Cisco Unified Communication Manager Quick and Useful Notes

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Digits Analysis and Closest-Match Routing

Cisco Unified Communications Manager uses closest-match logic to select the best pattern.

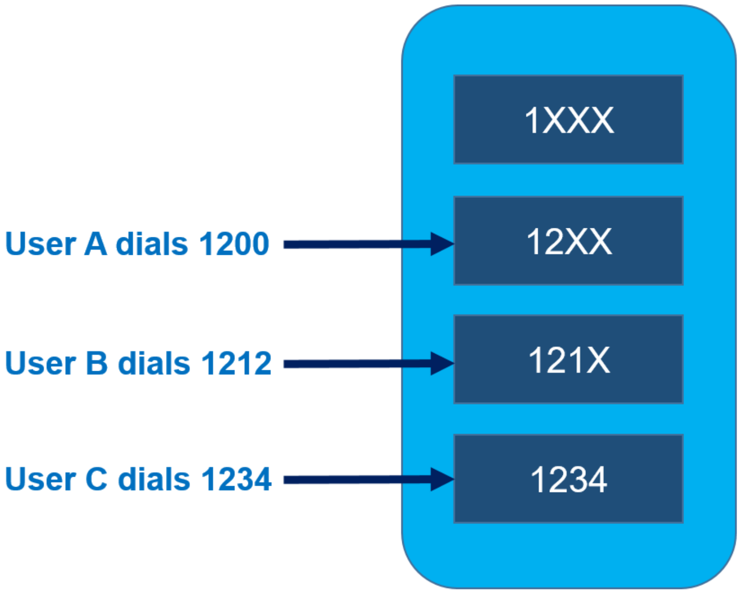
When multiple matching patterns are present, the best pattern is selected based on two factors:

It matches the dialed string.

AND

It matches the fewest strings other than the dialed string.

For example, the call-routing table includes the patterns **1XXX**, **12XX**, **121X**, and **1234**.



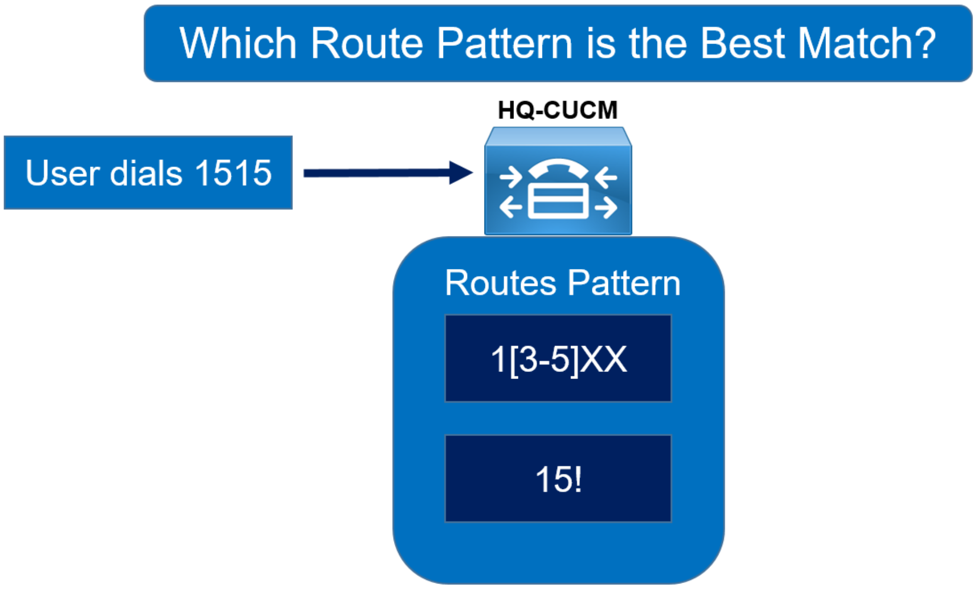
When User A dials the string **1200**, Cisco Unified Communications Manager compares it to the patterns in its call-routing table. In this case, there are two potentially matching patterns: **1XXX** and **12XX**. Both of these patterns match the dialed string, but 1XXX matches a total of 1000 strings (from **1000** to **1999**), whereas **12XX** matches only 100 strings (from **1200** to **1299**).
Therefore, **12XX** is selected as the destination of this call.

When User B dials the string **1212**, there are three potentially matching patterns: **1XXX**, **12XX**, and **121X**. As mentioned previously, **1XXX** matches 1000 strings and **12XX** matches 100 strings. However, **121X** matches only 10 strings. Therefore, **121X** is selected as the destination
of the call.

When User C dials the string **1234**, there are three potentially matching patterns: **1XXX**, **12XX**, and **1234**. As mentioned earlier, **1XXX** matches 1000 strings and **12XX** matches 100 strings.
However, **1234** matches only one string (the dialed string); therefore, **1234** is selected as the destination of this call.

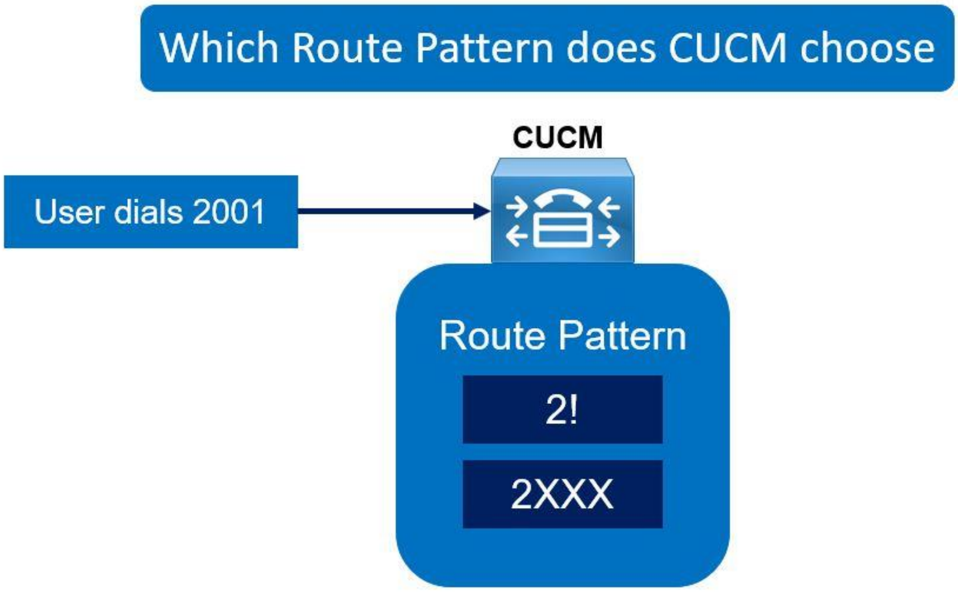
Let's take another example:

1[3-5]XX matches 200 numbers, and **15!** stands for unlimited possible numbers.
When dialing **1515,** the route pattern **15!** is the best match and it's used to route the call instead of the route pattern **1[3-5]XX**.



15! Route Pattern matches unlimited Digit Strings, but for the purposes of Closest Match Routing in this case, this matches 100 Digit Strings because you only consider the number of potential strings with the given number of digits dialed.

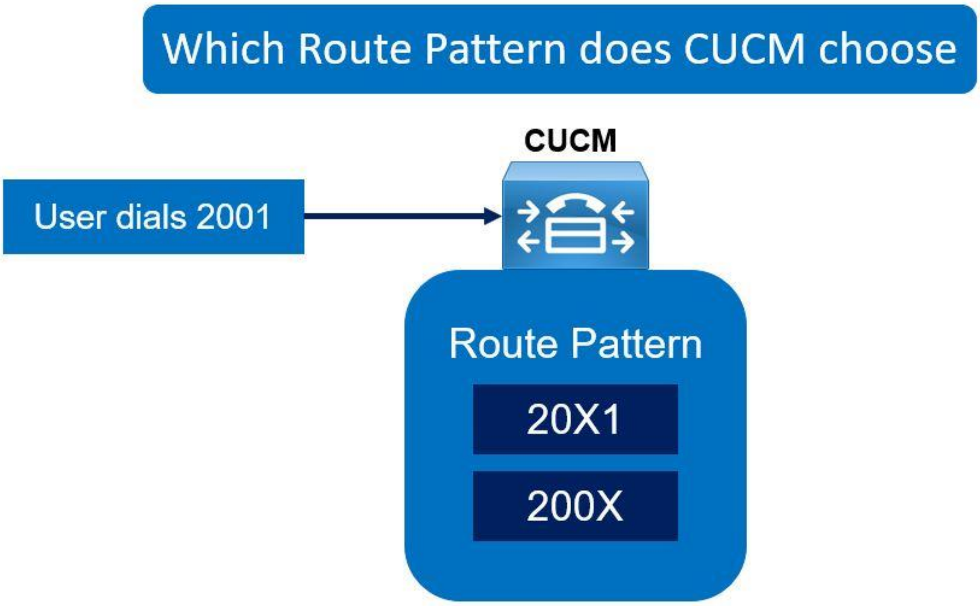
Another example with two Route Pattern **2!** And **2XXX**:



Route Pattern the **2!** pattern matches an infinite number of digits because the pattern matches a variable number of digits but for the purposes of closest-match routing, digit

analysis only evaluates the number of potential matches given the number of digits dialed. In other words, if a user dials **2001**, digit analysis treats the **2!** as **2XXX**, which means it also matches **1000** possible numbers.
In this scenario, it is non-deterministic which means that you don't know which Route Pattern will be used to route the call. To avoid this kind of behavior, you can use partitions and calling search spaces to give priority to one or the other, by the way it is not a good practice to have such ambiguity in your dial plan.

Another example with two Route Pattern **20X1** and **200X**.



The dialed number **2001** matches both Route Pattern **20X1** and **200X**, each matching 10 numbers. Which one does digit analysis choose? In this scenario, again it is non-deterministic, so we don't know which one will be used.

Conclusion, we need to take into consideration the overlapping patterns in order to know which pattern is a better match, in other how the call is routed.

Digit Manipulation On Cisco Unified Communication Manager

Digits manipulation on Cisco Unified Communication Manager is an important part of Dial Plan, as a voip engineer, mastering this concept is mandatory because the real requirements you meet in a deployment, today a multisite deployment is very popular for large enterprise, having different sites, means different Dial Plan at each site, the HQ site has the Directory numbers that starts with 1XXX for example and two remote sites with the directry numbers starts with 2XXX and 3XXX for example, to allow inter site dialing you plan to use 4 digits for intra-site dialing and 6 digits for inter-site dialing so that between sites, users dial a site access code followed by a site code and the destination's on-net extension (4 digits). This means digit manipulations must be implemented, another requirement when you need to manage PSTN dialing, so the calling parties of 4 digits must be converted to an E.164 format to be conform with PSTN dial plan standards, this means also that a digit manipulation must be involved.

To be able to answer these requirements, we need to know:

First Where: at which levels on Cisco Unified Communication Manager we can do Digits Manipulations?

- Route Pattern.
- route group at the route list details level.
- translation pattern.
- 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).

Second When: Which level of Digit Manipulation is appropriate to my requirement? this is the big deal.

To answer this question, you need to know two things:

Which Concept of Digit Manipulation is a part of routing process Number Transformation?

- Translation Pattern –called party transform directly influences routing decision
- Route Pattern –called party transform after routing decision
- Route Lists –called party transform after routing decision

Which Concept of Digit Manipulation is independent of 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)

Calling Transformation Pattern at the Phone Level

The Calling Transformation Pattern settings and available at the Phone Section on Cisco Unified Communication is sometimes confusing.

It allows the modification of the calling number or Number Presentation Transformation. If we look at the section, there are two options :

1. Caller ID For Calls From This Phone
2. Remote Number

The question that arises what is the difference and where to use these two options ?

First we can explain the purpose of the Calling Transformation Pattern at the phone level as follow :

When I call a number -> The number you see.

Let’s take an example.

The Phone-1 is registered with DN \+40854541001 while the Phone-2 is registered with DN \+21277882001.

Phone-1 calls Phone-2. Since we are using a Globalized Format for DNs, we want to use and display the calling number in 4 Digits for internal calls. In this case the calling device is \+40854541001 and Phone-2 wants to see 1001.

To do this, the Number Presentation Transformation settings provides the solution to modify the calling number by applying a Calling Transformation Pattern so that when it rings at the called device or the destination device.

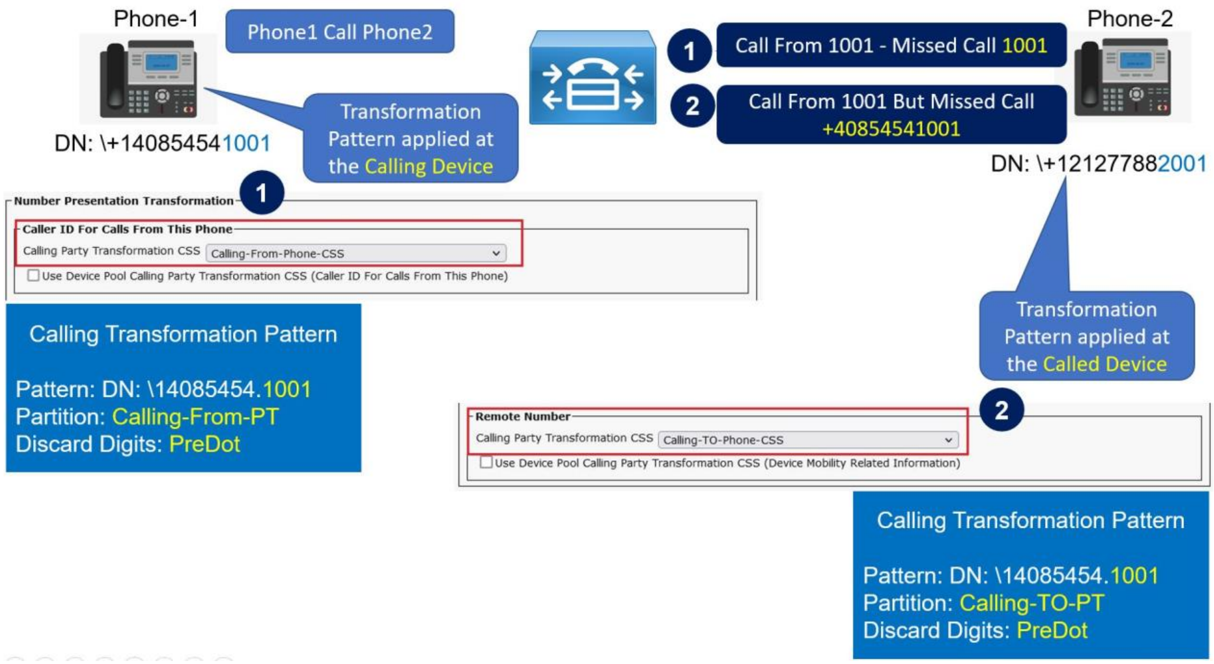
But, be carefull where to apply the Calling Transformation Pattern. As mentioned previsouly there are two options under the Number Presentation Transformation Settings :

1. Caller ID For Calls From This Phone : Apply the Calling Transformation Pattern at the Originating Device – Phone-1 in this case.

2. Remote Number : Apply the Calling Transformation Pattern at the Destination Device – Phone-2 in this case.

The difference between these two options is the way the missed calls are displayed .

As shown in the topology, The Transformation Pattern applied at the originating device will affect the Missed Calls presentation.
But when applying the Transformation Pattern at the destination device, the Transformation Pattern does not affect the Missed Calls presentation.


Phone-1
Phone1 Call Phone2
DN: \+140854541001
Transformation Pattern applied at the Calling Device
1
Call From 1001 - Missed Call 1001
2
Call From 1001 But Missed Call +40854541001
Phone-2
DN: \+121277882001
Number Presentation Transformation
1
Caller ID For Calls From This Phone
Calling Party Transformation CSS
Calling-From-Phone-CSS
Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)
Calling Transformation Pattern
Pattern: DN: \14085454.1001
Partition: Calling-From-PT
Discard Digits: PreDot
Transformation Pattern applied at the Called Device
Remote Number
2
Calling Party Transformation CSS
Calling-TO-Phone-CSS
Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)
Calling Transformation Pattern
Pattern: DN: \14085454.1001
Partition: Calling-TO-PT
Discard Digits: PreDot

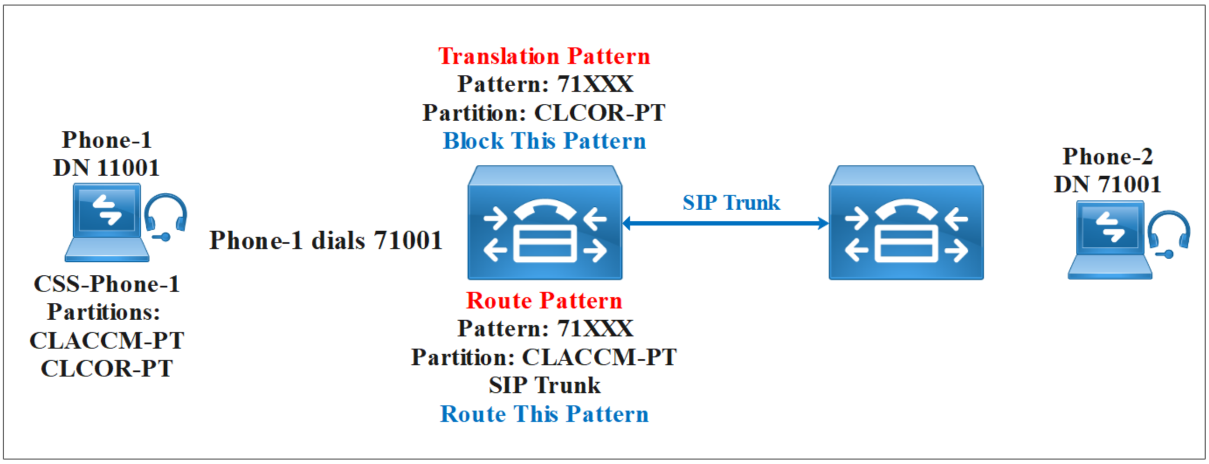
Route Pattern vs Translation Pattern vs Transformation Pattern

When talking about Dial Plan and Digit Manipulation Cisco Unified Communication Manager, three components play an important role to perform the transformation of the calling and called parties.

The Translation Pattern, Transformation Pattern and Route Pattern have a common word which is “Pattern”, the Pattern in the three components is checked by the CUCM to find a match of the Called Number when receiving a Call.

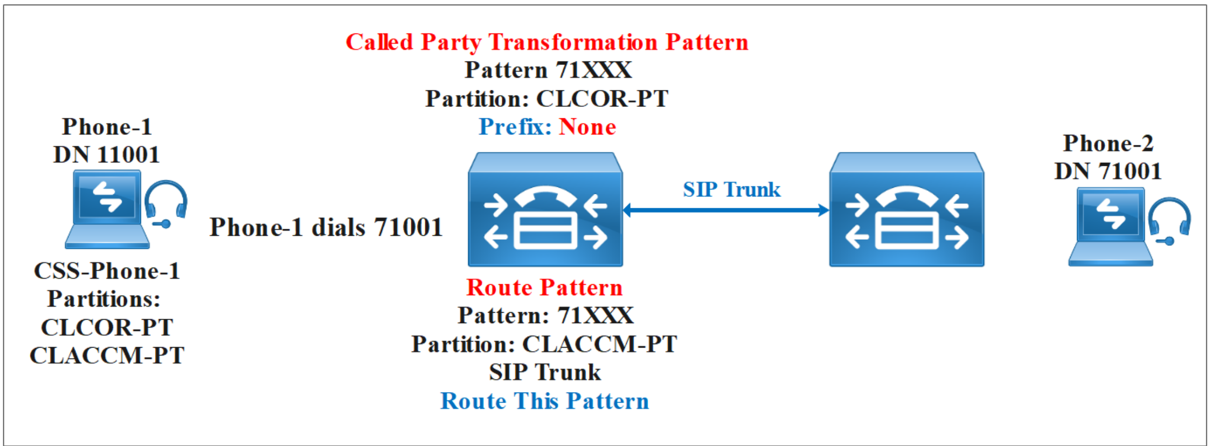
The priority to apply which component depends on the configuration of CSS and Partition, through the 4 use cases and examples below you will see when the translation pattern or transformation pattern or route pattern is applied first.

Case 1:



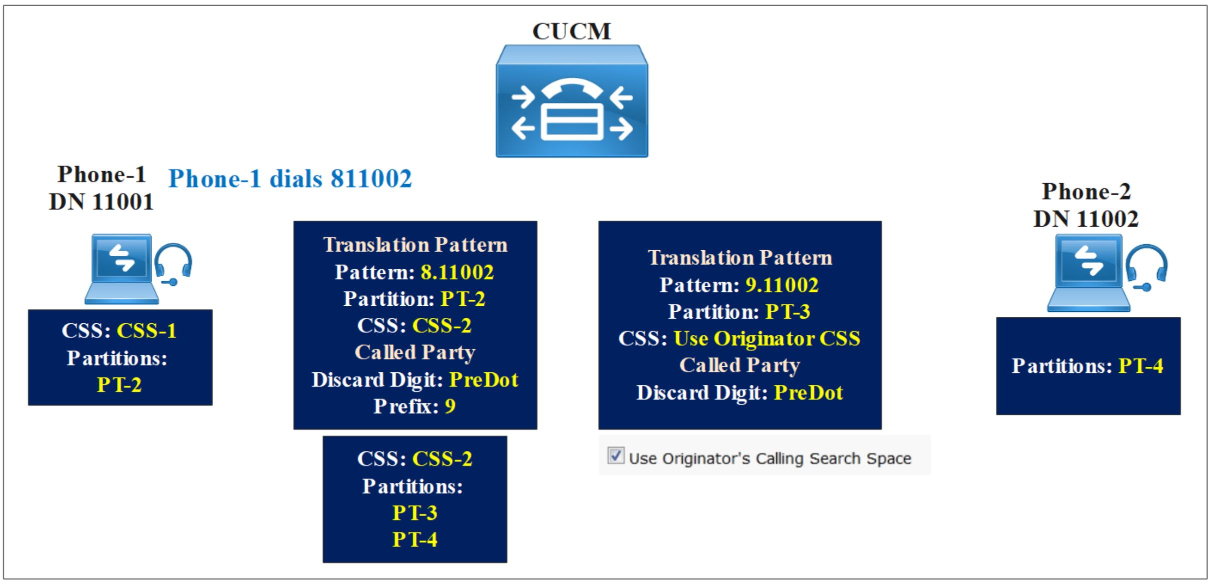
The **Phone-2** will ring because when we have equals match in this case a Translation Pattern and a Route Pattern then CUCM uses higher Partition in CSS (the Partition listed first). In this case **CLACCM-PT** is higher than **CLCOR-PT** so it will match the Route Pattern and will routes the call. So the **Phone-2** will ring because the RP is listed first in the partitions list.

Case 2:



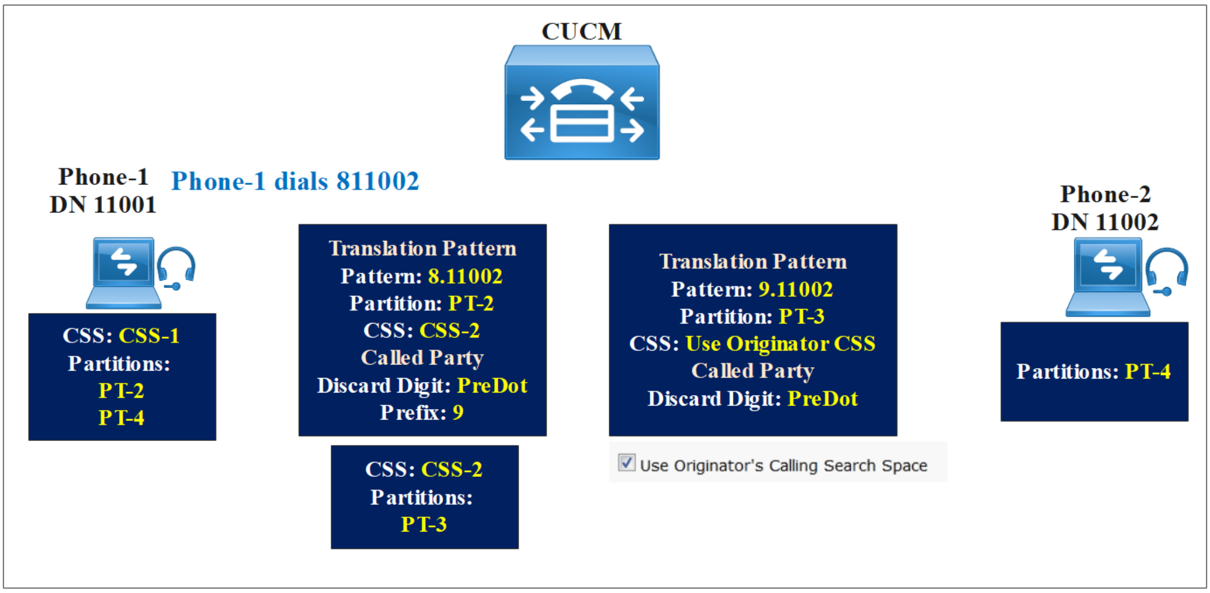
The **Phone-2** will not ring. The Called Transformation Pattern is applied after the route-pattern decision and after the Gateway or SIP Trunk Selection. Here what happen, the **Phone-1** CSS has access to the Called Transformation Pattern Partition, this Partition is listed first, the result is that there is a blackhole. and the call will not be routed.

Case 3:



Phone-2 will not ring. There are two sources of the call, **Phone-1** and the FIRST translation pattern **TP-1**, the Use Originator CSS option in the second translation pattern **TP-2** means that the CUCM will use the CSS of the phone, even if from the second translation pattern's perspective, the call comes from the first translation pattern.

Case 4:



The **Phone-2** will ring, Called Number **811002** hits the Translation Pattern **TP-1**, the **TP-1** translates the Called Number to **911002** and it has the **CSS-2** that includes the Partition **PT-3** of the second Translation Pattern **TP-2**, the new Called Number **911002** hits the **TP-2**, which uses originators CSS and **Phone-1** has a **PT-4** in its **CSS-1**. That allows **Phone-1** to call **Phone-2**.

SIP Method for Transfer Type: Release to Switch VS Supervise Transfer

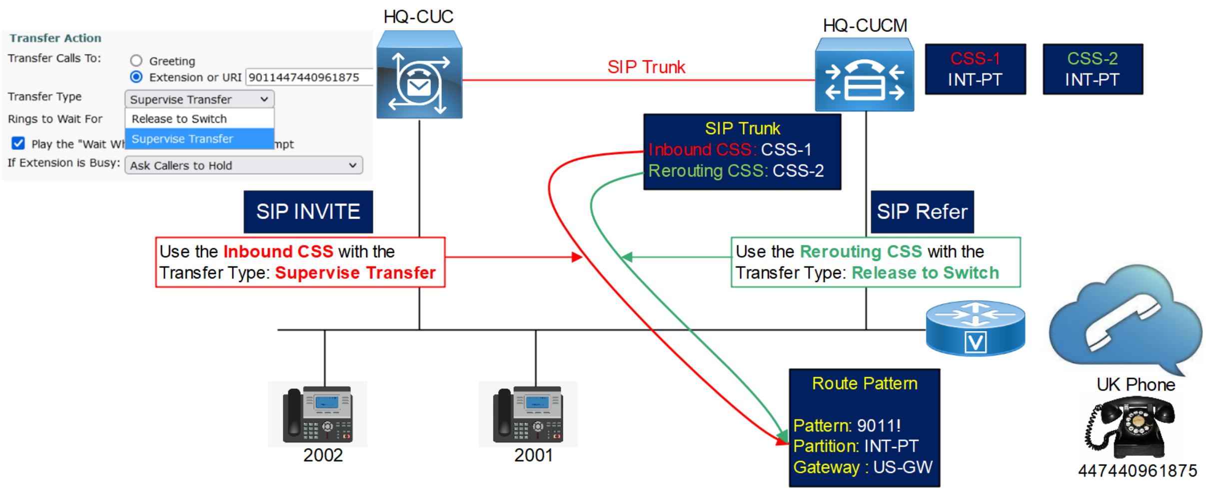
To prevent the toll fraud using the Transfer Rules on Cisco Unity Connection, we can do it through the Cisco Unity Connection with the Restriction Tables of on Cisco Unified Communication Manager.

On CUCM, I read per documentation that we need to modify the rerouting CSS in the SIP Trunk to include only the required partition, so it should not include the partition of the route pattern to PSTN International numbers.

I would like to say that it depends, because even the inbound CSS in the SIP Trunk must be taken into consideration, it depends to the Transfer Type you select in the Transfer Rules, the reason is:

Release to Switch: this type of transfer uses the SIP Refer method to reroute the call to the specified extension, it is sent in dialog in other words in the existing SIP conversation---> we need a Rerouting CSS in the SIP Trunk between CUCM and CUC.

Supervise Transfer: This type of transfer uses the SIP INVITE method, it is sent out of dialog, this means it's a new SIP conversation----> therefore we need the Inbound CSS.



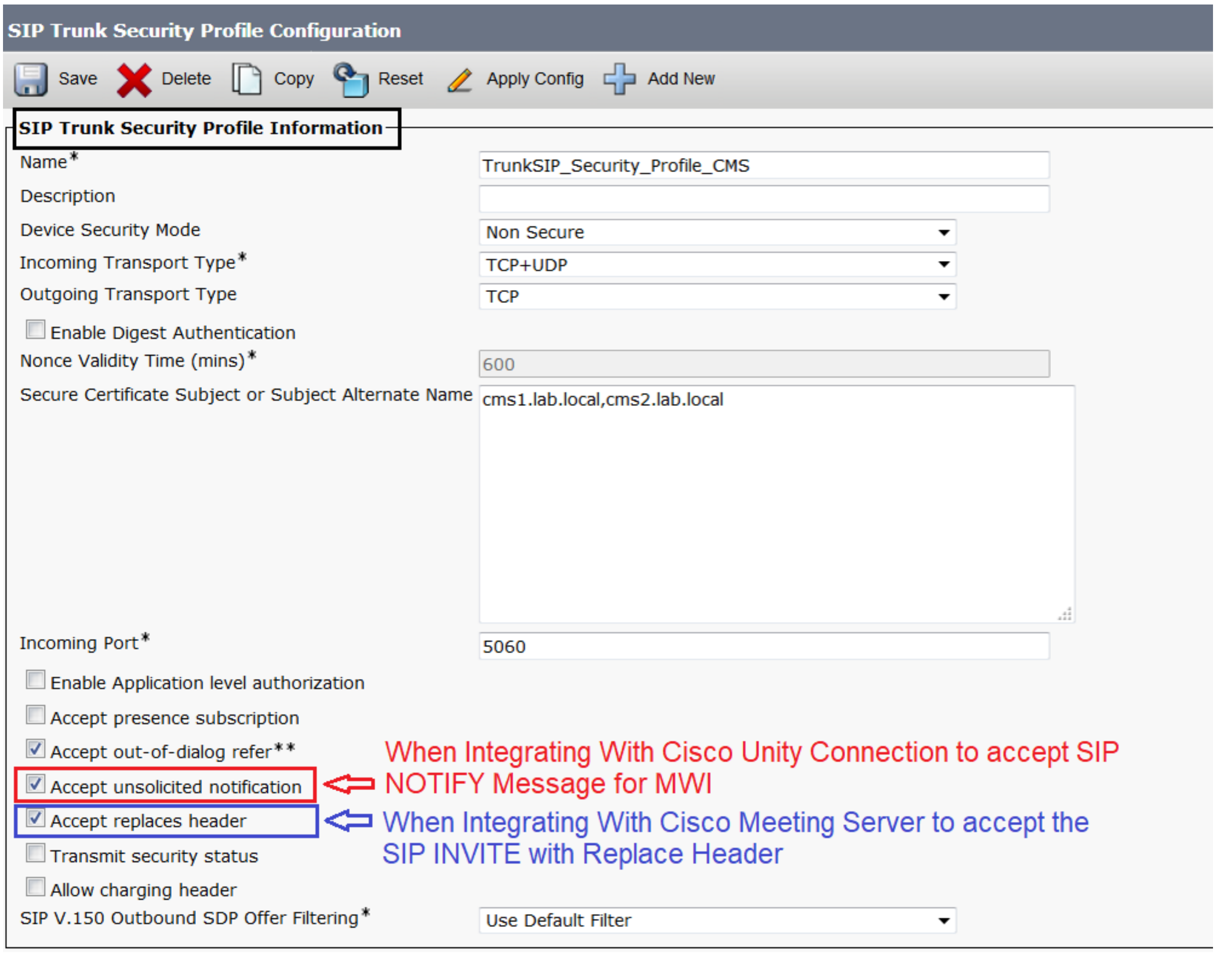
Accept Unsolicited Notification and Accept Replaces Header

Accept Unsolicited Notification

Common use case is when Cisco Unity Connection is integrated with Cisco CUCM. A caller leaves a message.
MWI (Message Waiting Indicator) is sent by Cisco Unity Connection to Cisco CUCM using a SIP NOTIFY message.
This SIP NOTIFY message is not part of an early SIP dialog or conversation.
In other words, Cisco CUCM receives a SIP NOTIFY without an early solicitation, neither requested nor solicited.
If the option Accept Unsolicited Notification is not checked. The CUCM sent a SIP Message 403 Forbidden to CUC.

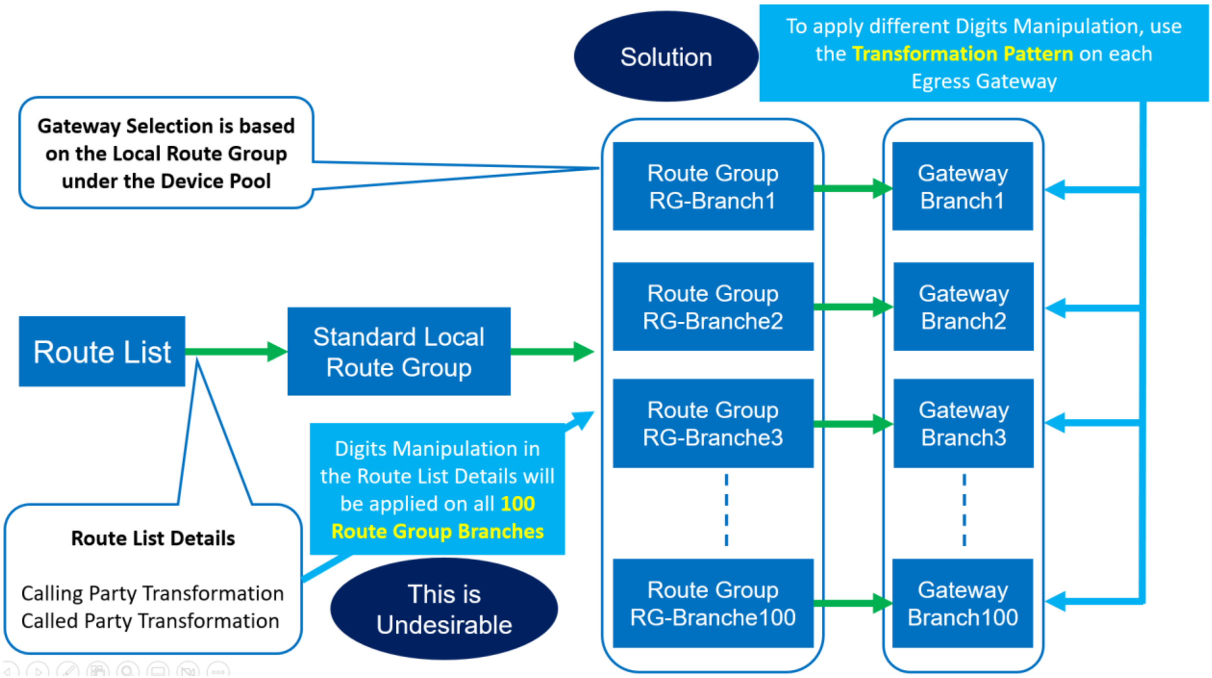
The Accept Replaces Header

Common use case is when integration a cluster of Cisco Meeting Server CallBridges, and the Call Bridge Group feature is enabled. When a conference is alreaty active on one callbridge node, and a second call to the same conference is routed to a second callbridge node. The first callbridge hosting the conference will request to retrieve the original SIP dialog between Cisco CUCM and the second callbridge.
This is done by sending a SIP INVITE with Replace header including the Call ID + TAG FROM + TAG TO that uniquely identify the SIP DIALOG (between CUCM and the second CallBridge).
If the option Accept Replaces Header is not checked. The CUCM sent a SIP Message 403 Forbidden to the first CallBridge.


SIP Trunk Security Profile Configuration
Save
Delete
Copy
Reset
Apply Config
Add New
SIP Trunk Security Profile Information
Name*
TrunkSIP_Security_Profile_CMS
Description
Device Security Mode
Non Secure
Incoming Transport Type*
TCP+UDP
Outgoing Transport Type
TCP
Enable Digest Authentication
Nonce Validity Time (mins)*
600
Secure Certificate Subject or Subject Alternate Name
cms1.lab.local,cms2.lab.local
Incoming Port*
5060
Enable Application level authorization
Accept presence subscription
Accept out-of-dialog refer**
Accept unsolicited notification
Accept replaces header
Transmit security status
Allow charging header
SIP V.150 Outbound SDP Offer Filtering*
Use Default Filter
When Integrating With Cisco Unity Connection to accept SIP NOTIFY Message for MWI
When Integrating With Cisco Meeting Server to accept the SIP INVITE with Replace Header

Transformation Pattern use case

One of the possible challenges that the local route group feature introduces is the inability to perform calling or called number transformations on a per-route-group basis. With a normal route group, you can manipulate calling and called party numbers by using the route list details configuration to modify numbers on a per-route-group basis, but with a local route group, these transformations apply to all the route groups configured as a local route group. For example, if you have route groups named Branch 1 PSTN through Branch 100 PSTN assigned as the local route group named Branch PSTN on 100 different device pools and you have a single route list that contains the Branch PSTN local route group, any transformations you perform on the route list details for the Branch PSTN local route group will be applied to all 100 route groups when they are used as part of this route list. If you want to be able to manipulate digits differently, depending on which of the branch PSTN routes you take, the transformation patterns is the solution.



Run on all active unified CM Nodes

To understand the “Run on all active unified CM Nodes” option on Route List and SIP Trunk, we need first to dissect the outbound call process without this feature.

When an outbound call is initiated from an endpoint and sent to CUCM cluster. One node will do digit analysis and finds a route pattern.

There are two scenarios before extending the call to the SIP Trunk.

1-The Route Pattern points directly to the SIP Trunk. In this case the calling device is the endpoint.

- If the endpoint is registered to the same node which is a member of the CUCM Group assigned to the SIP Trunk, then the node on which the endpoint is registered will be used to initiate the outbound call.

- If the endpoint is registered to a node which is not a member of the CUCM group assigned to the SIP Trunk, then the node will randomly distribute the calls accross the nodes in the Trunk’s CUCM group.

2-The Route Pattern points to the Route List. In this case the calling device is the Route List.

- If the Route List is not registered to node which is a member of the CUCM group assigned to the SIP Trunk, then the node will randomly distribute the calls accross the nodes in the Trunk’s CUCM group.

- If the Route List is registered to a node which is also a member of the CUCM Group assigned to the SIP Trunk, then the node on which the Route List is registered is used to initiate the outbound call.

According to this logic, there are two caveats when we have 8 Cisco CUCM for call processing or 16 in mega cluster:

1. The first caveat: The Route List is only active on one CUCM node, which is not recommended since all outbound calls will be initiated by the node where the Route List is registered.

2. The second caveat: if the SIP Trunk’s CUCM group is used, we can have max 3 CUCM servers to process outbound calls.

With the “Run on all active unified CM Nodes” option on either the Route List or the SIP Trunk, we can have max 8 nodes or 16 in mega cluster to process the outbound calls.

Activating “Run on All active Nodes” on the SIP trunk could lead to major outages, especially in big deployments as with many CUCM sunscribers or mega clusters and shared SIP trunks with Carriers, because almost all carriers will accept calls ONLY from the CUCM subscribers configured at the carrier’s end, and will reject calls from any other call manager subscriber. Most carriers will configure few CUCM subscribers at their end. The CUCM will not retry the call again if it got rejected (by default). The same applied to the Voice gateways if it accepts calls only from a trusted list or the CUCM IPs in the dial peers.

For example: If the Carrier is using max. three CUCM IP subscribers as the trusted IPs which are also in the CUCM group of the SIP trunk > everything will work fine. If You enable “Run on All Active nodes” on the Route List and the SIP trunk and the IP Phone initiating the call is registered to a CUCM that is not among these maximum three IPs > All calls will fail. In this case the “Run on all active unified CM Nodes” option must be disabled on both the Route List and SIP Trunk and the CUCM group configured in the device pool assigned to the SIP trunk should contain the same CUCM subscribers configured at the Carrier’s end or the Voice gateway.