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CCNP Collaboration Call Control and Mobility

CLACCM 300-815

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One difference between this translation pattern and the previous one is that Do Not Wait for Interdigit Timeout on Subsequent Hops is selected so that the subsequent match (which is the +! route pattern) does not have to wait for interdigit timeout. The other difference is that the digit discard instructions remove the trailing pound sign.

You might be wondering why the third and fourth translation patterns are needed since the patterns created so far already match the number in +E.164 format. These patterns are needed to ensure that the calling party number gets globalized. These patterns are configured as follows:

- **Pattern:** \+!
- **Partition:** NANP_International_PSTN
- **Calling Search Space:** Global_PSTN_Patterns_CSS
- **Calling Party Transformations:**
  - **Use Calling Party's External Phone Number Mask:** selected

Finally, the last translation profile needs to be configured as follows:

- **Pattern:** \+!#
- **Partition:** NANP_International_PSTN
- **Calling Search Space:** Global_PSTN_Patterns_CSS
- **Do Not Wait for Interdigit Timeout on Subsequent Hops:** selected
- **Calling Party Transformations:**
  - **Use Calling Party's External Phone Number Mask:** selected
- **Called Party Transformations:**
  - **Discard Digits:** PreDot Trailing-#

Each of these translation patterns has the calling search space Global_PSTN_Patterns_CSS configured. This calling search space includes only the Global_TEHO_Patterns partition. In a real deployment, this calling search space would also include other globalized route patterns to reach other destinations, even if they are not TEHO sites. For the sake of this example, the partition includes only a single route pattern that matches calls to the United Kingdom (country code 44) and routes them to a route list. The route pattern is configured as follows:

- **Pattern:** \+44.!
- **Partition:** Global_TEHO_Patterns
- **Gateway/Route List:** UK_DC1_DC2_LRG

Most of the parameters on the route pattern are left blank because there are no transformations being performed on the route pattern. Transformations to localize the number will be done either on the route list details for each route group or by using transformation patterns.

The route list contains three route groups, which will route the call to the U.K. PSTN trunk, the two centralized SIP trunks, and the local gateway, based on the local route group. From this point onward, you have several options for how to localize the number back to a form that the PSTN providers are expecting. You could use transformations on the route list details to perform the transformations, but this would cause problems for the local route group if you needed to apply different transformations at different branches that are using the local route group since the route list details configuration applies to the whole local route group.

Instead, you can use calling and called transformation patterns applied on the trunks or gateways. You need two sets of transformation: one to handle the calling and called party transformations for calls that egress through the U.K. PSTN and another set to localize the calling and called party numbers for egress through the U.S. PSTN. The first called party transformation pattern handles localization of the called party number for the United Kingdom:

- **Pattern:** \+44.!
- **Partition:** UK_Called_Xform_CSS
- **Called Party Transformations:**
  - **Discard Digits:** PreDot
  - **Prefix Digits:** 0

As you can see, this transformation pattern would remove the +44 and prefix a 0 to make the called number compatible with the U.K. PSTN. This converts the number +441632960286 to 01632960286. This transformation pattern would then be applied to the U.K. SIP trunk because the called party transformation calling search space is set to UK_Called_Xform_CSS.

Next, configure the following calling party transformation pattern to transform the calling party number:

- **Pattern:** \+.1!
- **Partition:** UK_Calling_Xform
- **Calling Party Transformations:**
  - **Discard Digits:** PreDot

This transformation pattern may or may not work, depending on how the PSTN carrier in the United Kingdom handles calling number information. It is possible that the carrier will accept the calling party number in +E.164 format, in which case this transformation is not needed. It is also possible that the carrier will not accept an international number for the calling party number, in which case you may have to resort to masking the number with a phone number owned by your company in the United Kingdom so that the called party will at least receive a phone number indicating that the call is originating from your company. If you need to do this, you can add a calling party transformation mask on the calling party transformation pattern with the number you want to send any time a TEHO call originates from the United States. To apply these transformations to the calling party number, you set the calling party transformation calling search space on the U.K. SIP trunk

to UK_Calling_Xform_CSS. Now calls dialed from the United States as international U.K. numbers will be routed via TEHO through the U.K. SIP trunk.

In case the call through the U.K. SIP trunk fails for some reason, the route list includes a second route group. Although not shown in Figure 4-31, this route group contains two SIP trunks: one in each data center. The provider for these two SIP trunks accepts the calling and called party numbers in +E.164 format, as shown in Figure 4-30, so for these two SIP trunks, no calling or called party transformation calling search space configuration is necessary.

If U.K. SIP trunk and both data center SIP trunks fail to route the call, the call is sent to the local gateway at the branch. This gateway is selected via the Local Route Group feature. Based on the device pool of the device, the route group that contains the gateway for that branch is selected. There are two more transformation patterns needed to convert the +E.164 format to a format expected by the local PSTN. Start with the called party number. The PSTN provider providing service to the local gateway you are interested in expects the calling and called party numbers to be in NANP format, which means the calling party number should be a 10-digit number, and the called party number should adhere to the standard NANP dialing habit. This means that for international calls, the number should be 011 + country code + national number. To transform the number, configure the following transformation pattern:

- **Pattern:** \+.[2-9]!
- **Partition:** US_Called_Xform_CSS
- **Called Party Transformations:**
  - **Discard Digits:** PreDot
  - **Prefix Digits:** 011

Notice that this pattern matches anything that starts with a + followed by 2 through 9. This represents any call to an international destination from the perspective of a U.S.-based caller. The pattern strips the + and prefixes the 011. This means that the number +441632960286 becomes 011441632960286, which is what the local provider is expecting.

Finally, configure the following transformation pattern to deal with the calling party number:

- **Pattern:** \+1.!
- **Partition:** US_Calling_Xform
- **Calling Party Transformations:**
  - **Discard Digits:** PreDot

Notice the position of the dot (.) in this case. The PreDot instruction discards the +1, leaving the 10-digit national number as the calling party number. Once you have put all this together, you will have a flexible TEHO design for calls to the United Kingdom.

Hopefully you can see how this process can be easily extended to other countries as well. If you want to add another TEHO destination, you just need to add a few more patterns to deal with calls to that country; users across your enterprise can then make use of it.

Now that we have covered numeric dialing in depth, it is time to return to the topic of placing calls using alphanumeric URIs.

Alphanumeric URI Routing

Alphanumeric URI dialing is becoming increasingly common, thanks to the adoption of video endpoints and video-capable soft clients, especially in environments where business-to-business (B2B) calling is enabled through Cisco Expressway. Alphanumeric dialing allows for calling between endpoints using a URI that resembles an email address, such as kydavis@cisco.com.

We briefly discussed alphanumeric URIs at the beginning of this chapter. As a reminder, a URI has two distinct components that are separated by the @ sign. The component to the left of the @ is sometimes referred to as the left-hand side (**LHS**), or the user portion, and the component to the right is referred to as the right-hand side (**RHS**), or the host portion. Unified CM generally matches URIs based only on an exact match. If there is no exact match for a full URI, Unified CM tries to route the call based on the host/RHS portion of the URI only.

Figure 4-32 provides a flowchart that outlines the process by which Unified CM determines how to route a URI-based call.

Let's walk through the flowchart. The first determination Unified CM makes when receiving a call in URI form is to check whether the user portion (LHS) of the URI is numeric (that is, only numbers). There is an asterisk next to that question in the flowchart because there is a caveat here: This decision is affected by the setting of the Dial String Interpretation parameter on the SIP profile of the calling device. There are three choices for this parameter:

- Phone Number Consists of Characters 0–9, *, #, and + (This is the default setting.)
- Phone Number Consists of Characters 0–9, A–D, *, #, and +
- Always Treat All Dial Strings as URI Addresses

The first option is the default setting, but if one of the other options is selected, then the "digits" ABCD are considered to be numeric or, in the case of the third option, the URI follows the NO path in the flowchart, regardless of whether the LHS is numeric. An additional caveat is that if the SIP request URI contains user=phone—for example, `sip:*1234@172.18.110.48;user=phone`—Unified CM will treat the dialed string as a number, regardless of the Dial String Interpretation setting.

Let's focus on the flow if the LHS is not numeric, which is the case for most alphanumeric URIs. The first thing that occurs is Unified CM searches for a match for the URI in the calling search space configured on the phone in the same way that it would search for a directory number or another pattern. URIs are configured on the directory number configuration page alongside the numeric pattern. In fact, you can configure up to five directory URIs on a directory number, and you can also configure Unified CM to automatically assign a directory URI to a line based on the directory URI configured on a user associated with the line, as shown in Figure 4-33.

Key Topic

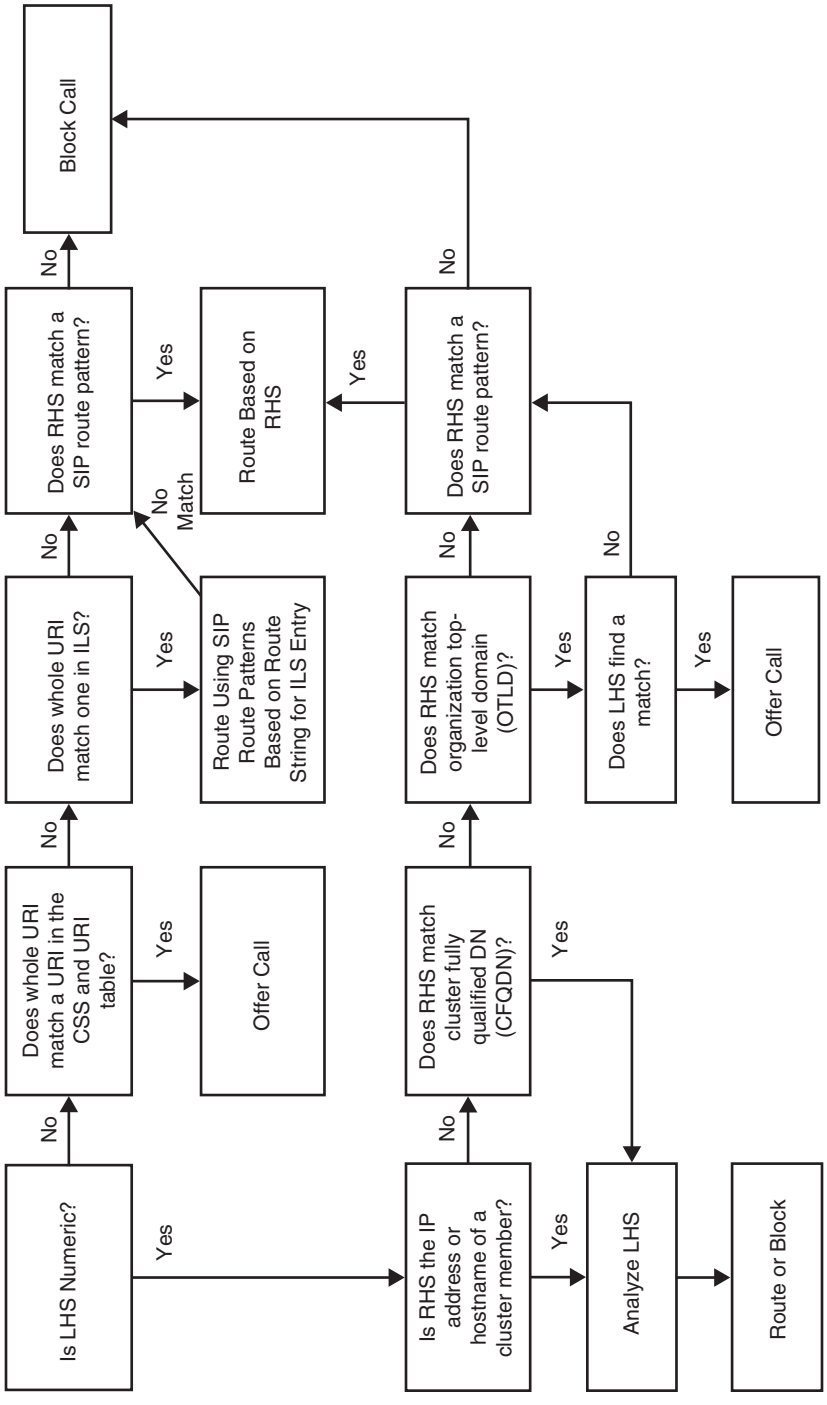

Figure 4-32 *URI Routing in Unified CM*

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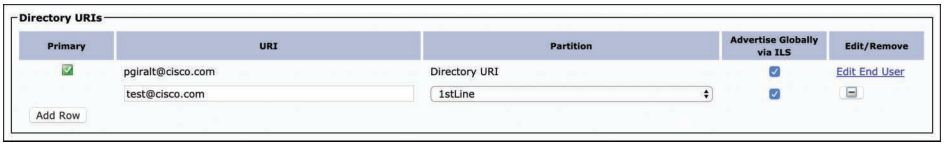

Figure 4-33 *Directory URIs Configured on a Line*

You can see in the figure that the first URI in the list cannot be modified. This is because it is imported from the associated user. Notice that the partition for the imported directory URI is just named Directory URI. This partition comes preinstalled in the system and cannot be deleted. You can include this partition in any appropriate calling search spaces to allow users to dial URIs in those partitions or you can configure the Directory URI Alias Partition parameter in the Enterprise Parameters configuration page (**Unified CM Administration > System > Enterprise Parameters**). If you select a partition for this parameter, any imported directory URI in the Directory URI partition is automatically placed into the Alias partition. If you add a new URI, the partition in which the URI is placed is configurable, as it would be for a directory number or for another pattern.

Also notice that there is an Advertise Globally via ILS checkbox next to each URI. Although we have not talked about ILS yet, it will be discussed in the next section. For now, just note the presence of this checkbox and know that it's a good practice to leave it enabled, even in a single-cluster environment, in case you ever need to expand your network to replicate URIs to other clusters.

If the calling search space on the calling device does not have the partition with the URI being dialed or the URI doesn't exist on the cluster at all, the call will move to the next step in the flowchart. If the URI is found in the calling search space, the call is extended to the device where the URI is configured.

It is important to note that URIs must be matched exactly as dialed to make a match—with one exception. By default, URI matching is performed in a case-sensitive manner. This means that if a line is configured as bob@example.com and a user dials Bob@example.com, the call will fail. Most administrators relax this policy to make URI matching case-insensitive so that bob@example.com, Bob@example.com, and BOB@EXAMPLE.COM all match the same URI (bob@example.com). In order to enable case-insensitive matching, you must set the URI Lookup Policy enterprise parameter, configured under **Unified CM Administration > System > Enterprise Parameters**, to Case Insensitive.

The next step is to look to see if the URI matches one in ILS, which is discussed in the next section. This is the full list of URIs that have been replicated from other clusters in the enterprise. The mechanisms by which ILS works are covered shortly; for now just know that Unified CM stores a list of all the URIs on all other clusters, and each of those URIs is associated with a value called a *route string* that identifies how to reach the cluster that "owns" that URI. If an entry in the ILS database is found with an associated route string, Unified CM attempts to route the call by searching for a SIP route pattern that matches the route string.

SIP route patterns are configured in Unified CM from the **Unified CM Administration > Call Routing > SIP Route Pattern** configuration page (see Figure 4-34).