What You Make Possible

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Designing Dial Plans for Enterprise Unified Communications

BRKUCC-2008

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BRKUCC-2008 Abstract

- This intermediate session provides detailed dial-plan design guidelines for each of the Cisco IP Telephony deployment models based on Cisco Unified Communications Manager, with recommended best practices to help ensure successful, scalable deployments.
- This session covers the various dial-plan tools available in Cisco Unified Communications Manager, such as route patterns, translation patterns for digit manipulation, calling party transformations for localisation and globalisation of calling party information, dial-plan interaction with PSTN gateways and URI dialing.
- This session also covers how to best use these tools to deal with real-world deployments. The main focus of the session is on system design, with some implementation aspects. This session is aimed at network planners and designers and telephony analysts and assumes a working knowledge of the Dial Plan functionality in Cisco Unified Communications Manager

Agenda

- Introduction
- Call Routing Recap
- Developing a Global Dial Plan – Call Routing
- Developing a Global Dial Plan – Number Presentation
- URI Dialing

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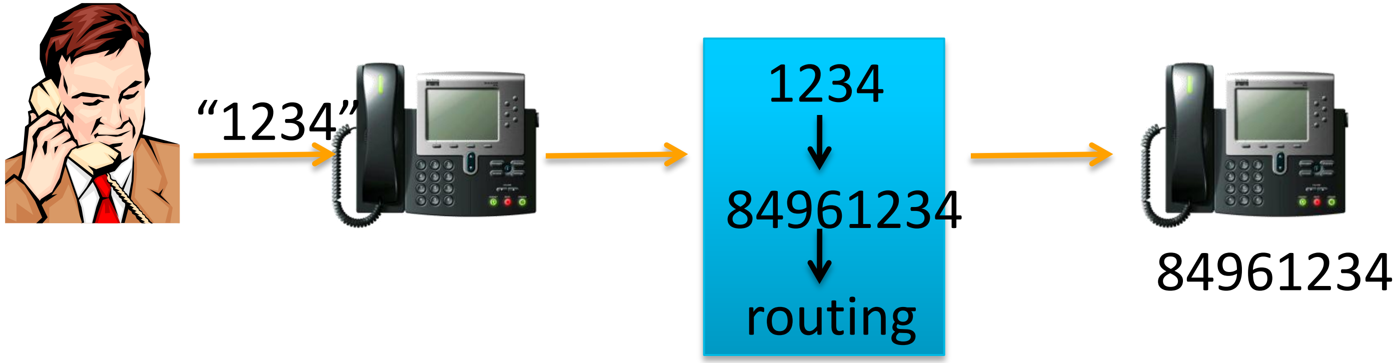
Remember

- Best and most important tools for dial plan design:
  - Pencil
  - Paper
  - Whiteboard
- Dial plans are not a new concept
- IP did not really change the fundamentals of dial plan design
- Dial Plan recommendations are not a monolith
  - Take what you need
- Keep it simple!

What Is a Dial Plan About?

- Mapping from dialed destinations to connected endpoints
- Concepts that are part of dial plans
  - user input
  - mapping of user input to routable format (transformations)
  - routing / routing restrictions (class of service)
  - call presentation
  - numbering plans



User Input / Dialing Habits

- Users dial using common habits: Dialing Habits
- Different formats for types of destinations
  – colleague next door
  – local, national, international
  – Inter-office (abbreviated on-net, forced on-net)
  – Voicemail
  – other services
- Especially external dialing habits are country-specific
  – 9 or 0 for outside line
  – Format of national numbering plan (fixed/variable length etc.)

Example Dialing Habits in Europe

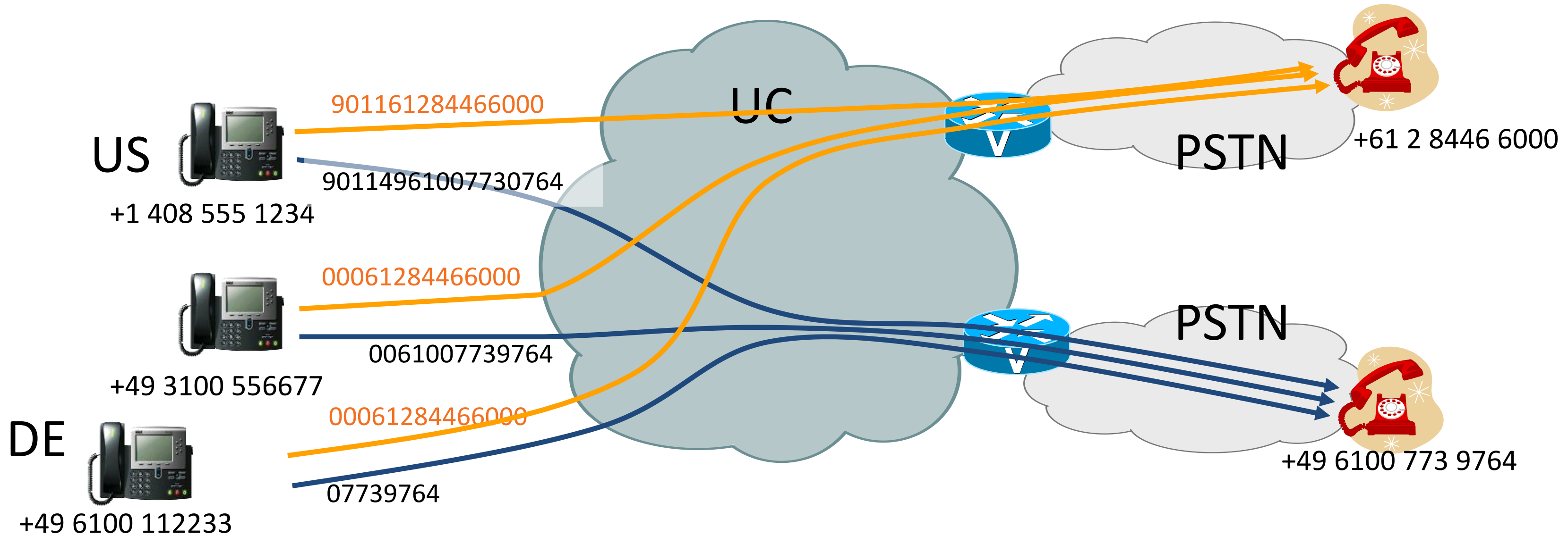
- “0” (or “9”) to get an outside line (trunk access)
- Any number starting with 1-8 is generally internal
  - But please stay clear of “112” → standardised emergency number
- National numbers need a “0” in front of the area code:
  - 0 – Trunk access
  - 0 – Escape for area code (Italy: “0” is part of the area code)
  - 69 – Area code of Frankfurt
  - Dial 0-0-6-9-... From inside the enterprise to Frankfurt
- international numbers are typically prefixed by “00”:
  - 0 – Trunk access
  - 00 – Escape for country code
  - 39 – Country code of Italy
  - → Dial 0-0-0-3-9-... From inside the enterprise to Italy

Enterprise Specific Dialing Habits

- Typically dialing habits for local, national, international calls are given/agreed based on a given domain/country
- In addition need to agree on how to dial:
  - Private numbers (on-net)
  - Intra-Site
  - Services (voicemail, meet-me, call park, pick-up ...); non-DIDs
- Do we also need to support “+”-dialing?
  - application support
  - number portability

Dialing Domain

- Enterprise call controls need to be able to support different national dialing behaviours (different dialing habits)
- Groups of users sharing common dialing habits need to be treated identical
- Definition: Dialing Domain = Group of users/devices sharing common dialing habits to reach identical destinations



Dial Plan vs. Numbering Plan

- Dial Plan: scheme to define mapping from dialing habits to destinations
- Numbering Plan: scheme to number entities (phones)
  - unique number per entity → e.g. (+)E.164, private numbering
    allows for single numbering domain
  - overlapping numbering → e.g. unique per site
    requires partitioned numbering domains

Dial Plan vs. Numbering Plan

- Dial plan might support various dialing habits
  - local call: 0 – number
  - national call: 00 – number
  - international call: 000 – number
  - abbreviated on-net: 8<7-digits>
  - +E.164: +E.164 string
- Enterprise Numbering Plan might follow one of the above dialing habits (e.g. abbreviated on-net)

... but does not necessarily have to!

Class of Service

- Common term to describe the permissions of users on communication systems
- COS includes
  - permission to reach certain destinations
  - voicemail access
  - reachability from outside
  - call forward restrictions
- Enterprise dial plan is the tool to implement required classes of service
- Important: make sure to start dial plan design with full view of required classes of service

E.164 geographic numbers

Background

<table>
<tr><td rowspan="2">CC
1-3 digits</td><td colspan="2">NSN
max 15-n digits (n=number of CC digits)</td></tr>
<tr><td>NDC
Defined by nat. numbering plan</td><td>SN
Defined by nat. numbering plan</td></tr>
<tr><td colspan="3">max 15 digits</td></tr>
</table>

- ITU Recommendation E.164 describes the “Numbering Plan of the International telephone service”
  - CC: Country Code
  - NSN: National significant number
  - NDC: National destination code
  - SN: Subscriber number
  - NDC+SN = NSN: National significant number
- National numbering plan left to national authorities
  - documented at http://www.itu.int/oth/T0202.aspx?lang=en&parent=T0202
  - US: fixed length, NSN 10 digits
  - DE: variable length, NSN 4-13 digits

+E.164 Notation and “Numbers”

- ITU Recommendation E.123 describes the “Notation for national and international telephone numbers, e-mail addresses and Web addresses“
  - “+” signifies the international prefix
  - Example: +14085551234
- Numbers in global directories should be in +E.164 format
  - global form including country code
  - leading “+”
  - no trunk access codes included: +44 (0) 208 1234 1243 is NOT a valid +E.164 number!
  - universal use
- Benefits of +E.164 “Numbers” in dial plans
  - unique by definition
  - no overlap with any other dialing habit (“+”)

Overlaps

- Dialing habits need to avoid overlaps to avoid interdigit timeout (T302, default: 15s)
- No overlap between:
  - Outside access code & intra-site (UK: No 9xxx DN)
  - on-net access code & intra-site (Cisco: No 8xxx DN)
  - on-net access code & outside access code (on-net: No 0 or 9)
- on-net and outside access code reduce the numbering space available for intra-site dialing
- Overlaps have to be avoided in the planning phase
- If overlapping dialing habits are defined this can not be resolved later

Private Numbering Plan
(abbreviated on-net dialing)

- Pro
  - Possibly shorter inter-site on-net dialing
  - Fixed length instead of possibly variable length inter-site on-net dialing
  - Can be re-used for VM subscriber IDs
- Con
  - National dialing to known sites can be forced on-net; no NEED for private numbering
  - Private numbers are only useable inside the enterprise
  - Will people actually use them?
  - Steering digit for private numbering reduces the set of available numbers
  - Planning and maintenance effort
- Is it worth it?

Guidelines for Private Numbering Plan

- Typical format:
  - <access code> - any digit or “*”
  - <site id> - Might be a hierarchical scheme including regional attributes
  - <extension> - Intra-site on-net extension
- Example: 8-496-1234
  - 8 – Access code
  - 496 – Site id (site 6 in Germany)
  - 1234 – Local extension
- Make sure to reserve space (what if we get more than 9 sites in Germany)
- Make it extensible (think “Shannon coding”)
- Changing an established private numbering is VERY hard

External Numbering Plan Requirements

- Providers dictate format for Calling/Called Party Numbers on trunks
- Technology:
  - ISDN: Concept of Type (national, international, subscriber) and Number
  - SIP: Only Number; typically +E.164
- PBX interconnect (Q.SIG)
  - End-to-end support for numbering used on existing PBX systems
  - Uniform across all systems?

What to Use as DNs?

- Options:
  - Intra-site extension: Requires per-site partitions
    Example: 9764
  - Unique abbreviated on-net extension (private numbering plan)
    Example: 8 496 9764
  - +E.164: Unique; “+” to avoid overlaps
    Example: \+49 6100 773 9764
  - E.164: Unique; how to avoid overlap?
    Example: 49 6100 773 9764
  - National number (10-digit US)
    What if you need to expand to global plan?
- Number transformations in UCM allow to map between numbering schemes

+E.164 DNs and Non-DIDs

- If Non-DIDs and DIDs share a common partition Non-DIDs need to be assigned using "unallocated" spaces to avoid overlaps
- International:
  - –Unallocated: http://www.itu.int/pub/T-SP-E.164D
  - –+0: reserved, possibly create hierarchical numbering scheme starting with +0
- National:
  - –Unallocated ranges in national numbering plans: http://www.itu.int/oth/T0202.aspx?parent=T0202
  - –Caution: national numbering plans might get changed (example: renumbering in Italy in 1998)
- Completely different space: e.g. numbers starting with "*"
- Reminder: Don't mix dialing habits and numbering requirements

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+ Sign Support

What It Is: Concept

- +E.164 support includes the use of + to wildcard international access codes AND to avoid overlap between globalised numbers and other ranges (e.g.: calls to India (+91XXXXXXXXX) and NANP toll calls (912125551234)
- Supporting the + sign allows UCM-based systems to route calls based on an universal non-site (country) specific format
- + can be used in all dialable patterns
  - DN
  - Route Pattern
  - Translation pattern
  - ...
- Most phones support +-dialing: 7925/21 from day one, newer phones starting with phone firmware 9.1.1

Number Transformations

- Two Concepts:
  - Implicit – as part of routing process
    - Translation Pattern
    - Route Pattern
    - Route Lists
  - Explicit – Transformation after routing decision
    - Incoming Calling/Called Party Settings on gateways, trunks (or device pools)
    - Calling/Called Party Transformation CSS on gateways, trunks (or device pools)
    - Calling Party Transformation CSS on phones (or device pools)

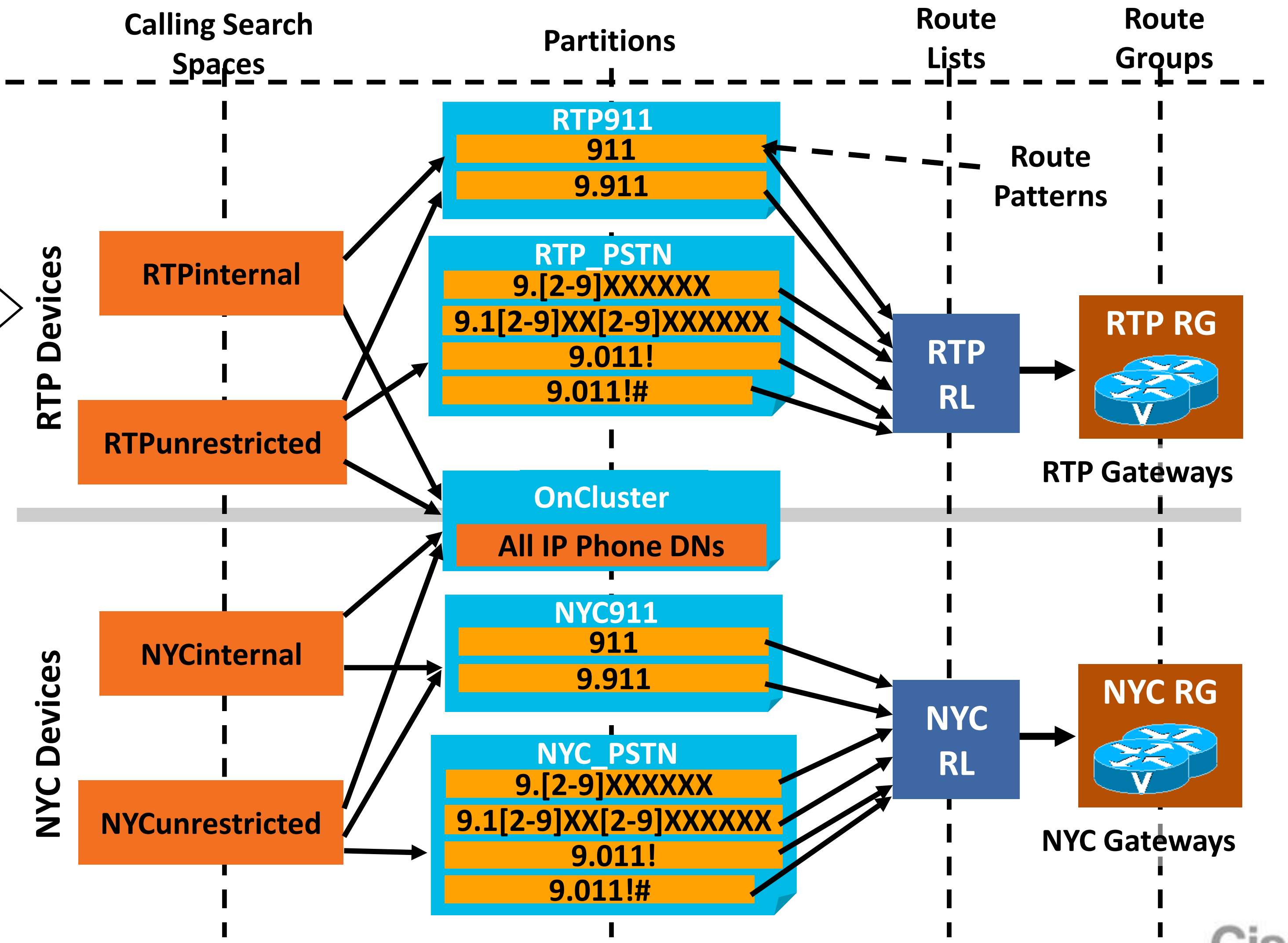
Building Classes of Service

Simple CSS Approach for Centralised Deployments

Device CSS Dictates:
- Class of Service
- Path Selection

 # CSS = N x C

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



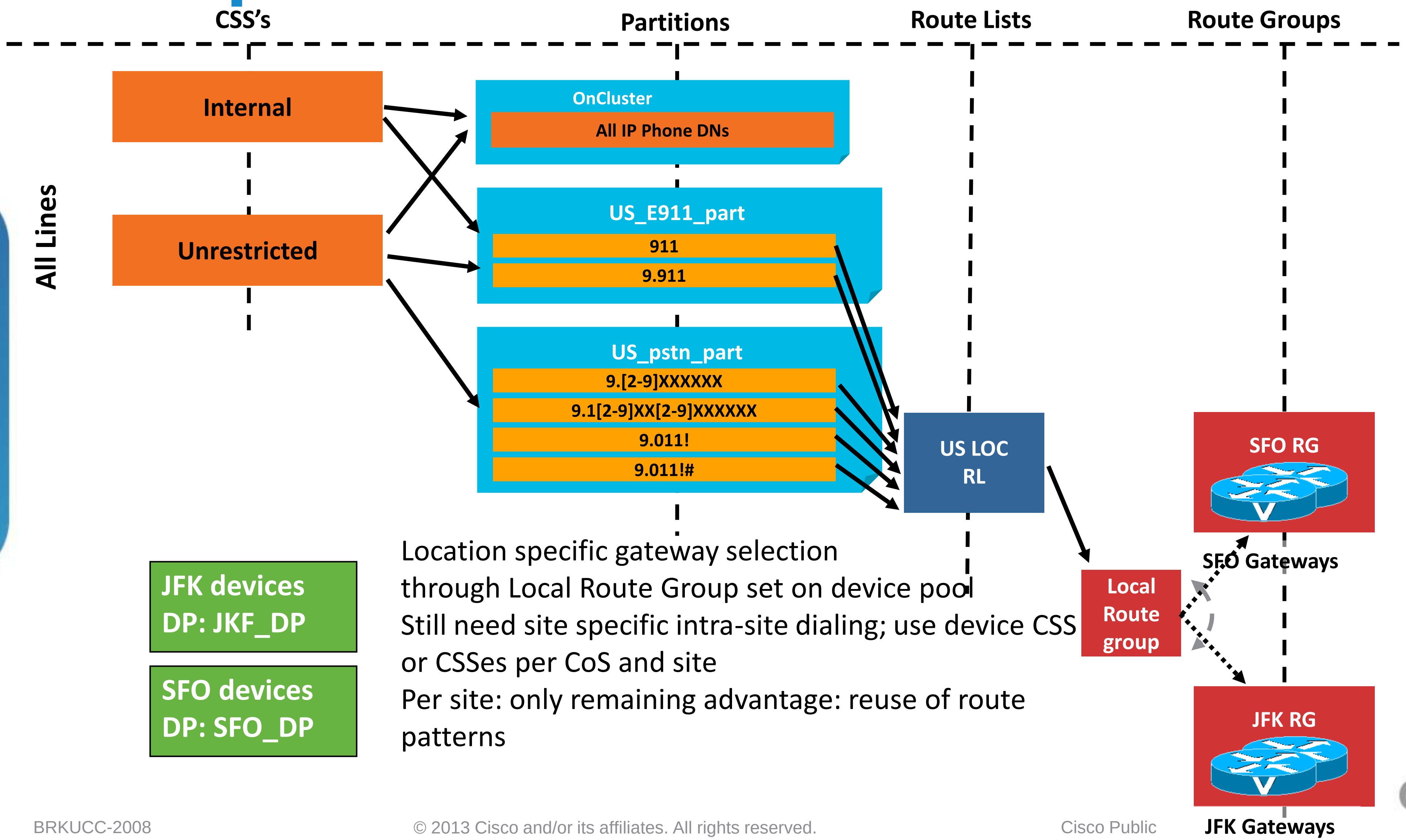
Local Route Group

What It Is: Concept

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

Simple CSS Approach with Local Route Group

Removing site specific route patterns



Location specific gateway selection through Local Route Group set on device pool

Still need site specific intra-site dialing; use device CSS or CSSes per CoS and site

Per site: only remaining advantage: reuse of route patterns

Simple CSS Approach with Local Route Group

LRG benefits

- LRG offloads the site specific path selection from the route pattern
- No requirement for site specific route patterns
- Limitation/Caveat:
  - only one LRG per device pool
  - if you need site specific route lists (primary/secondary route group), this approach will not work w/o adding site specific route patterns

LRG and Number Transformations

- Using LRG moves normalisation of calling and called party numbers to device level
  - Local context and numbering requirements of route group members unknown at the routing level
- OTOH Q.SIG only picks up number transformations at the routing level
- → LRG can only used with Q.SIG trunks if the calling and called numbering format of the PBX systems is implemented end-to-end in UCM
  - Transformation of calling and called numbers into the PBX numbering plan to be implemented using Translation and Route patterns

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Requirements

- Dialing Habits
  - 4-digit intra-site
  - + dialing for dialing from directories
  - US sites
    - 9 + 7-digit for local calls
    - 91 + 10-digit for national calls
    - 9011 for international calls
  - German sites
    - 0 for local calls
    - 00 for national calls
    - 000 for international calls
- Number presentation on phones in shortest possible format

Requirements

- Routing
  - Forced on-net
  - Local gateways in every site
  - TEHO for international calls
- Classes of Service
  - Internal: Allowed to call all on-net destinations
  - National: Only national off-net destinations
  - International: No restrictions

Requirements

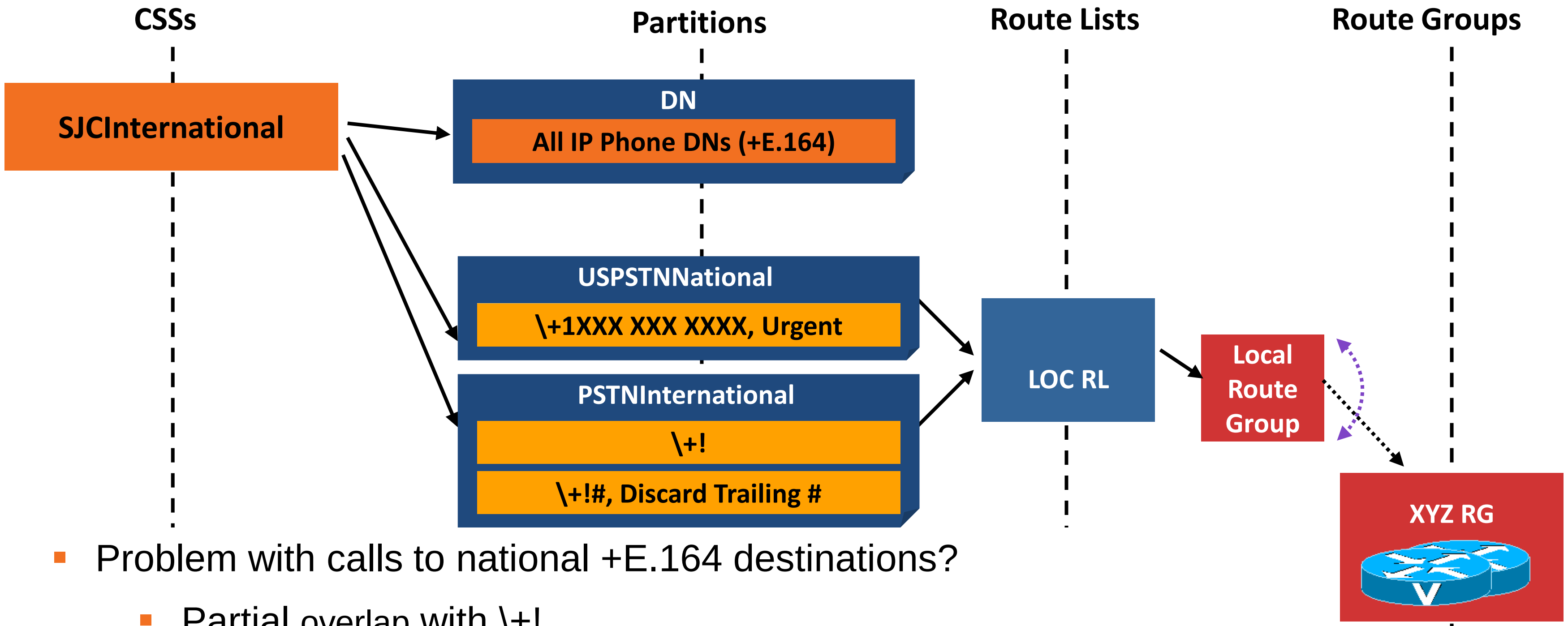
- Sites
  – ESC: +4961007739XXX
  – STU: +49710023911XXX
  – SJC: +14085551XXX
  – DFW: +19725551XXX

DN Format

- Single partition for all DNs
- Requires unique DNs
- We don't have an abbreviated on-net numbering plan
  - ... and don't want to create one from scratch
- +E.164 or E.164?
- Let's start with +E.164 DNs
- Will it work with just line CSS and LRG?

CoS International

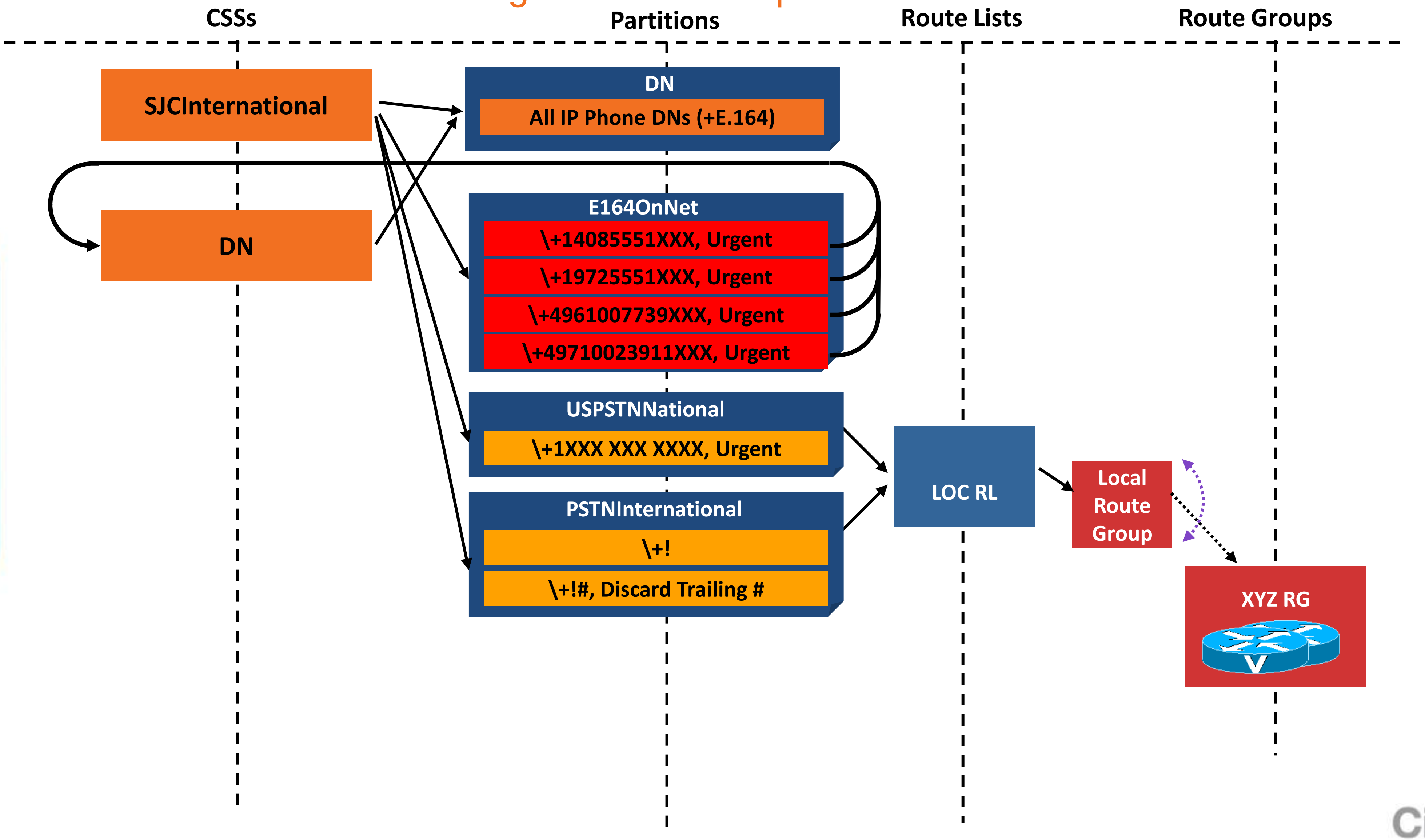
+E.164 Destinations



- Problem with calls to national +E.164 destinations?
  - Partial overlap with \+!
  - Solution: Make \+1XXX XXX XXXX urgent
- Other problems?
  - DNs are non-urgent patterns
  - \+! has partial overlap with all DNs
  - Solution: We need urgent patterns for all on-net destinations to avoid overlap with \+!

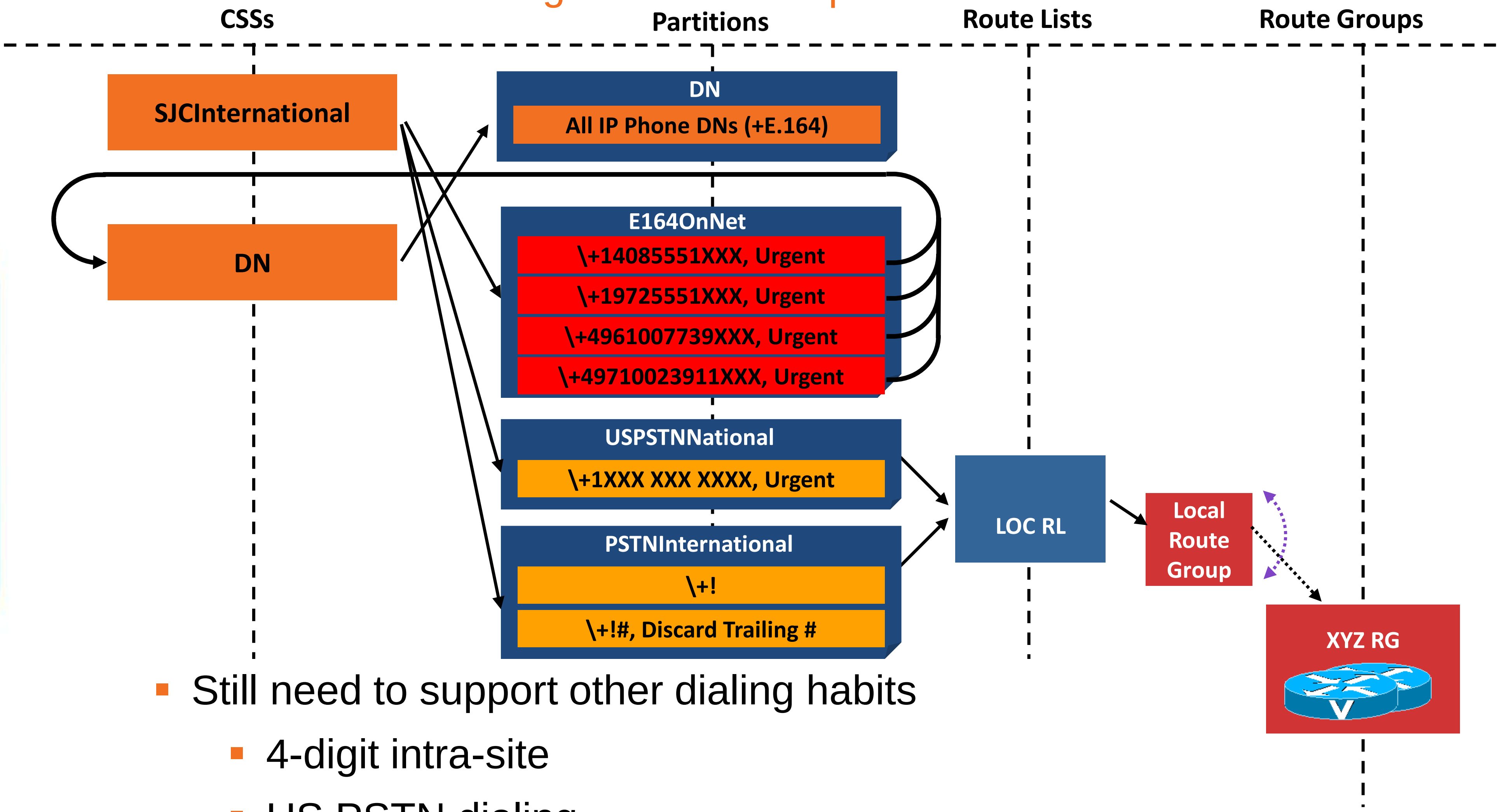
CoS International

+E.164 Destinations Avoiding Partial Overlap



CoS International

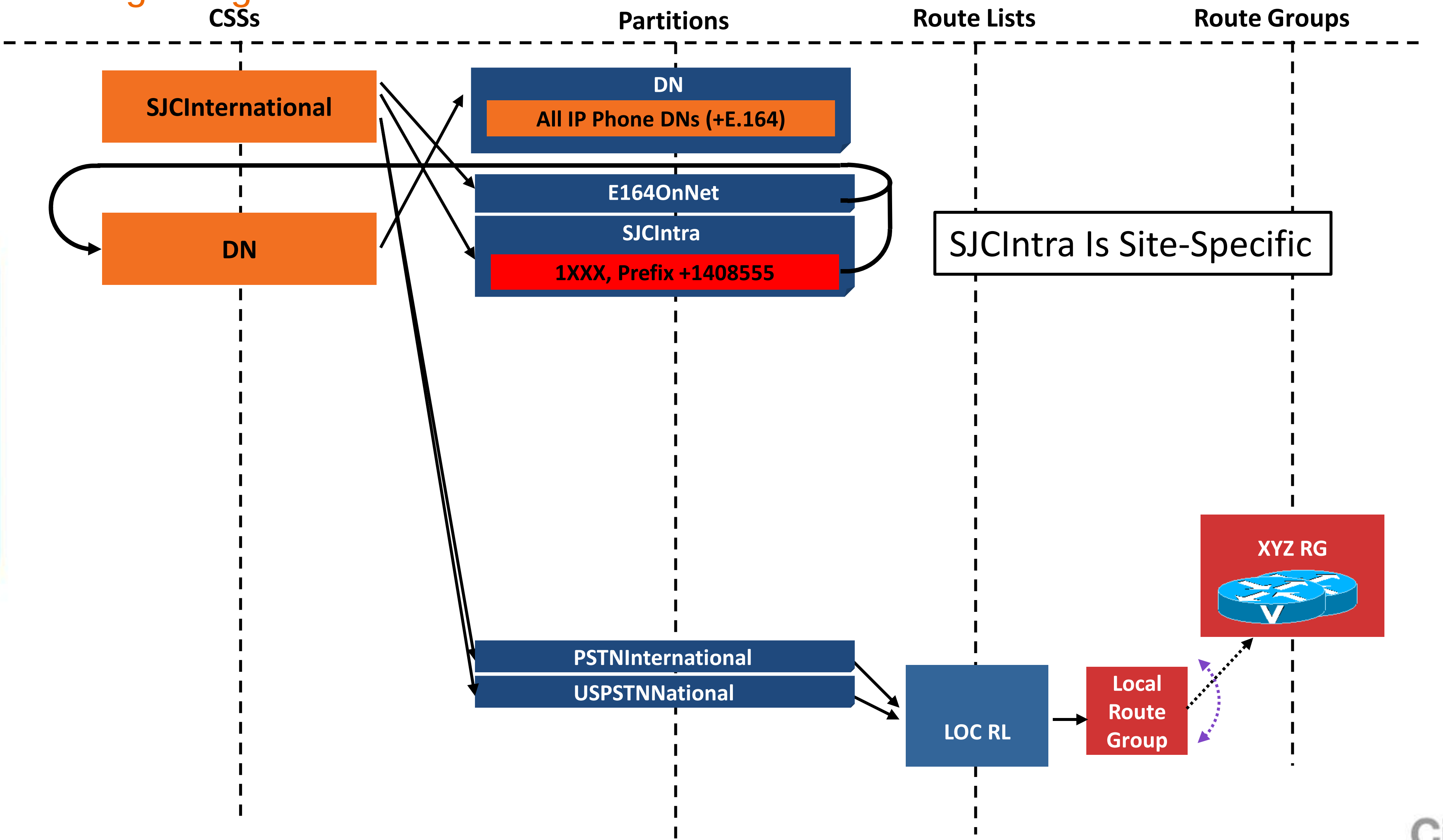
+E.164 Destinations Avoiding Partial Overlap



- Still need to support other dialing habits
  - 4-digit intra-site
  - US PSTN dialing

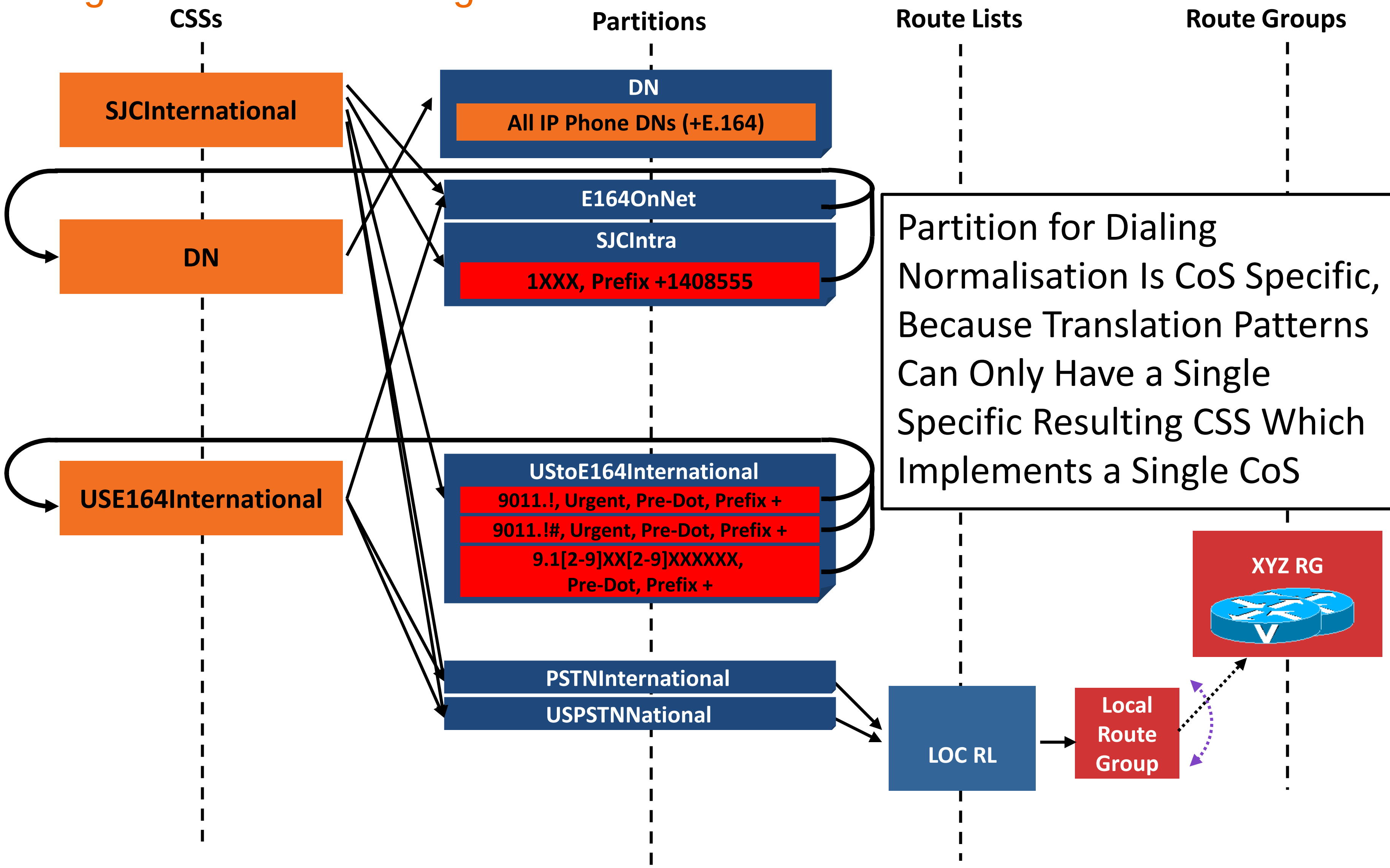
CoS International

Adding 4-Digit Intra-Site



CoS International

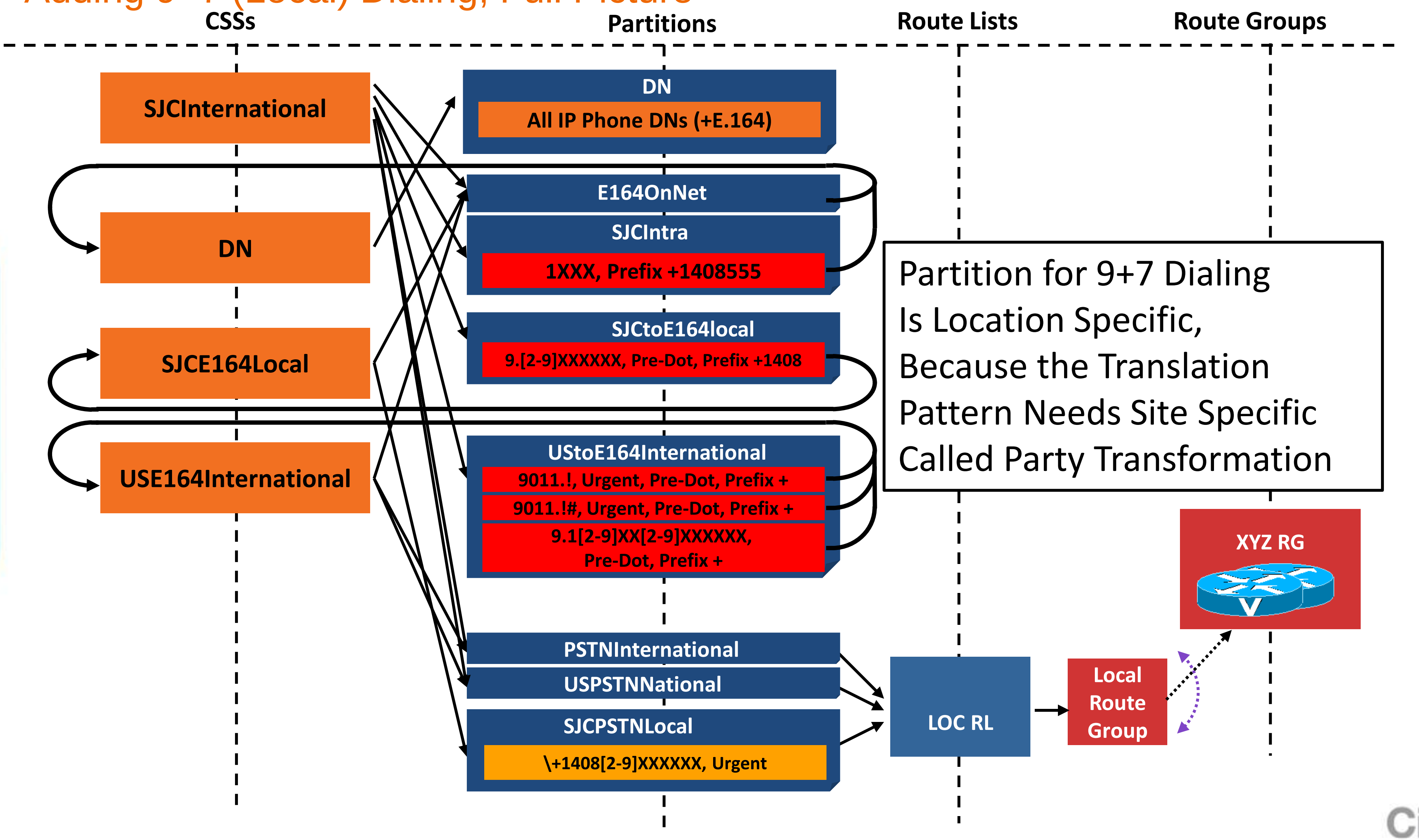
Adding International Dialing



Partition for Dialing Normalisation Is CoS Specific, Because Translation Patterns Can Only Have a Single Specific Resulting CSS Which Implements a Single CoS

CoS International

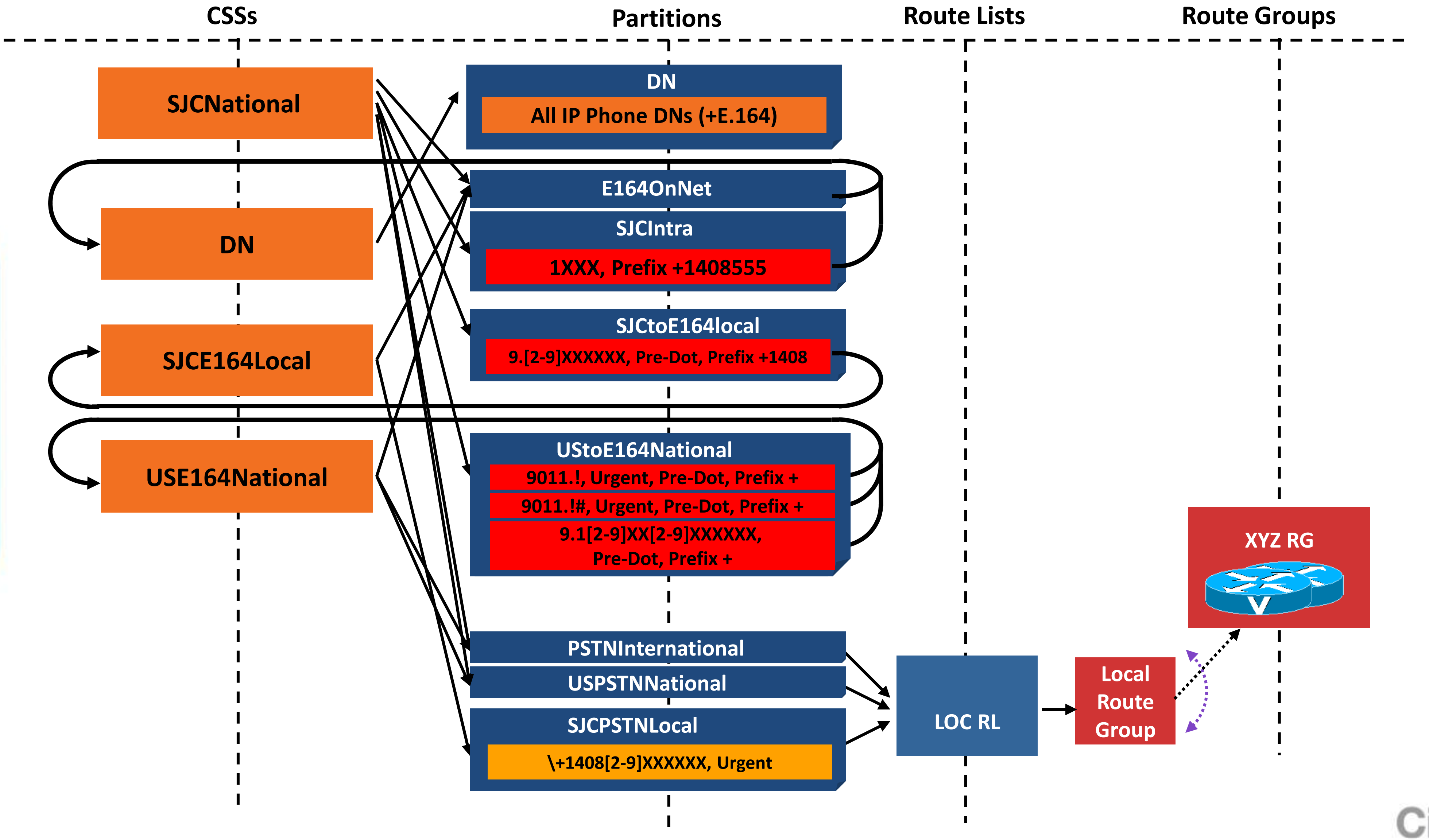
Adding 9+7 (Local) Dialing; Full Picture



Partition for 9+7 Dialing
Is Location Specific,
Because the Translation
Pattern Needs Site Specific
Called Party Transformation

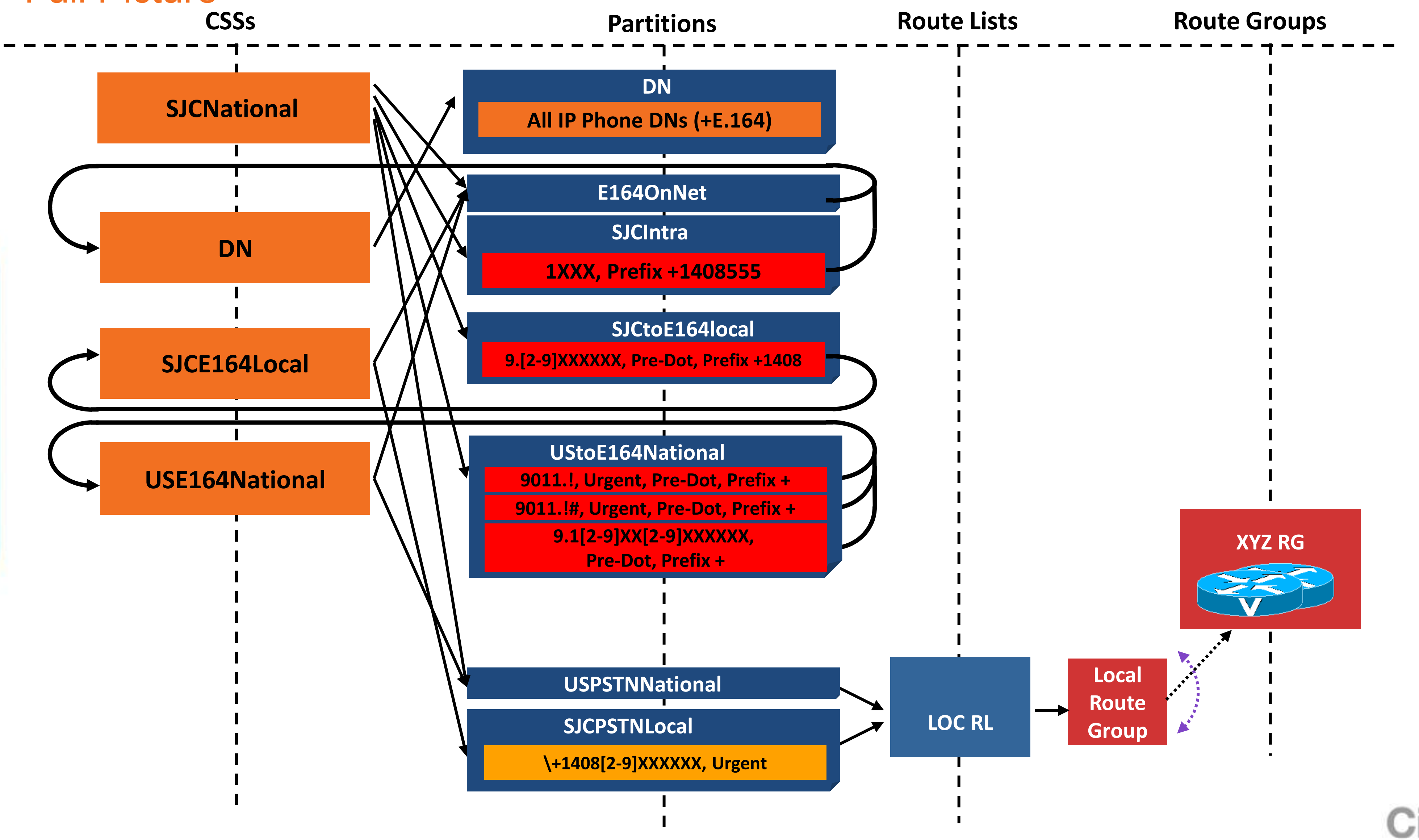
CoS National

Based on CoS International



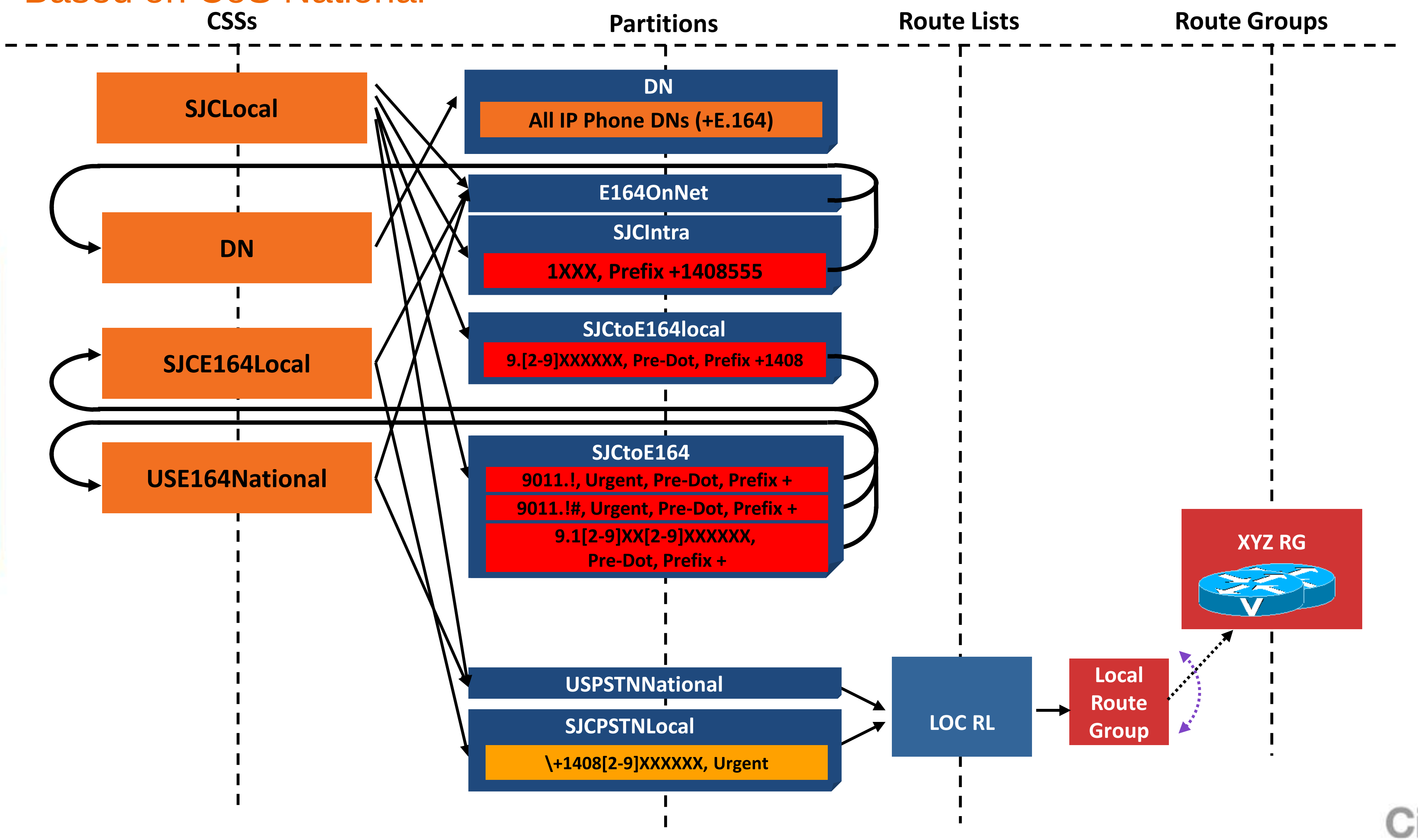
CoS National

Full Picture



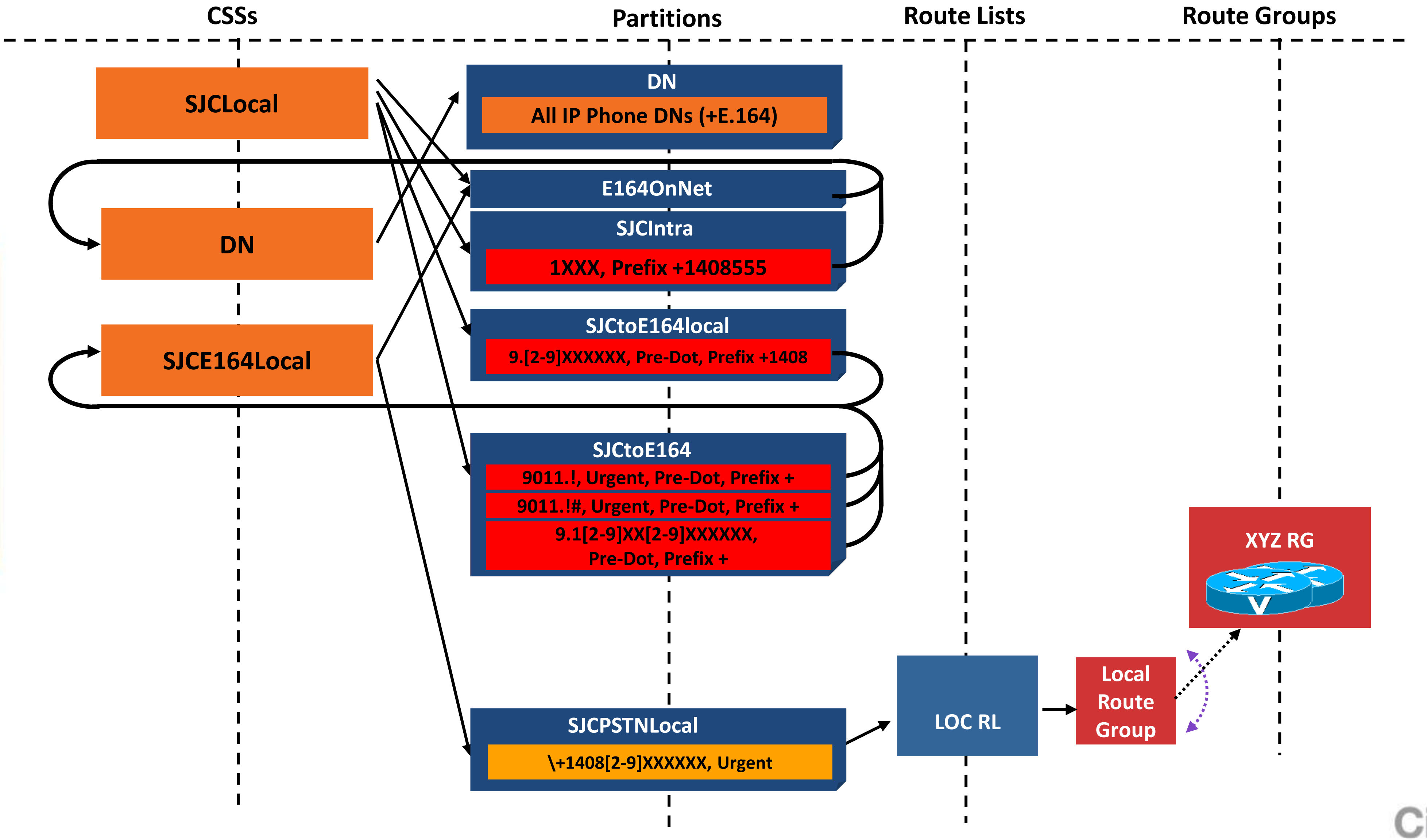
CoS Local

Based on CoS National



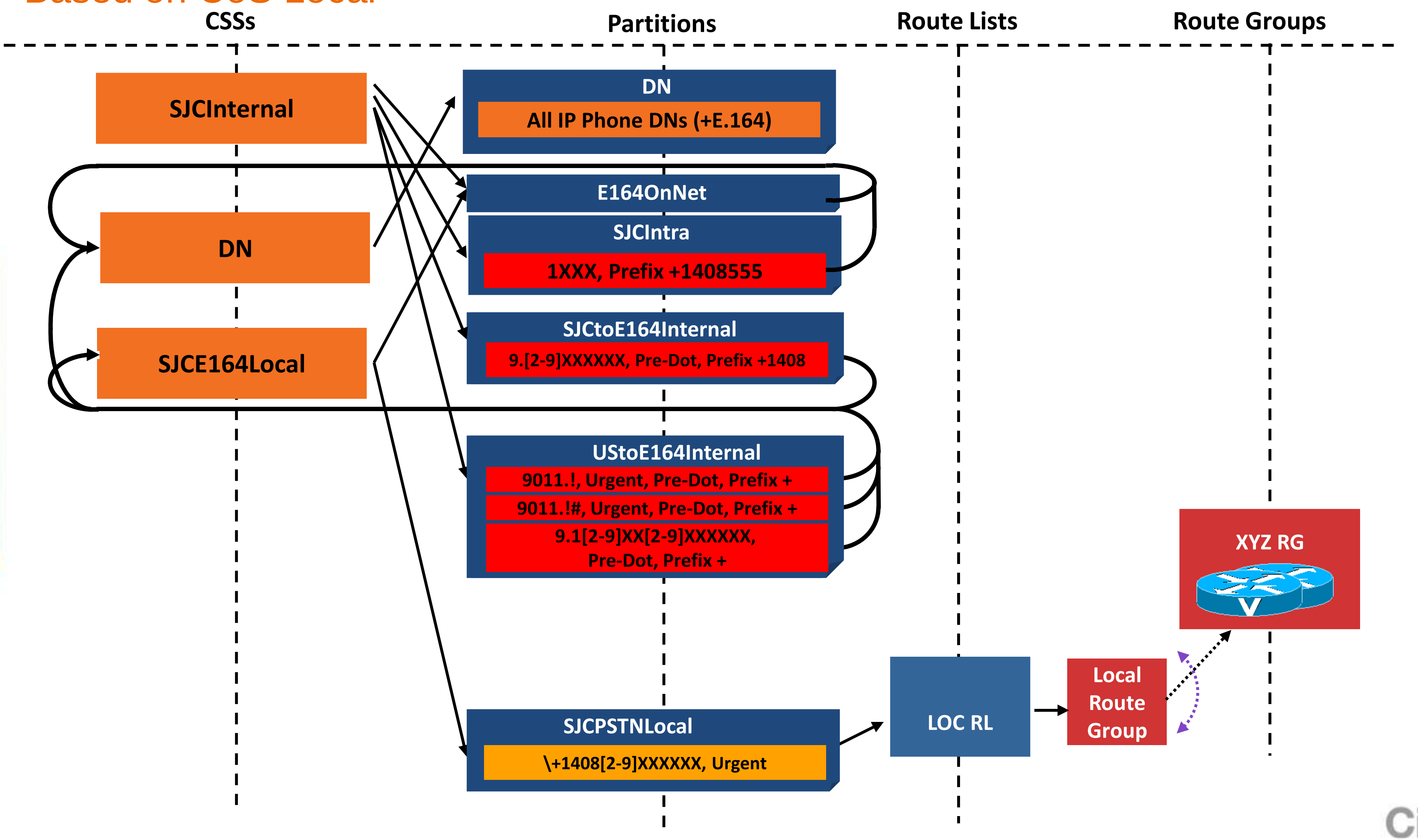
CoS Local

Full Picture



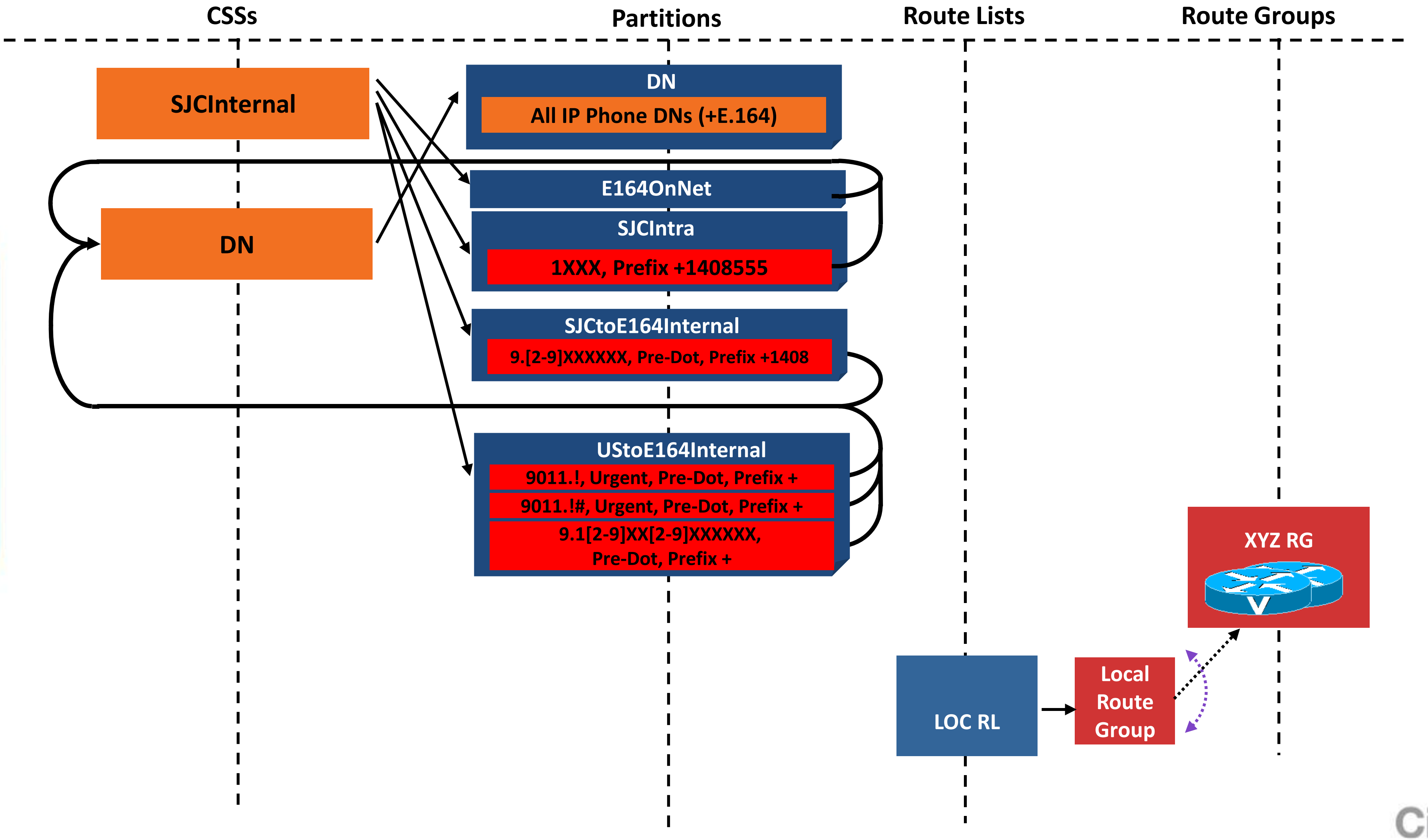
CoS Internal

Based on CoS Local



CoS Internal

Full Picture



Remember

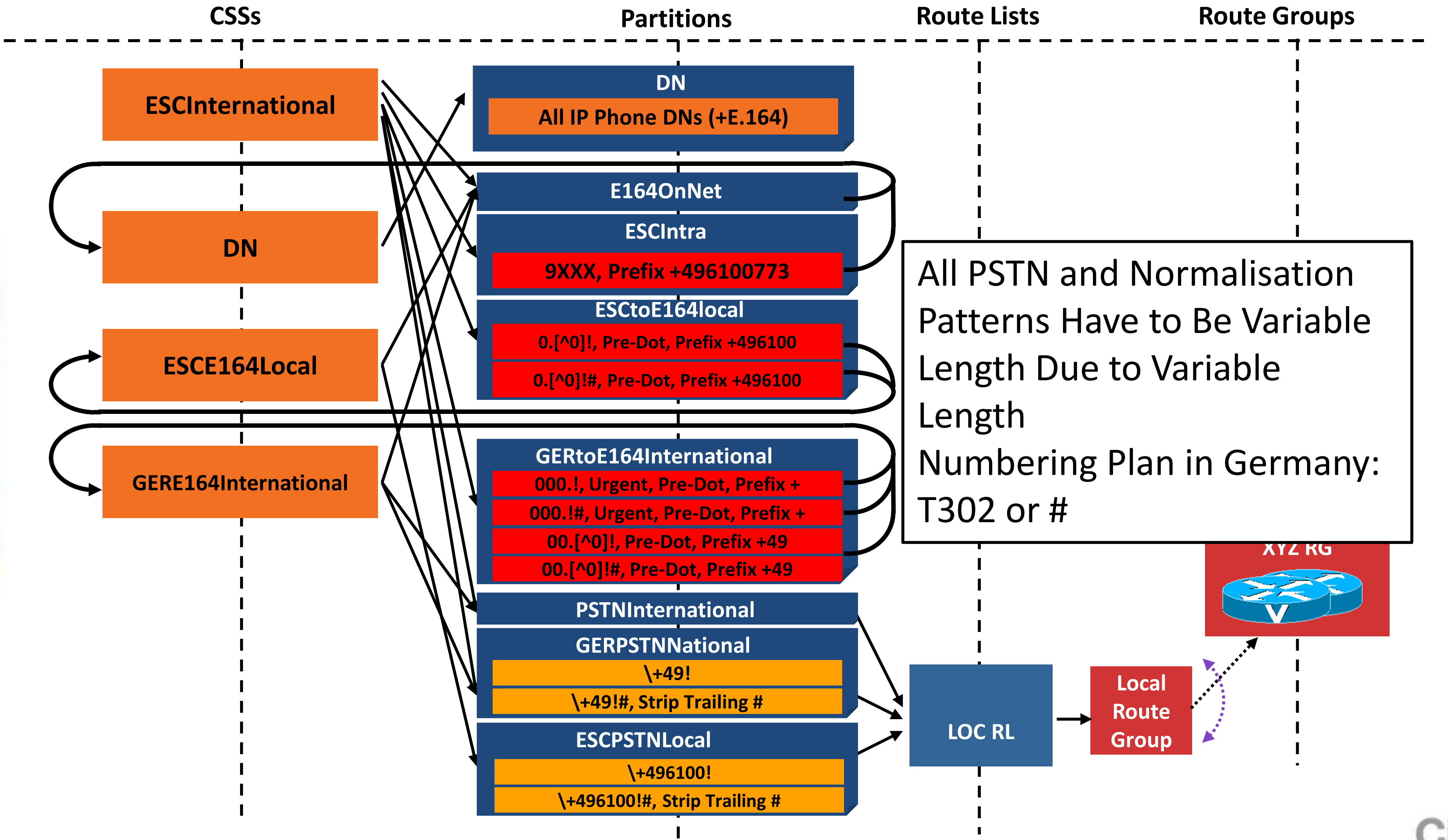
- Translation patterns used to normalise dialing to +E.164
  - Because TPs' resulting CSS implements new CoS (does not inherit the initial CoS), we need normalisation per CoS
- Non urgent DNs: Need to create urgent translation patterns to avoid T302 based on overlap between DNs and variable length PSTN route patterns

Other Dialing Domains (Germany)

- Dialing normalisation needs to be adapted to national dialing habits
- Need to create:
  - GERtoE164International
  - GERtoE164National
  - GERPSTNNational
- Site specific dialing normalisation and local dialing normalisation also need to reflect national dialing habits

CoS International (Germany)

Full Picture



Inbound Routing on Gateways

- Internal DNs are +E.164
- Format of received called party number is provider and technology depending
- Route after globalising to +E.164 on ingress
- Options
  - Incoming Called Party Settings: Prefixes and CSSes per number type (not on MGCP gateways and SIP trunks)
  - Inbound calls CSS; Translation Patterns to get to +E.164

Inbound Routing on Gateways

Incoming Called Party Settings

- H.323 Gateway, H.323 trunk
- Prefix or transformation CSS per type
  - Transformation CSS not used for call routing only for number transformations!
- Example: PSTN gateway in site ESC

Incoming Called Party Settings

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

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
National Number	+49	0	< None >	✓
International Number	+	0	< None >	✓
Unknown Number	Default	0	< None >	✓
Subscriber Number	+496100	0	< None >	✓

Emergency Calls

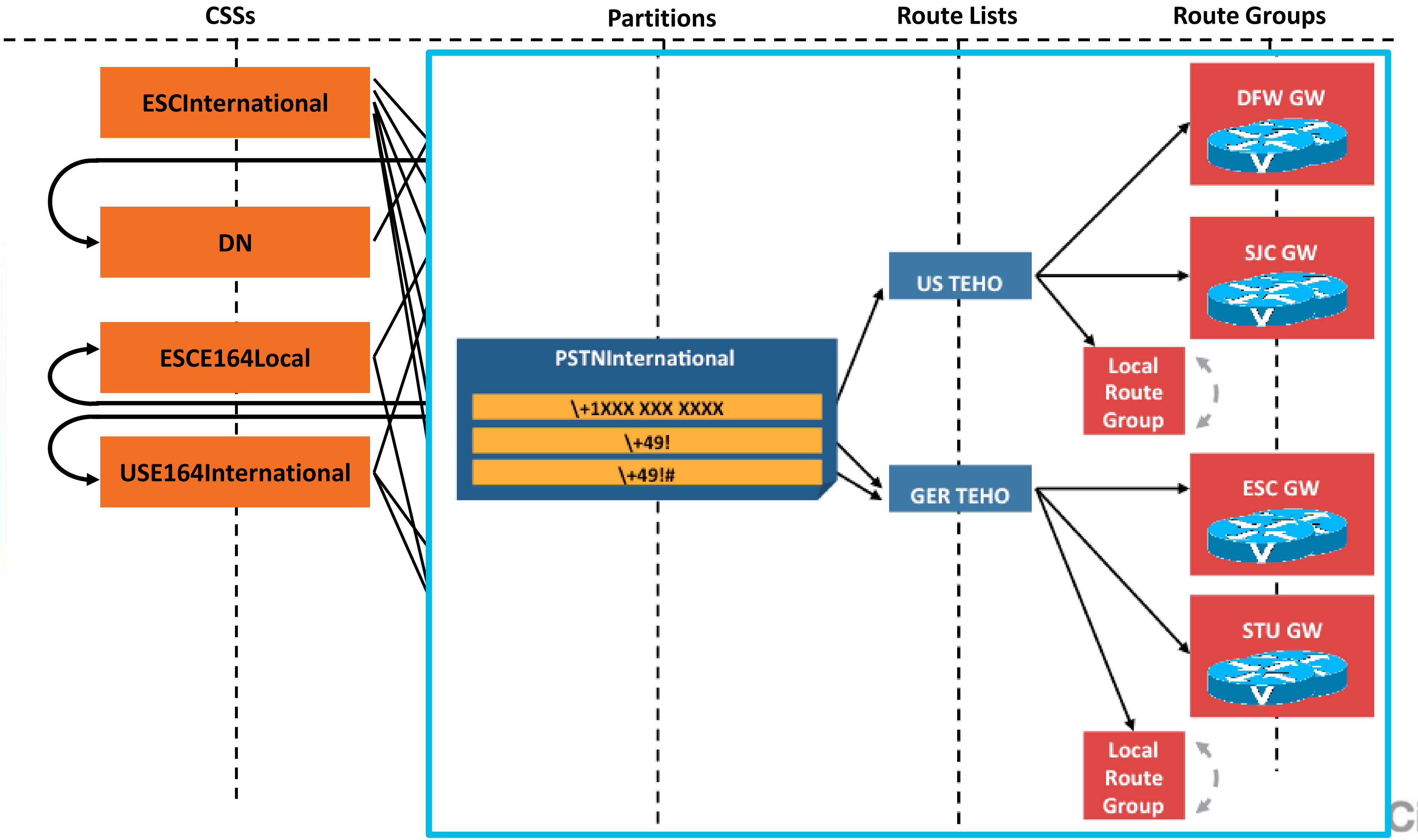
- Emergency Calls need to be enabled for ALL classes of service
- Emergency Calls need to be routed through an egress gateway local to the caller
- Different Emergency Numbers:
  – US: 911
  – Europe: 112
  – Other...
- Options:
  – Put emergency pattern in device CSS
  – Add emergency partition to all CoS CSSes

Tail-End-Hop-Off

- Business case for national TEHO difficult
- Caller ID preservation?
  - CLIP No Screening
- National restrictions for international TEHO?
- TEHO implemented through specific route pattern overlays

International TEHO

Full Picture



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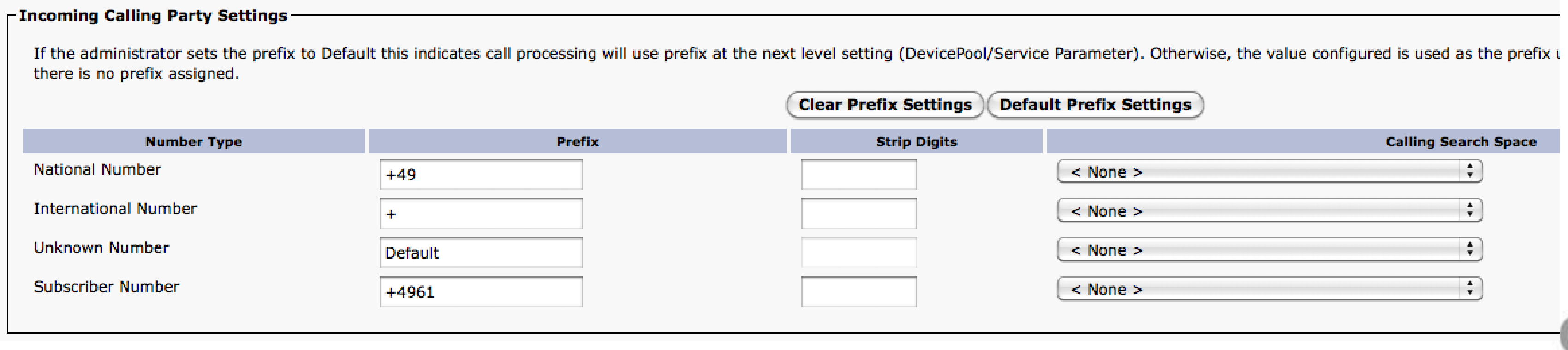
Calling/Called Number Transformations

What It Is: Concept

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

Globalise on Ingress

- Goal is to get to +E.164
- Service Parameter:
  - Prefixes per type for H.323, MGCP and SIP (unknown only)
  - Not recommended
- Device Pool
  - Prefixes or CSSes per number type
- Gateway/Trunk
  - Prefixes or CSSes per number type (only “unknown” on SIP trunks); Example: Gateway for ESC


Incoming Calling Party Settings

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there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	+49		< None >
International Number	+		< None >
Unknown Number	Default		< None >
Subscriber Number	+4961		< None >

Localise on Phones

- Transform Calling Party Number to shortest possible format
- Example for SJC phones (+1 408 555 1XXX):

Calls from	Display as
+1 408 555 1XXX	1XXX
+1 XXX XXX XXXX	91 XXX XXX XXXX or XXX XXX XXXX
+XX...	9011XX... or +XX...

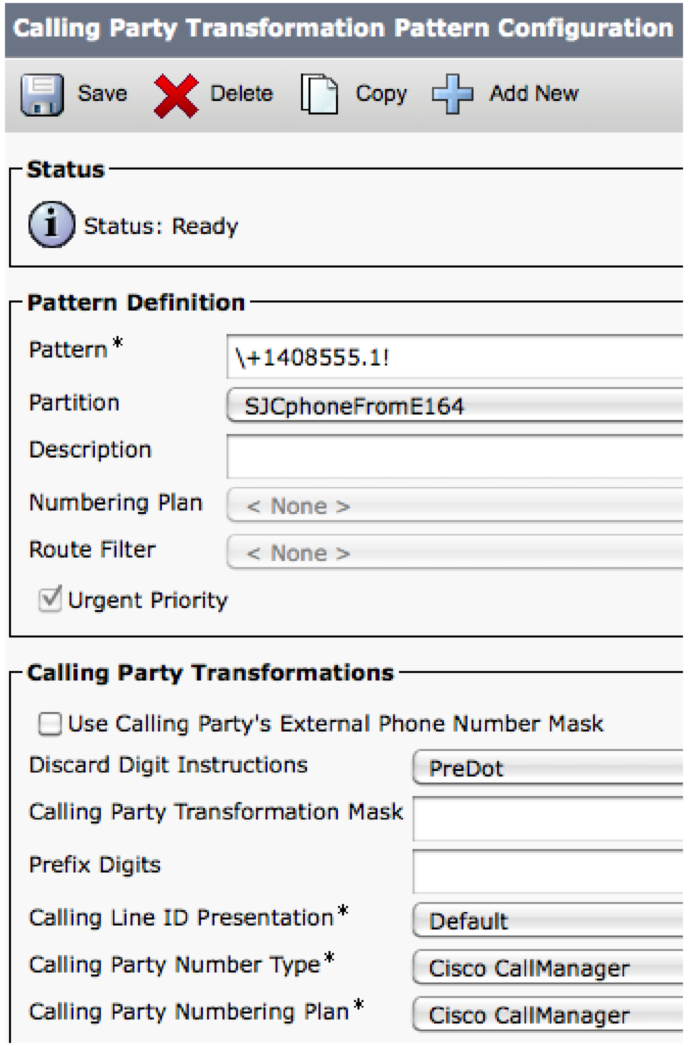
Is This a Problem?

- Callback from missed calls directory goes to pre-transformation number! (globalised number)
- Displayed number does not need to be dialable

Number Transformations

Calling Party Transformation Pattern

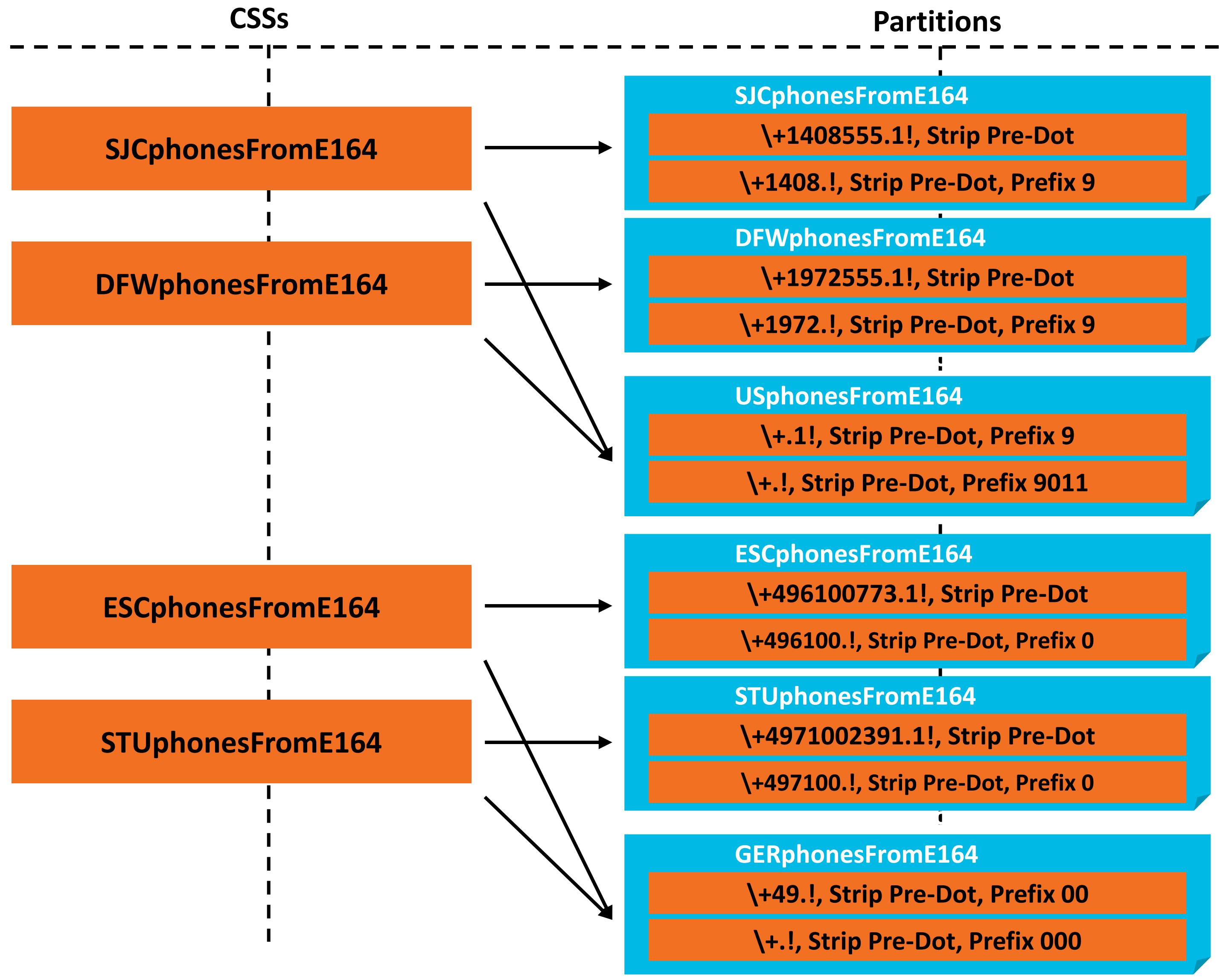
- Similar to translation pattern, but matches on calling (not CALLED) party number
- Only allow calling party transformations
- No impact on call routing
- Addressed by partitions and CSSes (like regular patterns)



Calling Party Normalisation

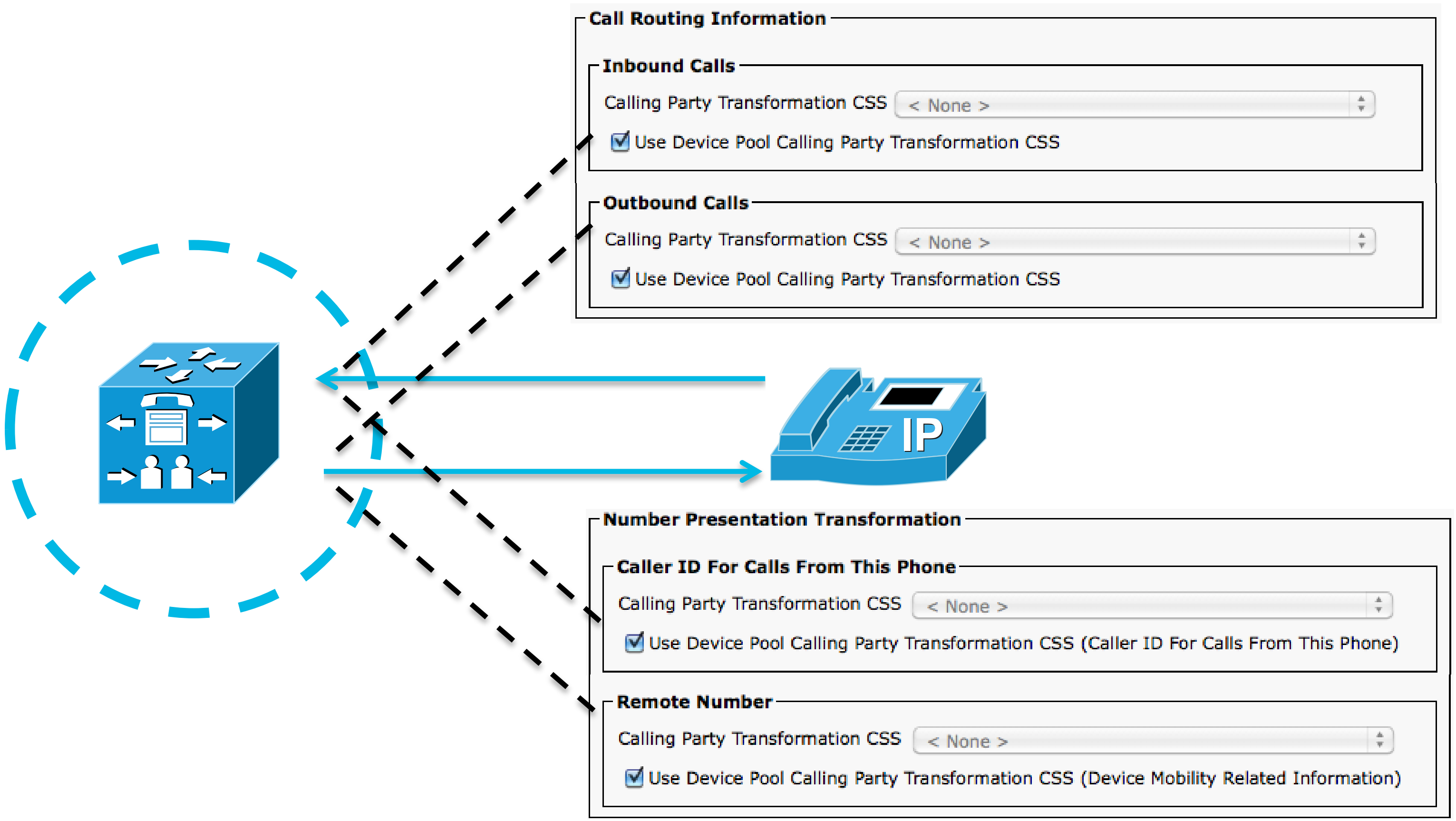
From +E.164 to Shortest Presentation

For Your Reference



Endpoint Calling Party Transformations

Naming conventions



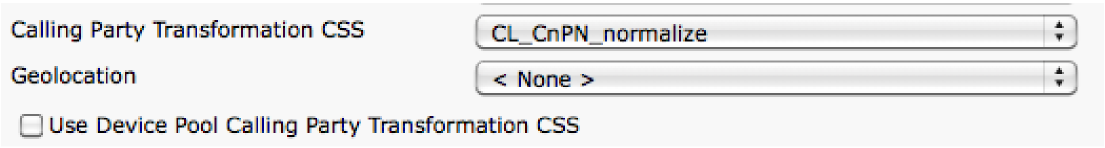
Version 9.0

Naming of transformation CSSes on endpoints changed with version 9.1

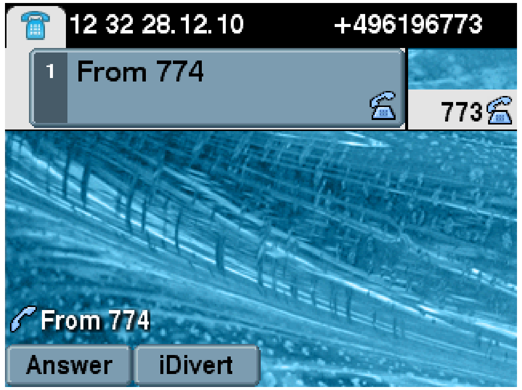
Version 9.1

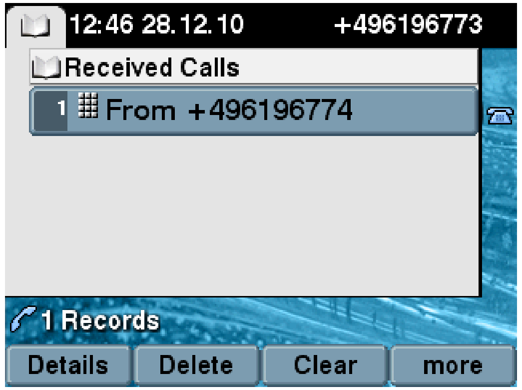
Phone Directories

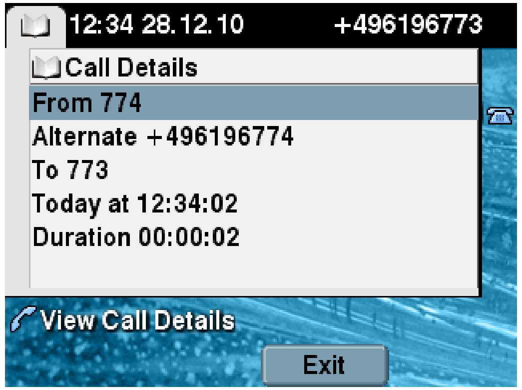
- Calling Party Numbers are transformed using phone’s (or device pool’s) calling party transformation CSS



- But: pre-transformation number is stored in missed calls directory and used for callback
- Concept: Pre-transformation calling party numbers should be „globalised“
  → globalise on ingress, localise on egress
- Globalised numbers (pre-transformation) have to be routable! (supported dialing habit)



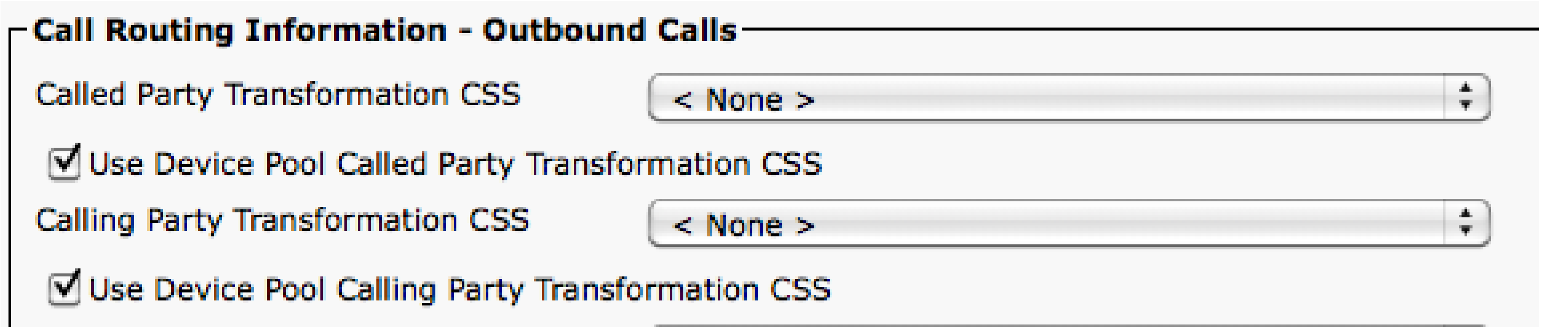




Egress Called Party Normalisation

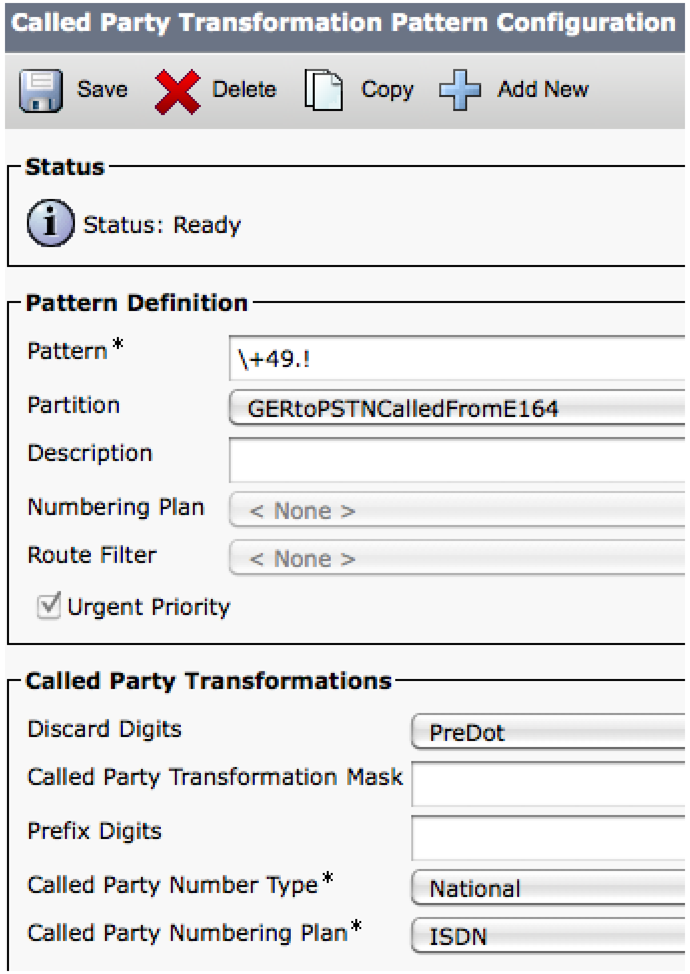
Gateways / Trunks

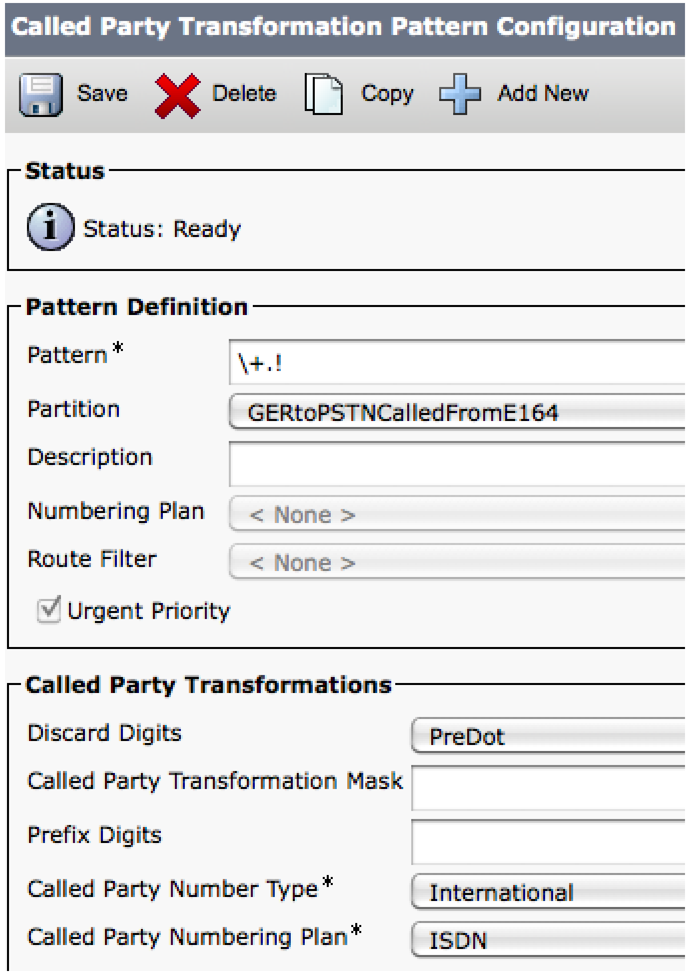
- required format for calling party numbers typically defined by the provider
- use Calling Party Transformation CSS for outbound calls
- Caveat: device level transformations have no effect on Q.SIG APDUs



Egress Called Party Normalisation

Example: German PSTN Gateway





Egress Calling Party Normalisation

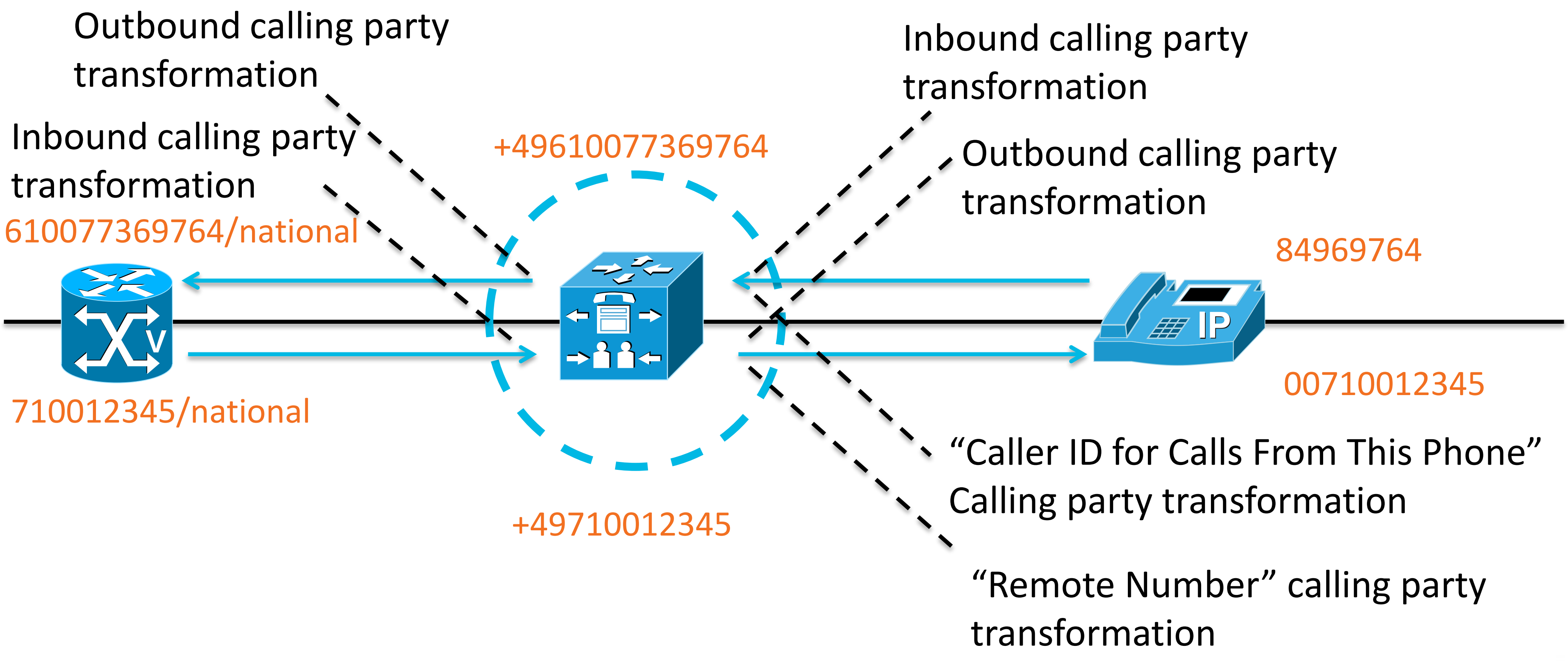
Gateways / Trunks

- Like called party normalisation, but use CALLING party transformation patterns and CSS!
- When using the device pool calling party CSS make sure that device pool is not shared by phones and gateways (typically require different transformations)
- Optional:
  - Filter non-DIDs and send dummy instead
  - Implement screening, if number does not match the number range assigned to the trunk by the provider

End-to-End Calling Party Transformations

Inbound / Outbound calls

Version 9.0



Version 9.1

Agenda

- Introduction
- Call Routing Recap
- Developing a Global Dial Plan – Call Routing
- Developing a Global Dial Plan – Number Presentation
- URI Dialing

SIP URI

what Is It?

- SIP URIs identify communications resources
- general form: sip:user:password@host:port;uri-parameters?headers
- user is optional, but CUCM does not support URIs w/o user
- uri-parameters and headers are optional
- password not recommended
- host: fqdn, ipv4 or ipv6; CUCM does not support ipv6
- user is case sensitive, host is case insensitive:
  - Jkrohn@cisco.com != jkrohn@cisco.com
- 7 bit ASCII only
- example: sip:jkrohn@cisco.com:5060

LHS/user RHS host port

URI routing/dialing

- Why
  - Native dialing method in SIP based video equipment
  - Extend support for SIP video endpoints registered with Communications Manager
  - Unambiguous dialing from directories
  - better integration with other call controls where URI dialing is the native dialing habit (e.g. VCS)
  - Enables easier B2B video call routing
- Limitations
  - URIs can not be used for PSTN calls (as long as there’s no mapping to E.164)
  - Limited endpoint support (+E.164/numbers might still be the native format)

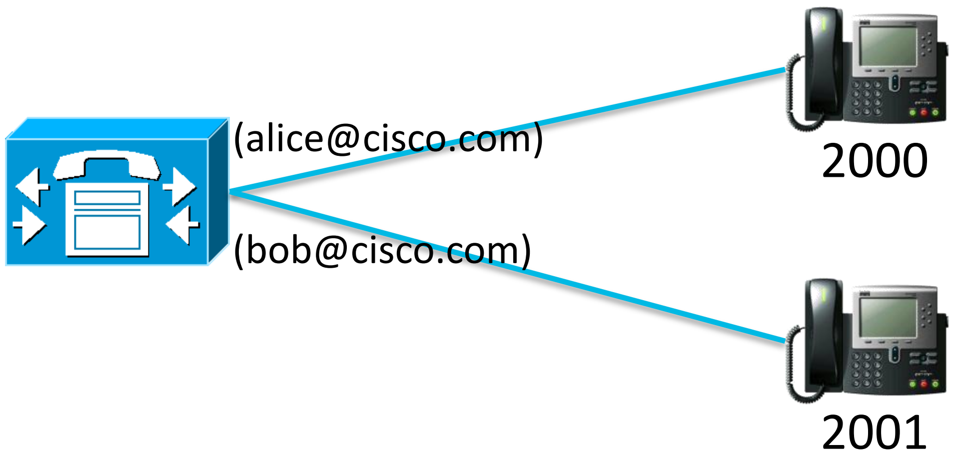
Endpoint Support

- Only a subset of phones support URI dialing
  - 99xx, 8961 phones (except transfer, conferencing, forwarding)
  - Jabber for windows will support URI dialing with 9.0(2)
  - Video Endpoints
- Directory lookups on CUCM currently will always return numbers; dialing from corporate directories will always dial numbers
- All phones can be called via an alpha URI (, because URI is mapped to a DN)

URI Dialing

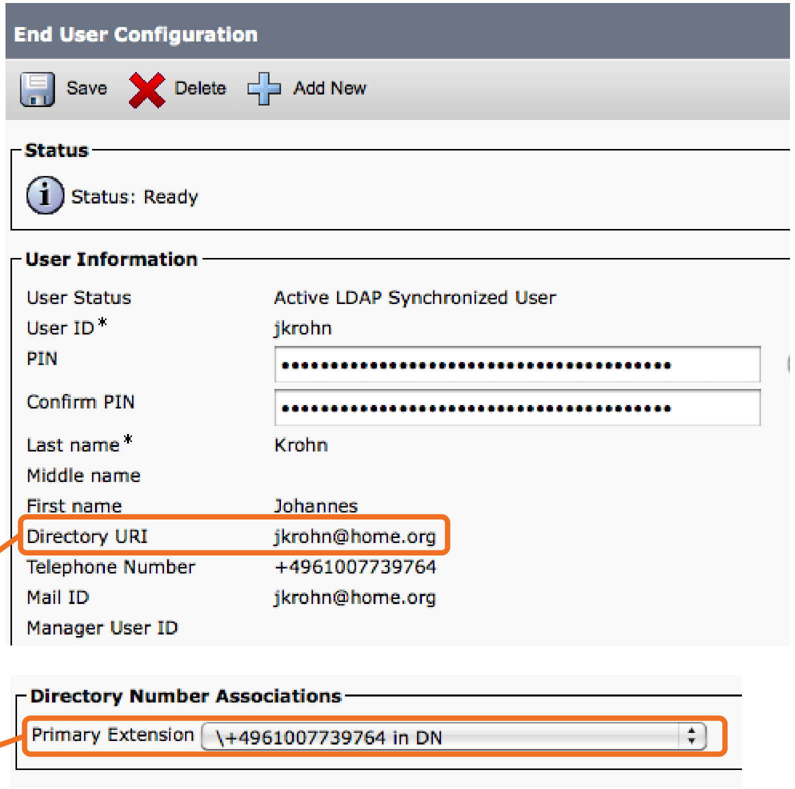
The Concept

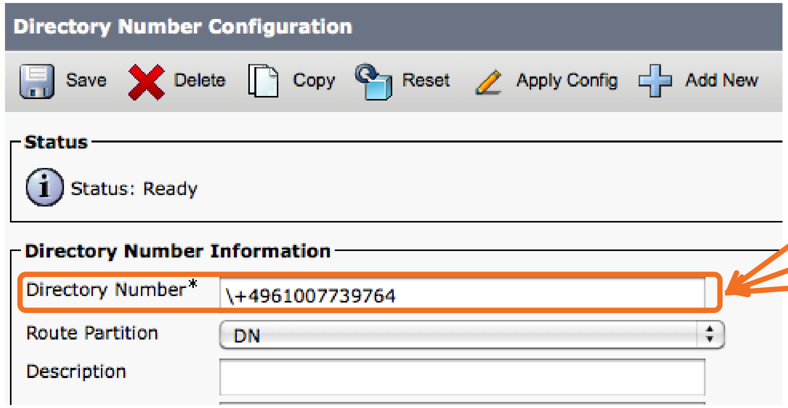
- In CUCM all endpoints will still have a DN
- Alpha URI can be associated with DN on any device (not only SIP)
- Phones always register via the DN (do not necessarily even know that there is an associated alpha URI)

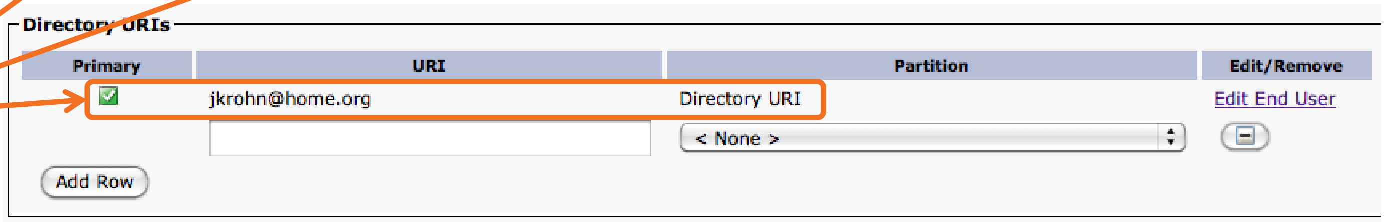


URIs and Directory Numbers

- Up to 5 URIs can be configured per DN
- Enduser’s directory URIs are assigned to directory numbers based on enduser’s primary extension; partition “Directory URI” (cannot be changed/deleted)
- other URIs can be in any partition; no need to have them in the same partition as the DN



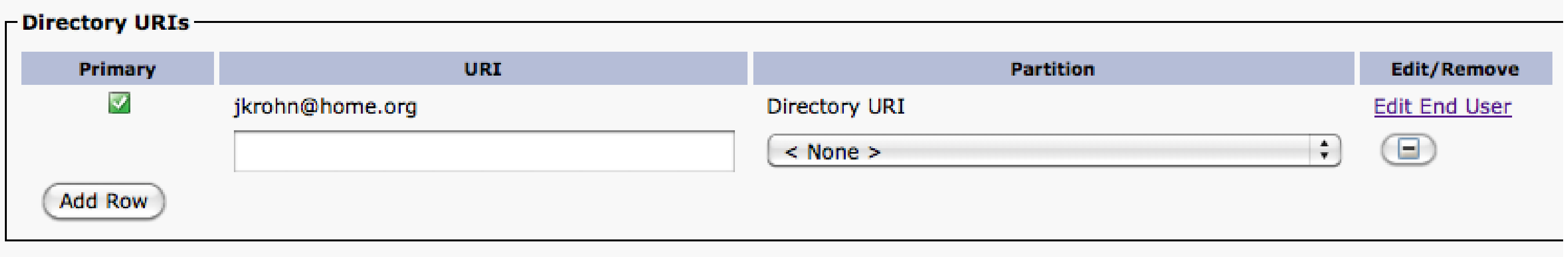




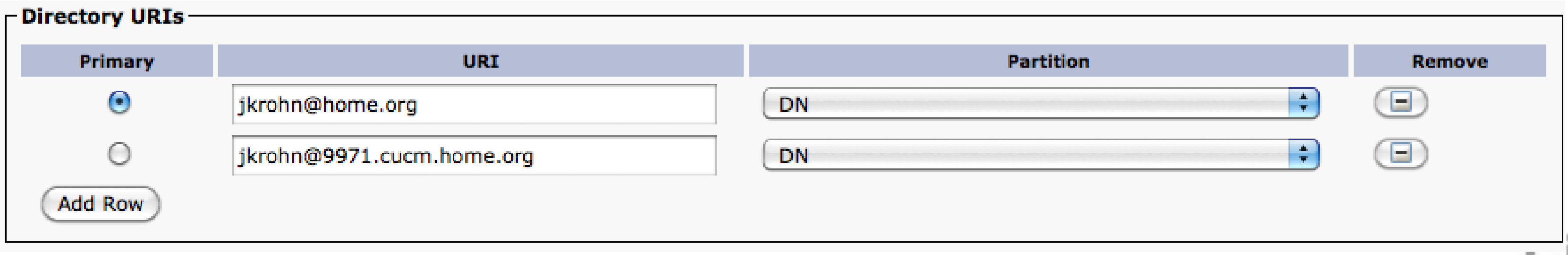
URIs and DNs

Primary URI

- One URI associated with DN is marked the primary URI
- Auto-generated URI based on user's primary extension will always be the primary URI



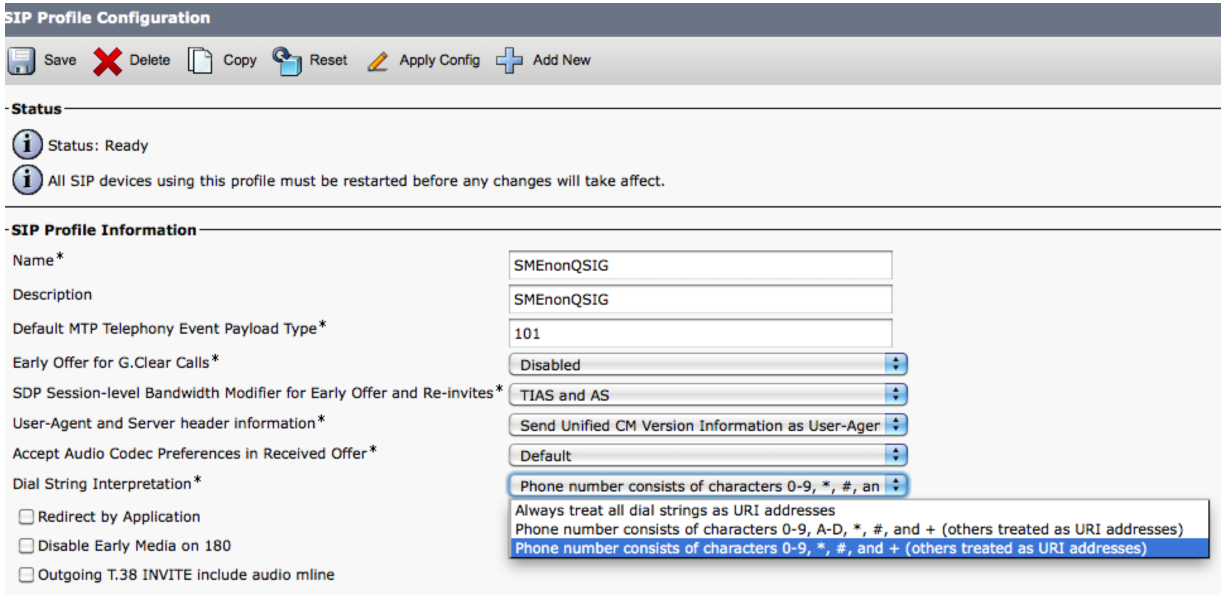
- If no auto-generated URI exists one of the other URIs can be marked "primary"
- Primary URI will be used URI identity for calls from/to this line



Alpha URI vs. Number

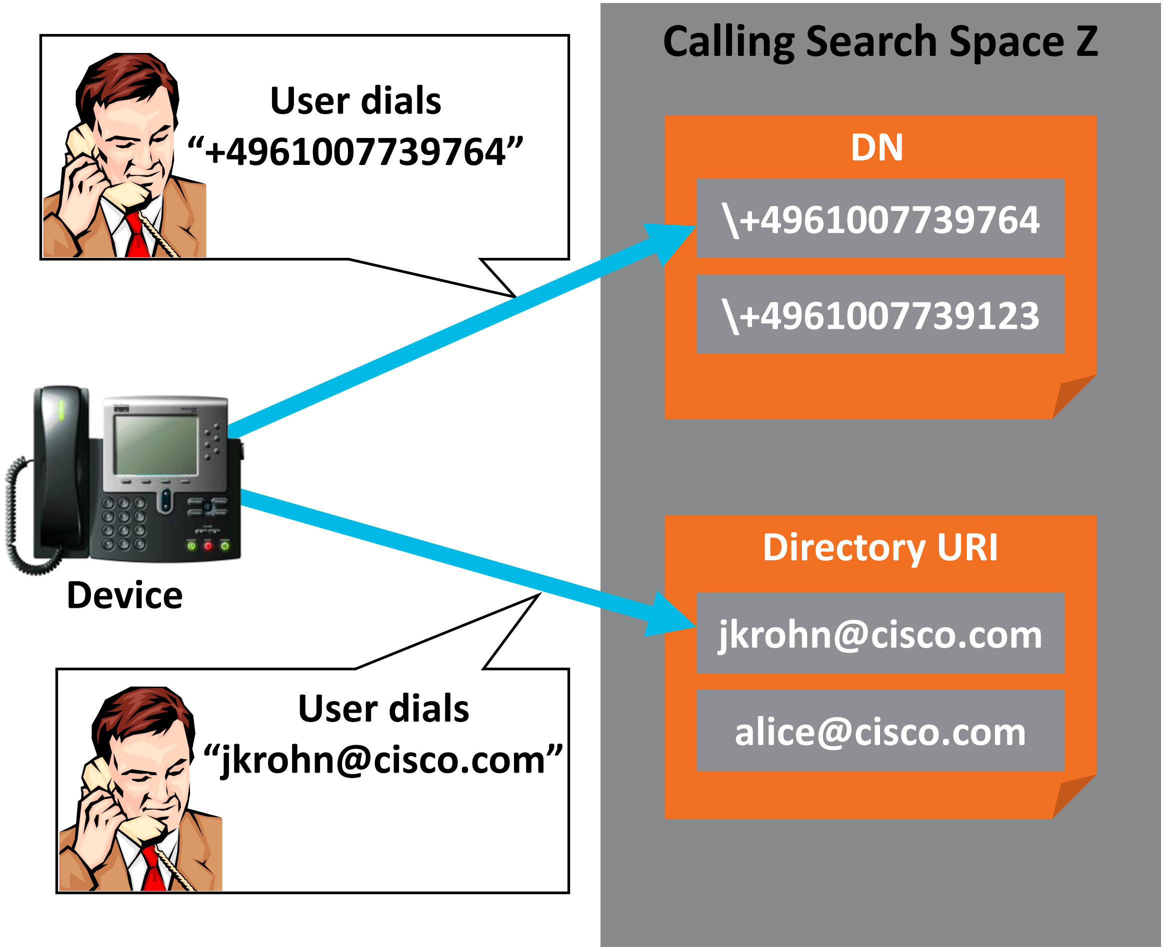
How to Differentiate Between a Number and an Alpha URI

- Dialed "numbers" can contain: +, 0-9, *, A-D
- SIP Profile now has "Dial String Interpretation" setting
- relevant for calls from endpoints and trunks
- Default: 0-9, * and + (Recommended)
- Recommendation: use un-ambiguous alpha URIs
- "user=phone" tag in request URI forces treatment as numeric URI



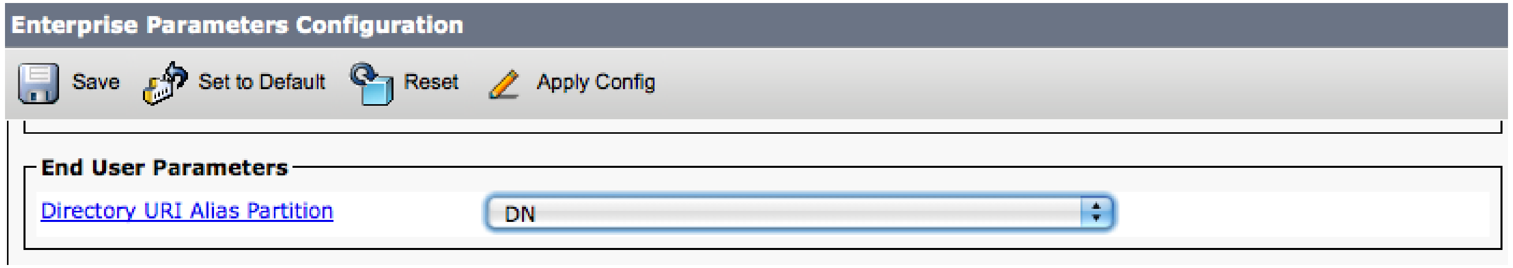
Calling URIs

- URIs can be called if the URIs' partition is member of calling CSS
- CSSs can contain DN and URI partitions
- partitions can contain DNs and URIs
- CSS/partition logic for URIs is identical to DN logic

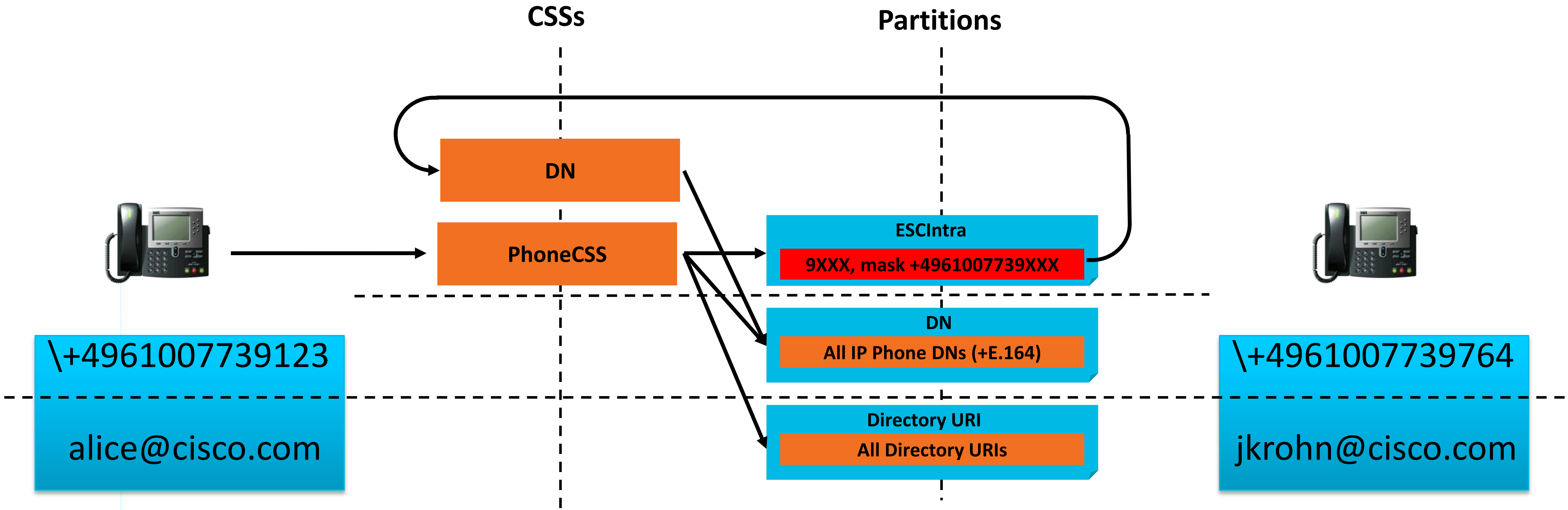


Directory URI Partition Alias

- Autogenerated directory URIs are in partition “Directory URI”
- “Directory URI” partition is predefined and can not be changed/deleted
- to be reachable this partition needs to be member of calling identity’s CSS
- An already existing partition can be defined as alias for “Directory URI” partition
  → URIs in Directory URI partition can be reached by all CSSes which have the alias partition
- Good candidate: already existing DN partition



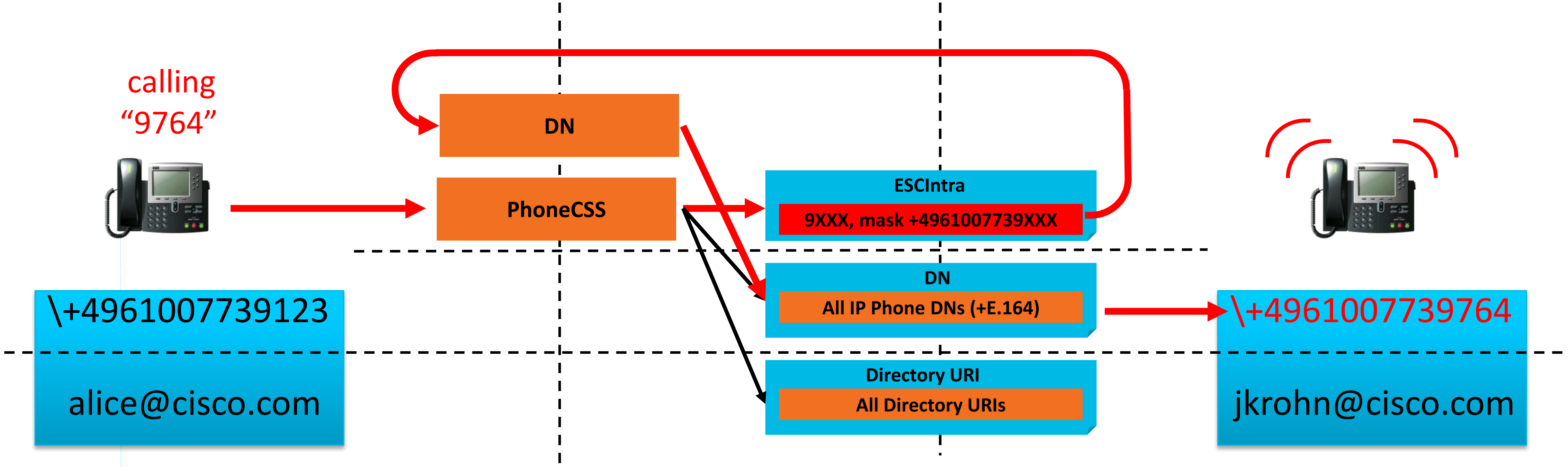
Independent Call Routing



- Typical dial plan e.g. has translation patterns to transform intra-site dialing to DN format
- This translation pattern might also have calling party transformations

Independent Call Routing

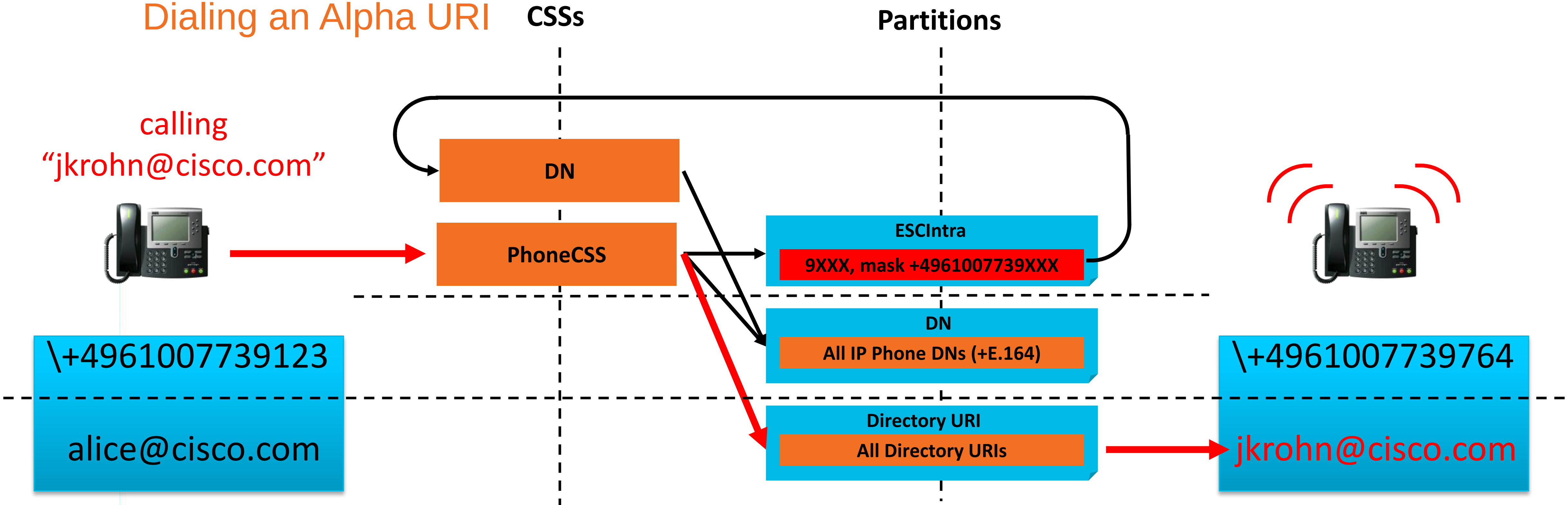
Dialing a Number



- Intra-site dialing is a two-step process (normalise and route)
- Normalisation translation pattern might impose calling party transformations (in addition to called party transformations)

Independent Call Routing

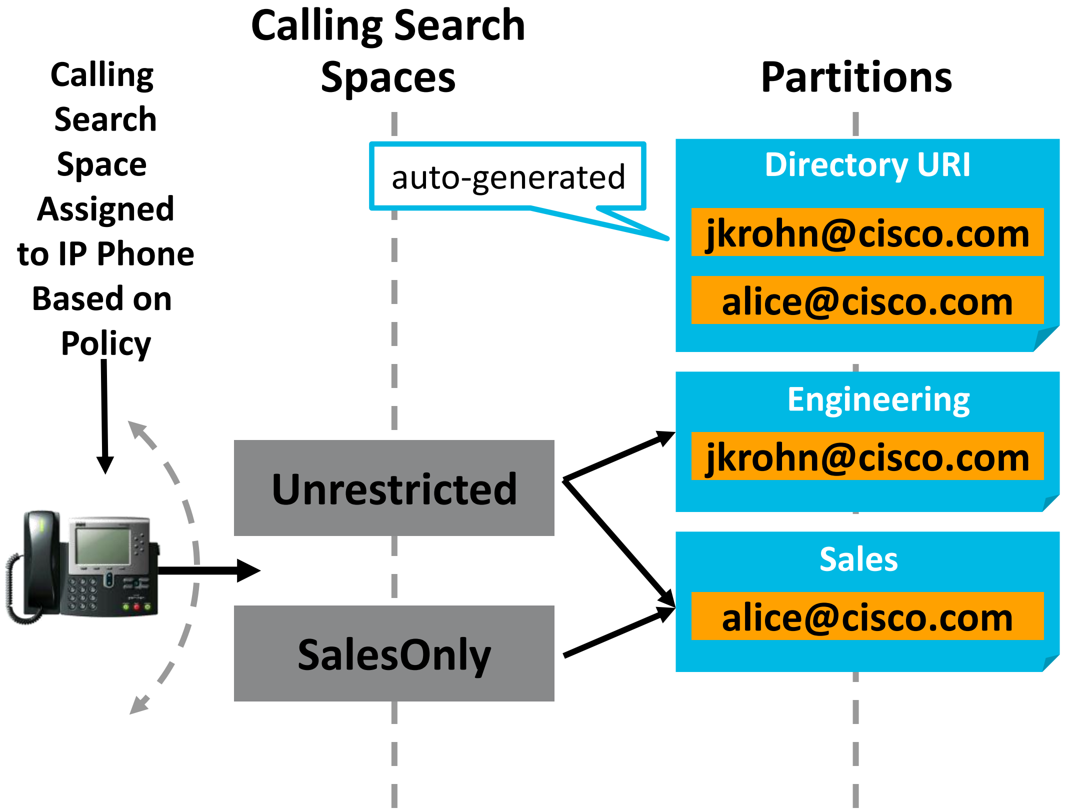
Dialing an Alpha URI



- Calling a URI takes a different path
- URI routing does not have the concept of translation patterns; no equivalence to block patterns
- Only option for calling party transformation is the outbound calls calling party transformation CSS on calling endpoint or calling endpoint’s device pool

Building CoS for URIs

- Default "Directory URI" partition will have ALL auto-generated user based URIs
- No way to differentiate different user groups based on auto-generated user based URIs
- If different user groups are required you need to explicitly provision the URIs in user group specific partitions and create appropriate CSSes

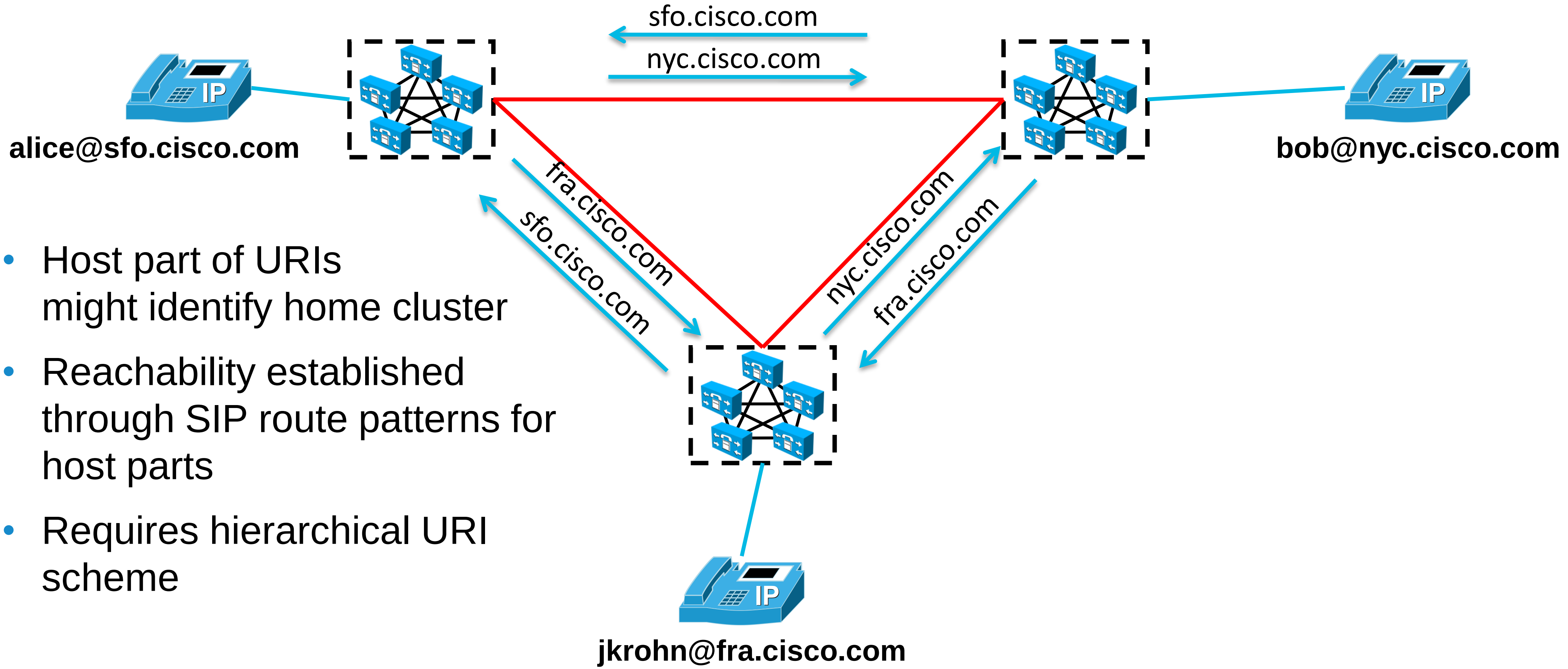


Directory URIs

Primary	URI	Partition	Edit/Remove
☑	jkrohn@home.org	Directory URI	Edit End User
	jkrohn@home.org	Engineering	⊟

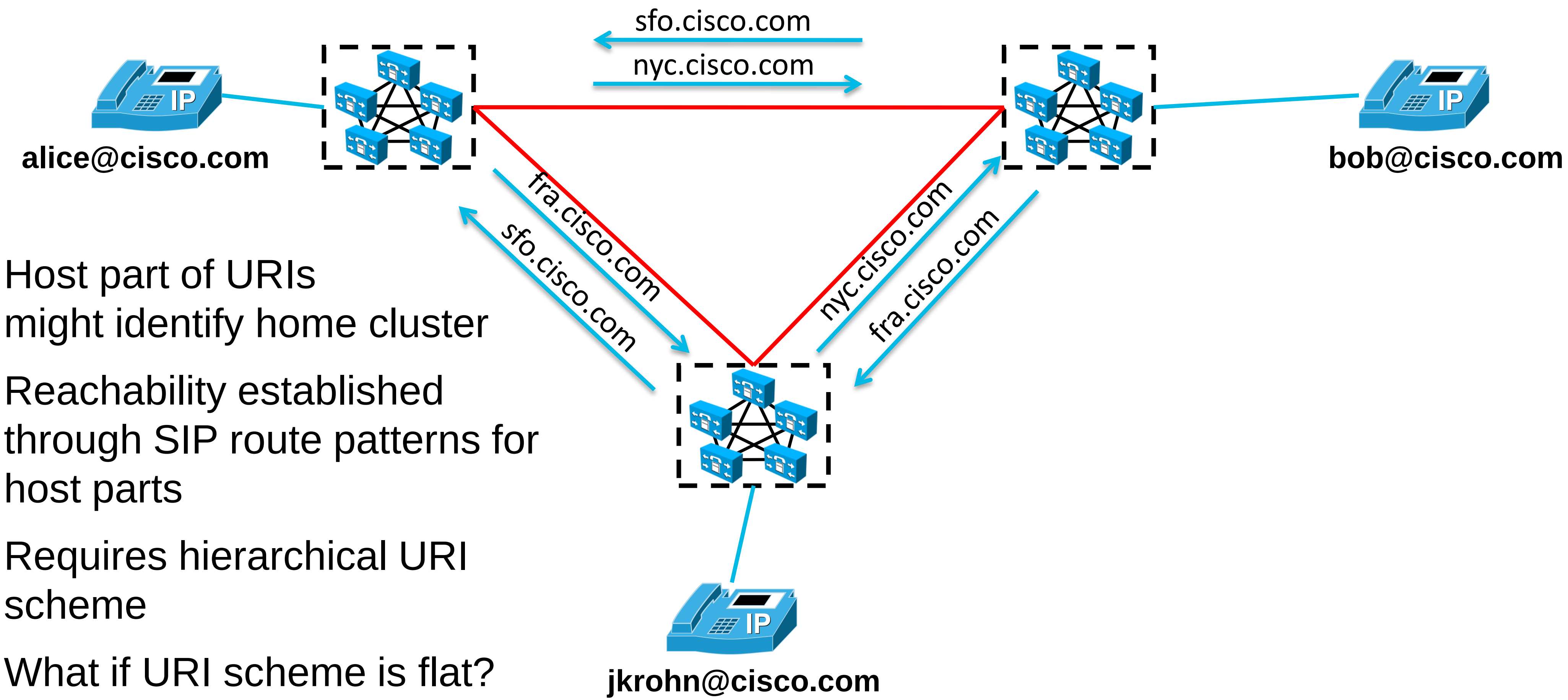
Add Row

Multicluster URI routing



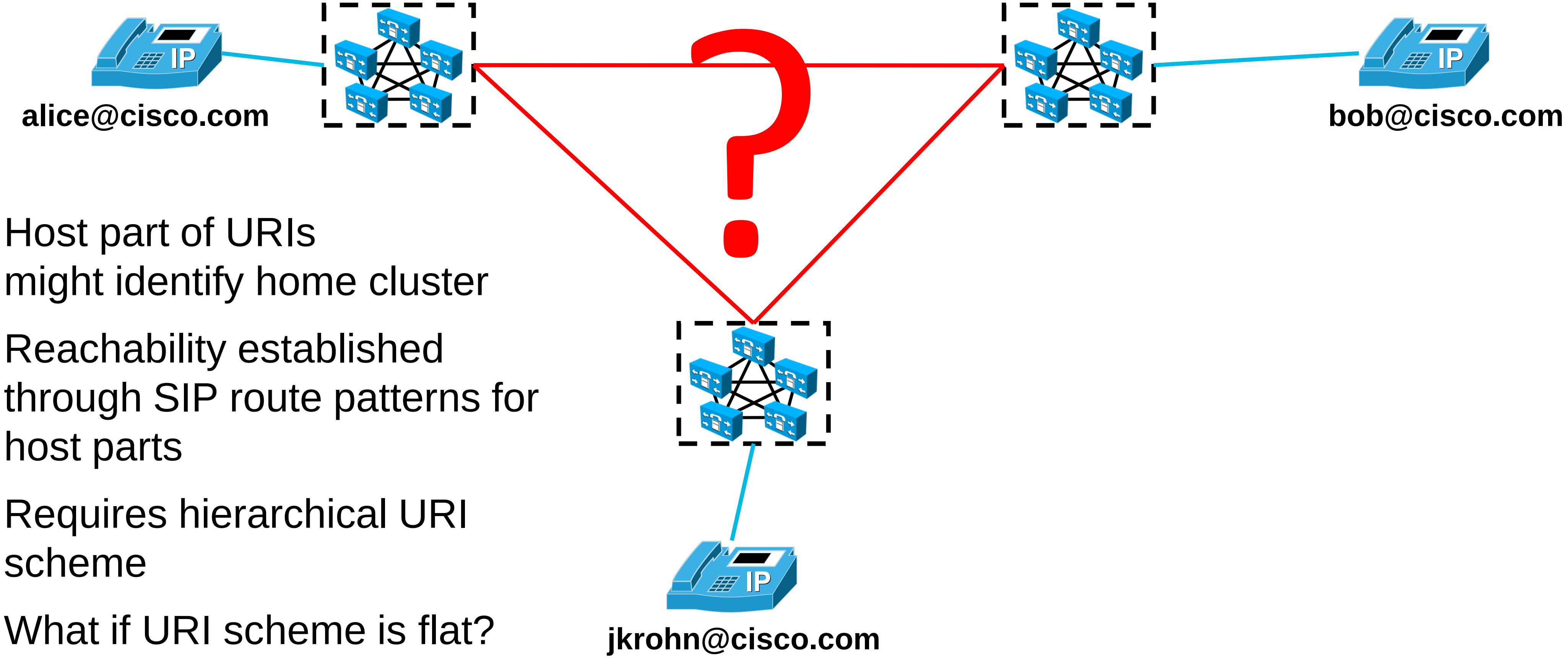
- Host part of URIs might identify home cluster
- Reachability established through SIP route patterns for host parts
- Requires hierarchical URI scheme

Multicluster URI routing



- Host part of URIs might identify home cluster
- Reachability established through SIP route patterns for host parts
- Requires hierarchical URI scheme
- What if URI scheme is flat?

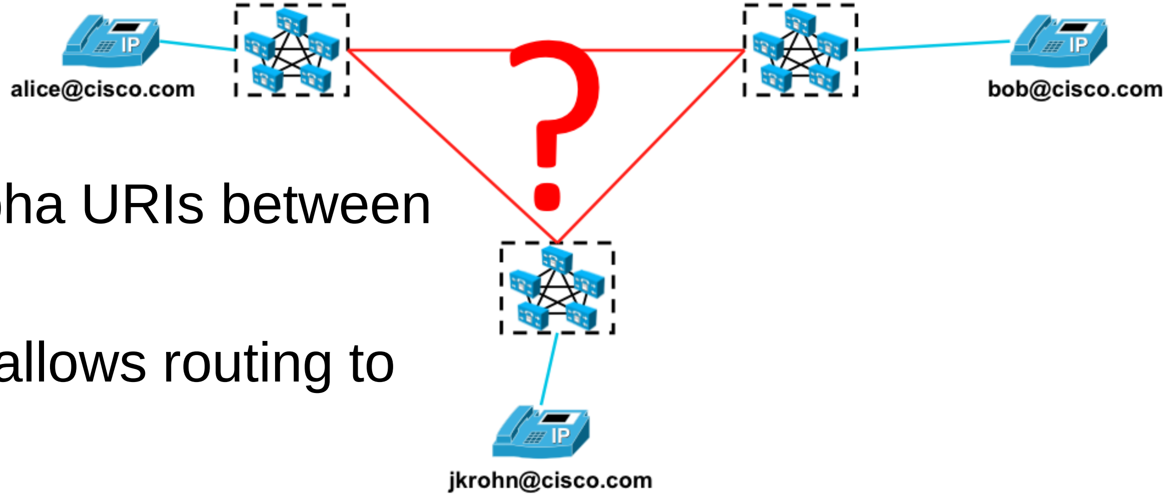
Multicluster URI routing



- Host part of URIs might identify home cluster
- Reachability established through SIP route patterns for host parts
- Requires hierarchical URI scheme
- What if URI scheme is flat?

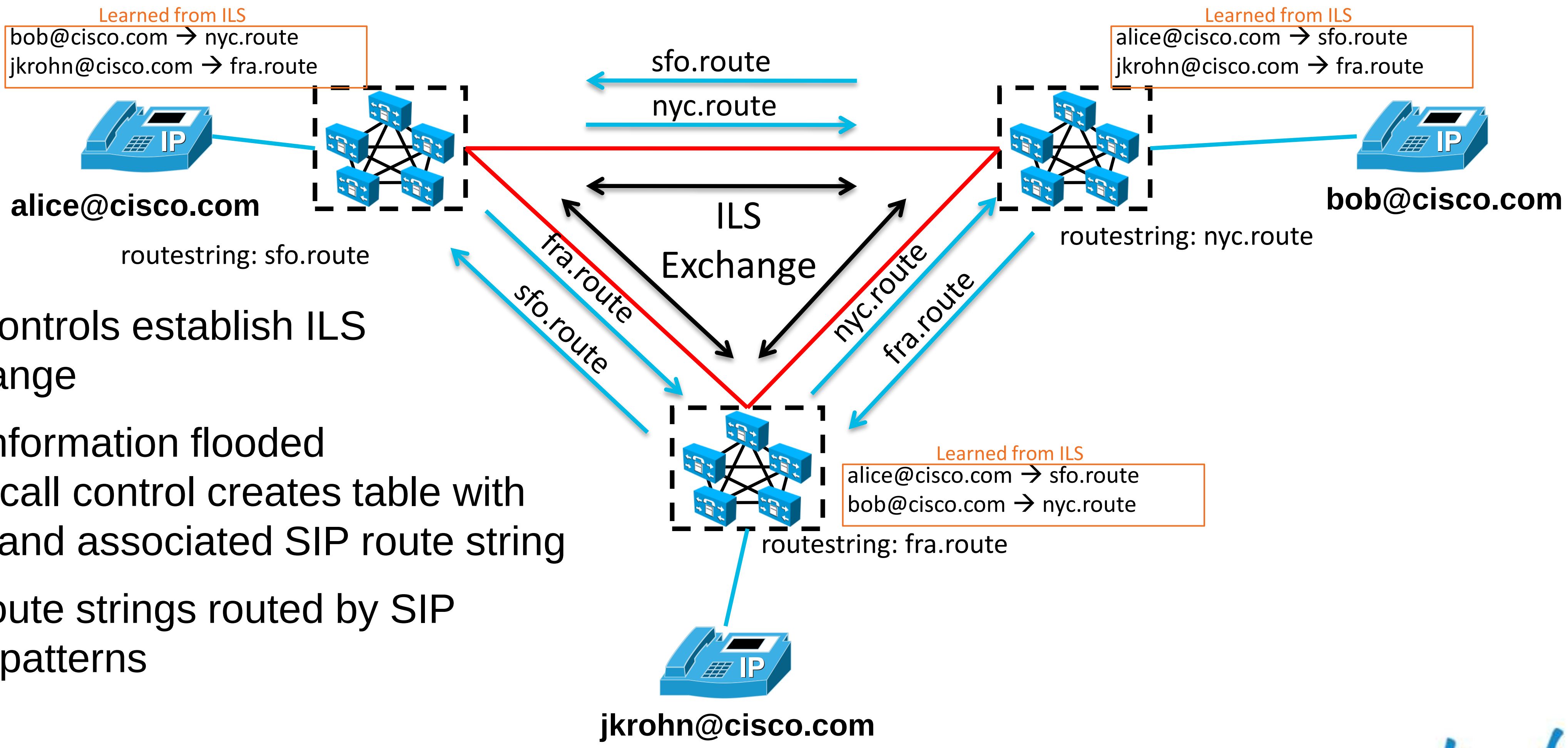
Intercluster Lookup Service (ILS)

Fundamental idea



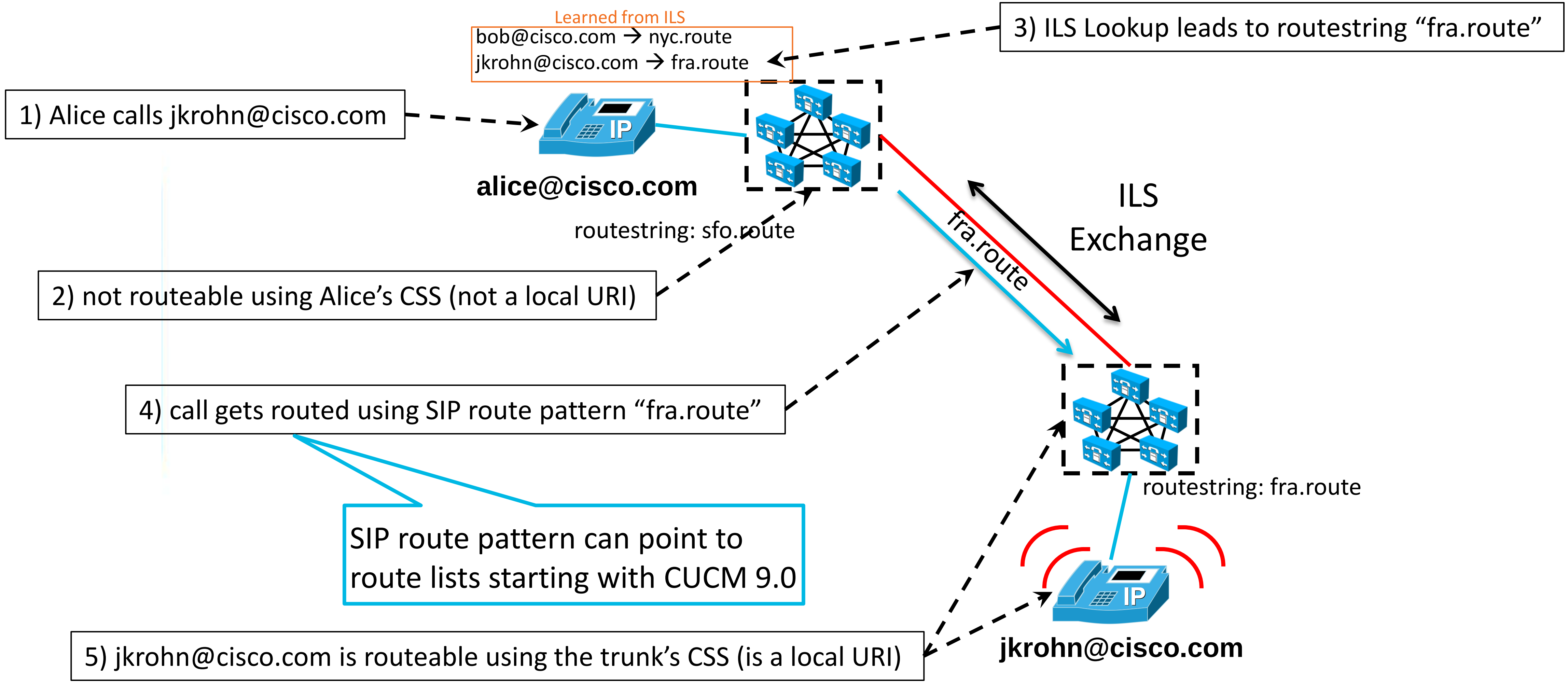
- Need mechanism that
  - allows propagation of individual alpha URIs between call controls
  - binds alpha URI with attribute that allows routing to URI's home cluster
- ILS
  - each call control replicates it's alpha URIs to it's neighbours
  - each call control also announces "SIP route string" together with the alpha URIs
  - "SIP route string" can be routed based on SIP route patterns → intercluster routing of alpha URIs not based on URIs' host part, but on SIP route string

ILS Learning

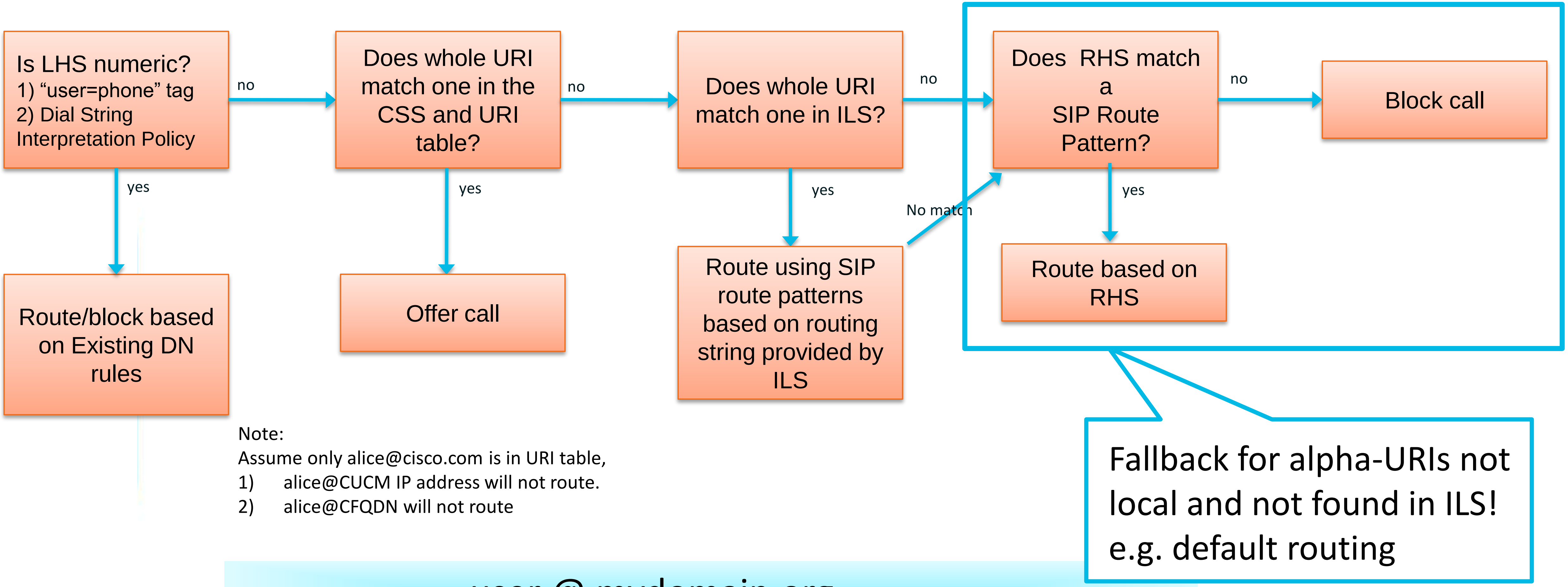


- Call controls establish ILS Exchange
- URI information flooded
  Each call control creates table with URIs and associated SIP route string
- SIP route strings routed by SIP route patterns

Routing Alpha URI Using ILS Information

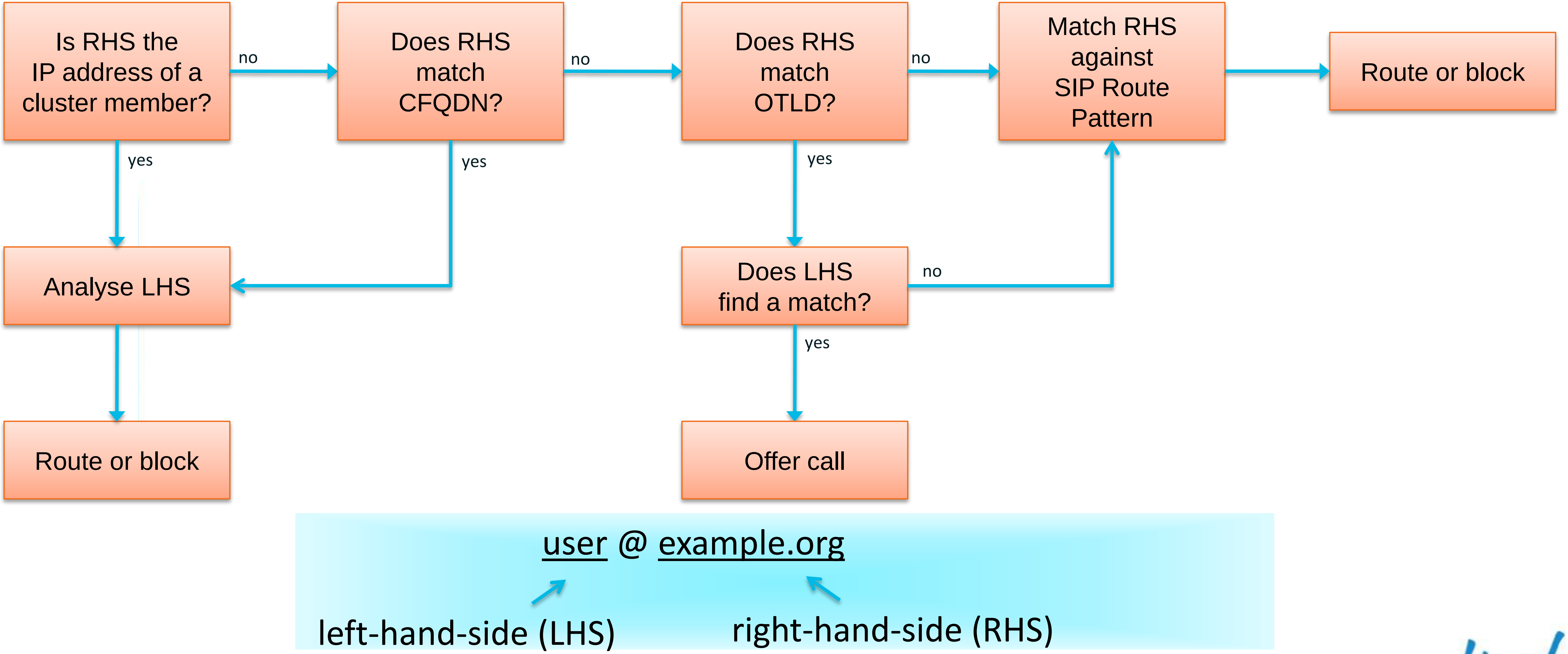


Routing Flowchart



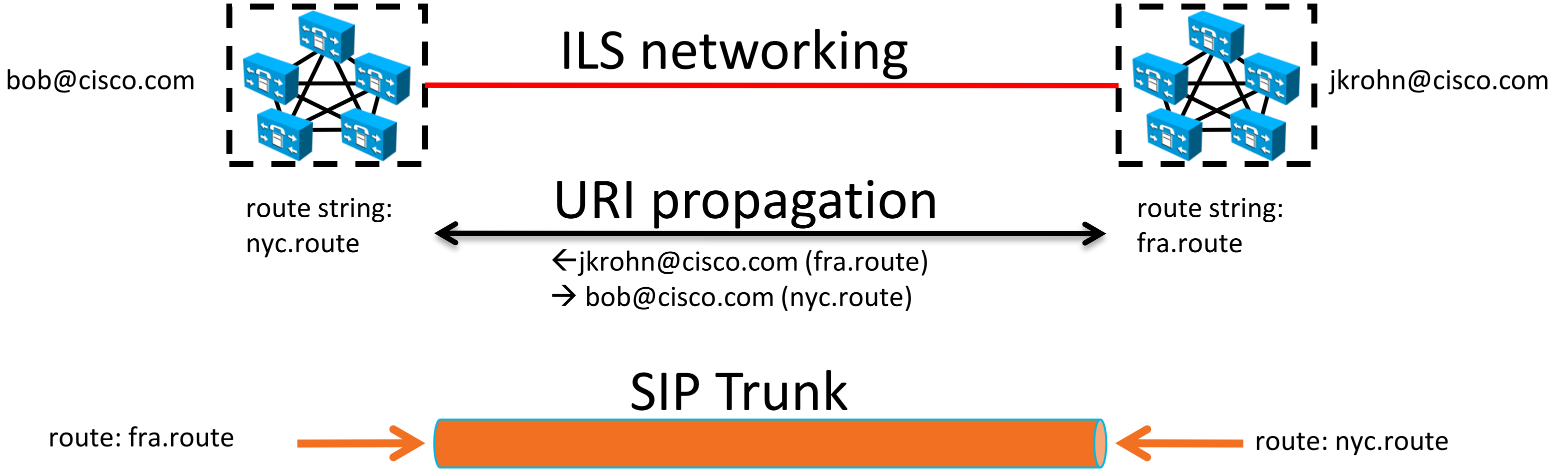
Note:
Assume only alice@cisco.com is in URI table,
1) alice@CUCM IP address will not route.
2) alice@CFQDN will not route

Numeric SIP Request Routing Flowchart



ILS Networking, URI Learning and Routing

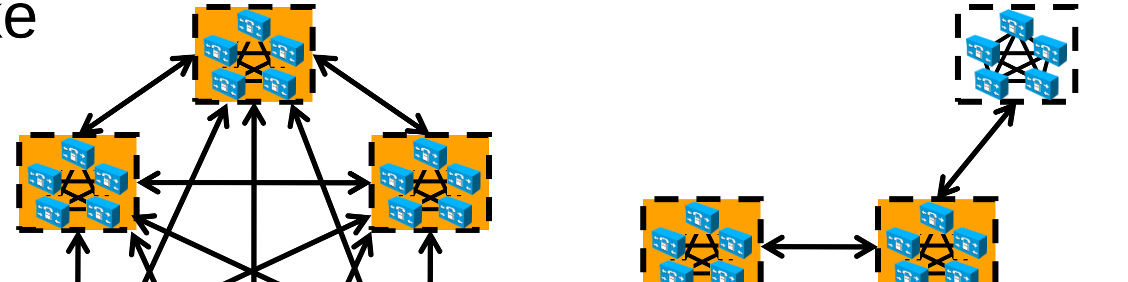
- Components of end-to-end URI dialing/routing
  - ILS networking
  - URI propagation
  - SIP trunk
  - SIP route pattern

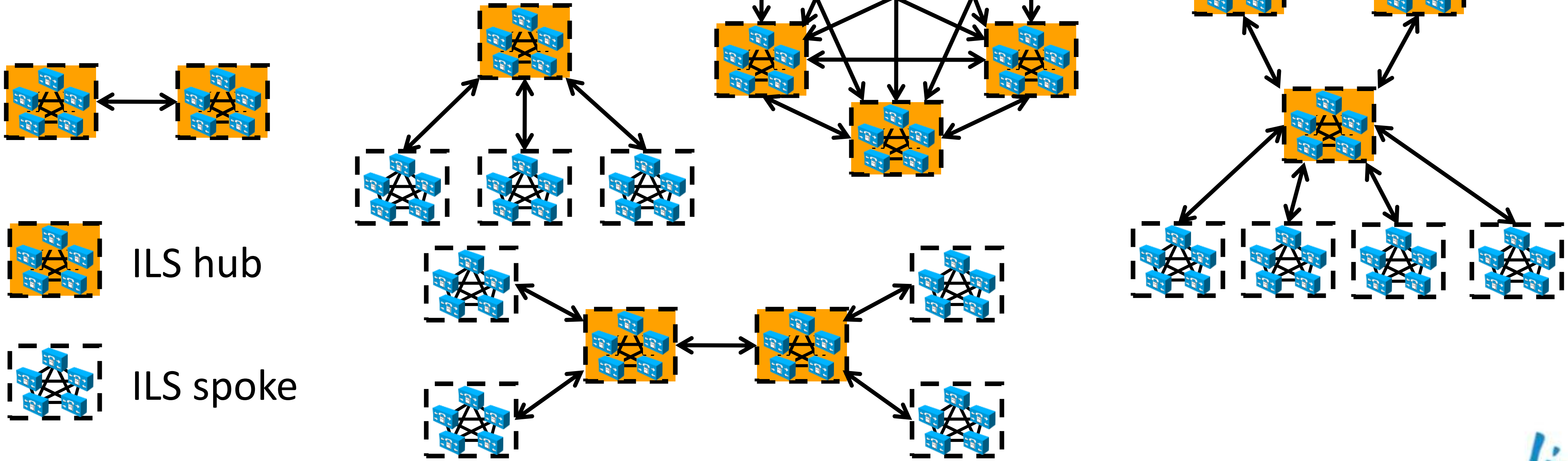


- SIP connectivity is foundation for call routing based on SIP route patterns
- ILS networking is foundation for exchange or URI reachability information
- URI propagation is enabled independent of ILS networking

ILS Networking

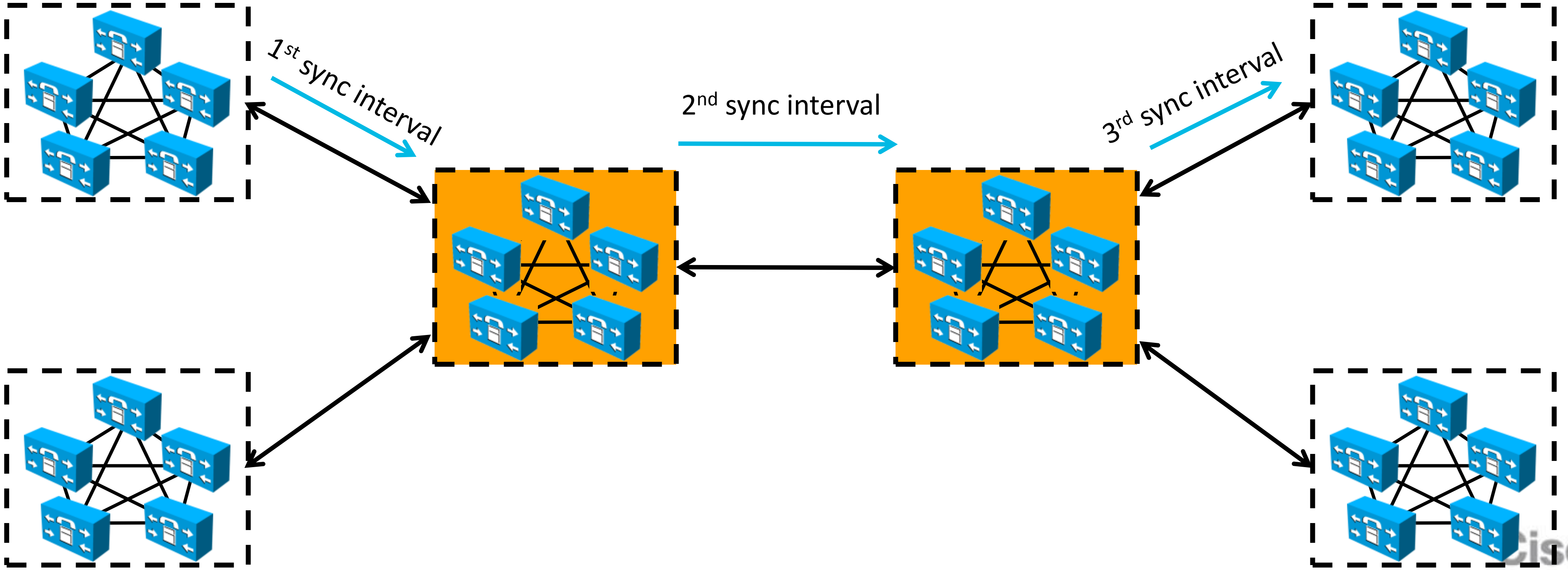
- Call Controls participating in ILS network form a hub & spoke topology
- Each Call Control is hub or spoke
- all hubs need to be full-mesh
- largest diameter is three hops





ILS URI Propagation

- Each call control keeps local copy of all URIs advertised by all other nodes in the ILS network
- Each call control periodically pulls in all changes in all URI catalogs advertised into ILS from directly connected call controls (interval 1-1440 minutes)
- URI catalog updates propagate through the ILS network hop-by-hop (remember: maximum diameter is three hops)

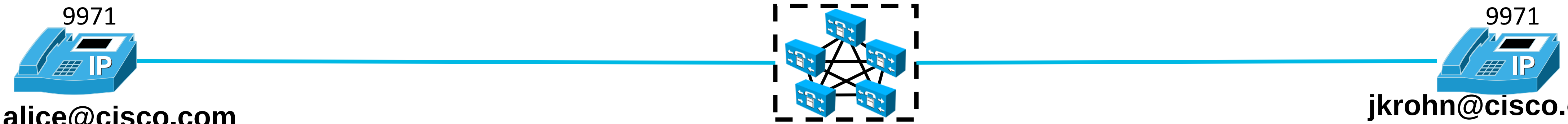


Identities

- In CUCM alpha URIs are assigned to DNs
- DNs are the “primary” identity
- devices register using DNs
- DN or alpha URI? What is the “correct” identity to be presented during calls?
  - mainly depends on the devices involved in the call
- “Blended Identity”: combination of DN and alpha URI
- CUCM can build missing piece:
  - DN → alpha URI: look at primary URI configured on DN
  - alpha URI → DN: search for DN that has the alpha URI as primary URI

Blending Identity

URI dialing to URI enabled phone



alice@cisco.com
+4961007739123

jkrohn@cisco.com
+4961007739764

1) Dial jkrohn[1] or jkrohn@cisco.com

2) jkrohn is an alpha URI!

3) blend DN for calling party:
+4961007739123 → "alice@cisco.com"

4) look up "jkrohn@cisco.com" in DA

5) blend called party:
jkrohn@cisco.com → +4961007739764

6) INVITE w/ RPID
<sip:alice@cisco.com;x-cisco-number: +4961007739123>

7) RINGING

8) blend alerting ID:
+4961007739764 → jkrohn@cisco.com

7) 180 RINGING w/ RPID
<sip:jkrohn@cisco.com;x-cisco-number: +4961007739764>

[1] if domain is missing CUCM will add OTLD configured in CUCM enterprise parameters

Blending Identity

URI dialing to non URI enabled phone

9971

alice@cisco.com
+4961007739123

7960

jkrohn@cisco.com
+4961007739764

1) Dial jkrohn[1] or jkrohn@cisco.com

2) jkrohn is an alpha URI!

3) blend DN for calling party:
+4961007739123 → "alice@cisco.com"

4) look up "jkrohn@cisco.com" in DA

5) blend called party:
jkrohn@cisco.com → +4961007739764

6) INVITE w/ RPID
<sip:+4961007739123@ip_of_cucm>

7) RINGING

8) blend alerting ID:
+4961007739764 → jkrohn@cisco.com

7) 180 RINGING w/ RPID
<sip:jkrohn@cisco.com;x-cisco-number: +4961007739764>

[1]if domain is missing CUCM will add OTLD configured in CUCM enterprise parameters

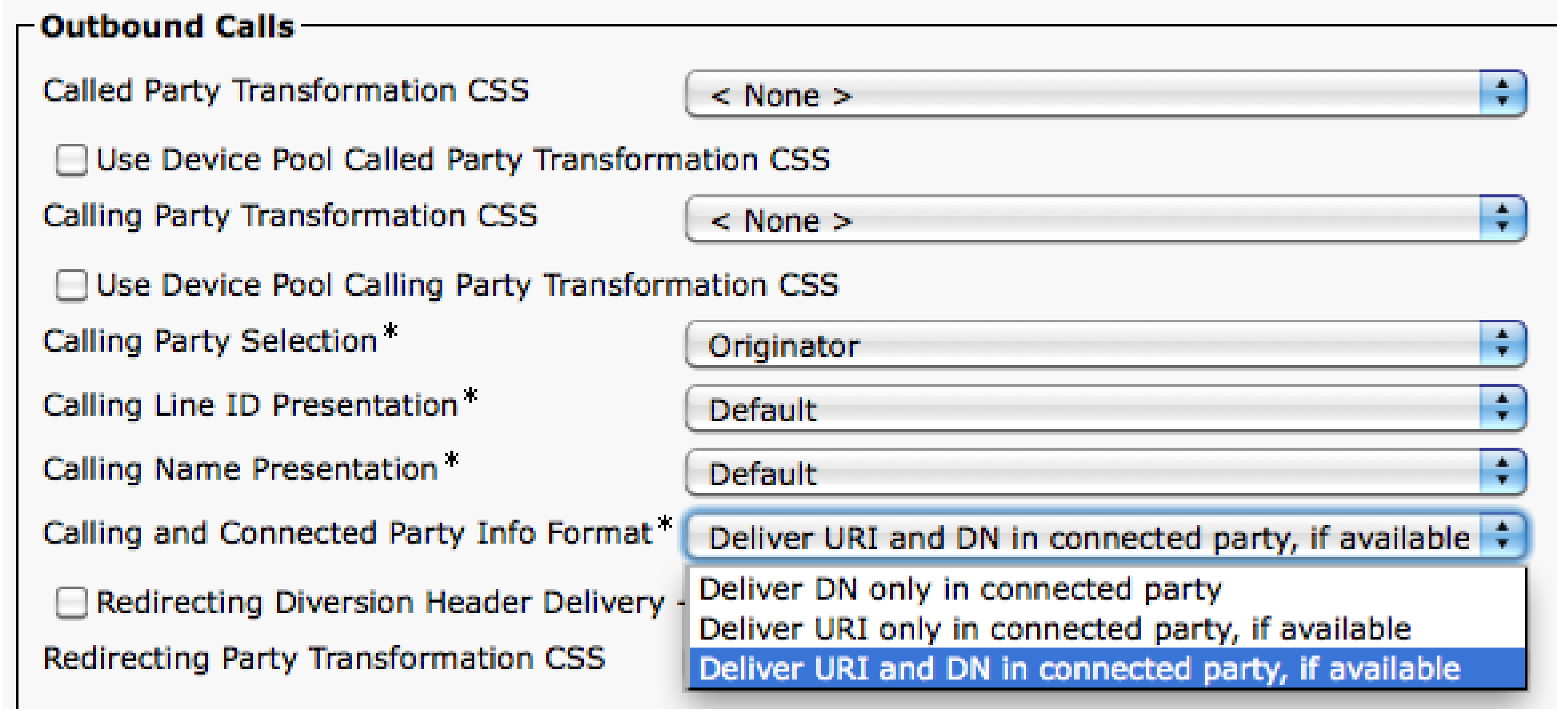
Blended Identity Delivery

- RPID carries both: alpha URI and number
  - Remote-Party-ID:<sip:jkrohn@cisco.com;x-cisco-number=+4961007739764>
- Headers affected:
  - Remote-Party-ID, Diversion, P-Asserted-ID (trunk only), P-Preferred-Identity (trunk only).

Blended Identity Delivery

Trunk Policy

- Policy on SIP trunks to define format for identity delivery
- Default: DN only (backward compatibility)
- Recommended: deliver URI and DN between clusters using URI dialing



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