# CUCM

## Route Patterns, Lists and Groups

**Route pattern**
Match dialed numbers
Digit manipulation
Points to a route list

**Route List**
Ordered list pointing to Route-groups

**Route Group**
Performs digit manipulation
Points to gateways
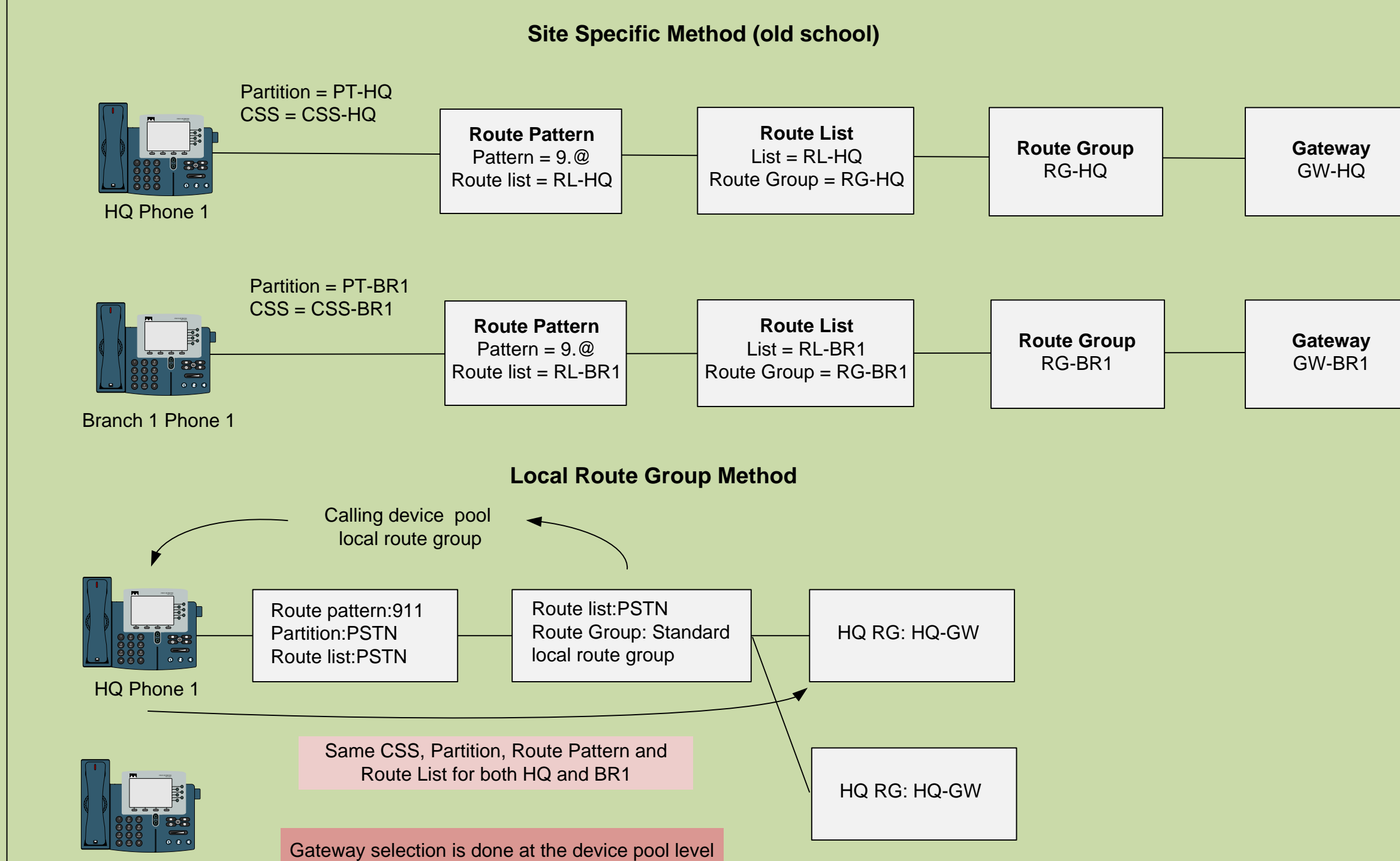The digit manipulation actually occurs (bound) on the route-list

**Route Pattern Wildcards**
@ = all NANP
X = any single digit
! = one or more digits 0-9
? = zero or more of the preceding digit
+ = one or more of the preceding digit
[ ] = enclose a range
- = used between brackets to denote a range
. = discard digit identifier
# = end of dialing sequence
\ = escape character (used with # dialing)

*(Diagram: Route Pattern → Route List → Route Group / Route Group → Gateway / Gateway)*

**Called Number Manipulation**
DDI (i.e. predot)
Mask (can add, delete or replace)
Prefix (add)

**Calling Number Manipulation**
External number mask
Calling party transformation mask
Prefix (add)

**Digit Manipulation**
If using more than one method in a route pattern or route-group, compounding takes place in order
Can take place in route pattern or route group (route list)
Use on route groups when different digit manipulation is required on each Gateway
Route lists supersede route patterns
If route pattern has pre-dot and route-list has a manipulation as well, the calling number displayed on the phone will be pre-dot
Called and Calling number manipulations are mutually exclusive. For instance, calling manipulation can occur on the route pattern and called on the route list

## Local Route Group

**Site Specific Method (old school)**

Partition = PT-HQ
CSS = CSS-HQ

HQ Phone 1

| Route Pattern | Route List | Route Group | Gateway |
| Pattern = 9.@ | List = RL-HQ | Route Group = RG-HQ | GW-HQ |
| Route list = RL-HQ | Route Group = RG-HQ | RG-HQ | |

Partition = PT-BR1
CSS = CSS-BR1

Branch 1 Phone 1

| Route Pattern | Route List | Route Group | Gateway |
| Pattern = 9.@ | List = RL-BR1 | Route Group = RG-BR1 | GW-BR1 |
| Route list = RL-BR1 | Route Group = RG-BR1 | RG-BR1 | |

**Local Route Group Method**

Calling device pool local route group

HQ Phone 1

| Route pattern:911 | Route list:PSTN | HQ RG: HQ-GW |
| Partition:PSTN | Route Group: Standard | |
| Route list:PSTN | local route group | |

Branch 1 Phone 1

Same CSS, Partition, Route Pattern and Route List for both HQ and BR1

HQ RG: HQ-GW

Gateway selection is done at the device pool level

## Calling and Called Party Transformation Patterns (egress)

**Calling Party**
Calling number is modified as call is sent out Gateway
Other methods depend on the called number. Transformation patterns occur irrespective of called number.
Transformation pattern overrides any other modification (wins over RP, RL)
If different calling number modifications are needed based on called number (subscriber, international, etc) use the RP, RL method not transformations
Use transformation pattern when calling number modification needs to be based on locale/Gateway

**Called Party**
Called party transformations are usually used when a number is globalized to modify it to a value that the PSTN can route. Frequently the + is stripped
Uses a Partition and CSS that gives access to the translation pattern. Partition and CSS are not used for calls

**Calling and Called Party Transformations**
Destination party always invokes transformation pattern
For inbound calls (PSTN to IP Phone) **phone** calling transformation pattern CSS is used. No called transformation since phone is final destination
For outbound calls (IP Phone to PSTN), **Gateway** transformation patterns (both calling and called) are used

**Example Calling Party Transform Pattern Table**

| Calling Party Transformation Pattern | Partition | Digit Manipulation | Calling # Type |
|---|---|---|---|
| 5XXX | PT-ANI-SJC-GW | Prefix 408777 | subscriber |
| 1XXX | PT-ANI-SJC-GW | Prefix 415888 | national |
| 1XXX | PT-ANI-SJ0-GW | Prefix 415888 | subscriber |

## Plus Dialing

Purpose is to universally use full E164 formatted numbers
Dependant on globalization/localization
Must have route pattern that matches the + character
Use the Called Party Transformation Pattern and CSS to
Example – \+1415.! PT-DNIS-GW-HQ
DDI PREDOT
Type= Subscriber
CSS= CSS-DNIS-HQ-GW
Dial peers strip the + character from the calling number since it's an illegal field in the IOS in the lab. Re-insertion may be necessary

## Globalization/Localization (ingress)

**Example – call comes into SF gateway with ANI of 415 555 1212**

May want to store the ANI of missed/received calls in a globalized format (E164) like +1 415 555 1212

May want to display the ANI in localized format like 555 1212

**Globalization**
Purpose is to store the number in globalized format
Globalize the number at the Gateway using **Incoming Calling Number Prefix**
Example – type subscriber: prefix +1415
Affects the received/missed called directory on the phone

**Localization**
Purpose is to display the number is a localized format
Localization of the incoming calling number to display the globalized number is done on the phone **calling party transformation** using the **CSS**. Example – \+1415.! DDI-PREDOT
Affects the displayed number on the phone

Destination always invokes calling party transformation so for ingress, this is done on the phone

## High Availability

**AAR**
Used for on cluster endpoint to endpoint communications when there is WAN congestion
Location or RSVP CAC uses PSTN to complete call
Since intracluster calls are normally 4 or 5 digits, called number must be expanded to match route pattern
Dedicated CSS for AAR. Example is lobby phone might not be able to make local calls but with AAR this is needed. AAR CSS applied on the device (phone)
The external number mask of the called party is used to replace the dialed digits
AAR group is set at the line level
Dial prefix (Predot instructions) are added on the AAR group
AAR must be enabled in call manager service parameters
AAR should not be used with TEHO

**Configuration steps**
1. Enable AAR in service parameters
2. AAR PT/CSS
3. AAR specific RP
4. AAR group
5. External number mask
6. Restart CCM

**SRST**
Two part process:
1. Phones must register with CME
2. Dial peers must exist on CME
Two methods on IOS router:
1. Call-manager fallback – easy but limited features
2. Telephony-service – more features
CUCM Configuration steps
1. Set SRST reference under system tab
2. Assign SRST reference to device pool
3. Reset the phone
4. Check that phone settings have SUB, PUB and SRST IP in correct order
See CME SRST section for more information

**CFUR**
If phone that is in the CUCM cluster tries to call a phone that is SRST mode, the SRST phone shows up as unregistered
Under DN in CUCM there are a Call Forward Unregistered for both internal and external calls

## Dial Plan and Call Routing Notes

Calling number manipulation for plus dialing
Goal is to add +1 to missed calls but have ringing display localized (i.e. no + and proper ANI length based on calling number)
"missed" - use calling party transformation
"ringing" - use incoming calling party settings on Gateway

Translation Patterns
Can be used to manipulate digits on incoming calls.
Example would be a MGCP Gateway where digit stripping is not possible and significant digits cannot be used on CUCM for whatever reason.

Example:
Translation Pattern : 6178631XXX
Called Party Transformations
Called Party Transform Mask : 1XXX

Common IOS translation rules

Prepend + to any string
voice translation-rule 111
rule 1 /^.*/ /+\0/

Remove area code/prefix
voice translation-rule 222
rule 1 /.../ //

add area code/prefix (212 example)
voice translation-rule 333
rule 1 // /212/

just last 4 digits
voice translation-rule 444
rule 1 /.*\(....\)$/ /\1/

Globalizing/Localizing Internal Calls
Create a PT strictly for the phones – assign all phones to this partition
Create a translation CSS – assign the phone PT
Add a PT for each GW
Add this PT to the CSS on each phone
Add a translation pattern that adds a +1 to each

DN    Translation Pattern    Phone
PT=phone-pt   Pattern =XXXX    PT=pt-hq-tp
   PT = pt-hq-tp    CSS=css-hq-int
   CSS = css-tp
   Prefix digits = +1

## SRST (CME)

When in SRST mode, appropriate dial peers, translations and COR must be configured on the gateway

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# CUCME/IOS

## Dial Peers

Two type of dial peers; POTS and voice-voice
POTS – PSTN, PBX, Telephone, etc.
Voice-Network – Gateway, CUCM, etc.

Dial peers and call legs



**Inbound (POTS)**
**Incoming-called number** – matches the called number from the phone.
Normally use -
**Direct-inward dial** – looks for a destination pattern that matches the DNIS
**Port** – physical port

**Outbound (VOIP)**
**Destination pattern** – matches the called number from the phone
**Session-target** – specifies the endpoint for the peer

### Digit Manipulation

**Digit Striping and Prefixes**
POTS by default strips matching the destination pattern. To change this, use either no **digit strip** or **prefix** commands
**No digit strip** disables automatic digit stripping
**Prefix** adds digit in front of the dialed string

**Forward Digits**
Controls how many digits are stripped before the dialed string is passed on. Only supported on POTS interfaces

**Number expansion**
Used to expand an extension number into dialable number such as a full E.164 number. Not likely a lab subject so no detail included here

**Voice Translation Rules**
Translation rules are used to change ANI, DNIS or the numbering type of a call
A translation rule which can consist of multiple entries is created then applied to either the called or calling number

3 step process
1. Create voice translation rule
2. Apply rule to profile for either called or calling number
3. Apply to dial peer in either the incoming or outgoing direction

**Voice Translation Rule Patterns**

| Wildcard | Definition |
|---|---|
| \ | Any single digit |
| 0 to 9, *, # | Any specific character |
| [0-9] | A range or sequence of characters |
| * | Match none or more |
| + | Match one or more |
| ? | Match none or one |

**Basic Voice Translation Rule Examples**

voice translation-rule 1
rule 1 /123/ /999/
! matches 123 and replaces with 999

voice translation-rule 2
rule 1 /^0+/ /909/
! matches any combination starting with zeros (0 ,00, etc) and replaces with 909

**Number Slice**
Number slice is used when a portion of the matched number needs to be copied to the replacement

| Character | Description |
|---|---|
| \ | In the match pattern, indicates where to slice up the number |
| \ | In the replacement pattern, indicates where to copy the sets to keep |
| () | Indicates which sets in the matched number to keep |

| Character Usage | Description |
|---|---|
| (a\) | Keep expression 'a' |
| b\ | Ignore expression 'b' |
| \1 | Copy the first set into the replacement number |

**Format**
/ (x\) y\ (z\) / /\w\1\2/
Split the matched number into three sets of x, y and z. The backwards slash indicates the places to slice up the number. The brackets indicate which sets to reuse in the replacement pattern. The w represents additional digits to be inserted into the replacement number

Two examples:

voice translation-rule 2
rule 1 /^7...\(5555\)/ /900442321411\1/
! starting with 7 then any 3 digits, keep 5555

voice translation-rule 3
rule 1 /^\(3...\)$/ /3214\1/
! any four digits starting with three prepend a 3214 and keep the 3 and last three digits

## MGCP

| | CUCM | | IOS |
|---|---|---|---|
| Choose device, gateway | | **Task description** | **Command** |
| Choose model of router and MGCP | | Create ISDN PRI with needed timeslots and enable the MGCP service | controller T1 0/0/0 pri-group timeslots 1-3,24 service mgcp |
| The domain name must match the FQDN of the router. Choose the CUCM group. Select the module with the WIC and the subunit | | Provide PRI back haul to CUCM | interface Serial0/0/0:23 isdn bind-l3 ccm-manager |
| Configure the WIC. Select PRI and other options as needed. | | Provide redundant CUCM to MGCP service | ccm-manager switchback immediate ccm-manager redundant-host 10.10.210.10 ccm-manager mgcp ccm-manager fax protocol cisco |
| | | Enables MGCP service on router Mandatory command that points to CUCM Allows DTMF symbols to be sent out of band Fax t38 error correction Bind control and media to loopback | mgcp mgcp call-agent 10.10.210.11 service-type mgcp version 0.1 mgcp dtmf-relay voip codec all mode out-of-band mgcp fax t38 ecm mgcp bind control source-interface Loopback0 mgcp bind media source-interface Loopback0 |

## H.323 Gateway

| CUCM | | IOS |
|---|---|---|
| Add into CUCM using 'h323 gateway' | | **Command** |
| Device name should have IP address of Gateway | Set source address for H.323 | Interface Loopback 0 H323-gateway voip bind srcaddr 10.10.110.1 |
| | Provide Codecs needed | Voice class codec 1 Codec preference 1 g711u Codec preference 2 g729r8 |
| | Set misc H.323 values | Voice class h323 1 h225 timeout tcp establish 3 |
| | Configure voice translation rules | |
| | Configure dial peers for call routing | |

## Gatekeeper Admission



## H.323 Gatekeeper

Gatekeeper resolves telephone number
Also accepts bandwidth requests
If enough bandwidth, GK sends destination IP back to requesting Gateway
A local zone is defined as a group of H.323 devices that a Gatekeeper controls. Can have multiple GK per zone for redundancy
GK's do not register with GK's in remote zones.
Location request messages are sent
Technology prefix – tech prefixes are used to group by type or class or to define a pool of Gateways. These are used when no registered E.164 number matches the call

Register CUCM to Gatekeeper

| CUCM |
|---|
| Device – Gatekeeper – Add IP address of GK |
| Device – H.225 Trunk – Add the trunk |

| CME |
|---|
| Under interface – h323-gateway voip |



**Gatekeeper**

| Task Description | Command |
|---|---|
| Define router as Gatekeeper and begin configuration | Gatekeeper |
| Create local zone named NY in domain cisco.com with RAS address 10.1.5.1 using port 1719 | Zone local NY cisco.com 10.1.5.1 1719 |
| Assign the 212 area code to the NY prefix | Zone prefix NY 212........ |
| Bring up gatekeeper | No shutdown |
| Create the default technology prefix for the gatekeeper | gw-type-prefix 1#* default-technology |
| Applies CAC on calls outside of NY zone to 16 Kbps | Bandwidth interzone zone ny 16 |

**Gateway**

| Task Description | Command |
|---|---|
| Select the interface | Interface loopback 0 |
| Configure the loopback interface as the Gateway | H323-gateway voip interface |
| The gatekeeper-id must match the zone name | H323-gateway voip id ny |
| Optionally, name the Gateway for the Gatekeeper | H323-gateway voip h323-id ny@cisco.com |
| Dial-peer example with RAS target | Dial-peer voice 91 voip destination-pattern 9T |