



Quick answers to common problems

Cisco Unified Communications Manager 8: Expert Administration Cookbook

Over 110 advanced recipes to effectively and efficiently configure and manage Cisco Unified Communications Manager

Tanner Ezell

[PACKT] enterprise 
PUBLISHING professional expertise distilled

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BIRMINGHAM - MUMBAI

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Preface

Cisco Unified Communications Manager 8: Expert Administration Cookbook is filled with many advanced recipes to effectively and efficiently configure and manage Cisco Unified Communications Manager.

This book intends to serve as a quick reference for consultants and administrators to quickly address and resolve common problems while providing design insights. Coupled with clear instructions and plenty of screenshots, this book will help in implementing new features and improving the existing architecture. This practical cookbook will help familiarize the readers with various aspects and conventions of Cisco's Unified Communications Manager solution.

What this book covers

Chapter 1, Call Routing, Dial Plan, and E.164, will expose you to call routing with an emphasis around local route groups and E.164. You will learn how to implement least cost routing, Tail End Hop Off, and various call routing technologies available.

Chapter 2, Call Admission Control, focuses on call admission control features, the components that make them up, as well as how to reroute calls when enough bandwidth is not available.

Chapter 3, Media Resources and Music On Hold, focuses on the multimedia aspects of the platform; you'll learn how to set up Music on Hold and upload custom audio files. You'll also learn how to configure media-related devices and their functions such as Media Termination Points and Transcoders.

Chapter 4, Tracing and Troubleshooting Tools, will expose you to some of the most common tools used to debug and troubleshoot issues on the platform, including the Real-Time Monitoring Tool.

Chapter 5, Device and Unified Mobility, focuses specifically on mobility for devices and end users. You'll learn to configure single number reach, two state dialing, and how to configure mobility-related features.

Chapter 6, User Management, will teach you how to manage end-user permissions, roles and user groups, and how they might apply to end users and administrators alike. You'll learn about LDAP integration and authentication as well as how and when to apply filters.

Chapter 7, User Features, focuses on commonly requested and demanded features for users, including Meet-Me conferencing, directed call park, as well as user niceties such as custom ringtones and backgrounds.

Chapter 8, Advanced Features, explains advanced features of Unified Communications Manager, specifically focusing around extension mobility, call recording, and monitoring along with the introduction of geolocations and logical partition.

Chapter 9, Securing Unified Communications, provides common configuration information for securing a Unified Communications Manager cluster. It also includes configuration for phones and conference resources over SRTP.

Chapter 10, Serviceability, Upgrades, and Disaster Recovery, aims to cover configuration of alarms and tracing, along with configuration of the three versions of SNMP. It also covers the backup and restore process for the Unified Communications Manager publisher.

Chapter 11, Bulk Administration Tool, introduces the Bulk Administration Tool. We will learn to generate CSV files with and without the `bat.xls` spreadsheet, as well as cover the fields required for some of the most common items that are bulk provisioned including devices, user device profiles, analog gateways, and mobility users.

What you need for this book

- ▶ Cisco Unified Communications Manager 8.5
- ▶ Cisco IP Communicator 8

Who this book is for

If you are a Cisco Unified Communications Administrator or Engineer looking forward for advanced recipes to perform important administration tasks, then this is the best guide for you. This book assumes familiarity with the basics of Cisco's Unified Communications Manager architecture.

Conventions


In this book, you will find a number of styles of text that distinguish between different kinds of information. Here are some examples of these styles, and an explanation of their meaning.


Code words in text are shown as follows: " The pattern `\+ [^1]` will match any E.164 number that does not start with a one."

Any command-line input or output is written as follows:

```
dial-peer voice 200 voip
service CCM
incoming-called number 13400
destination-pattern 13400
session target ipv4:192.168.1.5
codec g711ulaw
dtmf-relay h245-alphanumeric
novad
```

New terms and **important words** are shown in bold. Words that you see on the screen, in menus or dialog boxes for example, appear in the text like this: " Add a new route list that will serve as the link to the local route groups (**Call Routing** | **Route/Hunt** | **Route List**)."

 Warnings or important notes appear in a box like this.]

 Tips and tricks appear like this.]

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1

Call Routing, Dial Plan, and E.164

In chapter 1 we dive straight into the dial plan with recipes on E.164 globalization, call routing, and call restrictions. We will cover:

- ▶ Implementing local route groups with device pools for E.164 call routing
- ▶ Implementing E.164 route patterns and partitions
- ▶ Implementing E.164 called and calling party transformations
- ▶ Implementing least cost call routing using Tail End Hop Off
- ▶ Implementing call restrictions with line blocking patterns and calling search spaces
- ▶ Implementing short dial numbers
- ▶ Implementing time-of-day call routing
- ▶ Implementing Forced Authorization Codes
- ▶ Implementing Client Matter Codes

Introduction

In this chapter, we will focus on implementing local route groups, device pools, route patterns, and various other call routing technologies with a specific focus on building an E.164 compatible dial plan. All the recipes in this chapter require administrator access to the **Unified Communications Manager (UCM)**. It is strongly recommended you get comfortable performing these recipes in a lab environment before implementing them into production.

Even if you're not interested in E.164, the recipes in this chapter can be applied to building any style of dial plan while utilizing some of the feature benefits to make dial plan management easier than before.

Implementing local route groups with device pools for E.164 call routing

To simplify call routing and dial plan management, local route groups provide a logical way to process calls according to settings specific to the device pool of the originating device.

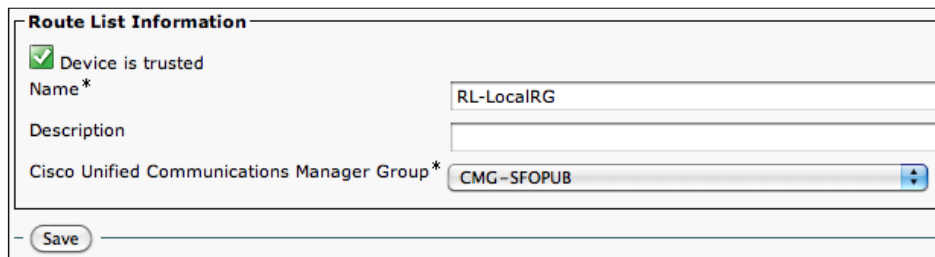
Getting ready

This recipe assumes you have a gateway or trunk device configured.

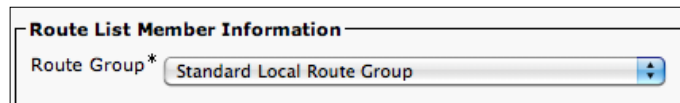
How to do it...

To implement a local route group for use with a device pool, perform the following:

1. Add a new route list that will serve as the link to the local route groups (**Call Routing | Route/Hunt | Route List**).
2. Click on **Add New** to add a new route list.
3. Type in a name and select a Call Manager Group in the drop-down with which the route list will be associated:



4. Click on **Save**.
5. Once the page reloads, click on **Add Route Group** and a new page will open.
6. Select **Standard Local Route Group** from the **Route Group** drop-down menu then click on **Save**. You will be returned to the **Route List** page:



7. Finally, click on **Save** to save the Route List.
8. Add a new route group containing the gateway or trunk (**Call Routing | Route/Hunt | Route Group**).

9. Find and select your gateway or trunk under the **Find Devices to Add to Route Group** section. Then click on **Add to Route Group**. You should now see the device in the **Selected Devices** list:

Current Route Group Members


Selected Devices (ordered by priority) * SIPT-SFO.VG01 (All Ports)

▼ ▲ Revers

Removed Devices***

▼ ▲

Route Group Members

 [SIPT-SFO.VG01](#)

10. Click on **Save**. The device will show up under **Route Group Members**.
11. Assign the route group you created in the previous step to the device pool by navigating to the device pool (From the menu, **System** | **Device Pool**) configuration page and selecting the route group from the **Local Route Group** drop-down under the **Device Pool Settings** section:

Device Pool Settings

Device Pool Name* DP-SFO

Cisco Unified Communications Manager Group* CMG-SFOPUB

Calling Search Space for Auto-registration < None >

Adjunct CSS < None >

Reverted Call Focus Priority Default

Local Route Group **RG-SFO-PSTN**

Intercompany Media Services Enrolled Group < None >

12. Click on **Save**.



These changes will not take effect until you *reset* the devices in the device pool.

How it works...

Prior to the introduction of local route groups in CUCM, dial plans relied on route patterns pointing to specific gateways and route lists in site-specific partitions. By utilizing local route groups with device pools we can simplify call routing and reduce the number of route patterns needed throughout the system, thereby making the overall system simpler and maintenance easier.

There's more...

When a call is placed on the system it matches a route pattern that informs the system where to send the call, typically a route list containing trunks and gateways. When the system is told to send the call to a route list containing the Standard Local Route Group, the egress gateway is determined by information pulled from the device pool settings of the device that initiated the call, and routes it accordingly.

Implementing E.164 route patterns and partitions

An advantage of an E.164 dial plan is that it requires only a single route pattern to make it all work, though additional route patterns are still needed to allow users to dial using traditional dialing and TEHO. Here we will create the route partition to be used by the E.164 route pattern.

How to do it...

To create the route pattern to support an E.164 dial plan, we will do the following:

1. Add a new partition, which will be globally accessible, by clicking **Add New** on the **Partitions** page located in the **Class of Control** submenu under the **Call Routing** menu.

Partition Information	
To enter multiple partitions, use one line for each partition entry. You can enter up to 75 partitions; the names and descriptions can have up to a total of 1475 characters. The partition name cannot exceed 50 characters. Use a comma (',') to separate the partition name and description on each line. If a description is not entered, Cisco Unified Communications Manager uses the partition name as the description. For example: << partitionName >> , << description >> CiscoPartition, Cisco employee partition DallasPartition	
Name*	PT-Global-E164, Clusterwide E.164 Partition

2. Enter in a partition name and a description in the text box and then click on **Save**.

3. Add the E.164 Route Pattern and assign the Route List to it (**Call Routing | Route/Hunt | Route Pattern**).
4. Click on **Add New**.
5. Enter \+.! for the **Route Pattern** and select the route partition previously created in the **Route Partition** drop-down:

Pattern Definition	
Route Pattern*	\+.!
Route Partition	PT-Global-E164
Description	Clusterwide E.164 Route Pattern
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	RL-LocalRG (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern

6. From the **Gateway/Route List*** drop-down, select the route list containing the Standard Local Route Group.
7. Ensure that the **Call Classification** is **OffNet** and the **Route Option** is set to **Route this pattern**.
8. Click on **Save**.

How it works...

When an E.164 number is dialed, the system will match it against the route pattern. The purpose of this pattern is to get the call to route to the local gateway or trunk where number normalization occurs, before sending the call out to the local gateway. **Call Classification** is set to **OffNet** for this pattern because we expect any calls that match this pattern to be routed out to the PSTN.

There's more...

Implementing a successful dial plan requires a few considerations from a technical perspective as well as a user experience standpoint.

Dial plan considerations and partitions

Partitions are a crucial part of both the dial plan and the implementation of calling restrictions. Having a well designed partition scheme can make management easier and it isn't difficult to implement. Some things to consider when planning your partition scheme are as follows:

- ▶ How many locations?
- ▶ Multinational?
- ▶ Will short dials (or hot numbers) be used?
- ▶ What about multinational dialing considerations?

Common system partitions

In most systems there are a few basic requirements from a partitioning perspective and at the very least we want to separate user directory numbers from system numbers. To accomplish this we might have the following partitions:

- ▶ PT-Line
- ▶ PT-System

If this is an E.164 dial plan, we want to separate the partitions from the rest of the system. That is why we also include:

- ▶ PT-Global-E164

Partitioning at a national level

In order to support a basic multinational dial plan we need partitions for dialing rules specific to each nation, for example:

- ▶ PT-US-DialPlan
- ▶ PT-UK-DialPlan

We would typically use these partitions for any patterns that reach the PSTN, including emergency and information services, as well as regular outbound calls.

Partitioning at a local level

If location specific dial rules are required, we might have partitions for each location. For example:

- ▶ PT-US-SFO-DialPlan

By doing this at the location level, we can allow for location specific short dials or dialing rules. For example, if we wanted to implement extension 4357 as a short dial to reach the local help desk, we would use a location specific partition such as that shown previously.

Dial plan considerations and route patterns

It's important to define how users will access the outside world based on what they are familiar with. In many corporations, dial plan rules exist to allow local calls to be dialed first with a 9 or 91, followed by seven or ten digits; other companies may require nine or ten digits for all calls. We call this seven digit and ten digit dialing, respectively.

Regardless of which dialing method is used, the setup is the same and thanks to E.164 you only need one route pattern to support all locations.

Seven digit dialing

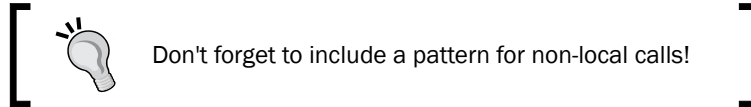
To implement seven digit dialing we will add another route pattern as explained earlier, which is the 9 . [2-9]XXXXXX pattern:

Pattern Definition	
Route Pattern*	9.[2-9]XXXXXX
Route Partition	PT-Global-E164
Description	Clusterwide E.164 7 digit dialing pattern
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	RL-LocalRG

Unlike the earlier example, we want to strip the 9 off and append a plus sign. This is necessary so the call will match the \+ . ! pattern before it can be routed to the local gateway or trunk:

Called Party Transformations	
Discard Digits	PreDot
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	+
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

In situations where you are not using an E.164 dial plan but want to implement seven digit dialing, you need to only put the pattern in a location specific partition and point the **Gateway/Route List*** to the appropriate route list or gateway. In this situation, you would not prefix the plus sign.



Ten digit dialing

To configure ten digit dialing, follow the previous steps and instead use a pattern of 9 . [2 - 9] xxxxxxxxxx.

Implementing E.164 called and calling party transformations

By using cluster-wide E.164 route patterns, number localization is no longer done on the route pattern. Instead, localization must occur prior to sending the call to the gateway or trunk device. This is accomplished with called and calling party transformations.

Getting ready

In this recipe we assume you already have the necessary partitions and calling search spaces for called and calling party transformations created. Refer to the *There's more...* section for an example partition and calling search space scheme.

How to do it...

To implement called and calling party transformation on either a gateway or trunk device, perform the following:

1. Add the calling party transformation pattern (**Call Routing | Transformation | Transformation Pattern | Calling Party Transformation Pattern**).
2. Add the transformation pattern appropriate to your environment and location:

Pattern Definition	
Pattern*	\+1.!
Partition	PT-US-Inbound-ANI
Description	Call normalization to remove +1 from inbound calls

3. Prefix any necessary digits and select the appropriate digit discard field. In the case of the previous example, **Discard Digits** is set to **PreDot** with no digits being prefixed.
4. Add the called party transformation pattern (**Call Routing | Transformation | Transformation Pattern | Called Party Transformation Pattern**).
5. Add the appropriate transformation pattern and any prefix digits necessary. In this case, we again choose **PreDot** for **Discard Digits** and set **Prefix Digits** to **9**. Refer to the *There's more...* section for further explanation if required.

Pattern Definition

Pattern*

Partition

Description

6. Navigate to the configuration page for the port or device.
7. On a MGCP controlled gateway, transformations are configured on a per port basis. The configuration page for the port is found by navigating to the configuration page for the gateway, then selecting the appropriate T1 port under the **Configured Slots, VICs and Endpoints** section as indicated in the following screenshot:

Configured Slots, VICs and Endpoints

Module in Slot 0

Subunit 0 0/0/0 T1PRI

Subunit 1

8. This is configured at the device level for trunks and gateways.
9. Next we apply the transformations to our trunks or gateways. Calling party transformations are configured under the section titled **Incoming Calling Party Settings**.



The type of device we are configuring will determine the fields available to us. On gateway devices we see **National, International, Unknown,** and **Subscriber**. On trunk devices we see **Incoming Number**.

10. Select the **Calling Search Space** that contains the partitions you assigned to the called and calling party transformation patterns and apply it to all applicable fields:

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="CSS-SFO-Inbound-ANI"/>	<input type="checkbox"/>

The previous screenshot is for an SIP trunk. Here we uncheck **Use Device Pool CSS** as we are not using the device pool for number transformation.

11. Finally, called party transformations are configured under different sections depending on the type of device.

On the gateways configuration page, this section is called **Call Routing Information - Outbound Calls**, and **Outbound Calls** for trunks.

Outbound Calls


Called Party Transformation CSS

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS

Again, we are not using the device pool for number transformation, so we uncheck boxes for both calling and called device pool transformations.

[

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The calling search space selected for the **Called** and **Calling Party Transformation CSS** must contain the partitions used when creating the transformation patterns.

How it works...

When a call enters through a gateway or trunk device, the calling and called party transformations are applied depending on transformation patterns available to the calling search space used:

- ▶ **Calling party transformation patterns:** In the example from the recipe you see a calling party transformation pattern using `\+1 . !`. As we explain in the example, we discard digits PreDot. We do this to normalize the number users see when their phone is ringing and connected, as users in the United States may not be accustomed to seeing +1 before the number.

Alternatively, say we have an office in San Francisco where users are accustomed to seeing only seven digits for local calls and ten for everything else. We still use the `\+1 . !` PreDot pattern to remove the +1 for calls but we add another pattern to strip the area code off. In this case that pattern would be `\+1415 . !`, stripping PreDot with a partition of PT-SFO-Inbound-ANI, or something similar. By doing this, calls from 415 numbers will show as seven digits on the display when ringing and connected.

- ▶ **Called party transformation patterns:** Prior to local route groups and called party transformations, you would prepare the dialed number to be sent to the gateway or route list on the route pattern itself. Called party transformation patterns would then be used to prepare the dialed digits to be accepted by the gateway, trunk, or PSTN. In many cases this involves stripping the plus sign and prefixing an access code before sending it out to the gateway or trunk to route the call to the PSTN.

How we modify the number depends on the type of gateways or trunks we are using. With MGCP gateways, we format the number so that it can be sent across to the PSTN. In some cases this means removing the plus, and appending or removing digits depending on what the carrier expects. For instance, if the carrier expects seven digits for local calls and 1 + 10 digits for long distance calls, we might strip the +1 and area code for local calls and strip only the plus for all other calls.

For gateways and trunks, access codes are typically configured to inform the gateway or trunk to send the call to the PSTN. Typically these are 9, or 91. In this situation we would strip the necessary digits and prefix the access code appropriate to the call. For example, say our carrier requires seven digits for local and eleven digits for long distance calls; assuming we require an access code of 9 for local and 91 for long distance calls, we might implement the following called party transformations:

- `\+1.!`
Partition: PT-SFO-Outbound-DNIS
Prefix digits: 91
- `\+1415.!`

Partition: PT-SFO-Outbound-DNIS

Prefix digits: 9

Now, when a call is made to +1 415 555 1234, for example, the transformation pattern will remove +1415 and append a 9, sending the call to the gateway or trunk as 95551234 where it would match a dial peer before being sent out to the PSTN. While it is possible to do these transformations on the gateway themselves, managing them in UCM provides a central point for configuration and can help reduce dial plan maintenance.

There's more...

Calling and called party transformations are primarily used to localize the ANI displayed for calls entering the system and localizing calls for the gateway before sending it out to the PSTN.

Partitions and calling search spaces for called and calling transformation patterns

In this recipe you will see a few partitions and calling search spaces that may not immediately make sense. In order to accomplish these transformations on a per location basis we have six partitions and three calling search spaces.

The partitions and calling search spaces we use are:

- ▶ CSS-SFO-Outbound-ANI
 - PT-SFO-Outbound-ANI
 - PT-US-Outbound-ANI
- ▶ CSS-SFO-Outbound-DNIS
 - PT-SFO-Outbound-DNIS
 - PT-US-Outbound-DNIS
- ▶ CSS-SFO-Inbound-ANI
 - PT-SFO-Inbound-ANI
 - PT-US-Inbound-ANI

If you have no need to localize the ANI on a per location basis you might have a single calling search space and partition instead:

- ▶ CSS-US-Inbound-ANI
 - PT-US-Inbound-ANI

These are only some suggestions; make sure you apply the appropriate calling search spaces and partitions for your cluster.

Implementing least cost call routing using Tail End Hop Off

Least Cost Routing (LCR) is not strictly limited to calls destined for the PSTN, instead LCR can be used to prevent OnNet calls from being routed OffNet. In this recipe we will cover both uses.

Getting ready

Before we begin this recipe it is helpful to have some information:

- ▶ DID ranges of locations for which we are implementing LCR
- ▶ Site codes of locations for which we are implementing LCR
- ▶ Local numbers per location for Tail End Hop Off

In this recipe we will implement LCR and Tail End Hop Off for calls destined to an office in San Francisco. We will assume the following:

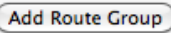
- ▶ DID Range for this location: +1 415 555 1000 to 1099
- ▶ Site code for this location: 11
- ▶ Local numbers for this location: 415 XXX XXXX

How to do it...

To implement Least Cost Routing for a location, we need to perform the following:

1. Add a new route list that will contain the route group with the gateway or trunk device local to the location for which we are implementing LCR, as well as the Standard Local Route Group:

Route List Information	
Registration	Registered with Cisco Unified Communications Manager SFO
IP Address	172.16.233.5
<input checked="" type="checkbox"/> Device is trusted	
Name*	RL-TEHO-SFO
Description	Route list for Tail End Hop Off to San Francisco
Cisco Unified Communications Manager Group*	CMG-SFOPUB
<input checked="" type="checkbox"/> Enable this Route List (change effective on Save; no reset required)	
<input type="checkbox"/> Run On All Active Unified CM Nodes	


Route List Member Information	
Selected Groups**	RG-SFO-PSTN Standard Local Route Group
	



The order here is important. Ensure the local route group is above the Standard Local Route Group in the list.

2. Add a new route pattern to send local calls to our new route list. Key fields to note here are **Route Pattern**, **Route Partition**, and **Gateway/Route List***:

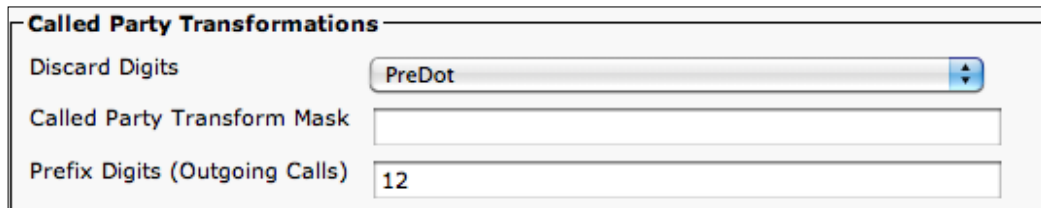
Pattern Definition	
Route Pattern*	\+1415XXXXXX
Route Partition	PT-Global-E164
Description	TEHO for SFO numbers.
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	RL-TEHO-SFO
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error
Call Classification*	OffNet
<input type="checkbox"/> Allow Device Override <input type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	

 Here we have unchecked **Provide Outside Dial Tone** as it is unused, but feel free to leave it checked.

- Next add a translation pattern (**Call Routing | Translation Pattern**, then click on **Add New**) that is responsible for converting E.164 numbers to their internal extensions.
 - Here the **Translation Pattern** must match only the DID range for the location. For our recipe the pattern is \+1415555.10XX. For the partition use something that is globally accessible, for example **PT-Global-E164**:

Pattern Definition	
Translation Pattern	\+1415555.10XX
Partition	PT-Global-E164
Description	Translate SFO DIDs to internal extension

For our pattern, it is necessary to set **Discard Digits** to **PreDot** and **Prefix Digits** to the site code—12 in this recipe.



The screenshot shows a configuration window titled "Called Party Transformations". It contains three fields: "Discard Digits" is a dropdown menu set to "PreDot"; "Called Party Transform Mask" is an empty text input field; and "Prefix Digits (Outgoing Calls)" is a text input field containing the value "12".

How it works...

Least cost routing with Tail End Hop off is accomplished by sending calls to locations where the call would cost the least. In addition to Tail End Hop Off, we can accomplish least cost routing by recognizing when a user dials the DID to another user on the same cluster by converting the E.164 number to the local extension and routing over the IP network.

There's more...

Once more we see the benefits of the logical nature of local route groups. By having localization settings at the gateway level, we don't have to worry about formatting and allow the local gateway to normalize the call as required by the PSTN. In the event that the call cannot be made through the gateway or trunk device at the local site, the call will fall back to the gateway or trunk device local to the originating caller.

Do remember that Tail End Hop Off is not legal in all countries. Check with local regulations before implementing it.

Implementing calling restrictions with line blocking partitions and calling search spaces

In this recipe we will be implementing class of service calling restrictions using partitions and calling search spaces, as well as exploring their design considerations.

Getting ready

For this recipe, preparation is key. We will need to determine the partitions, calling search spaces, and patterns to be blocked that will be appropriate to the environment. There is more information on this in the *There's more...* section of this recipe.