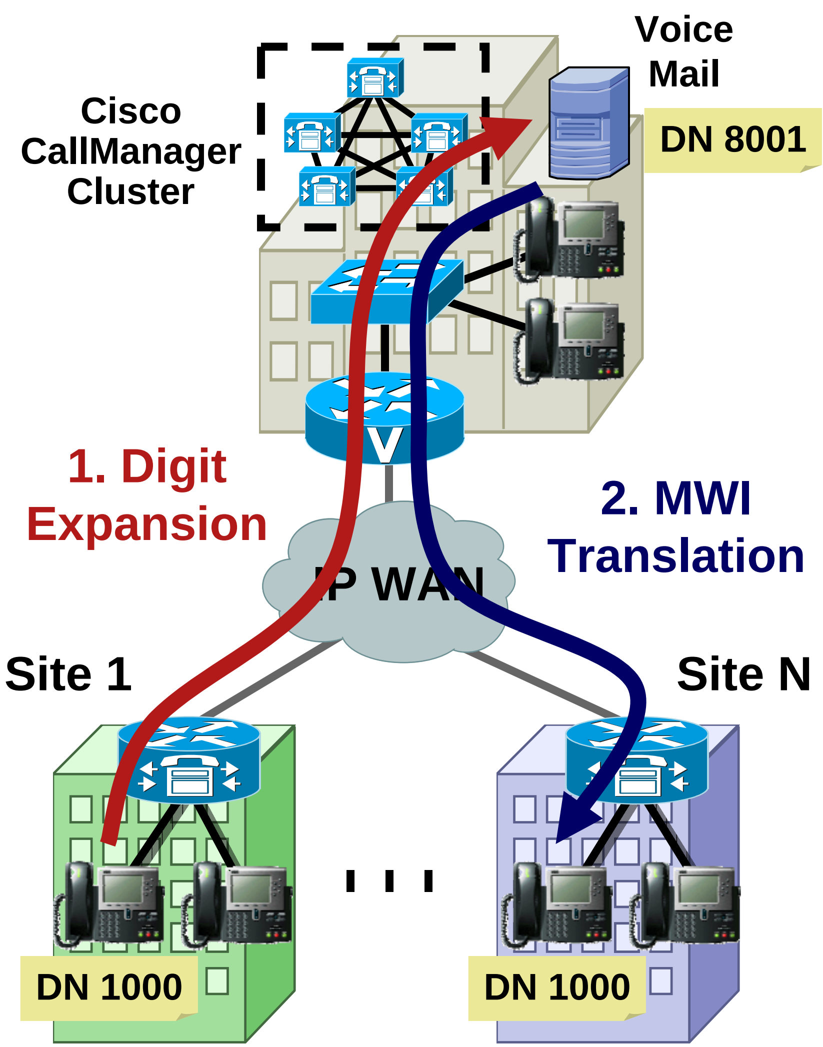
VLOD with Partitioned Addressing
Incoming PSTN/Gatekeeper Calls



VLOD with Partitioned Addressing
Gatekeeper Configuration

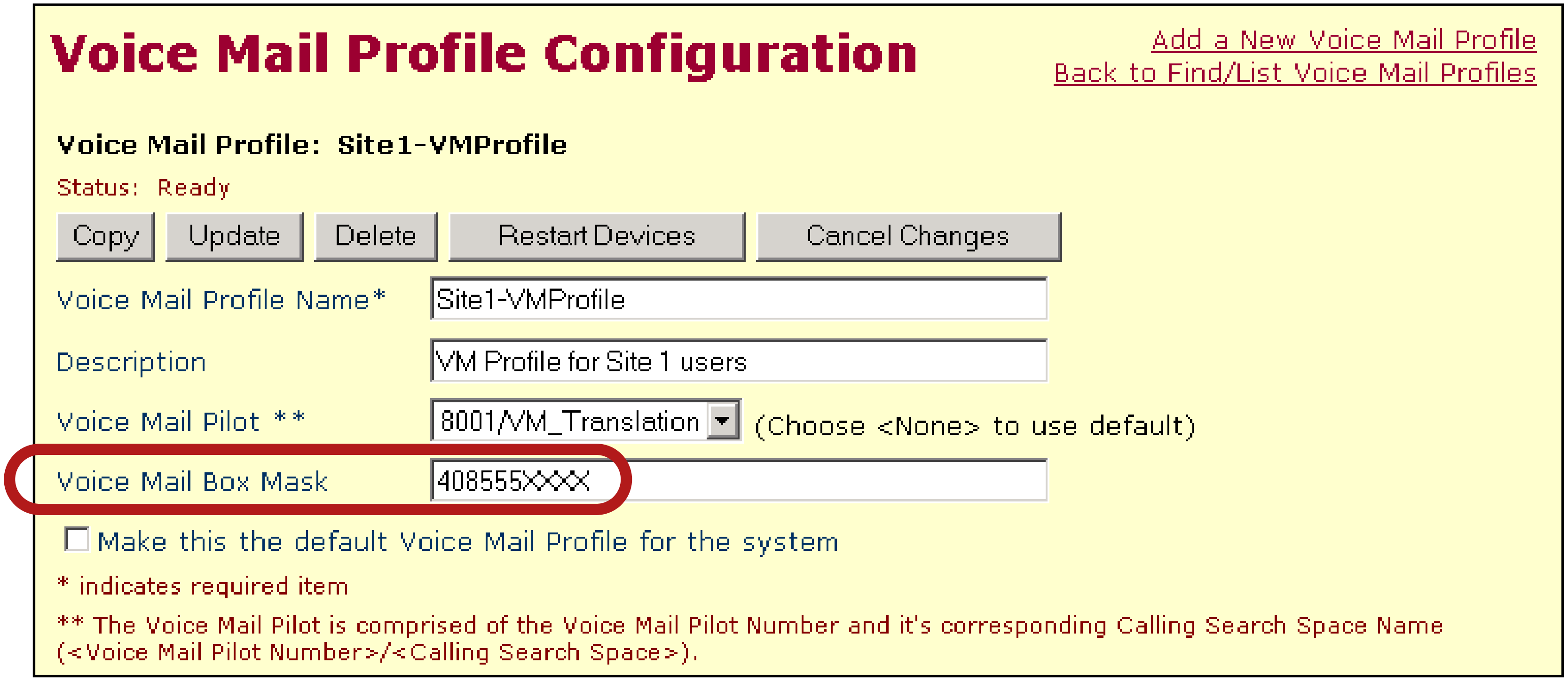
```
gatekeeper
  zone local US cisco.com 10.9.11.1
  zone local EU cisco.com 10.20.1.1
  no zone subnet US default enable
  no zone subnet EU default enable
  zone subnet US 10.9.11.2/32 enable
  zone subnet US 10.9.11.3/32 enable
  zone subnet EU 10.20.1.2/32 enable
  zone subnet EU 10.20.1.3/32 enable
  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



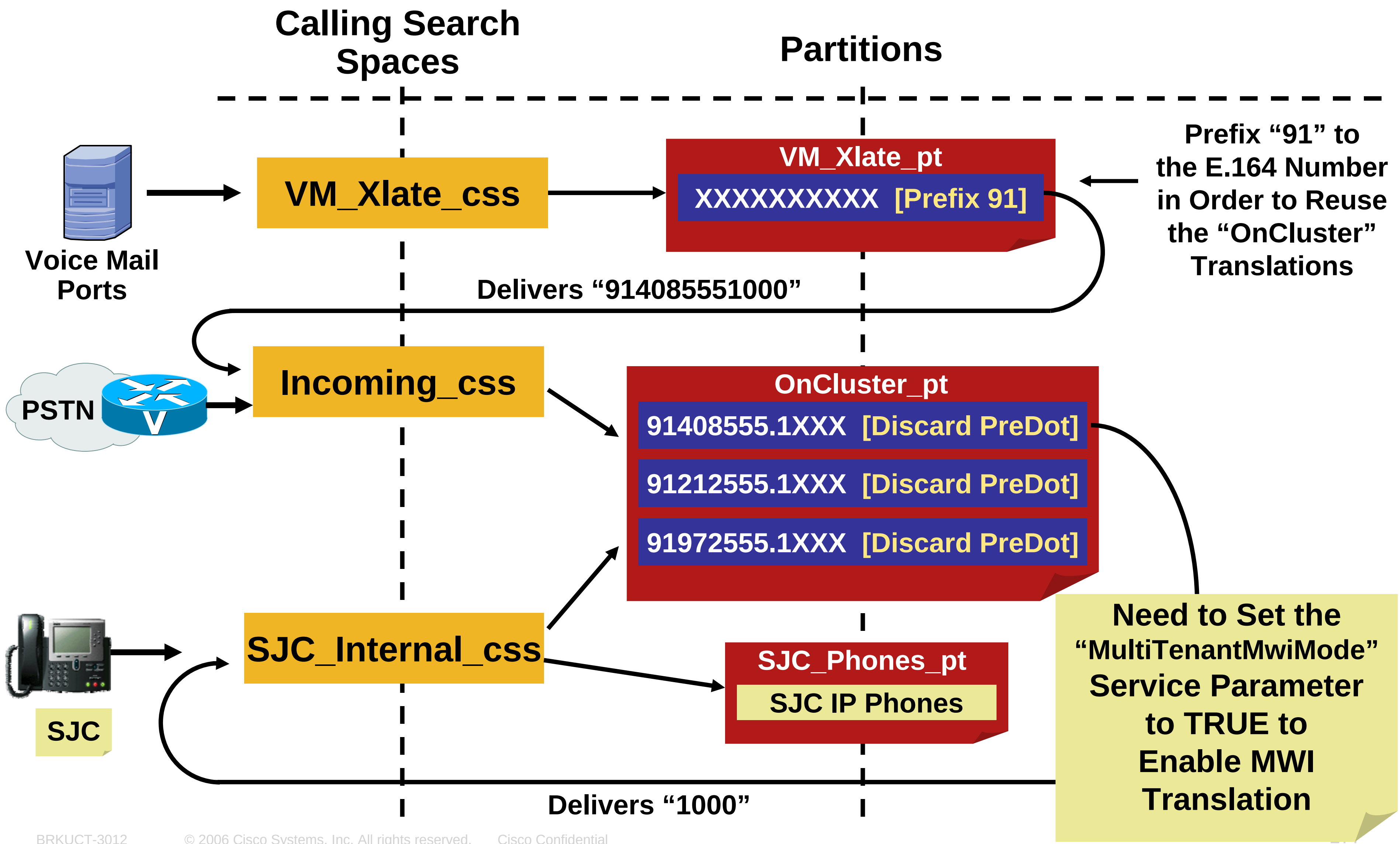
- Both SCCP- (Unity) and SMDI-based Voice Mail systems can be used
- Voice mail boxes need a unique DN
- Need to “expand” DNs when accessing VM
- MWI messages from VM system need to be “translated” to match appropriate DN/partition

VLOD with Partitioned Addressing
Voice Mail Integration: Digit Expansion



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

- Classes of Service for SRST (COR)
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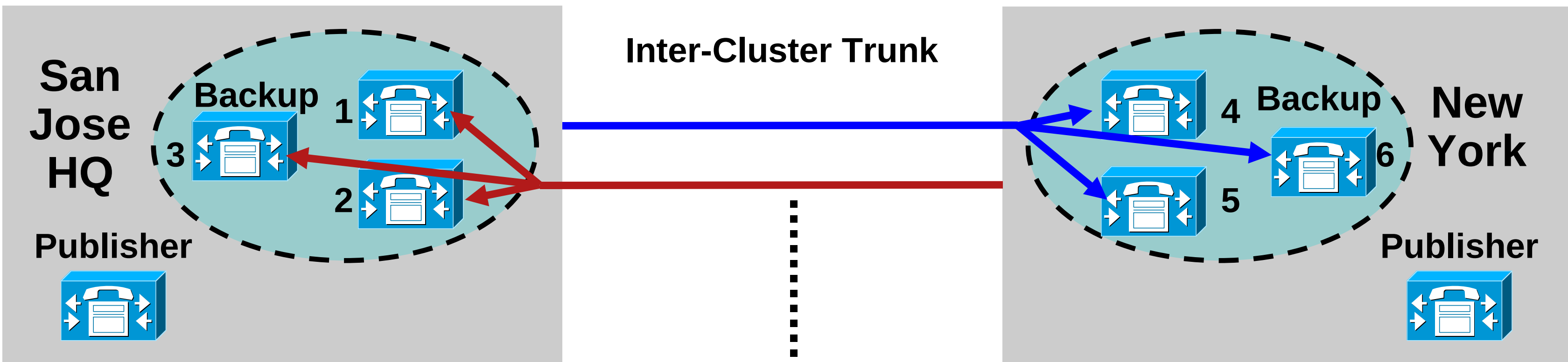
External Routes in Cisco Call Manager
Non-GK Controlled ICT



CM Group: SJC 1
SJC 2
SJC 3

CM Group: NYC 4
NYC 5
NYC 6

Remote Cisco CallManager Information	
Server 1 IP Address/Host Name*	NYC 4-IP Address
Server 2 IP Address/Host Name	NYC 5-IP Address
Server 3 IP Address/Host Name	NYC 6-IP Address

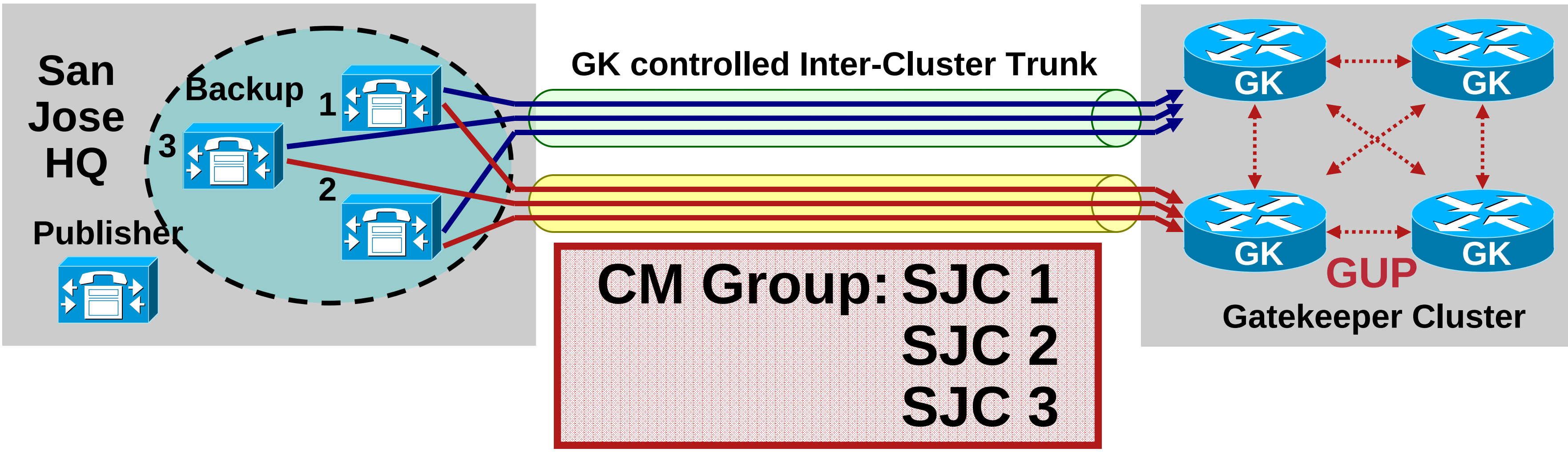
Remote Cisco CallManager Information	
Server 1 IP Address/Host Name*	SJC 1-IP Address
Server 2 IP Address/Host Name	SJC 2-IP Address
Server 3 IP Address/Host Name	SJC 3-IP Address

Redundancy is built into ICT (1 ICT needed instead of 3)

External Routes in Cisco Call Manager
Non-GK Controlled ICT

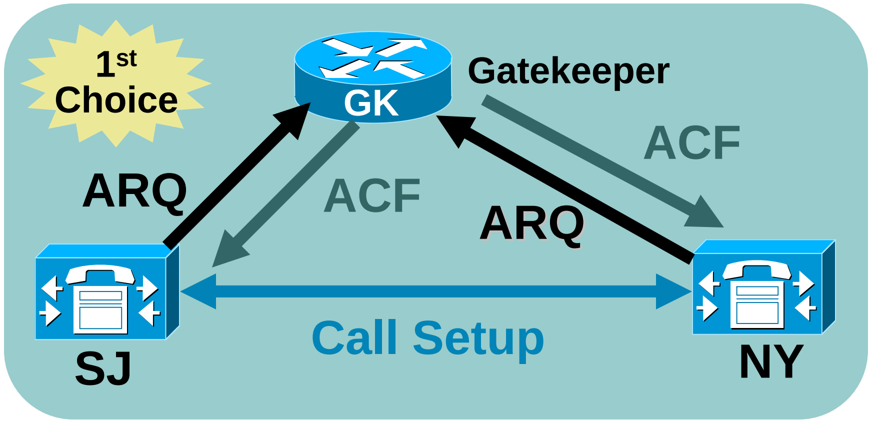
- Calls to a Non GK controlled inter-cluster trunk are load shared in a round robin fashion among the configured peer signaling addresses
- For example, the first call is routed to peer transport address one, next call to peer transport address two, third call to transport address three, fourth call to transport address one, and so forth

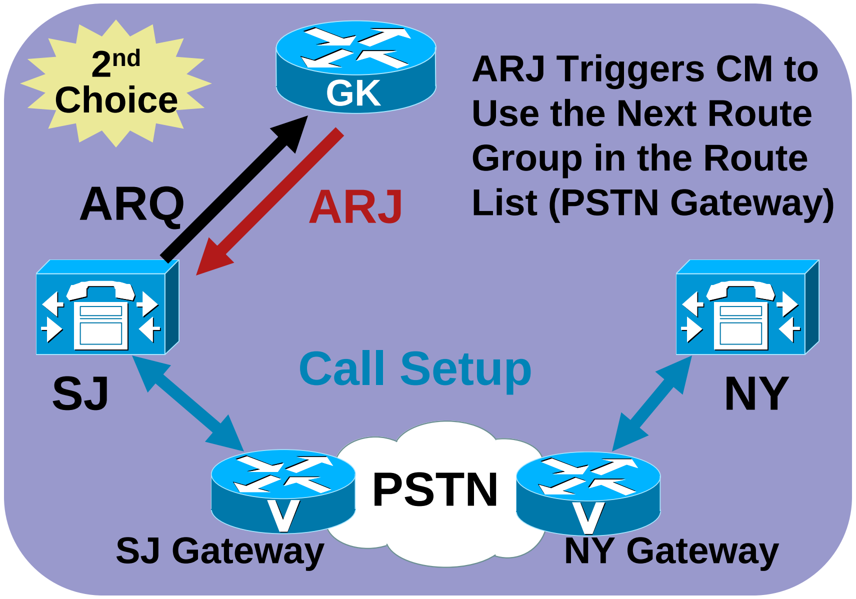
External Routes in Cisco Call Manager
GK Controlled ICT



- Easier Administration and Scalable (up to 100 Clusters)
- All Call Managers in CM Group register with GK, thus providing redundancy and load balancing
- Additional H.323 trunk defined for added redundancy when GK is not unavailable at initial registration or during reset

External Routes in Cisco CallManager
GK-Controlled Trunks: Automatic Reroute

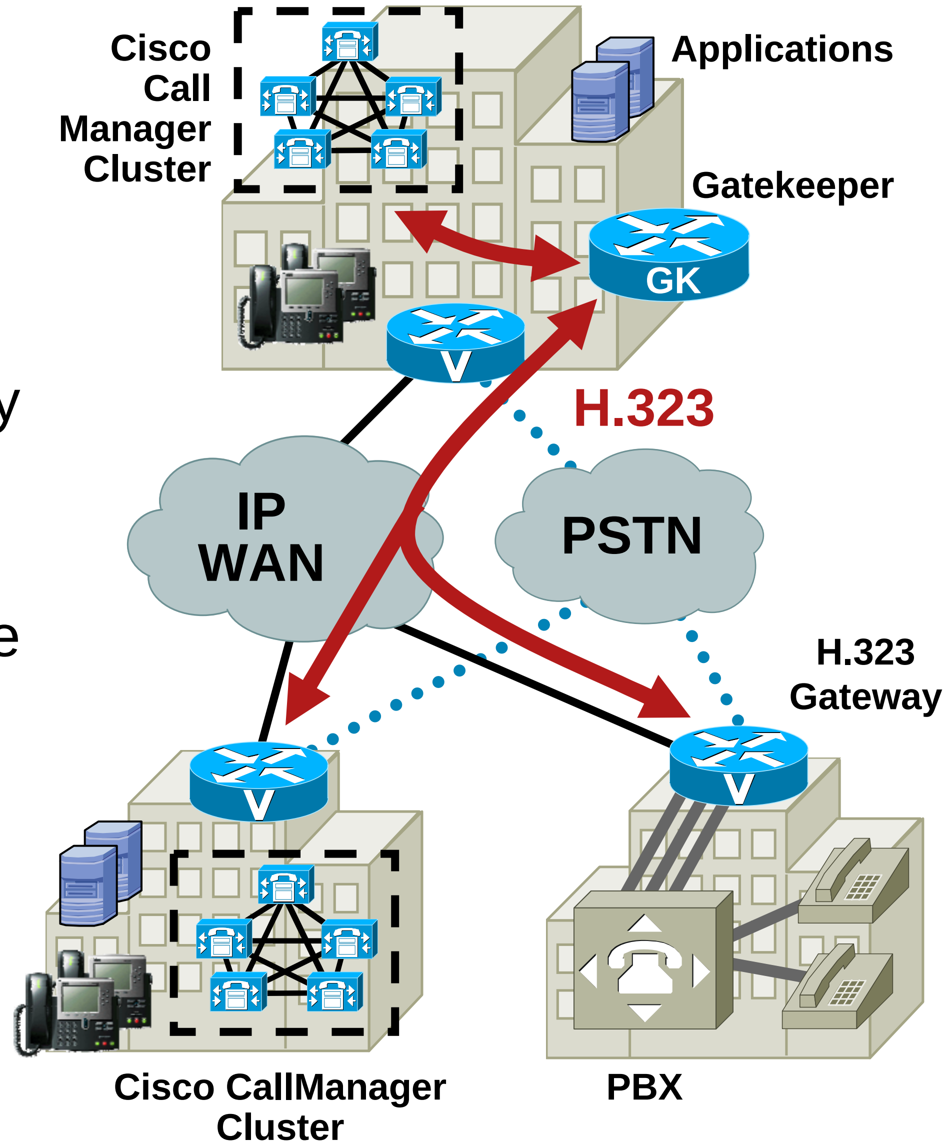





```
gatekeeper
  zone local SJC cisco.com
  zone local NYC cisco.com
  zone prefix SJC 140855534..
  zone prefix SJC 14085557...
  zone prefix SJC 131055598..

  [...]

  zone prefix NYC 16465551...
  zone prefix NYC 131255568..
  zone prefix NYC 120255524..

  [...]

  gw-type-prefix 1#* default-
                     technology
  bandwidth interzone zone SJC 480
```

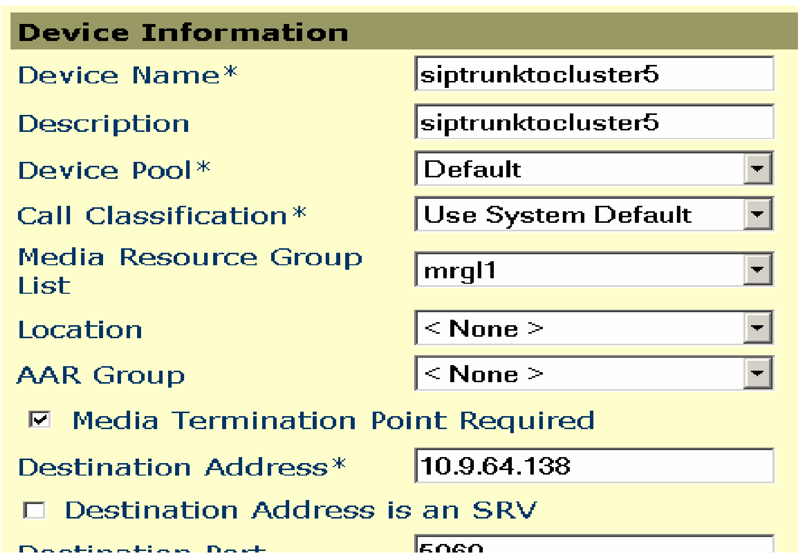
External Routes in Cisco Call Manager
H.225 Trunks

- Allows a mix-and-match of Cisco CallManager clusters and H.323 gateways
- Auto discovers if remote endpoint is H.323 gateway or Call Manager
- All calls across the WAN are controlled by the same gatekeeper
- Facilitates migration from toll-bypass networks



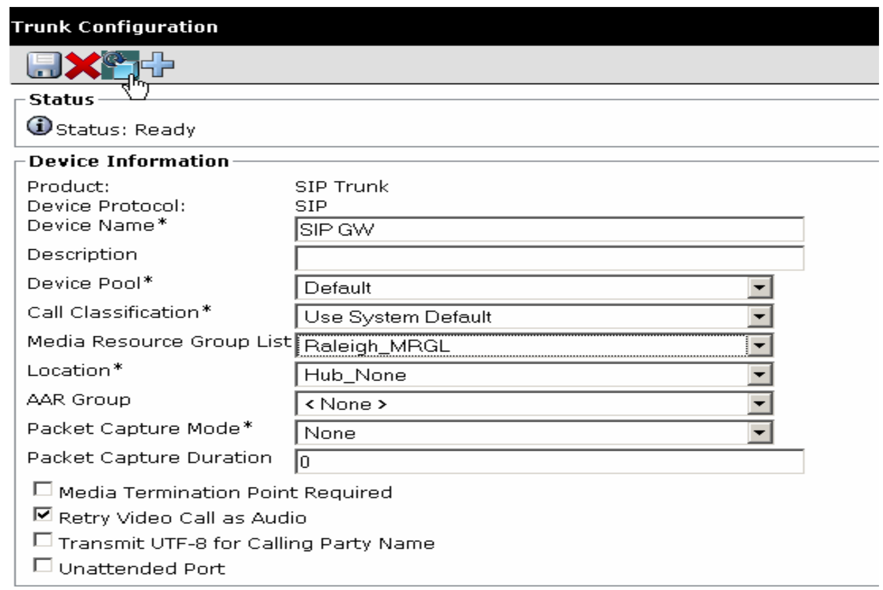
External Routes in Cisco Call Manager
SIP Trunks

CallManager 4.x SIP Trunk



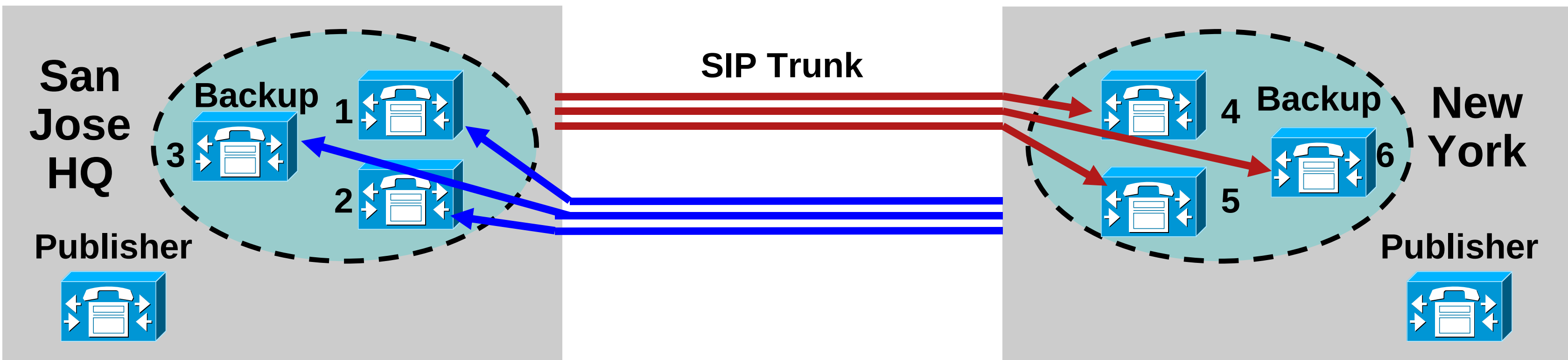
- Early-media only and s/w MTP is required.
- Only G.711 codec allowed.
- RFC2833 only
- No Video Support, Subset of SIP Messages

CallManager 5.0 SIP Trunk



- Delay-media (h/w – s/w MTP) and early-media (s/w MTP).
- MTP will be inserted dynamically if needed for OOB to 2833 conversion or early-media is used.
- RFC2833, KPML, Unsolicited-notify

SIP Trunks: Redundancy
Direct Integration



CM Group: SJC 1
SJC 2
SJC 3

CM Group: NYC 4
NYC 5
NYC 6

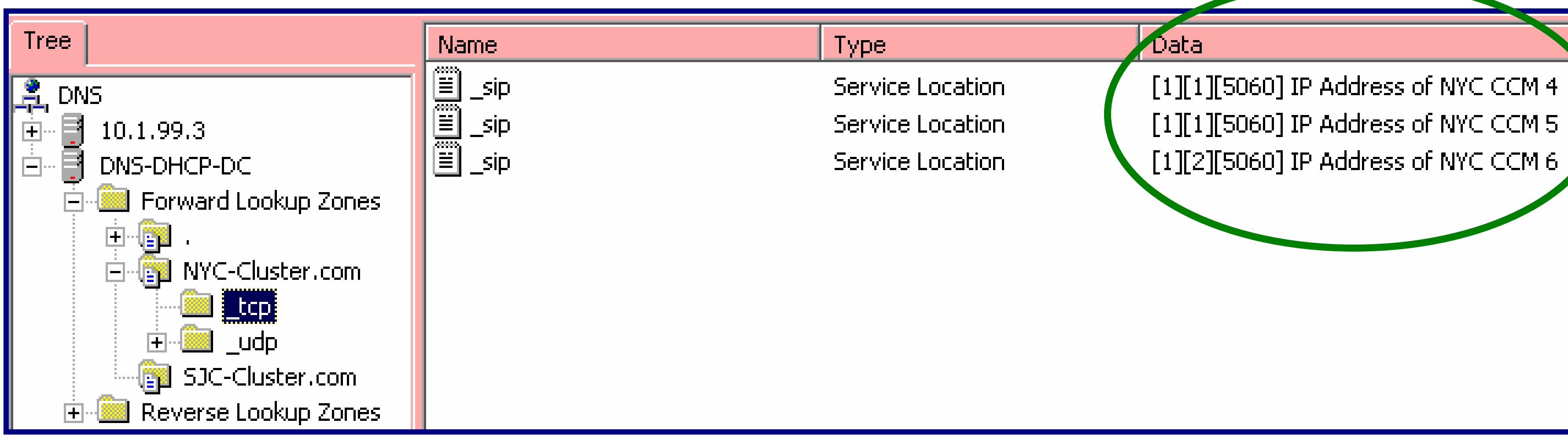
SIP Information

Destination Address* 10.1.104.2

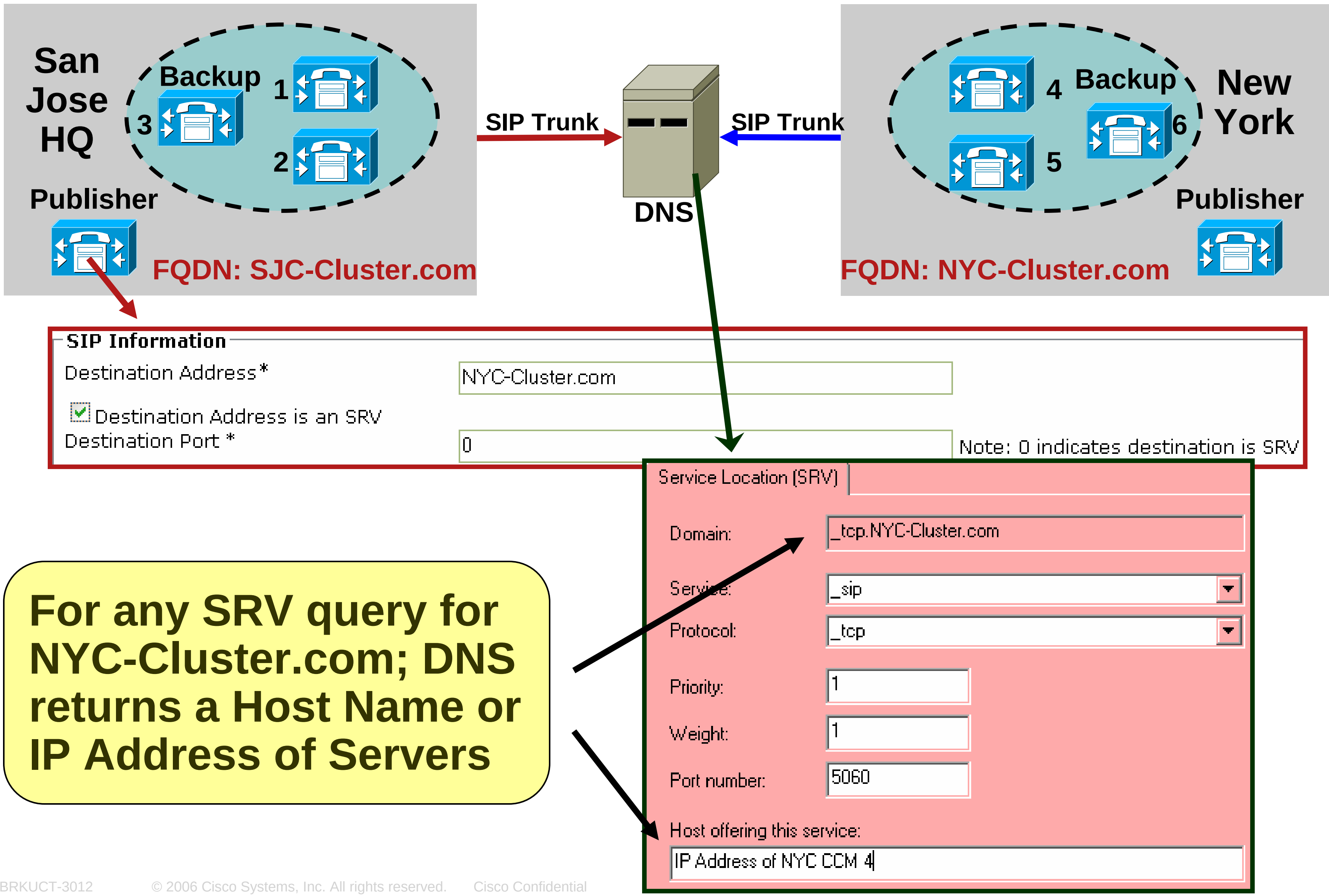
NO Redundancy built into SIP Trunk Configuration

SIP Trunks: Redundancy
DNS SRV Records

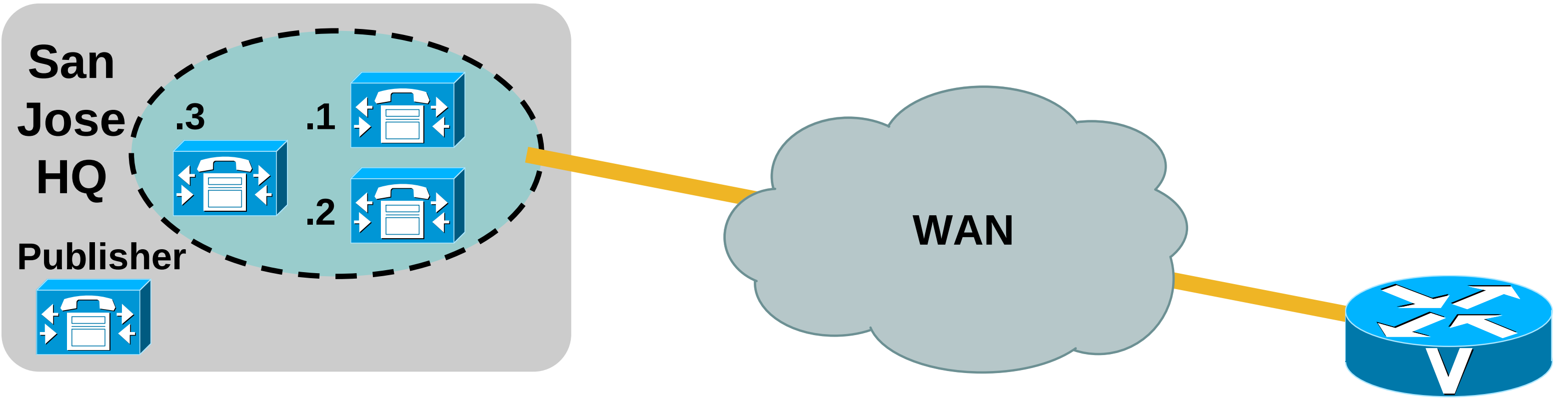
- Service (SRV) records allows:
  - Using several servers for single DNS domain
  - Designating some servers as primary and some as backups
  - Moving TCP/IP services from one host to other



SIP Trunks: Redundancy
DNS Integration



External Routes in Cisco CallManager
H.323 Gateways with Centralized Processing



Dial Peer Configuration

- Be sure to configure a dial peer for each CallManager server in the redundancy group/device pool assigned to the Gateway in CM
- Ensure that they match on both sides

```
dial-peer voice 1 voip
 destination-pattern 1...
 preference 1
 session target ipv4:10.10.10.1
dial-peer voice 2 voip
 destination-pattern 1...
 preference 2
 session target ipv4:10.10.10.2
dial-peer voice 2 voip
 destination-pattern 1...
 preference 3
 session target ipv4:10.10.10.3
```