Route Patterns, Lists and Groups

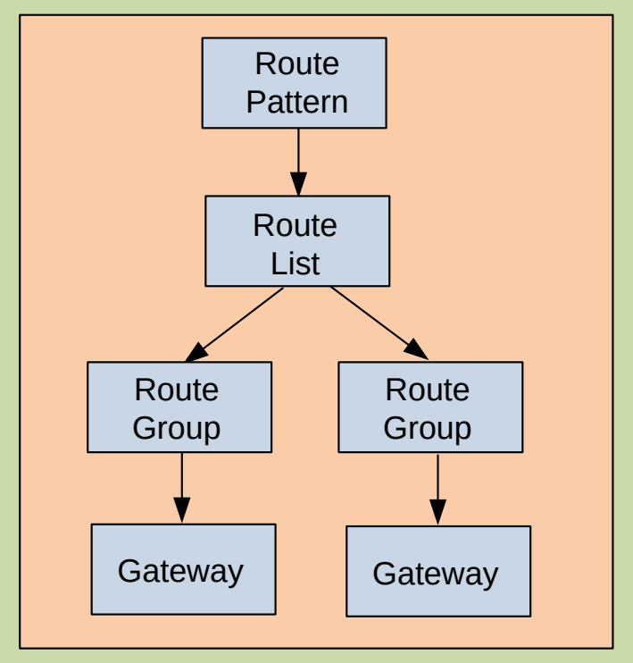
Route pattern
- Match dialed numbers
- Digit manipulation
- Points to a route list

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)

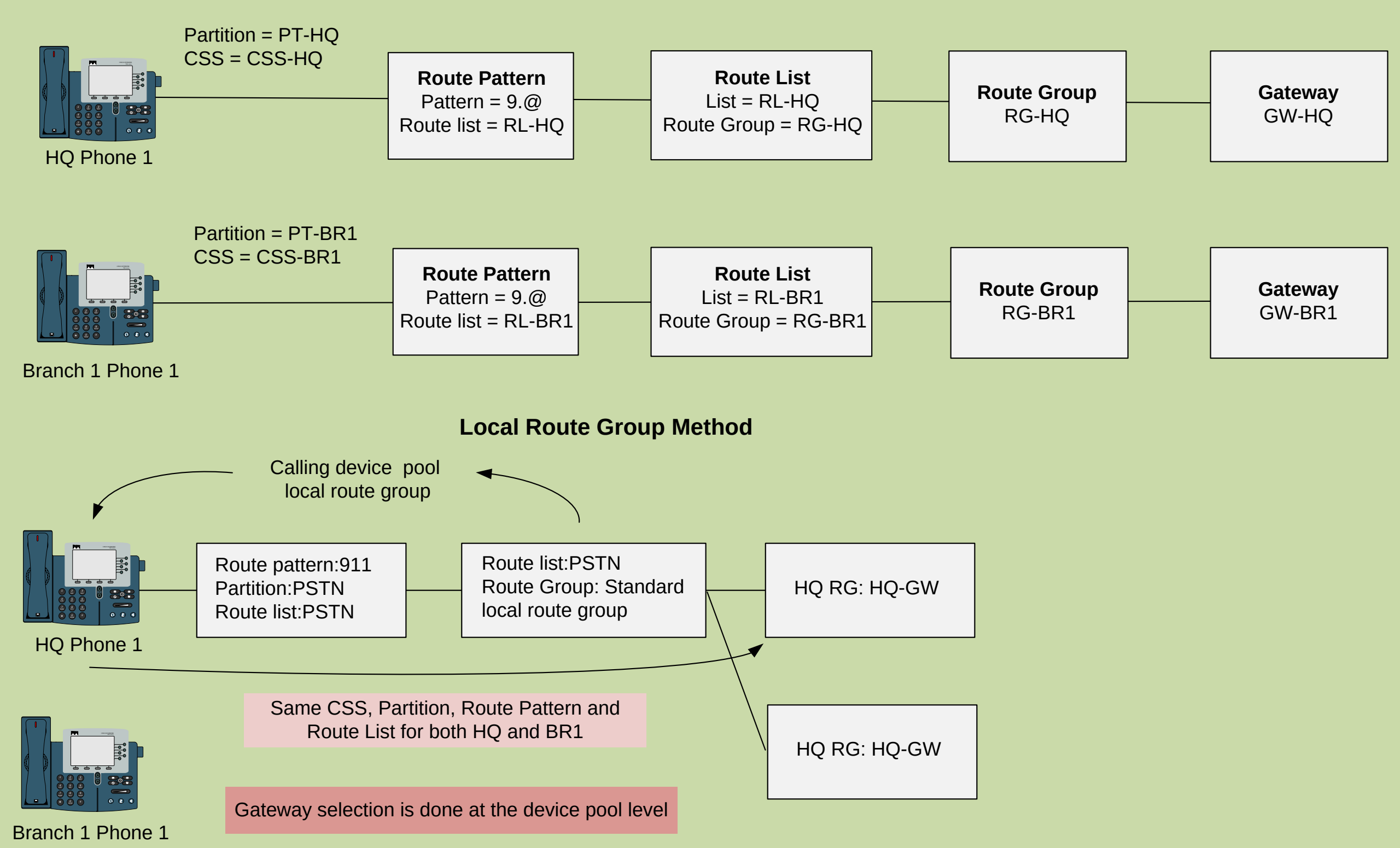


Called Number Manipulation
- DDI (i.e. predot)
- Mask (can add, delete or replace)
- Prefix (add)

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)

HQ Phone 1 — Partition = PT-HQ, CSS = CSS-HQ → **Route Pattern** Pattern = 9.@, Route list = RL-HQ → **Route List** List = RL-HQ, Route Group = RG-HQ → **Route Group** RG-HQ → **Gateway** GW-HQ

Branch 1 Phone 1 — Partition = PT-BR1, CSS = CSS-BR1 → **Route Pattern** Pattern = 9.@, Route list = RL-BR1 → **Route List** List = RL-BR1, Route Group = RG-BR1 → **Route Group** RG-BR1 → **Gateway** GW-BR1

Local Route Group Method

Calling device pool local route group

HQ Phone 1 → Route pattern:911, Partition:PSTN, Route list:PSTN → Route list:PSTN, Route Group: Standard local route group → HQ RG: HQ-GW / HQ RG: HQ-GW

Branch 1 Phone 1

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

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
- Uses a Partition and CSS that gives access to the translation pattern. Partition and CSS are not used for calls
- Recommended practice is to use a name in the PT/CSS that distinguishes this PT/CSS from others. Example: CSS-**ANI**-GW-HQ
- Calling party transformation pattern does not affect call routing. Call will never fail due to calling party transformation.

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-SFO-GW	Prefix 415888	subscriber

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

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

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 version 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 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

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
PT=phone-pt
CSS=css-tp

Translation Pattern
Pattern =XXXX
PT = pt-hq-tp
CSS = css-tp
Prefix digits = +1

Phone
PT=pt-hq-tp
CSS=css-hq-int

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 Call Forward Unregistered for both internal and external calls

SRST (CME)

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

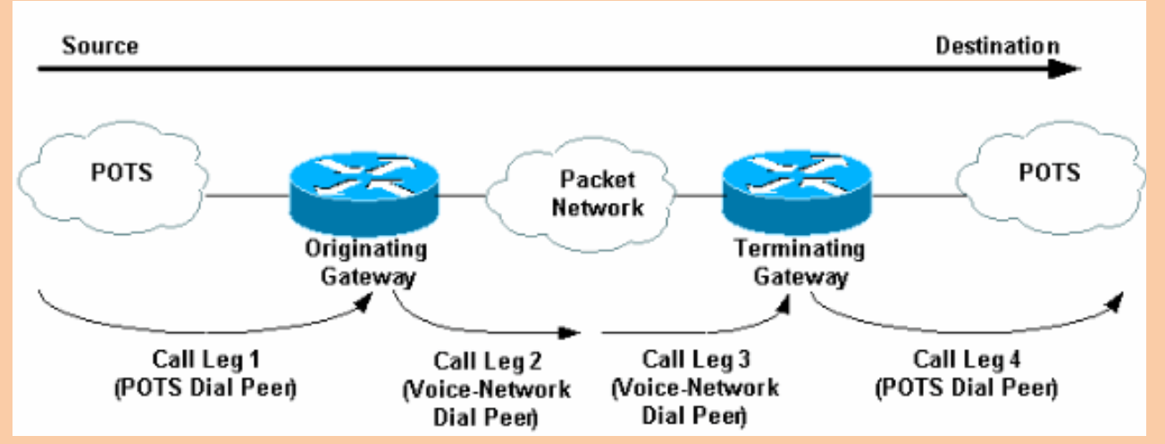
Call-manager-fallback	Telephony-service
ip source-address 1.1.1.1 port 2000	srst mode auto-provision all | dn | none
Max-ephones 1	Srst dn line-mode octo
Max-dn 1 dual-line	Srst dn template <tag>

CUCME/IOS

Dial Peers

Two type of dial peers; POTS and voice-network
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...\(5555\)/ /90044232141\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
```

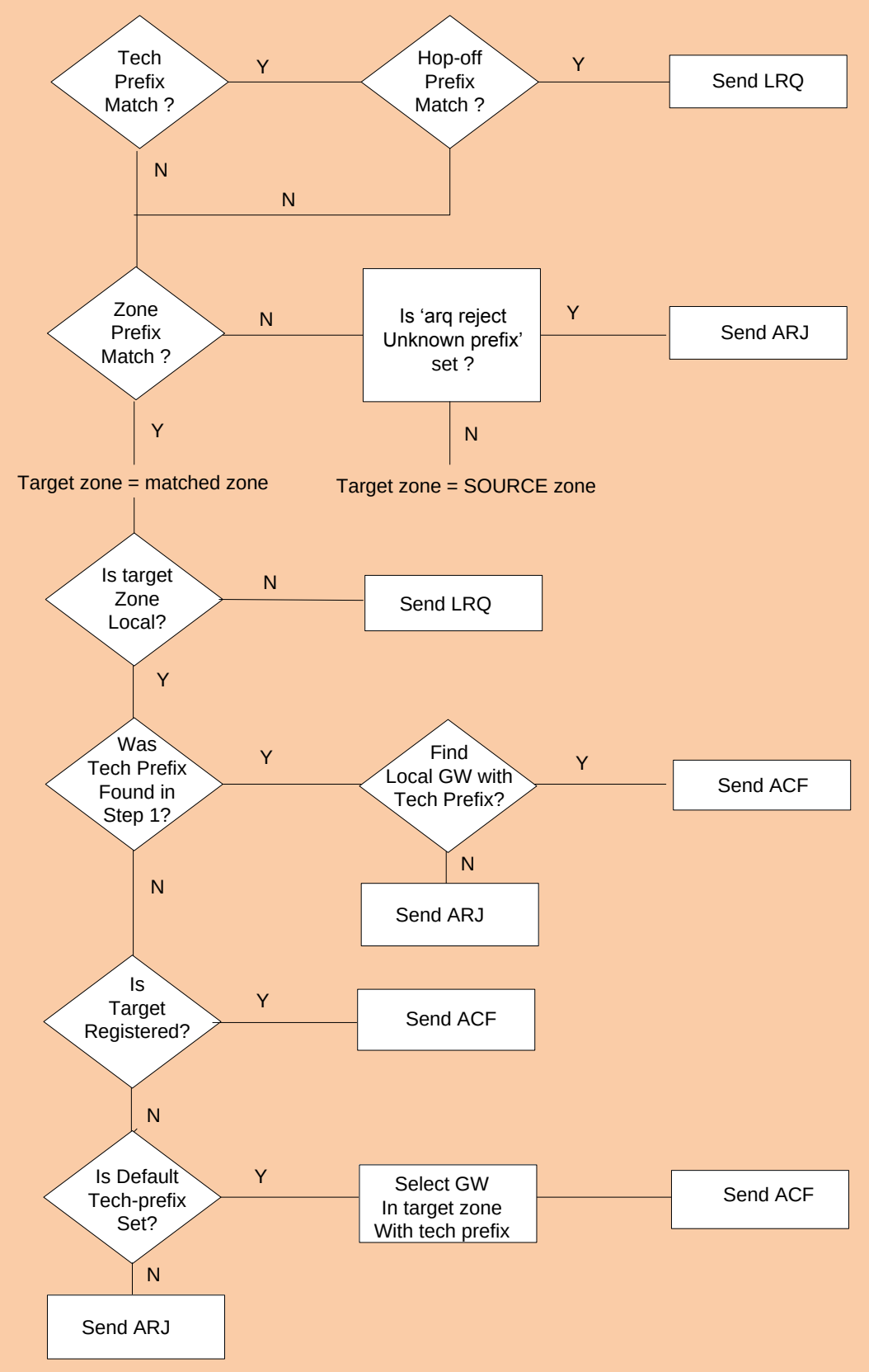
MGCP

CUCM
- Choose device, gateway
- Choose model of router and MGCP
- The domain name must match the FQDN of the router. Choose the CUCM group. Select the module with the WIC and the subunit
- Configure the WIC. Select PRI and other options as needed.

IOS

Task description	Command
Create ISDN PRI with needed timeslots and enable the MGCP service	controller T1 0/0/0 pri-group timeslots 1-3,24 service mgcp
Provide PRI back haul to CUCM	interface Serial0/0/0:23 isdn bind-l3 ccm-manager
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

Gatekeeper Admission



H.323 Gateway

CUCM
- Add into CUCM using 'h323 gateway'
- Device name should have IP address of Gateway

IOS

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	

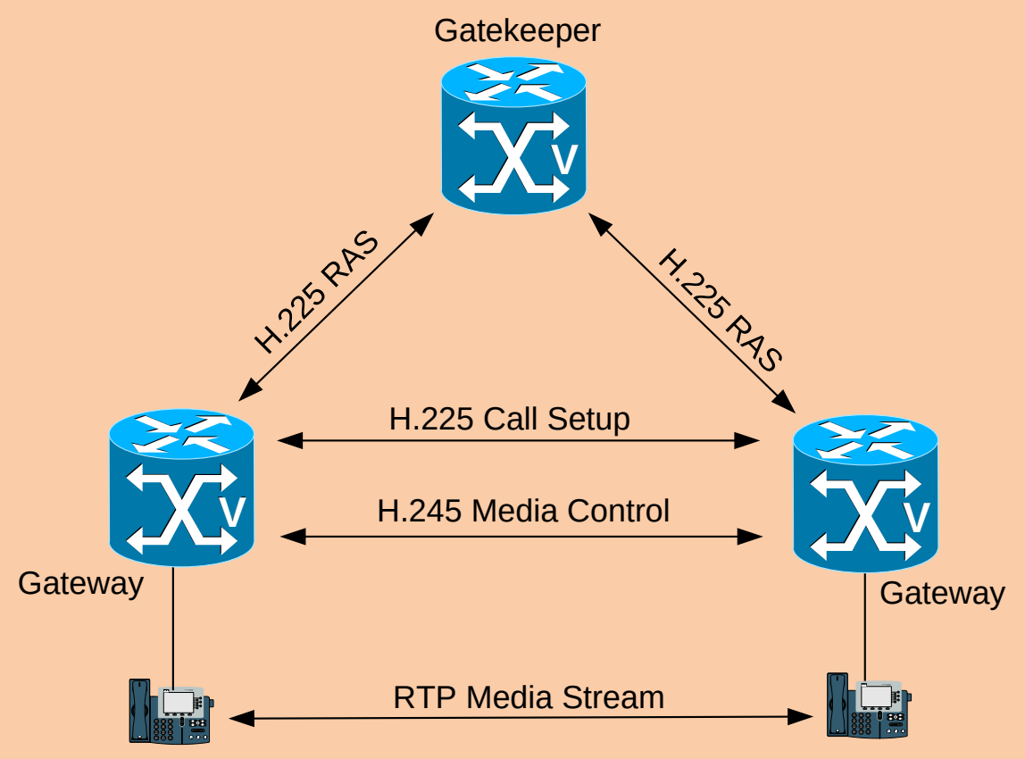
H.323 Gatekeeper

- 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
- Global - gateway



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 session target ras
Set the technology prefix	Tech prefix-1#