

Návrh číslovacího plánu, URI dialing

T-COL2 /L2

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Program

- Developing a Global Dial Plan
- URI dialing

Developing a Global Dial Plan

- „E.164“
- Number presentation

Dial Plan: Think Globally, Act Locally

- More than just a cute phrase: it actually applies to Unified Communications
- **Even a *local* only company will make calls to, or receive calls from, *international* locations**

Mobility users tend to travel: their mobile phone thus would be best equipped with global address books

You may have to click-to-dial international numbers from your CUPC (or other soft phone)

- ...

Dialing

- different types of dialed numbers
 - national
 - international
 - national on-net – national calls to known sites on-net
 - international on-net – intl. calls to known sites on-net
 - private numbers –private voice network (PVN)
 - intra-site – “office next door”
- who/what is dialing (is the source of the number)
 - users using the keypad** – typically want short numbers
 - applications, CTI** – number length irrelevant
 - directories** – number format in the directory?



(+)E.164 DNs

+E.164 DNs

- +E.164 DNs supported since CUCM 7.0
- Configured in CUCM admin GUI as “\+”
- **Pro:**
 - Unique by definition
 - Direct + dialing with DN partition in CSS
- **Caveats:**
 - CTI applications (including UCCE/UCCX)
 - Unity Connection does not support +E.164 (as primary DN)
 - ...
- Why not use E.164 DNs instead?

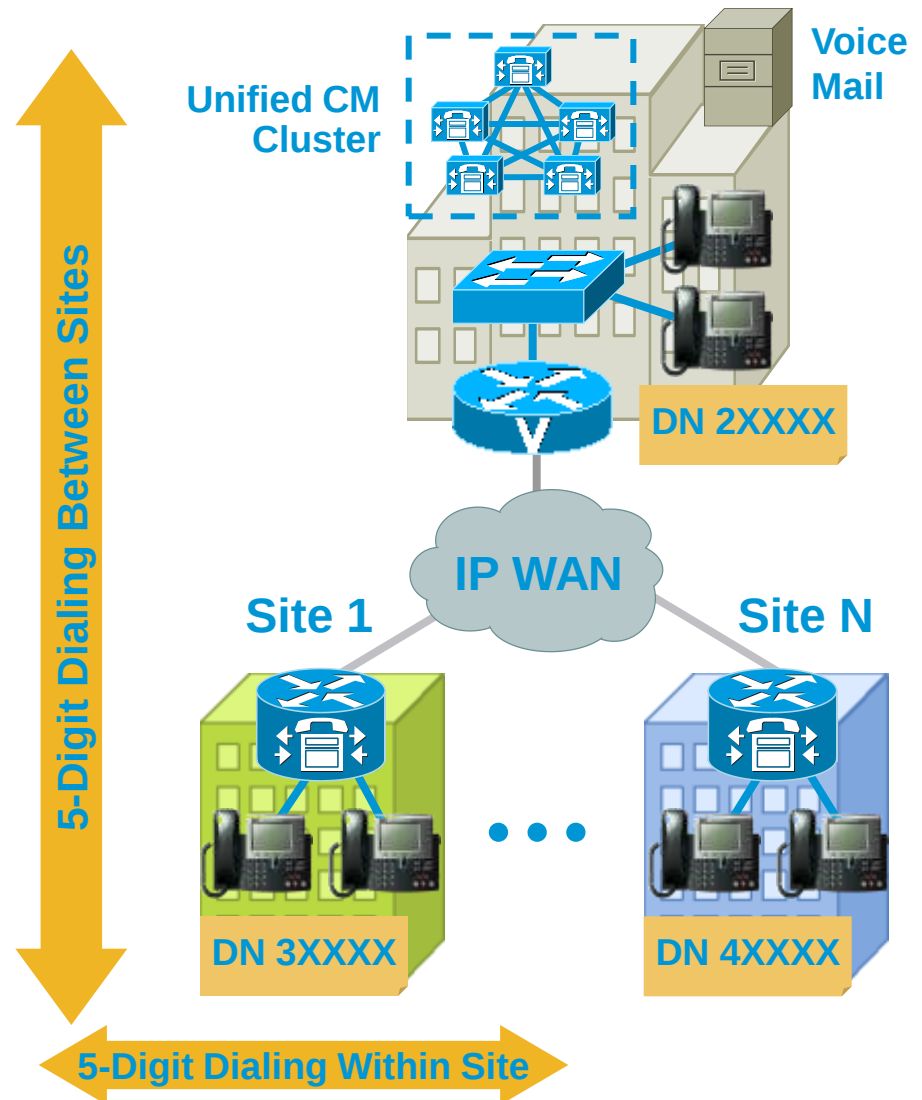
E.164 DNs

- Unique by definition
- Country code overlaps with intra-site dialing (e.g. 1xxx)
- Really?
 - E.164 dialing (w/o +) is not a supported dialing habit
 - the DN partition must not be addressed directly
- +E.164 DNs require translations from supported dialing habit to +E.164 anyway; why not translate to E.164 instead of +E.164?
- **Calling party number globalization**
 - set external phone number mask to +E.164
 - “use external phone number mask” on translation patterns

Choosing a Dial Plan Approach

Uniform On-Net Dialing

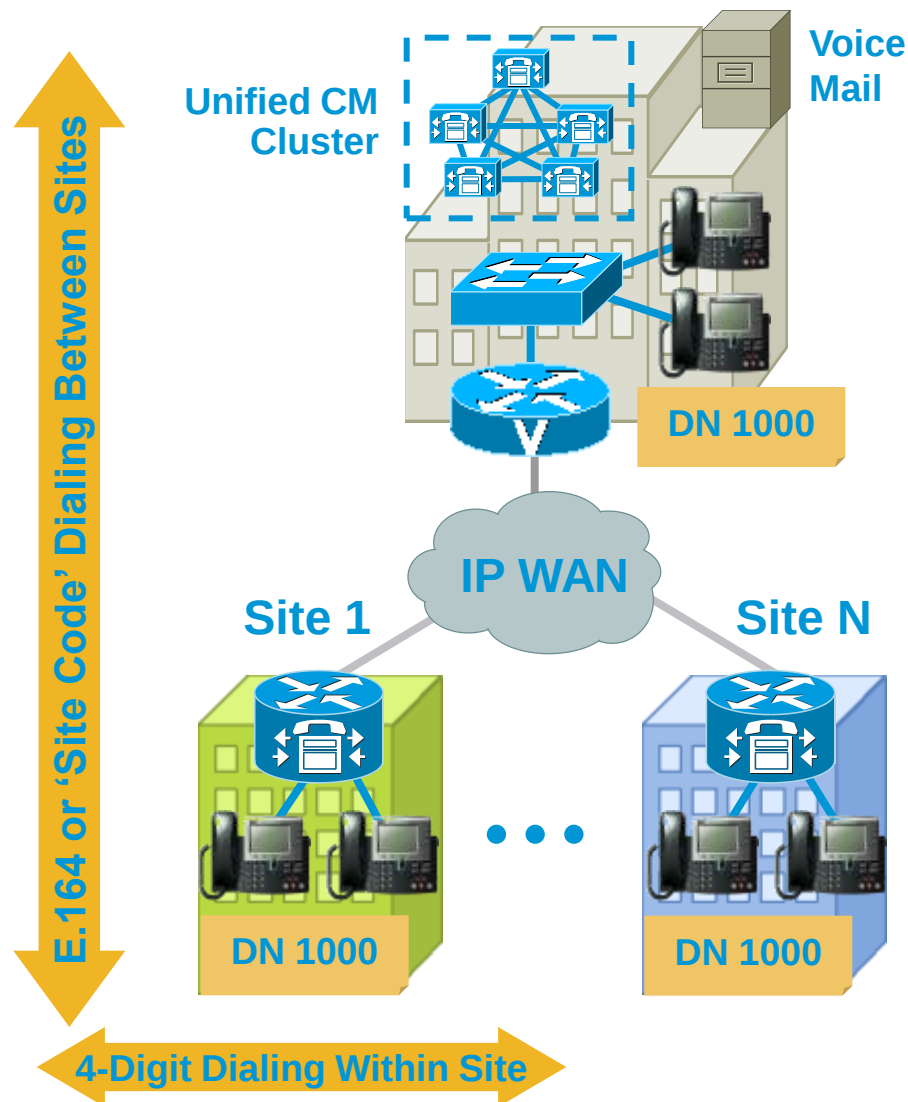
- Dialing within a site and across sites with same number of digits (e.g., five)
- Extensions are globally unique
- Easy to design and configure
- Limited scalability of the addressing method (number of sites, number of extensions)



Choosing a Dial Plan Approach

Variable-Length On-Net Dialing (VLOD)

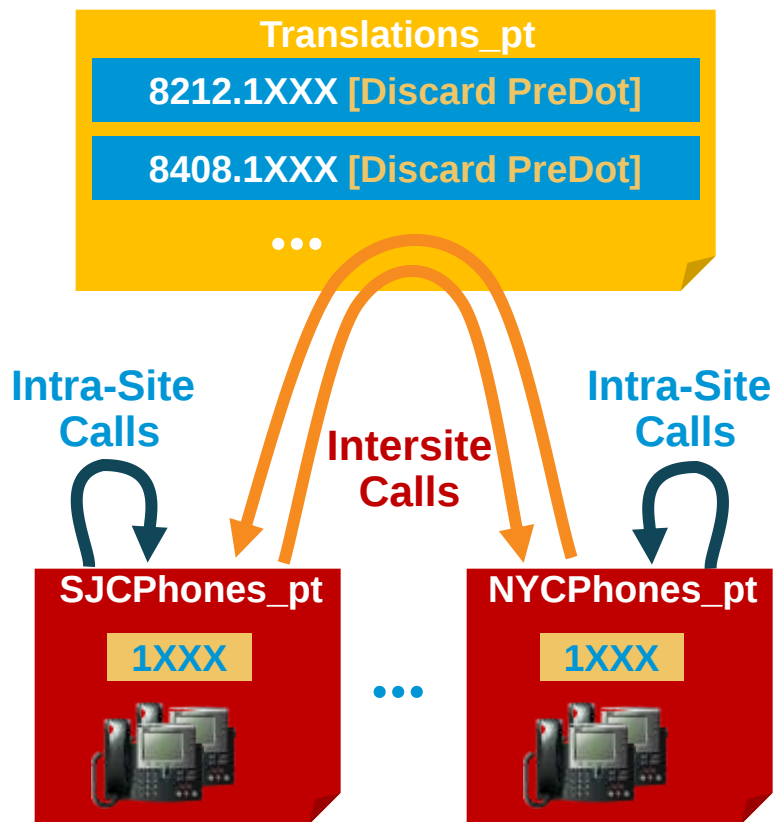
- Abbreviated dialing within a site (four or five digits)
- Identical extensions (e.g., 1000) may appear at different sites
- Intersite calls use an **escape code** (e.g., 9 + full E.164, or 8 + site code + extension)
- Easier scalability for large numbers of extensions and sites



Choosing a Dial Plan Approach

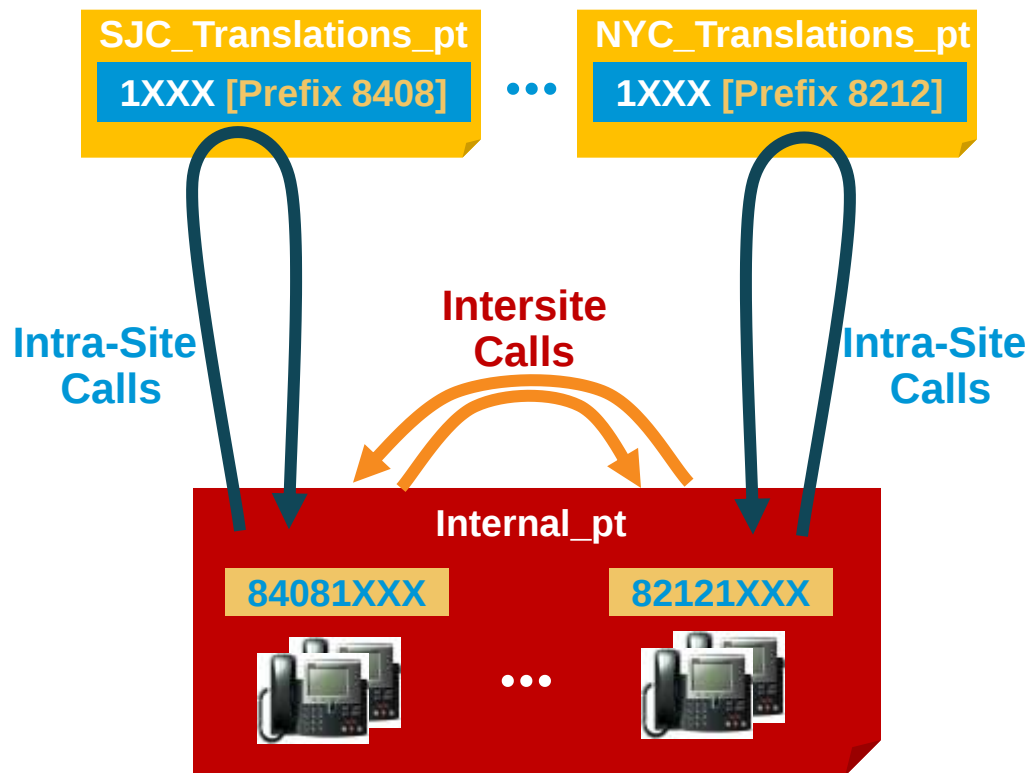
Addressing Methods for VLOD

Partitioned Addressing



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

Flat Addressing



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

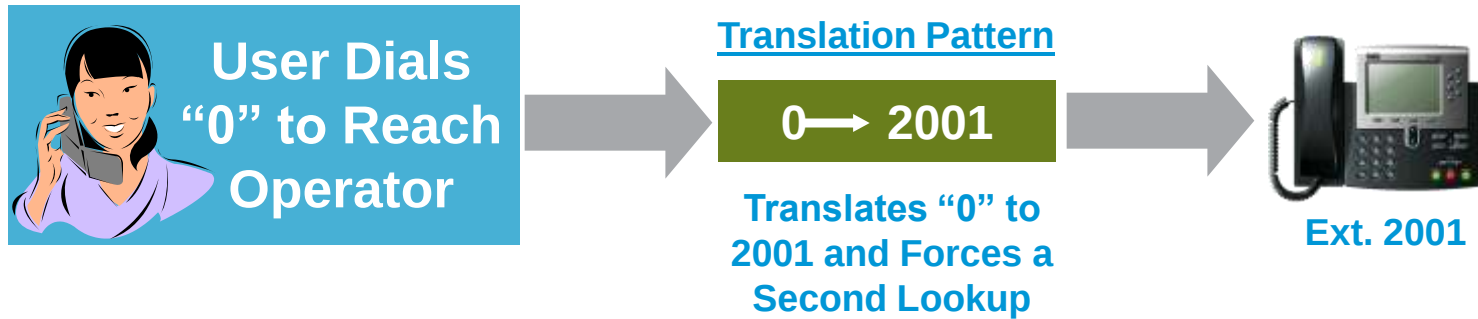
Types of Callable Patterns

- Directory Number
Extend call to registered device (phone, voicemail port etc.)
- Route Pattern
Modify calling and called party and start routing to an external route
- Translation Pattern
Modify calling and called party and continue to route using a different calling search space

Translation Patterns

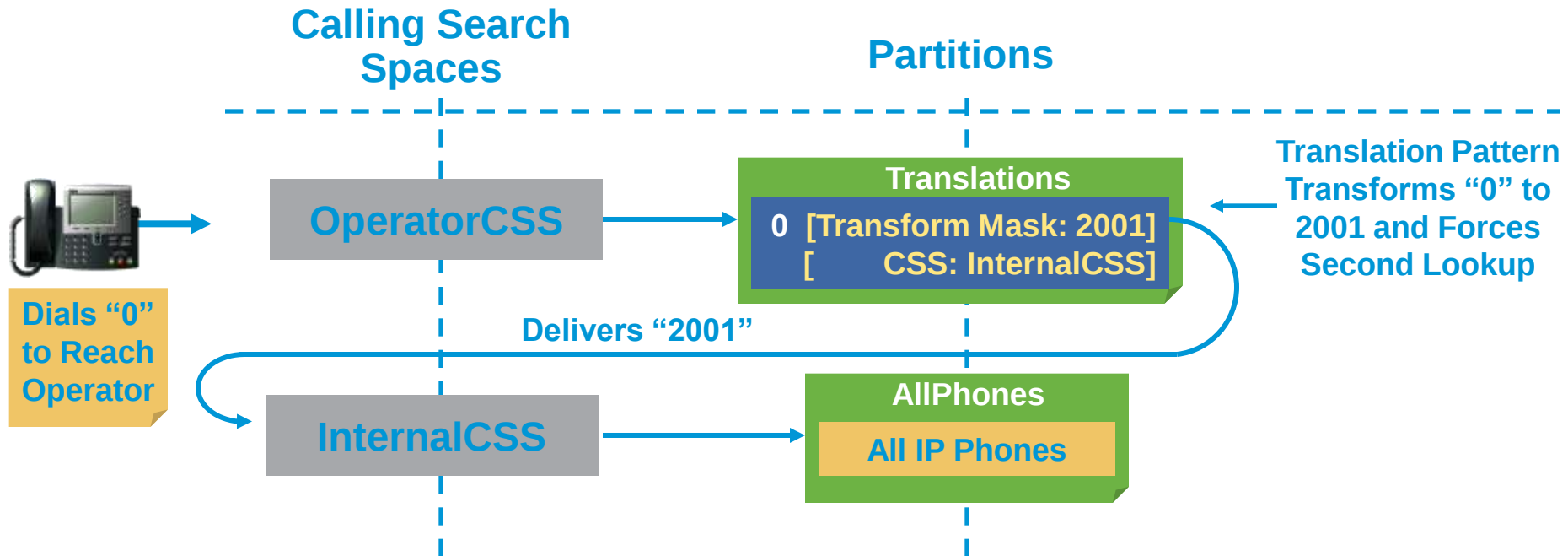
The Basics

- Match on dialed digits
- Perform calling and/or called party digit manipulation
- Force second lookup in Cisco Unified CM, using a (possibly different) calling search space



Translation Patterns

Call Flow



- Allows digit manipulation of called and calling party number
- Forces second lookup in Cisco Unified CM, using a (possibly different) calling search space

Number Transformations

- Two Concepts:

Implicit – **As part of routing process**

Translation Pattern

Route Pattern

Route Lists

Explicit – Transformation **after routing decision**

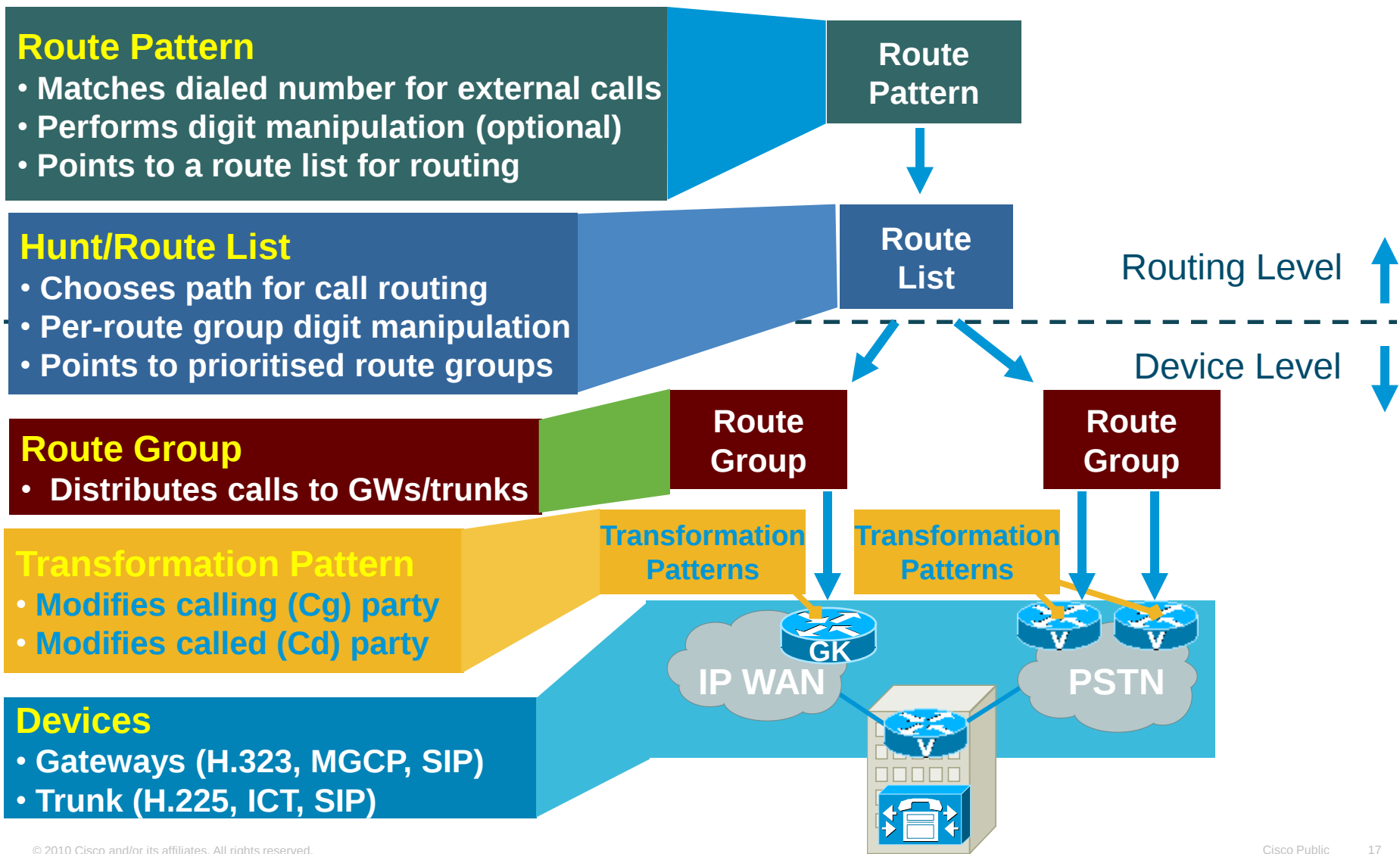
Incoming Calling/Called Party Settings on gateways, trunks (or device pools)

Calling/Called Party Transformation CSS on gateways, trunks (or device pools)

Calling Party Transformation CSS on phones (or device pools)

Number Transformations

Gateways: Calling and Called-Party Transforms

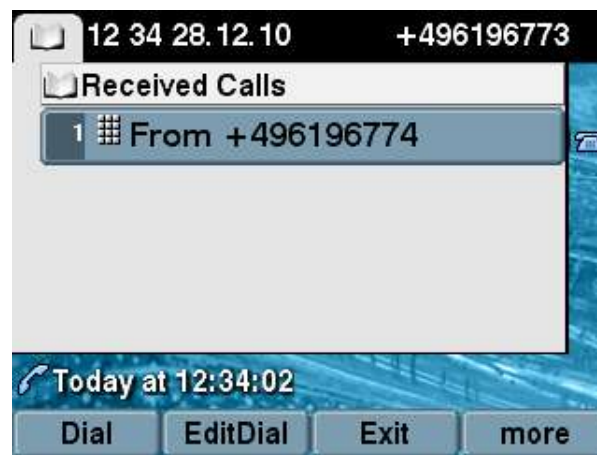




Calling Party Numbers

Where do they show up?

- Alerting plane
- calling party number in outgoing calls
- missed/received calls directory



General Concept for CnPNs

- globalize on ingress:
 - incoming calling party transformation CSSes on GWs
 - globalization of DNs has to use calling party transformations of a translation pattern in the path (e.g. use external phone number mask)
- localize on egress
 - outgoing calling party transformation CSS on GWs
 - calling party transformation CSS on phone

URI Dialing

What is a URI in CUCM 9.0?

- URI is a “uniform resource identifier”. Example: johndoe@cisco.com
- URIs are aliases for device numbers.
- A call to the URI behaves as if the call was made directly to the device number.

Why dial Alpha URIs?

- Always a VOIP call. (Calls to a number can be routed over the PSTN, but this isn't desirable because PSTN calls don't support Video.)
- Provides for better integration with URI centric call control systems like Tandberg's VCS and Microsoft Office Communicator Server (OCS).
- URIs can be used for a more reliable directory integration.

URIs and routing

- **Intra-cluster:** Calls to URIs on the local cluster are routed using partitions and search spaces. Look at the SRND for guidance.
- **Inter-cluster routing problem:**
 - URI calls between clusters can't be routed with prefix like phones can.
 - URIs on multiple clusters can have the same domain suffix so you can't route calls to the correct cluster using the domain suffix.
- **Inter-cluster routing solution:**
 - URIs are replicated between clusters using the Inter-cluster Lookup Service (ILS). When a call is made a lookup is done in ILS to figure out what SIP trunk to route the call out of.

URIs in caller ID

- A call from a device number includes a URI in the caller ID if one exists
- A call from a device number always includes the device number in the caller ID so it can be presented to a device that doesn't support URIs and those devices can return the call.

Alpha URI Dialing phone support

- Only a subset of the phones support dialing a URI. Currently supported phones:
 - Round Table (exceptions: transferring, conferencing, and forwarding today.)
 - E-20, E-90 (coming soon?)
 - Cisco Jabber for Windows (coming soon?)
- All phones support receiving an alpha URI call.

Features supported

- Transfer
- Forward
- Conference
- Callback (Callback to a URI call will be numeric.)
- Speed Dial
- Abbreviated Dial
- Redirecting (SIP 302)
- CDR
- CTI/JTAPI/TAPI (SIP/SCCP)
- UDS (URI lookup and query)
- SIP trunks (Inter-cluster, SME, CUBE, IME and CoRes)

Feature limitations

Round table phones:

Don't let you enter a URI will transferring, conferencing or configuring forward all.

H323. MGCP trunks/gateways and most phones:

Don't let you dial URIs.

Present the number as the caller.

Q.SIG trunks:

The caller ID may be numeric in some situations.

SRST:

Doesn't support URI dialing.

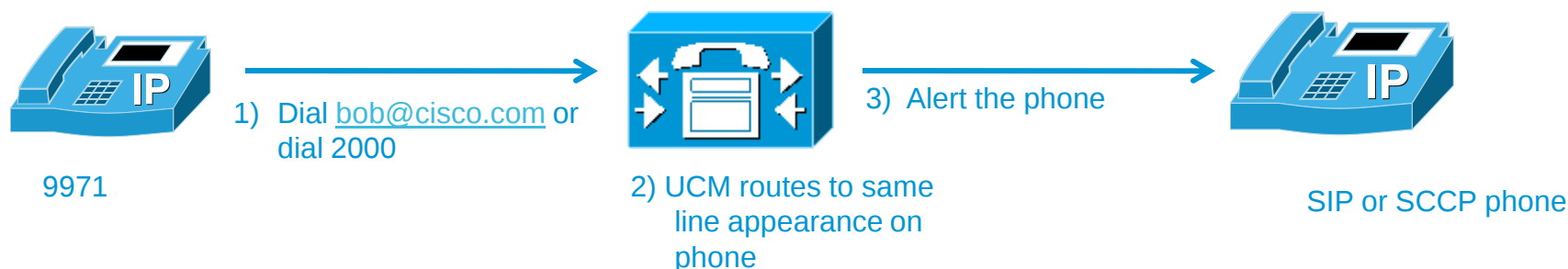
URI Intra-cluster routing

Basics of intra-cluster routing when dialing a URI



Overview – URI dialing

- A URI is an alias of a DN. You can dial either and reach same destination.
- URIs can be associated with a DN on any line device, SIP or SCCP.
- DN or Blended (URI and DN) identity can be delivered in the signaling for display/call history population.
- Phones have no notion of their associated URI. They still register with DN.



Supported URI formats

- Supported according to SIP RFC 3261, section 25.1

alice@cisco.com

alice@10.10.10.1

- user portion is case-sensitive, host portion is case-insensitive

e.g. Alice@cisco.com and alice@cisco.com are not the same

- explicit IPv6 host not supported

e.g. alice@[2001:420:8ff:6:250:56ff:fe8e:103c]

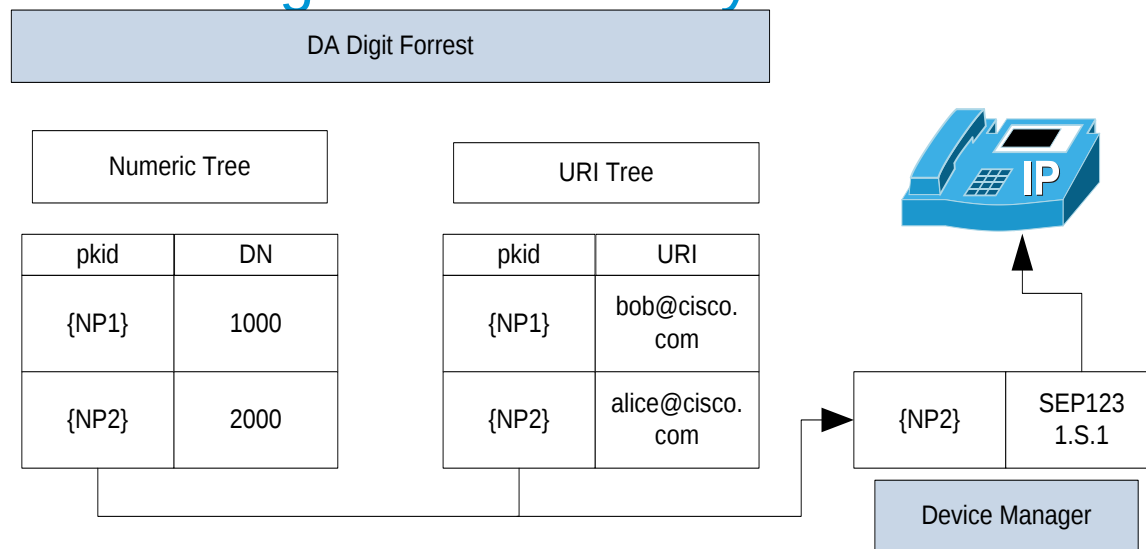
- URIs without a user name are not supported

e.g. sip:cisco.com

- 7-bit ASCII only

Storing URIs in Digit Analysis

- URIs are stored in separate tree in DA.
- URI is an alias of a DN. Both URI and DN point to same line appearance on the device.
- Components calling DA must indicate if a dial string is a DN or URI before sending to DA for analysis



Dial String Interpretation – Problem Statement

- Characters 0-9, A-D,*,+ span both the numeric and alpha-numeric sets, there needs to be a way to disambiguate when an incoming dial string is arbitrary.

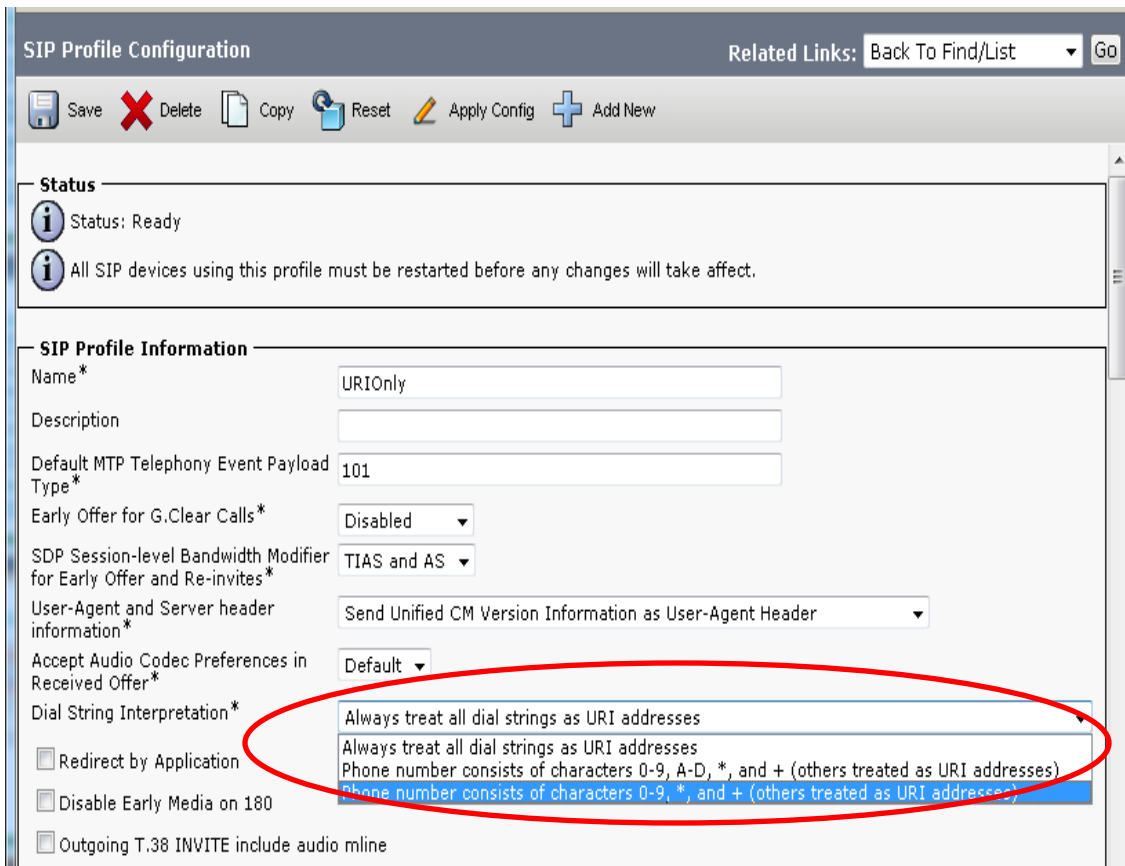
e.g., is the user portion of sip:1234@10.10.10.1 a DN or URI dial string?

- Two tiered solution

1) Phone includes “**user=phone**” tag on Request-URI when in keypad mode. This indicates the dial string **is a number**.

2) “**user=phone**” not present, UCM applies a configurable interpretation policy on the Request-URI.

SIP Profile setting, Dial String Interpretation



The screenshot shows the 'SIP Profile Configuration' page. The 'Dial String Interpretation*' dropdown menu is open, showing three options: 'Always treat all dial strings as URI addresses' (selected), 'Always treat all dial strings as URI addresses', and 'Phone number consists of characters 0-9, *, and + (others treated as URI addresses)'. A red circle highlights the dropdown menu.

SIP Profile Configuration Related Links: [Back To Find/List](#) [Go](#)

Save **Delete** **Copy** **Reset** **Apply Config** **Add New**

Status

Status: Ready

All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name* URIOOnly

Description

Default MTP Telephony Event Payload Type* 101

Early Offer for G.Clear Calls* Disabled

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites* TIAS and AS

User-Agent and Server header information* Send Unified CM Version Information as User-Agent Header

Accept Audio Codec Preferences in Received Offer* Default

Dial String Interpretation*

- Always treat all dial strings as URI addresses
- Always treat all dial strings as URI addresses
- Phone number consists of characters 0-9, *, and + (others treated as URI addresses)

☐ Redirect by Application

☐ Disable Early Media on 180

☐ Outgoing T.38 INVITE include audio mline

- Dial String Interpretation setting on SIP Profile page.
- Default is “0-9,*, and +” dial string is number.
- Best practice, provision non-ambiguous URIs

Delivering blended identity

- Blended identity is delivered to phones that indicate support for new x-cisco-number tag

e.g. Remote-Party-ID:<sip:alice@cisco.com;x-cisco-number=1000>

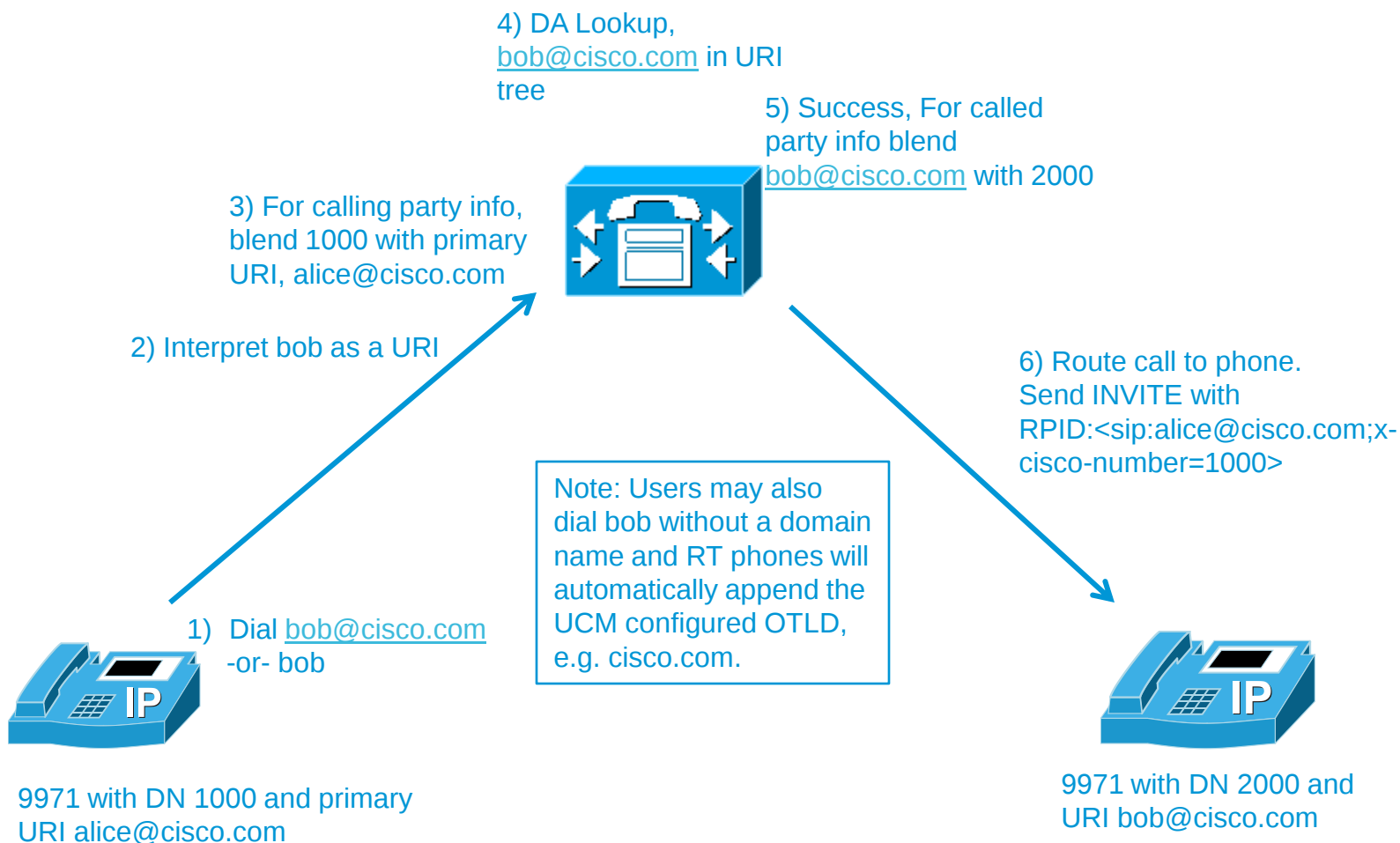
- Phones indicate support via new REGISTER/optionsind key

e.g. <x-cisco-number></x-cisco-number>

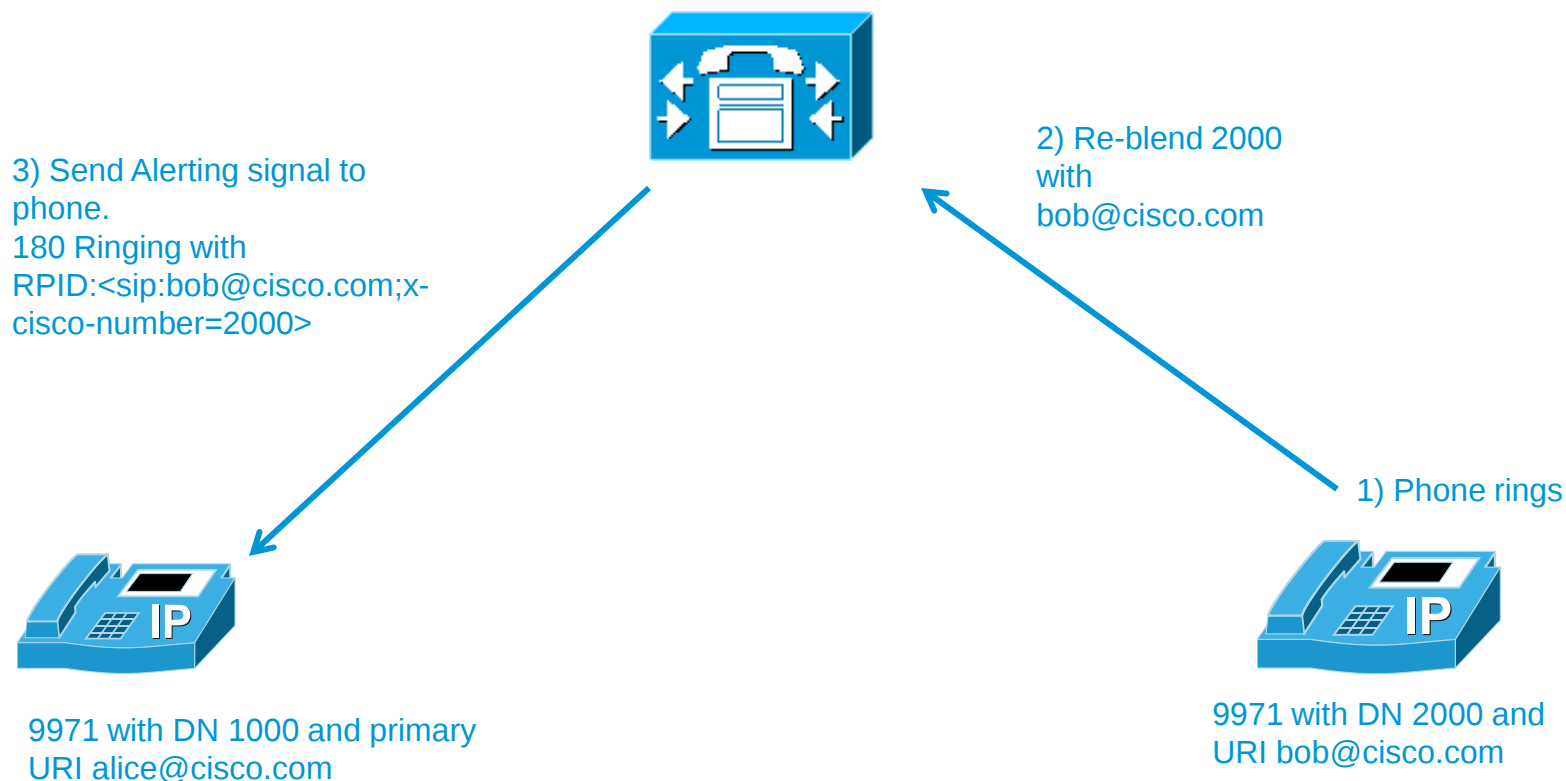
- -or- Configure phone model with new QED flag
- SIP Trunk, configure for blended delivery on trunk page (more on that later)
- Headers affected,

Remote-Party-ID, Diversion, P-Asserted-ID(trunk only), P-Preferred-Identity(trunk only), NOTIFY content for shared line.

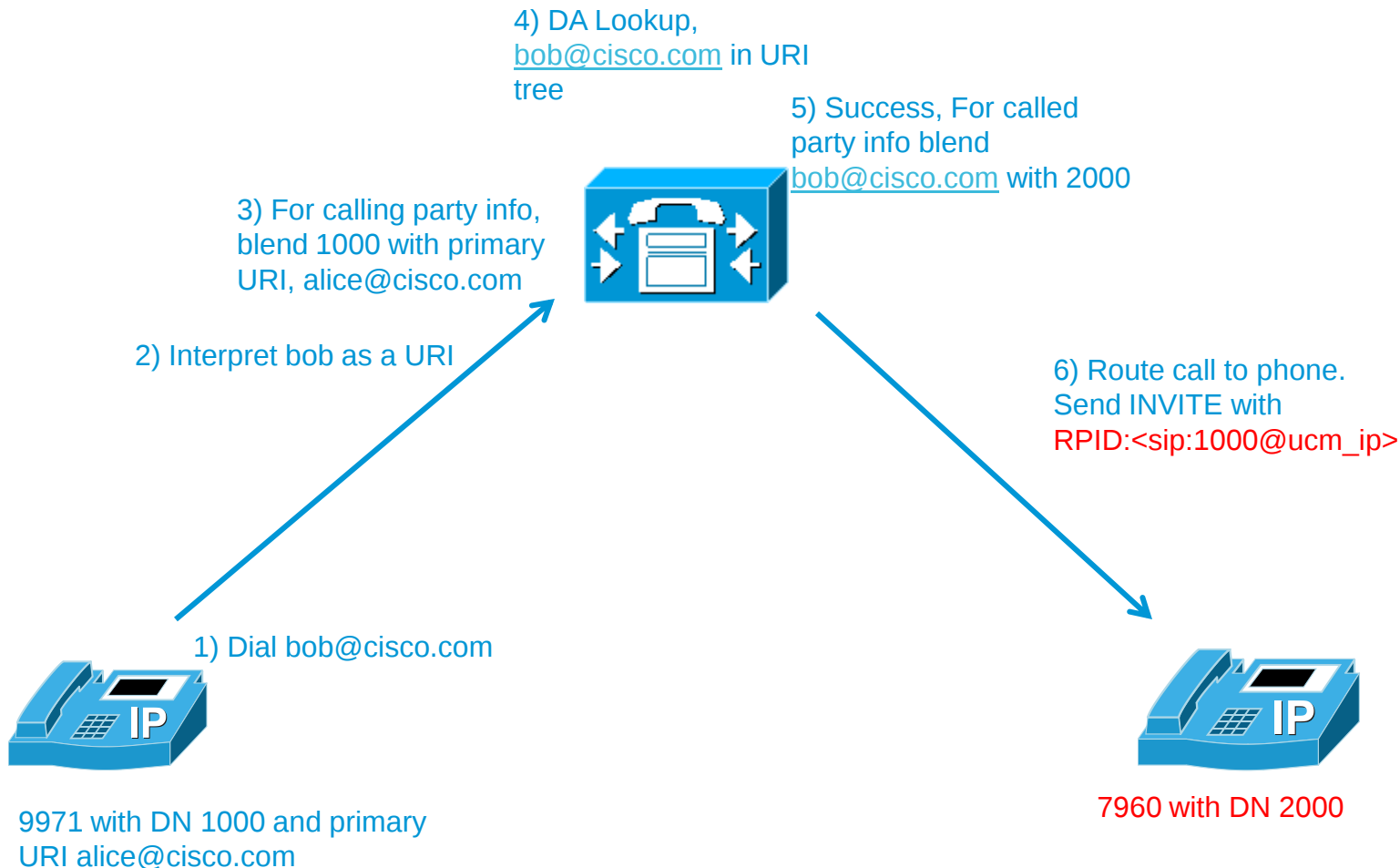
Call Flow Example – Outgoing call flow



Call Flow Example – Alerting Flow



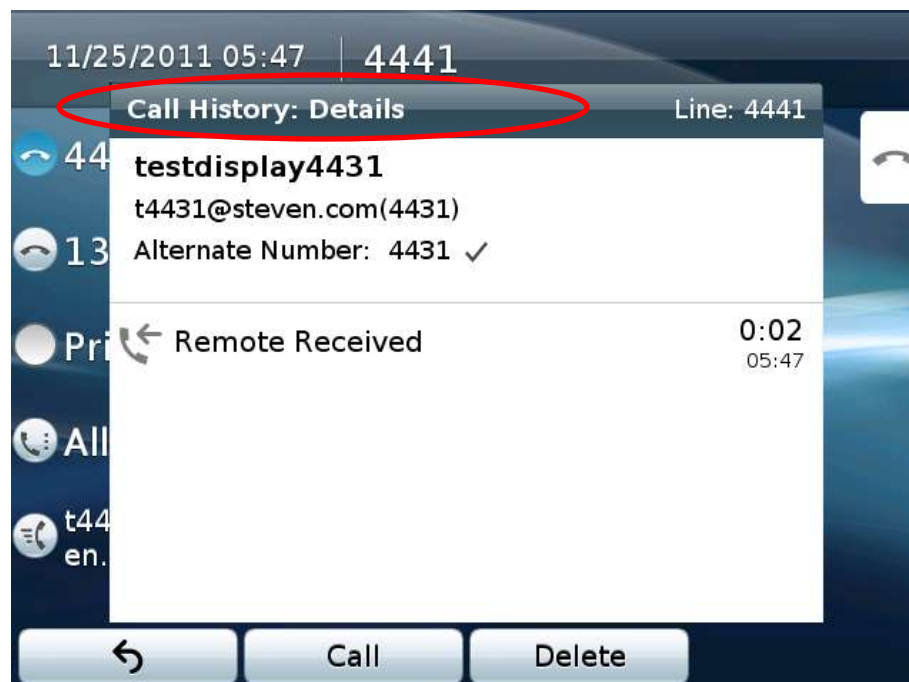
Call Flow Example – Outgoing call flow, calling legacy phone



Displaying blended identity on Phone

Phones that support blended identity consumption may use it for display and/or call history population.

- RT phones will give URI priority when both URI and DN are available.
- For display, RT phones will only display Alerting/Display Name and URI. (no DN)
- For call history population, RT phones will log both. URI will be preferred, DN is available in the details window.



URI Inter-cluster routing



Multiple Cluster

Inter-cluster lookup service

- Replication Overview:

A URI is owned by one cluster

Each cluster replicates its URIs and route string to its neighbors.

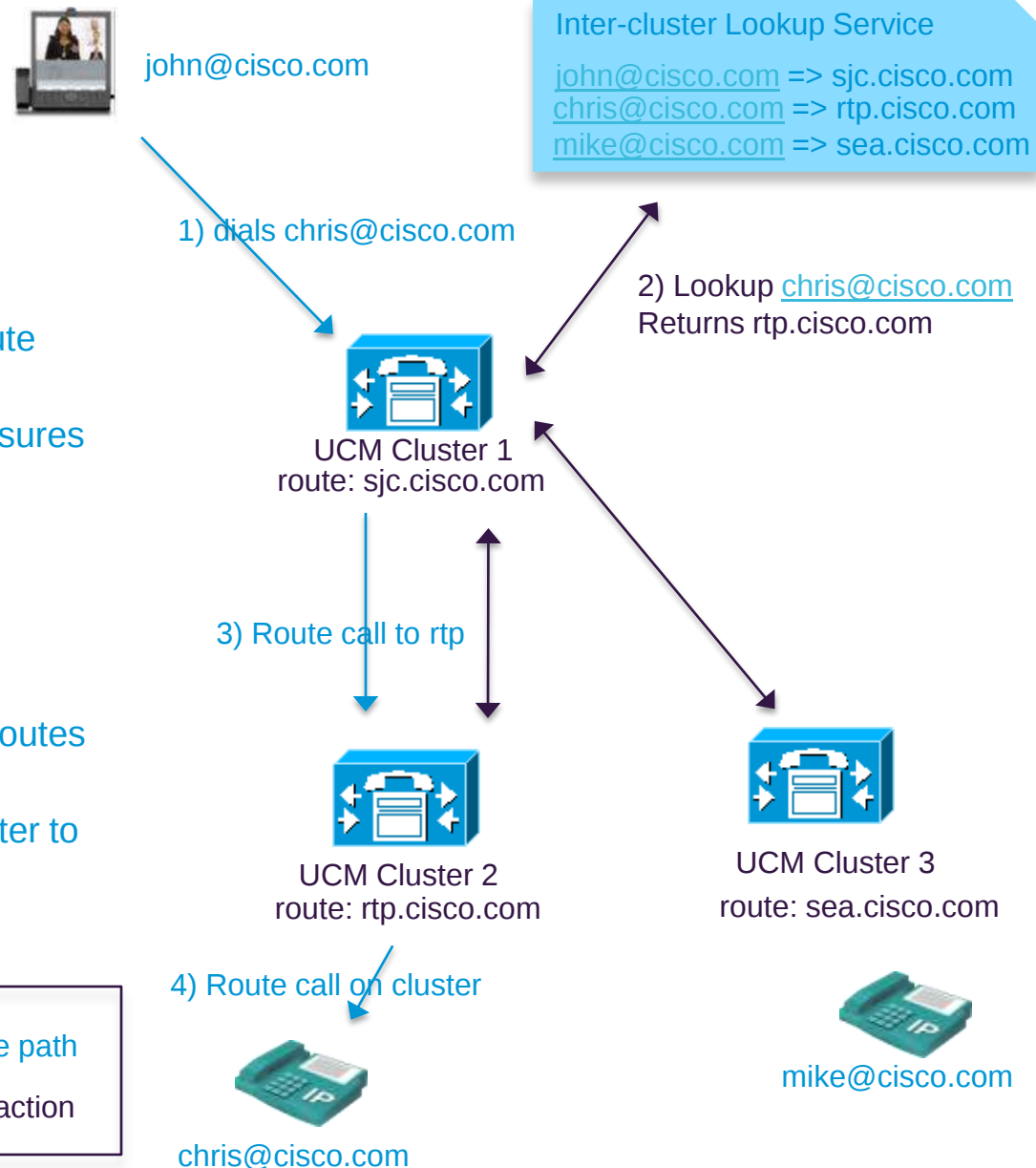
HUB and spoke replication topology ensures a fully connected network.

- Routing Overview:

Route string is retrieved for the URI

Route string is routed through the SIP routes to a SIP trunk

If necessary, process repeats from cluster to cluster until it gets to home cluster



Basic Provisioning



Add URI via Directory Number page

admin Search Documentation About Logout

System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help

Directory Number Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Status: Ready

Directory Number Information

Directory Number* 1000

Route Partition < None >

Description

Alerting Name

ASCII Alerting Name

☒ Allow Control of Device from CTI

Associated Devices SEP0024C4FCB030 SEP000830306EFF

Edit Device Edit Line Appearance

Dissociate Devices

Directory Number Settings

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

Calling Search Space < None >

Presence Group* Standard Presence group

User Hold MOH Audio Source < None >

Network Hold MOH Audio Source < None >

☐ Reject Anonymous Calls

Directory URIs

Primary	URI	Partition	Remove
☒	bob@disco.com	< None >	

Add Row

Done Local intranet | Protected Mode: Off 100%

- Any DN can have a URI associated/be dialed by it.
- Phones have no explicit knowledge of associated URI.
- Can add up to 5 separate URIs, only one is primary.
- The primary URI is the well known URI for a user. It's the URI that gets asserted when blending calling party identity and blended with the called party DN when a DN is dialed.
- Each URI can be in a separate partition and need not be in the same partition as the associated DN.

Manually Add URI via End User page – Work Flow

- Create End User with directory URI
- Associate End User with phone
- Associate primary DN with End User
- URI shows up in DN page as primary, non-deletable URI

Add URI via End User Configuration page

End User Configuration

Related Links: [Back to Find List Users](#) Go

Save

Status

Status: Ready

User Information

User ID* Bob

Password

Confirm Password

PIN

Confirm PIN

Last name* Smith

Middle name

First name

Directory URI bobsmith@cisco.com

Telephone Number

Mail ID

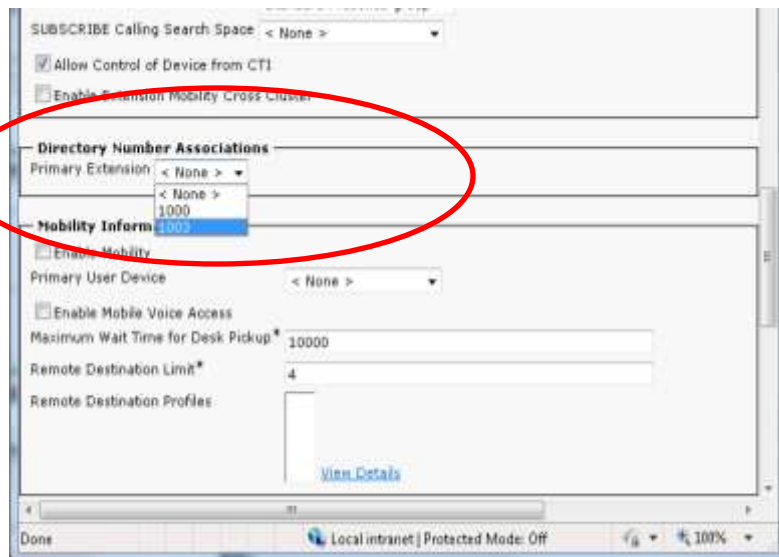
Manager User ID

Department

Done Local intranet | Protected Mode: Off 100%

- Create an end user and add a directory URI
- Associate End User with phone

Associate End User with Phone/Primary DN



SUBSCRIBE Calling Search Space: < None >

☒ Allow Control of Device from CTI

☐ Enable Extension Mobility Cross Cluster

Directory Number Associations

Primary Extension: < None > (dropdown menu showing 1000, 1001)

Mobility Inform 1000

☐ Enable Mobility

Primary User Device: < None >

☐ Enable Mobile Voice Access

Maximum Wait Time for Desk Pickup*: 10000

Remote Destination Limit*: 4

Remote Destination Profiles

[View Details](#)

Done Local intranet | Protected Mode: Off

- End User page, associate primary extension with end user.
- URI shows up in DN page.
- End User URI is always primary.
- End User URI gets put into non-deletable “Directory URI” partition.



User Hold MOH Audio Source: < None >

Network Hold MOH Audio Source: < None >

☐ Reject Anonymous Calls

Directory URIs

Primary	URI	Partition	Edit/Remove
☒	bobsmith@cisco.com	Directory URI	Edit End User
☐	charlie@disco.com	< None >	Edit End User

[Add Row](#)

AAR Settings

Voice Mail	AAR Destination Mask	AAR Group
AAR ☐ or		< None >
☒ Retain this destination in the call forwarding history		

Done Local intranet | Protected Mode: Off

LDAP Directory URI Synchronization

- Works the same as all other LDAP fields, LDAP values that have a corresponding CUCM value are synchronized
- LDAP Directory URI field synchronizes with Directory URI on the End User Configuration page
- Synchronized directory URIs are added to the default Directory URI partition
- Any user that has a primary extension and an LDAP directory URI will have that URI added to the list of routable URIs in the system

LDAP Directory Configuration

LDAP DirectoryRelated Links: [Back to LDAP Directory Find/List](#) [Go](#)

 Save

Perform Sync Just Once ☐

Perform a Re-sync Every* DAY

Next Re-sync Time (YYYY-MM-DD hh:mm)*

Standard User Fields To Be Synchronized

Cisco Unified Communications Manager User Fields	LDAP Attribute	Cisco Unified Communications Manager User Fields	LDAP Attribute
User ID	sAMAccountName	First Name	givenName
Middle Name	middleName	Last Name	sn
Manager ID	manager	Department	department
Phone Number	telephoneNumber	Mail ID	mail
Directory URI	☒ msRTCSIP-primaryuseraddress ☐ mail ☐ none		

Custom User Fields To Be Synchronized

New Numeric Transformations

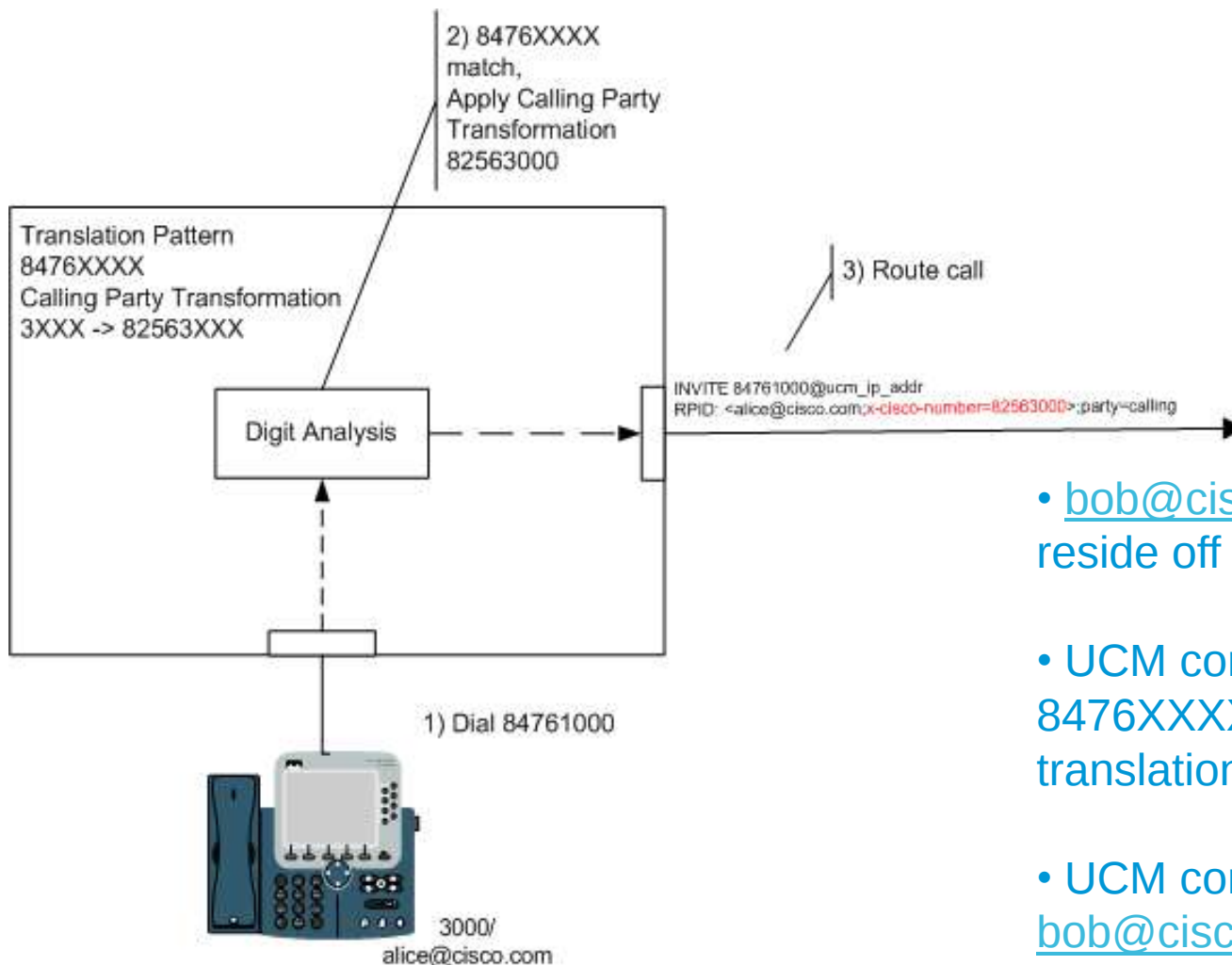
Inbound Calls, New Calling Party CSS



Problem Statement

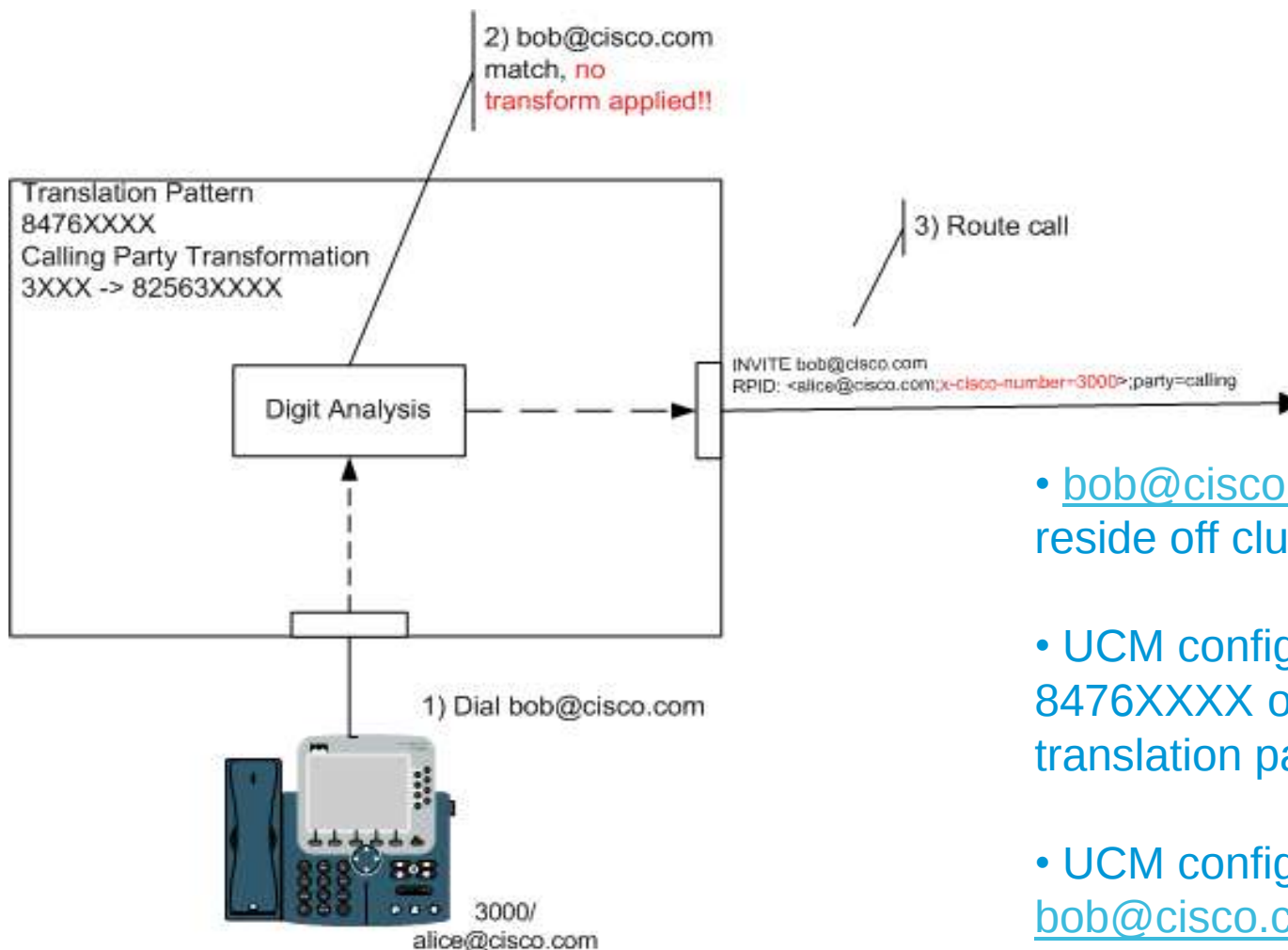
- URI is an alias of a DN. Expectation that you can dial either and reach same destination.
- URI routing can take a different route path through the system such that required digit transformations are not applied.
- Specifically, numeric translation patterns, which are a routing-level entity, can modify the calling and called party numbers.
- Deployments that use translation patterns to modify calling party numbers will want to add this CSS when using URI dialing feature.

Dial DN Example



- bob@cisco.com/4761000 reside off cluster.
- UCM configured to route 8476XXXX off cluster via translation pattern.
- UCM configured to route bob@cisco.com via ILS

Dial URI Example

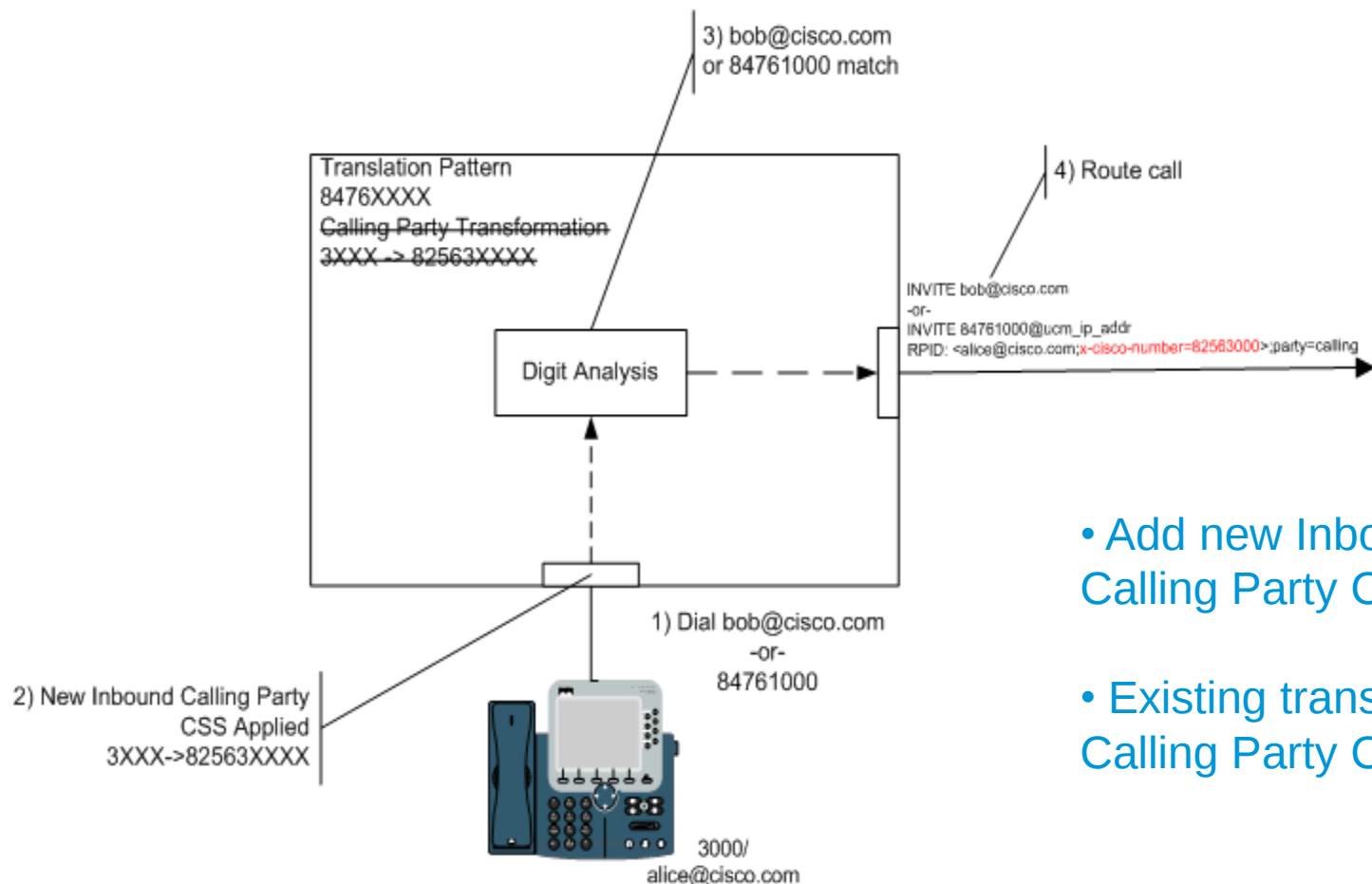


- bob@cisco.com/4761000 reside off cluster.

- UCM configured to route 8476XXXX off cluster via translation pattern.

- UCM configured to route bob@cisco.com via ILS

Dial URI or DN, new Inbound Calls, Calling CSS example



- Add new Inbound Calls, Calling Party CSS
- Existing translation pattern Calling Party CSS not applied

Phone Configuration Page, Calling Party Transformation CSS

The screenshot shows the Cisco Phone Configuration page. At the top, there are tabs for System, Call Routing, Media Resources, Advanced Features, Device, Application, and User. Below the tabs, the page title is 'Phone Configuration' and there is a 'Related Links' section with a dropdown menu showing 'Back To Find/List' and a 'Go' button. A toolbar contains icons for Save, Delete, Copy, Reset, Apply Config, and Add New. The main content area is divided into sections. The 'Call Routing Information' section is highlighted with a red box and contains two sub-sections: 'Inbound Calls' and 'Outbound Calls'. The 'Inbound Calls' section has a dropdown menu for 'Calling Party Transformation CSS' set to '< None >' and a checked checkbox for 'Use Device Pool Calling Party Transformation CSS'. The 'Outbound Calls' section has a similar dropdown menu set to '< None >' and an unchecked checkbox for 'Use Device Pool Calling Party Transformation CSS'. Below the 'Call Routing Information' section is the 'Protocol Specific Information' section, which includes a 'Packet Capture Mode*' dropdown menu set to 'None'.

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User

Phone Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

☐ Ignore Presentation Indicators (internal calls only)
☒ Allow Control of Device from CTI
☒ Logged Into Hunt Group
☐ Remote Device
☐ Protected Device****
☐ Require off-premise location

Call Routing Information

Inbound Calls

Calling Party Transformation CSS < None >

☒ Use Device Pool Calling Party Transformation CSS

Outbound Calls

Calling Party Transformation CSS < None >

☐ Use Device Pool Calling Party Transformation CSS

Protocol Specific Information

Packet Capture Mode* None

- Added new “Inbound Calls, Calling Party Transformation CSS” dropdown with Device Pool override
- Moved existing “Calling Party Transformation CSS” to new “Outbound Calls” section
- Grouped both into a “Call Routing Information” section

Device Pool Page, Calling Party Transformation CSS

The screenshot shows the 'Device Pool Configuration' page in a Cisco management interface. The page has a top navigation bar with tabs: System, Call Routing, Media Resources, Advanced Features, Device, Application, and User Management. Below the navigation bar is the 'Device Pool Configuration' section. It includes a 'Related Links' dropdown set to 'Back To Find/List' and a 'Go' button. Below this is a toolbar with icons for Save, Delete, Copy, Reset, Apply Config, and Add New. The main configuration area is divided into sections: 'International Number', 'Unknown Number', and 'Subscriber Number', each with a text input (Default), a numeric input (0), and a dropdown menu (< None >). Below these is the 'Phone Settings' section, which contains three sub-sections: 'Inbound Call Settings', 'Connected Party Settings', and 'Redirecting Party Settings'. The 'Inbound Call Settings' section is circled in red and contains a checkbox labeled 'Inbound Call Settings' and a dropdown menu for 'Calling Party Transformation CSS' set to '< None >'. The 'Connected Party Settings' section contains a dropdown menu for 'Connected Party Transformation CSS' set to '< None >'. The 'Redirecting Party Settings' section contains a dropdown menu for 'Redirecting Party Transformation CSS' set to '< None >'. The 'Inbound Call Settings' section is circled in red.

- Added new “Inbound Call Setting, Calling Party Transformation CSS” dropdown

- Overrides phone page setting, if checkbox checked

Otázky a odpovědi

- Twitter www.twitter.com/CiscoCZ
 - Talk2Cisco www.talk2cisco.cz/dotazy
 - SMS 721 994 600
-
- Zveme Vás na **Ptali jste se...** v sále **LEO**
1.den 17:45 – 18:30
2.den 16:30 – 17:00

**Prosíme, ohodnot'te
tuto přednášku.**

