Appendix C

Direct SIP with IP-PBX in Office Communications Server 2007 R2

What is Direct SIP with IP-PBX?

Direct SIP is the documented and supported way in which Microsoft® Office Communications Server 2007 R2 exchanges voice calls with qualified third-party on-premise devices such as Session Initiation Protocol to Public Switched Telephone Network (SIP/PSTN) gateways and Internet Protocol Private Branch eXchange (IP-PBX). In Direct SIP, an Office Communications Server's Mediation Server is directly connected to a SIP/PSTN gateway or an IP-PBX. Microsoft provides the Unified Communications Open Interoperability Program (OIP) for the qualification of third-party solutions for interoperability with Microsoft Office Communications Server. Under the OIP, Direct SIP is based on common industry standards, such as SIP over Transmission Control Protocol (TCP), Realtime Transport Protocol (RTP), and G.711. Direct SIP with a qualified SIP/PSTN gateway enables Office Communications Server to exchange calls directly with the PSTN, as well as with virtually any Time Division Multiplex (TDM) PBX, and (by using back-to-back SIP/PSTN gateways) with virtually any IP-PBX.

Direct SIP with IP-PBXIP-PBX is a variation in which the calls are exchanged over IP between the Mediation Server and an IP-PBXIP-PBX, without the use of back-to-back IP/PSTN gateways (meaning that exchanges take place in IP all the way, without transcoding between IP and TDM). This is done across an IP-to-IP connection over which the two systems converse in a standard manner (such as SIP over TCP, RTP, or G.711) as specified in the OIP's Direct SIP specifications. This kind of connection is referred to as a "SIP trunk" by most IP-PBXIP-PBX vendors.

Note Microsoft uses the term "SIP Trunking" differently from the term "Direct SIP with IP-PBX" and from the more generic term of "SIP trunk" described above. "*SIP Trunking*" (also occasionally called "SIP Trunking to Carrier") is the use of SIP and RTP to pass telephony traffic from the enterprise network edge to a network service provider over an IP connection, without traversing TDM or circuit networks. The term "Direct SIP" is used for cases where Office Communications Server is connecting to a SIP/PSTN gateway or IP-PBX, as qualified through the OIP.

Why Direct SIP with IP-PBX?

In the early days of the OIP program, most OIP qualified devices were SIP/PSTN gateways, and early deployments have virtually all been based on OIP qualified SIP/PSTN gateways. Without Direct SIP with IP-PBX, the supported way for Office Communications Server to exchange voice calls with an IP-PBX is to connect a Mediation Server with an OIP qualified SIP/PSTN gateway. In turn, this SIP/PSTN gateway's PSTN edge is connected to the IP-PBX through the PSTN edge of one of its IP/PSTN gateways or cards. This configuration is called "back-to-back IP/PSTN gateways".

The back-to-back IP/PSTN gateway approach relies on a mature and broadly supported telephony protocol between the two gateways (typically QSIG). It has the benefit of working every time with virtually every IP-PBX, including IP-PBXs that are based on protocols other than SIP (such as H.323). It is simple to implement and has proven ideal for pilot and small scale deployments, or for non-SIP compliant IP-PBXs.

However this approach requires more gateway ports (hence higher capital costs). It also increases media latency due to the double transcoding from IP to TDM and back to IP. As Office Communications Server Enterprise Voice deployments become more prevalent and grow beyond small scale deployments, many Office Communications Server customers who also have an IP-PBX deployed have been asking for Direct SIP with IP-PBX as a substitute to the back-to-back SIP/PSTN gateways approach.

Why is IP-PBX Only Now Becoming Qualified For Direct SIP?

Microsoft actually built the Direct SIP capability into the product and into the OIP from the start and was ready to support Direct SIP with IP-PBX that qualified in the OIP. However, testing showed that few if any IP-PBX were capable of meeting the standards-based specifications for Direct SIP interoperability and OIP qualification.

Specifically, Office Communications Server conforms to RFC3966. RFC3966 defines the taxonomy and syntax for phone numbers that is mandated by the SIP standard. While Office Communications Server is RFC3966 compliant, many of the most commonly deployed IP-PBXs in the market today are not. If two voice systems from different vendors do not format phone numbers the same way, Direct SIP interoperability will not work. Lack of RFC3966 compliance is the primary technical reason why those IP-PBXs could not qualify for Direct SIP in the OIP.

Note Throughout Microsoft documentation, you may find references to "E.164 numbers" to describe numbers such as +14255551212, which are globally routable phone numbers that begin with a plus sign (+). ITU's E.164 is a global scheme for normalizing globally routable numbers (and is used in particular for international numbering on the PSTN and increasingly on mobile phone networks). While not incorrect, the use of the term E.164 to designate those numbers can lead to confusion because E.164 does not mandate starting the number with a plus sign because traditional phones and PBXs do not have a way to input, transmit or process a plus sign.

Most IP-PBXs can understand E.164 numbers but only when those numbers are expressed without the plus sign, as dial strings. Typically, such IP-PBXs do not support the plus sign in the REQUEST and TO header fields of a SIP message. They may tolerate the plus sign in the FROM header field of a SIP message, but generally are not able to process it: numbers that have a plus sign in the FROM field cannot be dialed back to Office Communications Server from the IP-PBX.

Numbers such as +14255551212 are actually RFC3966 compliant global numbers. RFC3966 defines the telephone number Uniform Resource Identifier (URI) (tel: URI) scheme that is used by SIP. For globally routable phone numbers, RFC3966 is based on E.164, which it clarifies by requiring the leading plus sign. It also adds support for non-globally routable numbers (also called local or private numbers) that are not defined in E.164. RFC3966 requires a "Phone-Context" in the tel: URI; the "Phone-Context" removes any ambiguity as to which system or network entity the local number refers to, and may enable the system to translate or normalize between local and global numbers.

What is Changing?

With Office Communications Server 2007 R2, Microsoft is introducing the capability for Office Communications Server to exchange calls with non-RFC3966 compliant IP-PBX systems, and in particular with SIP-based IP-PBXs from Cisco. As a consequence, Microsoft now supports Office Communications Server deployments in Direct SIP with IP-PBX between Office Communications Servers and specific versions of Cisco Unified Communications Manager.

This change gives administrators the ability to set up Office Communications Server so that it can directly interoperate with qualified IP-PBXs using E.164 globally routable telephone numbers without the plus sign at the beginning that is required by RFC3966. Additionally, Office Communications Server will be able to interoperate with IP-PBX within a private dialing plan, exchanging locally routable private numbers. This last point is the not the primary focus of the present document.

In Office Communications Server 2007 R2, the changes are built in, but are turned off by default and require a WMI setting to be turned on, as well as the configurations that are explained in the following sections.

Direct SIP Support for Office Communications Server 2007

The changes are accessible to users of Office Communications Server 2007 through update packages that are described in the following Microsoft Knowledge Base articles: http://support.microsoft.com/kb/952783/; http://support.microsoft.com/kb/952780/; http://support.microsoft.com/kb/953659/; http://support.microsoft.com/kb/957707/.

The updates need to be installed on the RTM (6362.0) version or on previously updated installations and are available for Office Communications Server Enterprise and Standard Editions.

You must install Server. msp for the following servers, however, you do not have to restart the server:

- Standard Edition Server
- Enterprise Edition Server (front end)
- Proxy Server
- Director Server
- Edge Server
- Forwarding Proxy

For the Mediation Server role, apply **MediationServer.msp** and **UCMARedist.msp** to the Mediation Servers that will be connected to the IP-PBX. You will have to restart services after you apply **UCMARedist.msp.**

For Office Communicator 2007, apply **Communicator.msp**. (version 6362.97) and restart Communicator.After the updates, you will have to create or edit a configuration file on Office Communications Server 2007 to enable the changes. The use of a configuration file is instead of applying the WMI setting as in the case of Office Communications Server 2007 R2. You must create or edit the configuration file, MediationServerSvc.exe.config, in the Mediation Server directory where the MediationServerSvc.exe file is installed. By default, that directory is C:\Program Files\Microsoft Office Communications Server 2007\Mediation Server.

Here is a sample configuration file where the value of `RemovePlusFromRequestURI` has been changed from the default of `NO` to `YES`:

```
<?xml version="1.0" encoding="utf-8" ?>
<configuration>
        <appSettings>
                <add key="RemovePlusFromRequestURI" value="YES" />
        </appSettings>
</configuration>
```

After this is completed, restart the Mediation Server.

Note this configuration file is also used to turn Transport Layer Security (TLS) on or off in the Mediation Server existing functionality. Therefore, the file may already exist on a particular Mediation Server computer. By default, if the file does not exist or does not have the GatewayTLS setting, TLS is turned off.

Which IP-PBX Does Microsoft Currently Support Direct SIP?

Simultaneously with the changes, Microsoft announced that it had successfully tested the IP-PBX shown in Table C-1 against the test matrix that is part of the OIP requirements for Direct SIP.

Table C-1 IP-PBX successfully tested by Microsoft for Direct SIP

IP-PBX vendor	Product	Versions tested
Cisco	Unified Communications Manager 6.1	6.1.1.3000-2
Cisco	Unified Communications Manager 5.1	5.1.3.1000-12 5.1.3.3000-5
Cisco	Unified Communications Manager 4.2	4.2(3)sr3a

Microsoft now supports Office Communications Server interoperating in Direct SIP with the above IP-PBX from Cisco. While not tested, other versions of Unified Communications Manager starting with 4.1(3)SR7 are expected to comply.

This is in addition to IP-PBXs that have qualified in the OIP. For the most current list of OIP qualified IP-PBXs, see http://technet.microsoft.com/UCOIP.

Besides Cisco Unified Communications Manager, Alcatel, Avaya, NEC, Nortel, Siemens and several other vendors do not support RFC3966 in their most common releases. The changes in Office Communications Server should facilitate the qualification of these vendors' IP-PBXs in the OIP. We highly recommend that you validate in your own environment prior to making a final topology decision.

How Do the Changes Work?

These changes enable interoperating with a qualified IP-PBX using Direct SIP with IP-PBX, across an IP connection between the Mediation Server and the IP-PBX. This IP connection is typically called a "SIP trunk" on the IP-PBX side.

Figure C-1 illustrates the environment using Cisco Unified Communications Manager 5.1. Note the use of a Media Termination Point (MTP) on the Unified Communications Manager, where the signaling and media from the SIP trunk is terminated. MTP may or may not be required depending on the specific versions of Unified Communications Manager and the Cisco devices used. For more information about the use of the MTP, see the Cisco documentation.

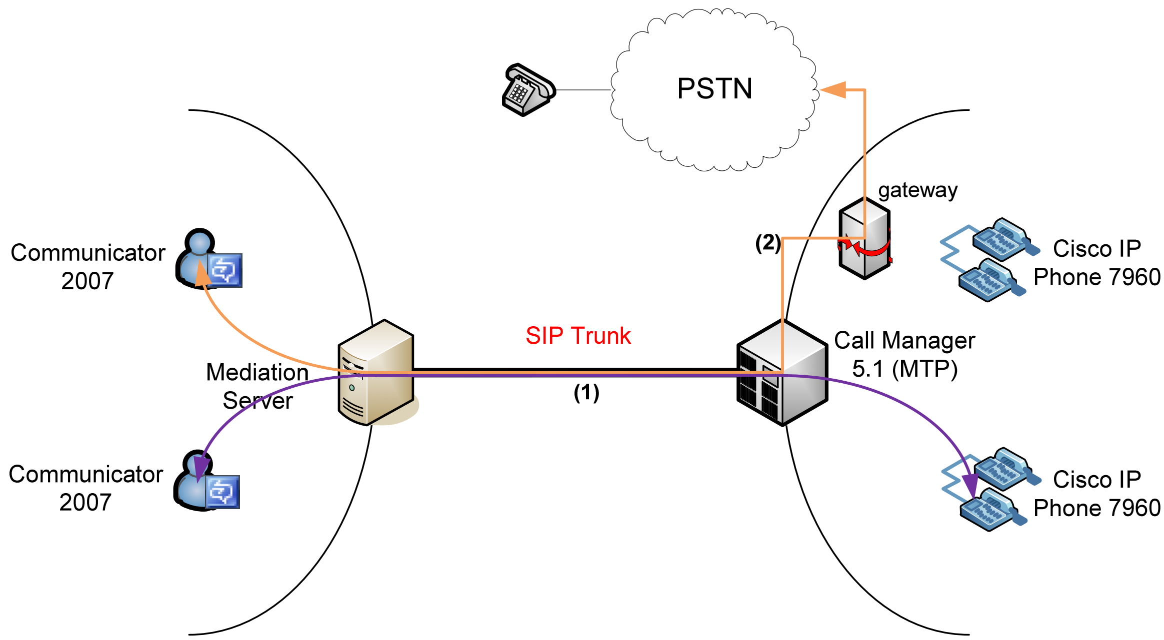


Figure C-1 Office Communicator environment using Direct SIP

While this is not a requirement, Figure C-1 assumes, as do most of the explanations later in this document, that all PSTN interconnectivity is handled by the IP-PBX. Calls from the PSTN to Office Communications Server Enterprise Voice users will first be presented to the IP-PBX, which passes the calls to Office Communications Server across the SIP trunk. That is the most common deployment case for small deployments. For larger deployments, it is often preferable for each system to handle its own PSTN interconnection directly, and reserve the Direct SIP interconnection for the exchange of calls between users of the two systems.

The changes in OCS for interoperability with Direct SIP are targeted at facilitating interoperability with an IP-PBX that, in its SIP messages across the SIP trunk, presents and receives dial strings. These dial strings might be E.164 numbers without a plus sign, or any other set of dial strings such as extensions.

Because of the IP-PBX requirement that SIP messages across the SIP trunk do not include the plus sign, appropriate dial strings must be used by the Office Communications Server Mediation Server when interacting with the IP-PBX. However the core internal logic of Office Communications Server with respect to phone numbers has to be preserved to prevent disrupting a range of typical Office Communications Server functions, such as publishing contact information within the presence document. Therefore the Mediation Server plays an important role role in a deployment that involves Direct SIP with a non-RFC3966 conforming IP-PBX.

The changes introduce the capability for Office Communications Server to normalize the FROM header so that a non-RFC3966 representation of an E.164 number (meaning an E.164 number that does not have a plus sign) is converted to a global number that conforms with RFC3966 and is placed in the P-Asserted-Identity (PAI) header field. The PAI header enables the user lookup functionality in Office Communicator. If the normalization process does not result in a global number, Office Communications Server will add a Phone-Context value of "enterprise" in the PAI header. Additionally, Office Communications Server bypasses the server normalization logic if the REQUEST URI header already contains a Phone-Context value of "enterprise."

In Office Communications Server 2007 R2, a new Mediation Server WMI setting called *RemovePlusFromRequestURI* is created to activate the new capabilities. This WMI setting can take a value of *YES* or *NO*; the default is *NO*. The following summarizes what is activated when the value is set to *YES*.

For outgoing calls (calls from the Mediation Server to the IP-PBX):

- The conversion from RFC3966 conforming numbers to dial strings (that can be interpreted by the IP-PBX) happens after the outbound routing logic of Office Communications Server is complete. This enables Office Communications Server to maintain the logic unchanged.
- The Mediation Server strips the plus sign from the REQUEST field and the FROM field in any outgoing invitation. (This results in dial strings that can be interpreted by the IP-PBX.)
- Because the Mediation Server copies the value in the REQUEST URI to the TO URI in the INVITE that is sent to the IP-PBX, the plus sign is also removed from the TO URI, and the TO URI is presented to the IP-PBX in a dial string format it can interpret.
- After these steps, all dial strings are presented to the IP-PBX across the SIP trunk in a format the IP-PBX can interpret.

For incoming calls (calls from the IP-PBX to the Mediation Server):

- The preexisting normalization rule capabilities in Office Communications Server are sufficient to normalize the TO field and route the call correctly.
- The FROM field and the REQUEST field also require a normalization rule that will insert a plus sign. The rule may be more complex and manipulate digits as well.
- If this normalization does not result in a global number, the Mediation Server will set the Phone-Context in the FROM field to *enterprise*
- Office Communications Server will then set the correct PAI value, allowing the contact model to work accurately with Office Communicator 2007.

Additionally, in environments where phone numbers in the Active Directory are entered as dial strings representing E.164 numbers without a plus sign, these numbers will be converted to RFC3966 compliant global numbers that are represented as TEL URI by the Address Book Service. The Address Book normalization rule converts these numbers into RFC3966 compliant numbers. Office Communications Server users in such scenarios will always have the *RTCSIP-Line* parameter configured with an RFC3966 compliant TEL URI.

As mentioned earlier in this document, those changes are not applied by default (or by running the updates for Office Communications Server 2007). Administrators must also perform configuration steps and add the appropriate normalization rules. Configuration is also required on the IP-PBX side to set up the SIP trunk, as well as to normalize numbers as necessary.

How Do I Set Up Direct SIP with an IP-PBX?

To set up the Direct SIP with IP-PBX interoperability, it is important that you first understand the dial plan and normalization rules on the IP-PBX side and make decisions about the number range allocation between Office Communications Server and the IP-PBX. Next you must provision and configure the SIP trunk from the IP-PBX to the Mediation Server. Last, set up Office Communications

Server, starting with Mediation Server and adding the appropriate location profiles and normalization rules.

Dial Plan and Normalization Rules on the IP-PBX Side

Enterprises' dial plans vary broadly. Therefore each implementation is unique, and the examples given in this document are not meant to cover every case.

Small or medium size enterprises (especially single-site ones) typically have a pre-existing internal dial plan based on short dialing extensions (generally 3 to 5 digits). Users are accustomed to dialing the short extensions to reach internal users and to dialing a prefix (such as 9 or 0) prior to dialing numbers external to the enterprise.

To directly reach an internal user from the PSTN, outside callers have to know the user's Direct Inward Dialing (DID) number, which is a publicly routable number that corresponds to the extension of the user to reach. They would access that DID by dialing a dial string (i.e. a series of digits). The PSTN and the IP-PBX are already configured with the appropriate transformation rules to convert the requested dial string to the corresponding extension. In general, this is done by removing the appropriate number of leading digits and placing the result in the TO field. For example, the DID (1) 425 555-1234 might be dialed as the dial string 14255551234 which would be converted to the extension 1234.

This conversion however is applied only to the TO field (also known as the Called ID or called number) of the call. The FROM field (also known as the Caller ID or caller number) is commonly either left unchanged from what the operator provided, or transformed into a dial string that enables simple callback to the original caller. For example, for an IP-PBX located in the United States, a U.S. Caller ID dial string of 14255551234 may be converted to 914255551234, and a French Caller ID dial string of 33169861234 may be converted to 901133169861234, where 9 is the prefix for external dialing on the IP-PBX and 011 the carrier mandated international dialing prefix in the United States. It can take a large number of rules on the IP-PBX to cover all possible cases, and those rules vary based on what numbering format the operator will present and require.

Larger enterprises that use extensions generally need either much longer extensions (sometimes as long as 6 or 7 digits) or internal prefixes for site to site dialing. This situation creates a risk of overlap, where a dial string received as Caller ID could be identical to a dial string representing an internal route or user. For that reason, large enterprises almost always implement Caller ID transformation rules such as the ones described in the previous paragraph, The exception to this is large enterprises that use E.164 numbering internally instead of extensions).

Therefore the enterprise's existing dial plan influences the format in which the IP-PBX will pass TO and FROM dial strings to Office Communications Server and what normalization rules will be required on Office Communications Server. In typical (but not all) cases, the IP-PBX will present to Mediation Server the following:

For calls from an IP-PBX user to an Office Communications Server user:

- The FROM field is a dial string representing an extension xxxx on the IP-PBX.
- The TO field is a dial string representing an extension yyyy on Office Communications Server.

For calls from the PSTN to an Office Communications Server user:

- The FROM field is a dial string representing the caller number as presented by the carrier, and transformed by the IP-PBX, as described earlier, starting with well defined patterns such as 91 or 9011 in the United States.
- The TO field is a dial string representing an extension yyyy on Office Communications Server, because the DID that was initially presented by the carrier was converted by the IP-PBX.

Therefore Office Communications Server should be provisioned with the appropriate normalization rules to handle the various dial string formats in the FROM field.

Conversely, in typical cases the Mediation Server will present the following to the IP-PBX:

- Both the TO and the FROM fields are dial strings representing full E.164 global numbers without a plus sign.

Therefore the IP-PBX should be provisioned with the appropriate translation rules to transform these strings into the appropriate formats for the IP-PBX and the PSTN.

DID or Extension Range Allocation

The simplest case for number range allocation is where specific ranges are dedicated to one or the other system. For example, in a four-digit extension plan, you could assign all extensions of the pattern 4xxx or 5xxx to Office Communications Server, leaving all other extensions assigned to the IP-PBX. Where possible, allocating ranges is preferable in order to simplify provisioning of users and routing patterns.

There are cases where simply allocating ranges is not possible. This can be because no unused range is available, or because users must retain their original DID as they migrate to Office Communications Server Enterprise Voice. In the case where all PSTN interconnectivity is handled by the IP-PBX as explained earlier, not allocating ranges does not change the way in which Office Communications Server is set up; all adjustments will have to be made on the IP-PBX. Typically that involves redirecting each DID from the IP-PBX to Office Communications Server. In our example hereafter we will show how that can be achieved.

Example: Setting Up Cisco Unified Communications Manager 4.2.1 for Direct SIP

This example demonstrates how to configure Direct SIP with Cisco Unified Communications Manager 4.2.1.

Main Assumptions for the Example

The following are the main assumptions for this example:

- This is a single site, located in Paris, France (country code 33, local code 1).
- All PSTN calls, incoming and outgoing, from and to Cisco Unified Communications Manager and Office Communications Server, use the preexisting PSTN interconnection trunks that are connected to and managed by the Cisco Unified Communications Manager. All calls to and from Office Communications Server go across the SIP trunk between Cisco Unified Communications Manager and Mediation Server.

> Note As discussed earlier, it is also possible (and eventually more effective, especially in large-scale deployments and, for example, with the use of SIP Trunking to Carrier) to have direct PSTN-out routes from Office Communications Server to the PSTN without going through the Cisco Unified Communications Manager. Also, in this example, redundant routes between Cisco Unified Communications Manager and Office Communications Server are not implemented; it is possible to do this but exceeds the scope of this example.

- All users are assigned a unique, externally routable DID from a local range and a unique 4-digit extension that matches the last 4 digits of their DID.
- Specifically, all DIDs for users on the Cisco Unified Communications Manager and Office Communications Server are in the format +3316986xxxx where xxxx is their extension.
- Users on the Cisco Unified Communications Manager are accustomed to dialing 4 digits for internal users, and this pattern will also be used when they dial out to users on Office Communications Server.
- Users on the Cisco Unified Communications Manager must dial 0 as a prefix to exit to the PSTN.
- In the first part of this example, the extension range 4xxx to 5xxx is allocated to Office Communications Server. In the advanced example, a configuration for situations where there are exceptions to the range allocation will be provided.

> Note It is notassumed that all numbers in the range are actually allocated on Office Communications Server; calls to unallocated numbers will be routed by Cisco Unified Communications Manager to Office Communications Server which will provide an appropriate answer in rejecting the call. The call will then be handled as setup on the Unified Communications Manager. This could include responding to the call with a “fast busy” dial tone, or sending the call to an auto-attendant, for example on Exchange 2007 Unified Messaging.

- All other numbers are allocated to the Cisco Unified Communications Manager. It is possible, but beyond the scope of this example, to create more complex, granular route patterns.

The following assumptions are also made about the formatting of the FROM and TO fields of calls:

The local PSTN carrier requires the following:

- The FROM field must be formatted as a 9-digit dial string xxxxxxxxx
- The TO field:
  - For domestic (i.e. destined to a DID in France) calls: the TO field must be formatted as a full 10-digit local French number (which is not an E.164 number): 0xxxxxxxxx
  - For international calls (i.e. destined to a DID outside of France): the TO field must be formatted as a 00 (the prefix for international dialing in France) followed by the E.164 dial string for the caller, for example 0014255551212

> Note This format may vary by locale, carrier, or even on the basis of the type of trunk used. Please verify the correct format for your environment.

For calls from a Cisco Unified Communications Manager extension xxxx to an Office Communications Server extension yyyy that are sent across the SIP trunk:

- The FROM field is the dial string xxxx
- The TO field is the dial string yyyy.

For calls from the PSTN to an Office Communications Server extension yyyy:

- The FROM field has already been formatted by Cisco Unified Communications Manager into a dial string that enables immediate dial back to the PSTN in France:
  - For domestic calls (meaning calls that. originated from a DID in France), the outside line prefix “0” is followed by the full ten-digit local French number (which is not an E.164 number) for example, 00xxxxxxxxx
  - For international calls (meaning calls that originated from a DID outside of France), the outside line prefix “0” is followed by “00” (the prefix for international dialing in France) and by the E.164 dial string for the caller, for example 00014255551212.
- The TO field is the dial string yyyy.

Step 1: Creating a Partition on Cisco Unified Communications Manager

First, create a partition named “OCSIncoming” as shown in Figure C-2.

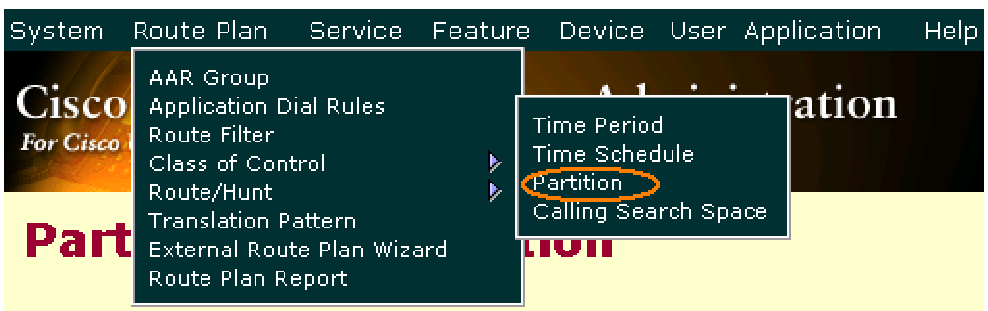


Figure C-2 Creating a partition

A Cisco Unified Communications Manager partition contains a list of route patterns. Partitions facilitate call routing by dividing the route plan into logical subsets that are based on organization, location, and call type. For more information about partitions, see "Partitions and Calling Search Spaces" in the Cisco Unified Communications Manager System Guide.

This partition enables the application of rules (such as route patterns and translation patterns) that are specific to the traffic coming into the Cisco Unified Communications Manager from the Mediation Server across the SIP trunk, as shown in Figure C-3.

Figure C-3 Partition Configuration dialog box

Note Whether a partition is required or not in a particular environment may depend on the rules required. Please verify with your Cisco Unified Communications Manager administrator.

Step 2: Creating a Calling Search Space on Cisco Unified Communications Manager

A Cisco Unified Communications Manager calling search space is an ordered list of route partitions. Calling search spaces determine which partitions (and in which order) are searched when Cisco Unified Communications Manager is attempting to complete a call.

In this example, a calling search space named "OCSIncoming" is created, as shown in Figure C-4, and assigned to the OCSIncoming partition that was created in Step 1, as shown in Figure C-5.

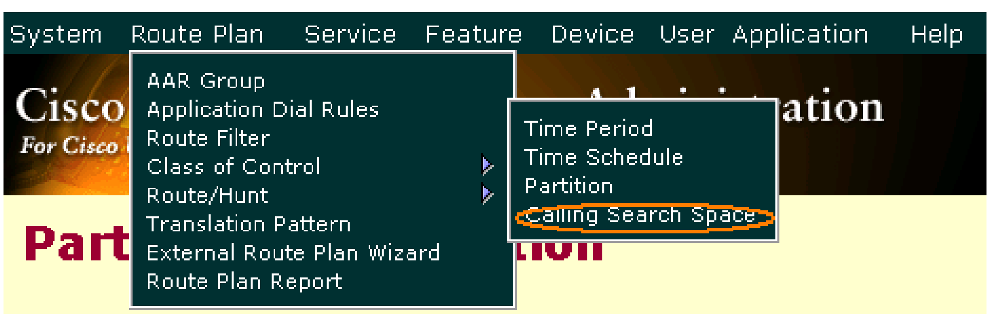


Figure C-4 Creating a calling search space

Figure C-5 Configuring a calling search space

Step 3: Setting up Translation Patterns for the Partition on Cisco Unified Communications Manager

The next step is to create translation patterns for the partition, as shown in Figure C-6. The translation patterns will be used for *inbound* calls to the Cisco Unified Communications Manager from Office Communications Server. Translation patterns manipulate dial strings before routing a call.

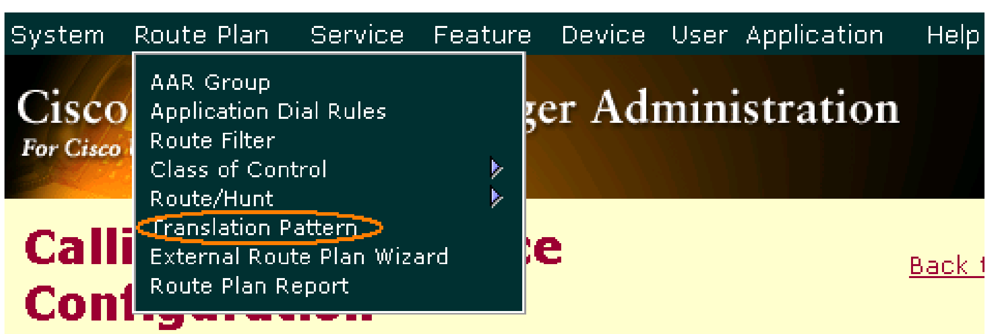


Figure C-6 Creating a translation pattern

Figure C-7 illustrates how to configure the translation pattern. The first translation pattern is called [^33]!. It will match any dial strings in the TO field that do not start with 33; as in are not destined to a French domestic PSTN number. This translation pattern will handle all calls from Office Communications Server that are destined to an international PSTN number. Note that the translation pattern is assigned to the OCSIncoming partition.

For the FROM field, the pattern retains the last 9 digits; therefore, it removes the country prefix (which is always 33 in this example) from the E.164 calling party dial string, and then presents it to the PSTN in the required format. In this example it transforms 3316986xxxx into 16986xxxx.

For the TO field, the pattern adds 000 as a prefix to the called party dial string. For example, it transforms 14255551212 into 00014255551212.

The reason it translates dial strings sent by the Mediation Server in a different manner for the TO and the FROM is the need for the TO field to start with the outside line prefix (in this example “0”) to obtain an outside line on the Cisco Unified Communications Manager, then the international prefix 00, and then the full E.164 dial string.

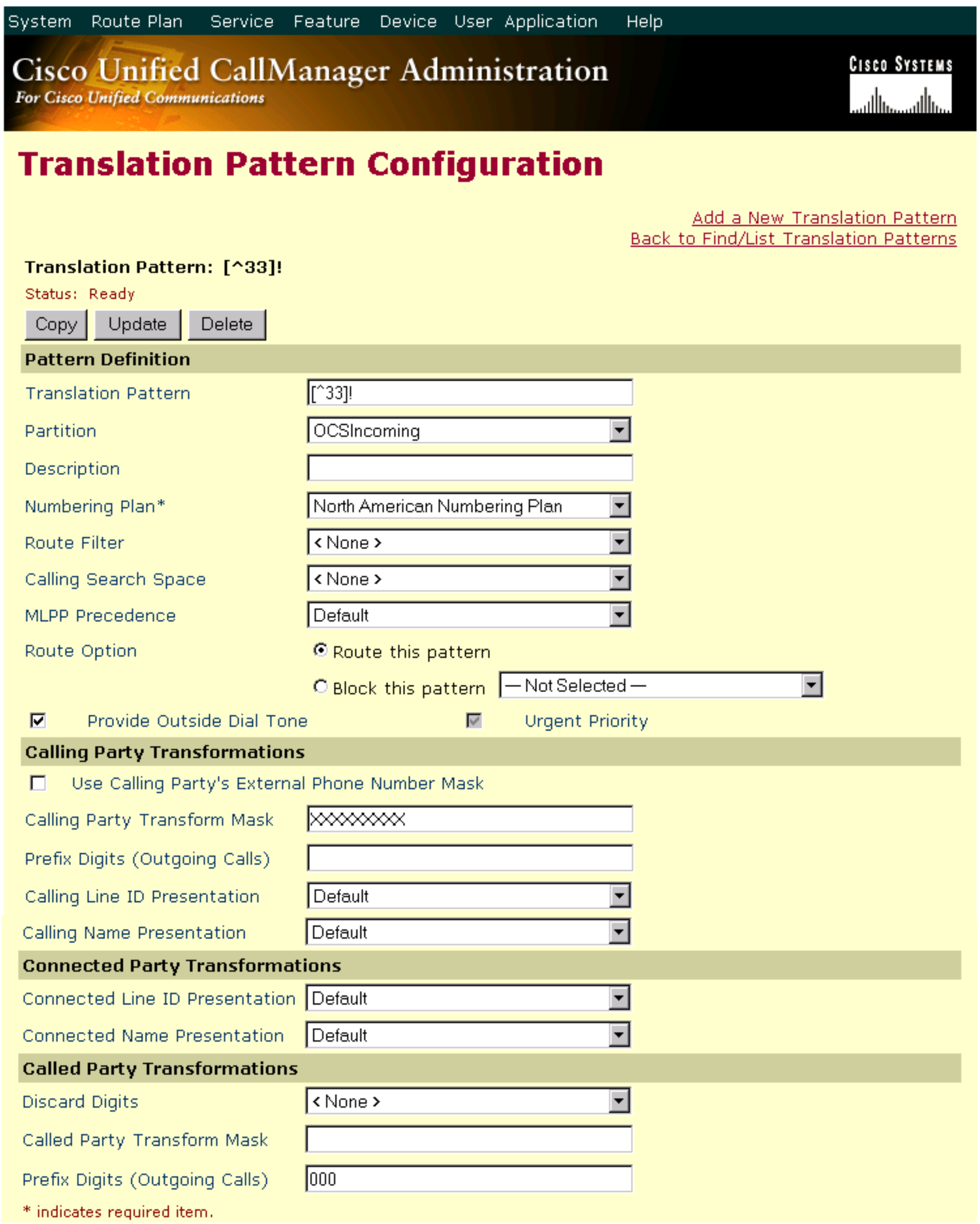


Figure C-7 Configuring translation pattern [^33]!

The second translation pattern is called 33.xxxxxxxxx. It will match any dial strings in the TO field that start with 33 followed by exactly nine digits. This translation pattern will handle all calls from Office Communications Server that are destined to a domestic (French) PSTN number. This translation pattern is also assigned to the OCSIncoming partition, as shown in Figure C-8.

For the FROM field: as for the previous translation pattern, we retain the last 9 digits, stripping the country prefix (which is always 33 in this case) from the E.164 calling party dial string, and presenting it to the PSTN in the required format. In this case it transforms 3316986xxxx into 16986xxxx.

For the TO field: The pattern strips the leading 33 (the digits prior to the dot) and adds 00 as a prefix to the remaining string. It transforms 33xxxxxxxxx into 00xxxxxxxxx, where the first 0 is the outside line prefix needed to obtain an outside line on the Cisco Unified Communications Manager.

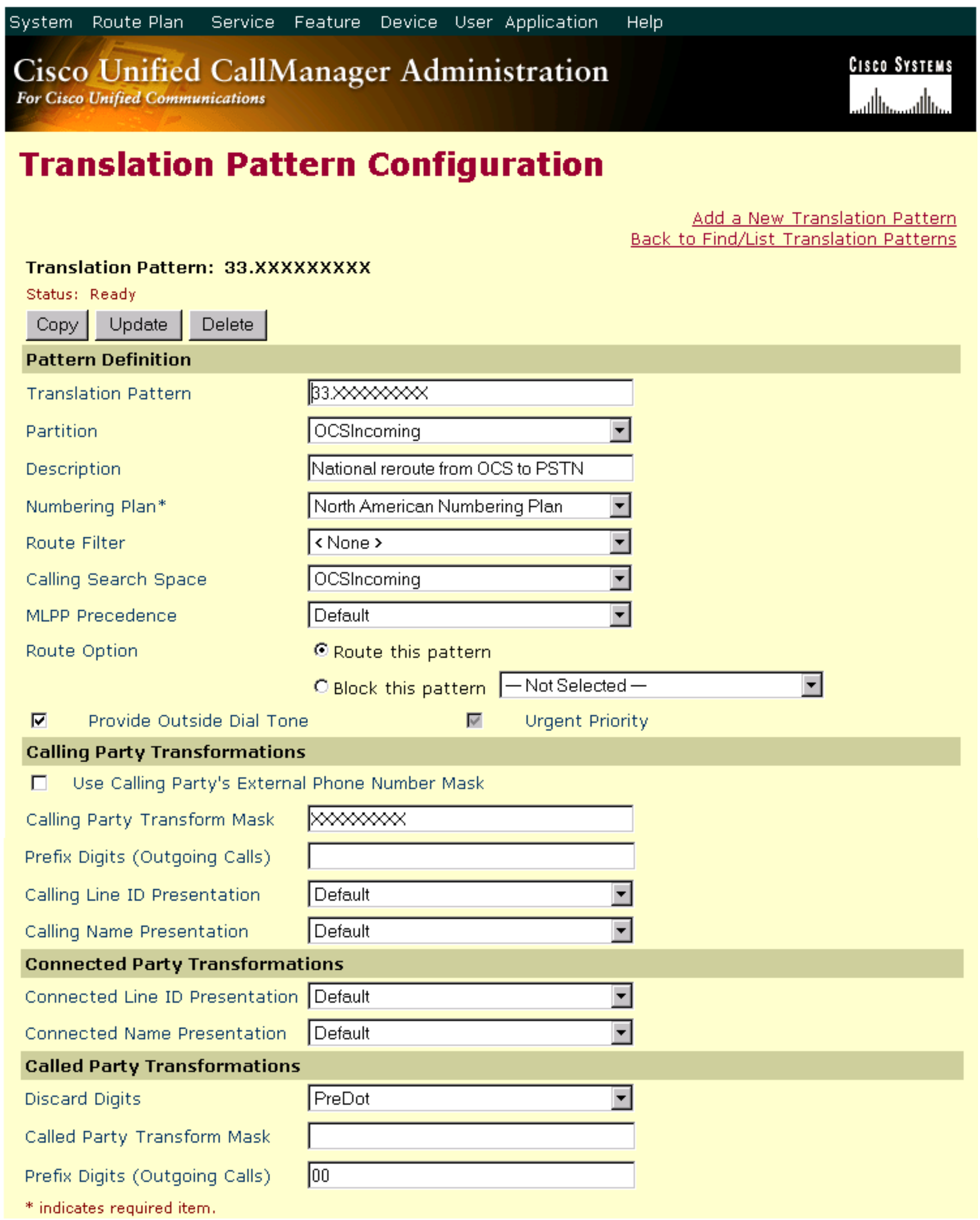


Figure C-8 Configuring translation pattern 33.xxxxxxxxx

The third translation pattern is called 3316986xxxx. This translation pattern will handle all calls from Office Communications Server that are destined to a Cisco Unified Communications Manager assigned number, i.e. calls internal to the company that should not be sent to the PSTN. Here again the translation pattern is assigned to the OCSIncoming partition (Figure C-9). This pattern will be applied to calls where the TO field is of the form 3316986xxxx, rather than the second translation pattern, because it represents a longer match. Cisco Unified Communications Manager selects translation patterns from a list on the basis of the longest match.

The translation pattern translates dial strings for calls sent by the Mediation Server where the TO field matches the pattern. It strips all leading digits from TO and FROM fields to retain the last 4 digits of both. Dial strings of the pattern 3316986xxxx will be translated to dial strings of the form xxxx that match the internal Cisco Unified Communications Manager dial plan. As can be seen below, this translation will be performed on both the called number and the caller number.

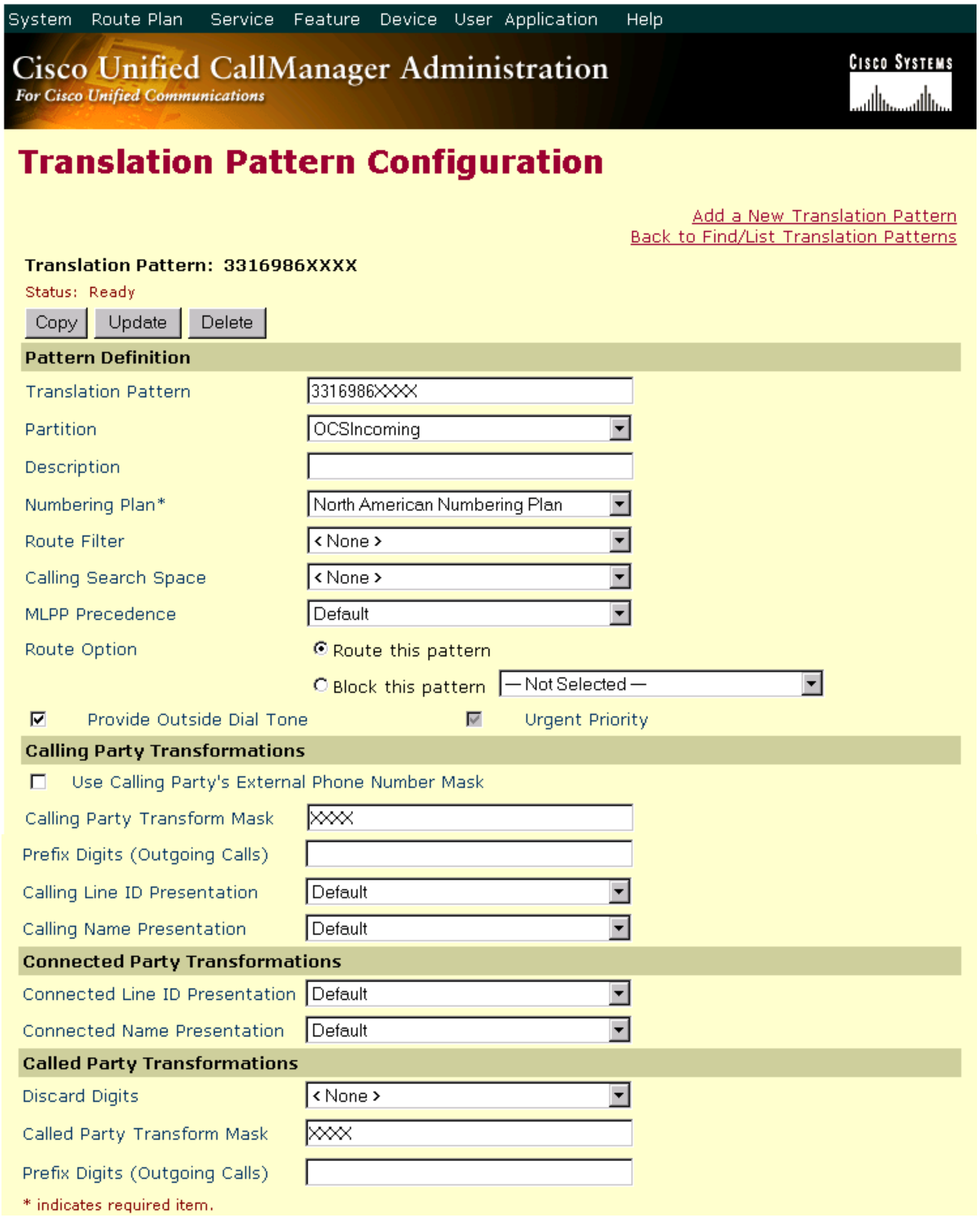


Figure C-9 Configuring translation pattern 3316986xxxx

Other translation patterns could be set up if appropriate. For example, for normalization of emergency numbers.

Step 4: Provisioning a SIP Trunk on Cisco Unified Communications Manager

Next, set up a SIP trunk on the Cisco Unified Communications Manager, as shown in Figure C-10, and assign the Calling Search Space for incoming traffic to the OCSIncoming partition that was created earlier.

Figure C-10 Creating the SIP trunk

As shown in Figure C-11, the trunk name in this example is “Trunk_to_OCS” and the Mediation Server’s external edge IP address is 192.168.0.105. Note the selection of TCP for transport, and the port number in Figure C-11.

System Route Plan Service Feature Device User Application Help
Cisco Unified CallManager Administration
For Cisco Unified Communications
Cisco Systems
Trunk Configuration
Add a New Trunk
Back to Find/List Trunk
Dependency Records
Product: SIP Trunk
Device Protocol: SIP
Status: Ready
Update
Delete
Reset Trunk
Device Information
Device Name* Trunk_to_OCS
Description Trunk_to_OCS
Device Pool* Default
Common Profile < None >
Call Classification* Use System Default
Media Resource Group List < None >
Location < None >
AAR Group < None >
Media Termination Point Required
Destination Address* 192.168.0.105
Destination Address is an SRV
Destination Port 5060
Incoming Port* 5060

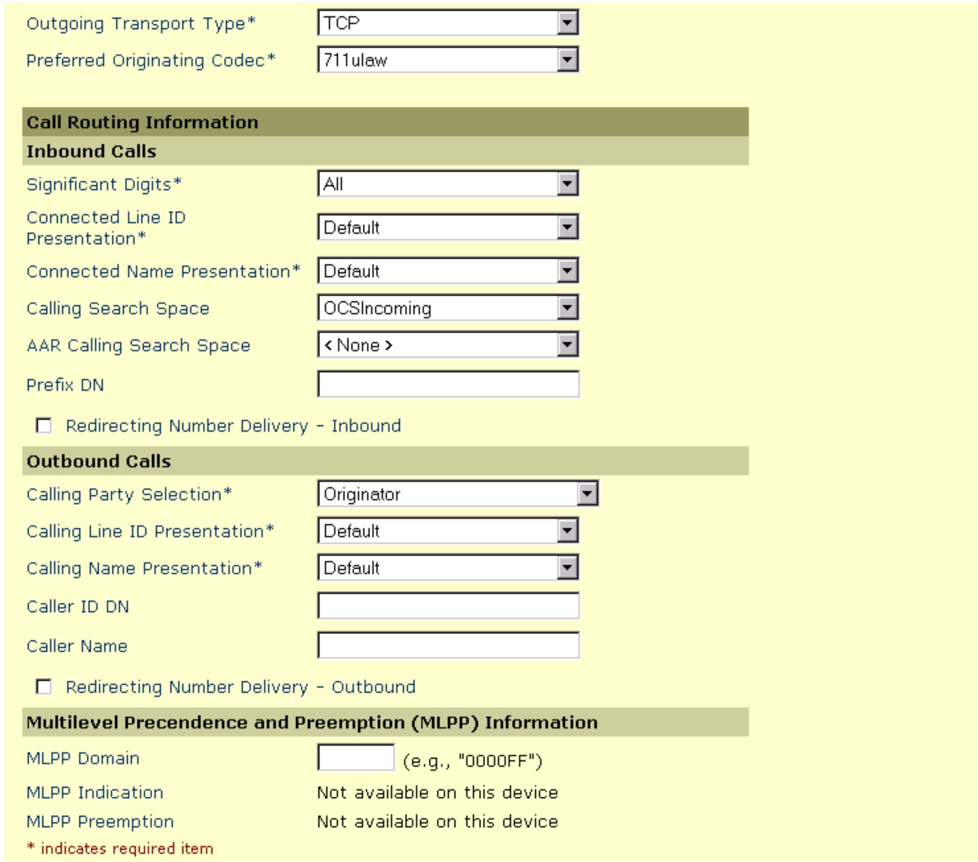


Figure C-11 Configuring the SIP trunk

Step 5: Setting Up a Route Pattern for the SIP Trunk on Cisco Unified Communications Manager

Route patterns are used for *outbound* calls from Cisco Unified Communications Manager to Mediation Server. They define what calls are sent to the SIP trunk based on matching the number in the TO field with a specific pattern. Route patterns can also perform transformations of the dial strings in both the TO and the FROM field.

A route pattern called [4-5]xxx is created to handle outgoing calls from Cisco Unified Communications Manager to Office Communications Server, as shown in Figure C-12. The route pattern is then associated with the Trunk_to_OCS SIP trunk that was created in Step 4. This route pattern instructs Cisco Unified Communications Manager to route to Mediation Server all calls destined to Office Communicator users (from both the PSTN and Cisco Unified Communications Manager users), on the basis of the match of the TO string with the pattern [4-5]xxx, as shown in Figure C-13. In this example,transformation of the strings in the TO and the FROM fields is not needed.

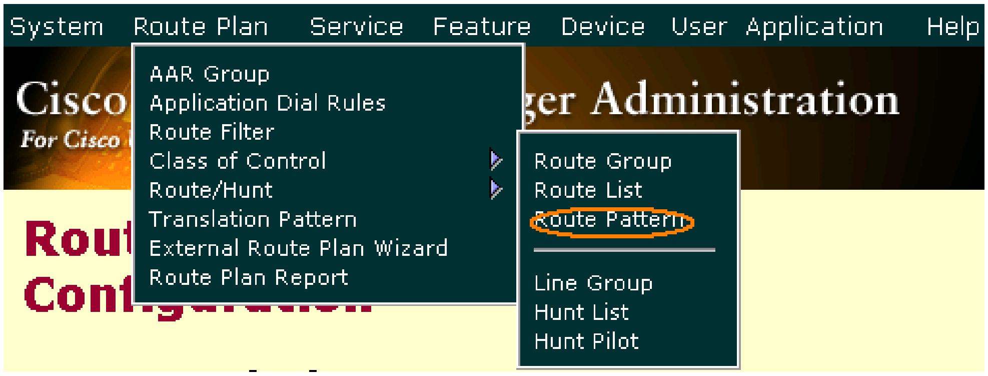


Figure C-12 Creating a route pattern

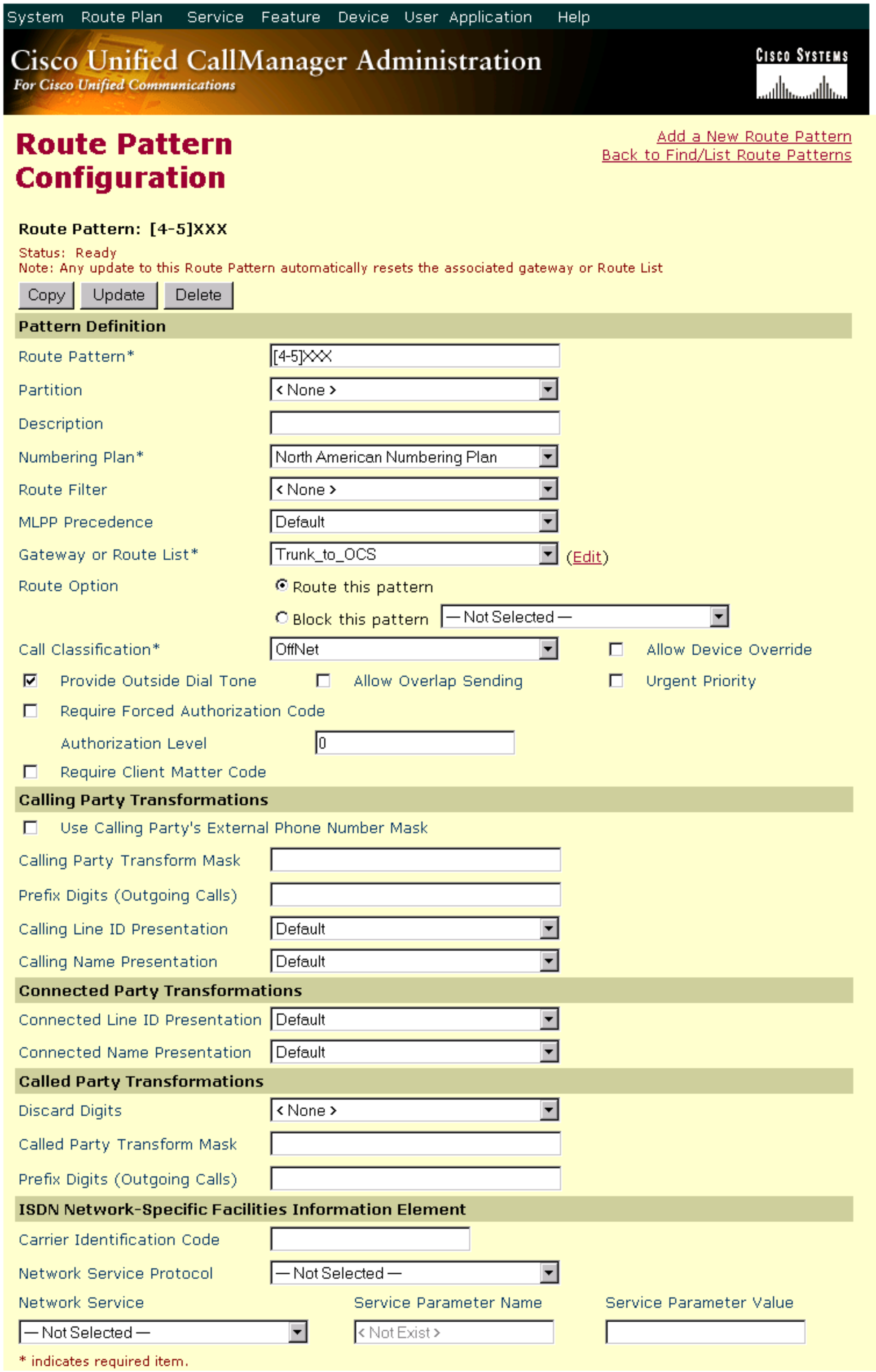


Figure C-13 Configuring the route pattern

Step 6: Setting Up Office Communications Server for Direct SIP

After Cisco Unified Communications Manager is configured, you will need to set up Office Communication Server. Configuring Office Communications Server for Direct SIP involves the following steps:

- Configuring the Mediation Server to connect to the SIP trunk
- Editing the Mediation Server Location Profile
- Configuring the Mediation Server Normalization Rules for Cisco Unified Communications Manager 4.2.1

Configuring the Mediation Server to Connect to the SIP trunk

In this example, the Mediation Server's outside edge IP address is 192.168.0.105, and the Cisco Unified Communications Manager's IP address is 192.168.0.110 (the Cisco Unified Communications Manager listens by default on its server IP address).

Open the Properties dialog box for the Mediation Server that you want to configure, as shown in Figure C-14.

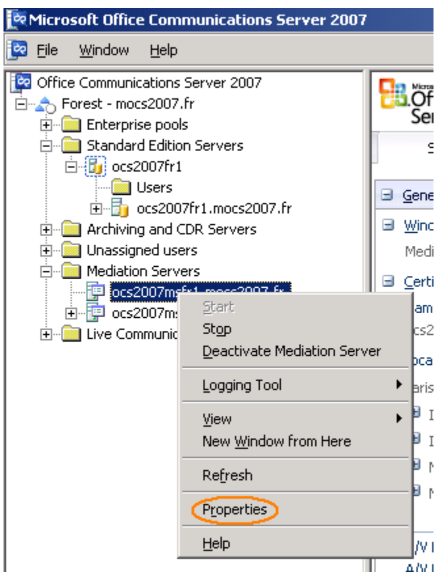


Figure C-14 Opening the Properties dialog box

The Cisco Unified Communications Manager's IP address is inserted in the PSTN Gateway next hop section on the Next Hop Connections tab in the Properties dialog box. Note the selection of the ports that correspond to the selection made on the SIP trunk, and the IP address of the Unified Communications Manager in the PSTN Gateway next hop IP address field, as shown in Figure C-15.

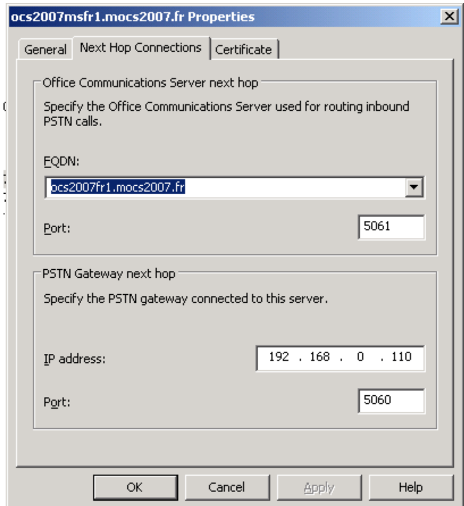


Figure C-15 The Next Hop Connections tab in the Mediation Server Properties dialog box

On the General tab, note the selection of a Default location profile for the Mediation Server (in this example, the location profile is called "Paris-LP.mocs2007.fr"). This location profile will be edited later in this configuration process , as shown in Figure C-16.

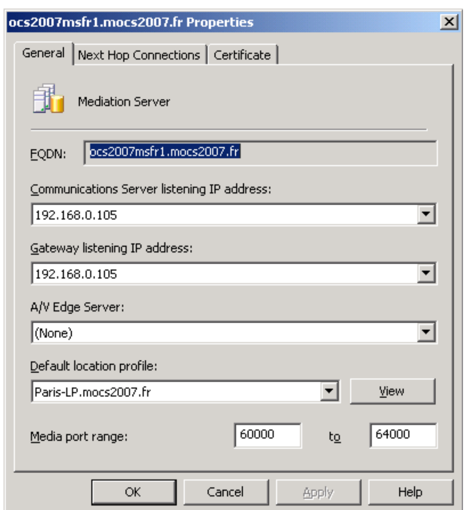


Figure C-16 The General tab in the Mediation Server Properties dialog box

Editing the Mediation Server Location Profile

Next, edit the location profile used by the Mediation Server to include the appropriate normalization rules for the SIP trunk. In this example, we will add or edit the normalization rules Internal number NR, International, and National to the Paris-LP.mocs2007.fr location profile.

Open the Voice Properties for the forest that you want to configure, as shown in Figure C-17.

Figure C-17 Voice Properties

Select the default location profile chosen in figure C-16 (in this example Paris-LP.mocs2007.fr) and click Edit, as shown in Figure C-18.

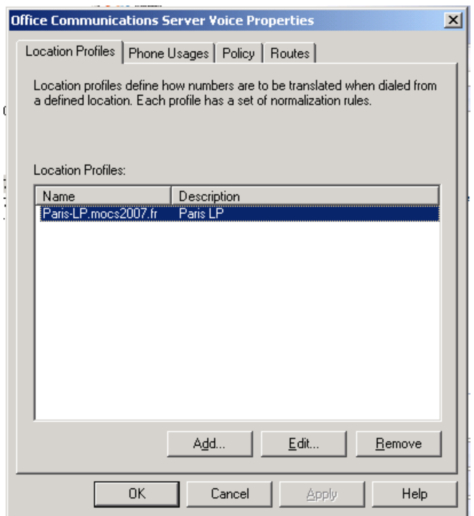


Figure C-18 Location Profile

Add to the Location Profile the following normalization rules: Internal number NR, International, and National, as shown in Figure C-19.

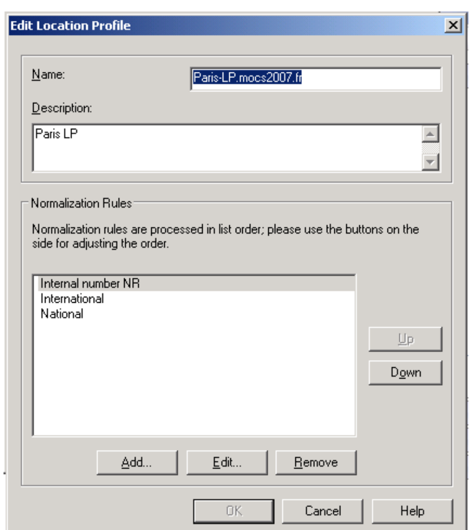


Figure C-19 Normalization rules

Configuring the Mediation Server Normalization Rules for Cisco Unified Communications Manager 4.2.1

The next step is to configure the normalization rules to normalize the dial strings coming across the SIP trunk from Cisco Unified Communications Manager into Office Communications Server. In this example, we configure the three normalization rules added previously in the Location Profile: Internal number NR, International, and National.

Internal number NR

The Internal number NR normalization rule transforms a 4-digit dial string xxxx into an RFC3966 compliant global number +3316986xxxx. The rule will be applied to inbound traffic to Office Communications Server from the Cisco Unified Communications Manager. The phone pattern regular expression in this scenario is ^([0-9]{4})$, and the translation pattern regular expression is +3316986$1.

Because of the Route Pattern configuration for the SIP trunk that was configured earlier in this example, calls that are passed to the Mediation Server by the Cisco Unified Communications Manager must have a TO field that matches the route pattern [4-5]xxx. Therefore, the normalization rule Internal number NR will translate the TO field to RFC3966 global numbers that should match Enterprise Voice enabled users of Office Communications Server.

In addition, calls that are passed to Mediation Server across the SIP trunk may have a FROM field that matches the dial string pattern xxxx. This will be the case when the calls have been dialed from an extension of the Cisco Unified Communications Manager. The normalization rule will also translate that FROM field to an RFC3966 global number, enabling querying against Active Directory for the name and SIP URI of the dialer. When the dialer is a user of Office Communications Server for Instant Messaging and Presence, having their SIP URI enables advanced scenarios; for example, the call recipient could redirect the incoming phone call to reply with an instant message.

Last, the normalization rule can also be used if users dial a 4-digit dial string in Office Communicator. The rule will normalize that dial string into a global number. If that number can be matched against the phone number of a user who is enabled for Unified Communications, it will be associated to a SIP URI and routed within Office Communications Server. If there is no match, the call will be routed to the appropriate external route—in this example, to the Cisco Unified Communications Manager, as shown in Figure C-20.

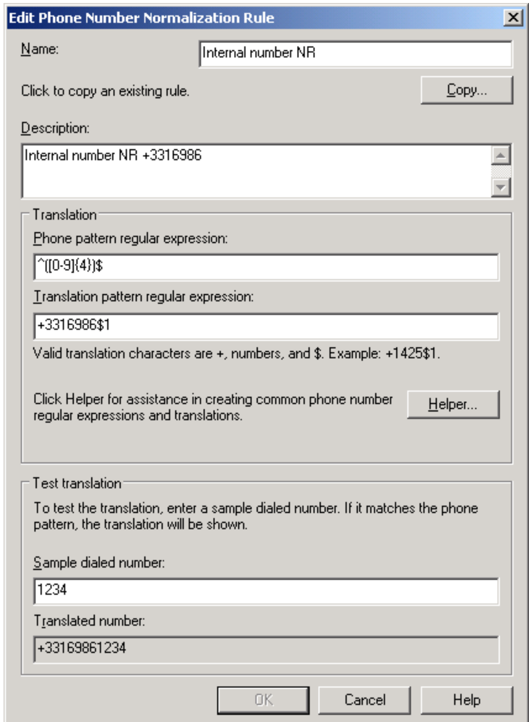


Figure C-20 Editing the Internal number NR normalization rule

International

The International normalization rule transforms a dial string that starts with 000 into an RFC3966 compliant global number by removing the 000 and adding a plus sign. The rule will be applied to inbound traffic to Office Communications Server from the Cisco Unified Communications Manager. The phone pattern regular expression in this example is ^000(d*)$, and the translation pattern regular expression is +$1.

For calls passed by Cisco Unified Communications Manager across the SIP trunk, only the FROM field should contain dial strings that start with 000. This will be the case when the calls have been dialed by a PSTN user outside of France, after the Cisco Unified Communications Manager has added the prefix 0. The normalization rule will translate that FROM field to an RFC3966 global number for use by Office Communications Server, as shown in Figure C-21.

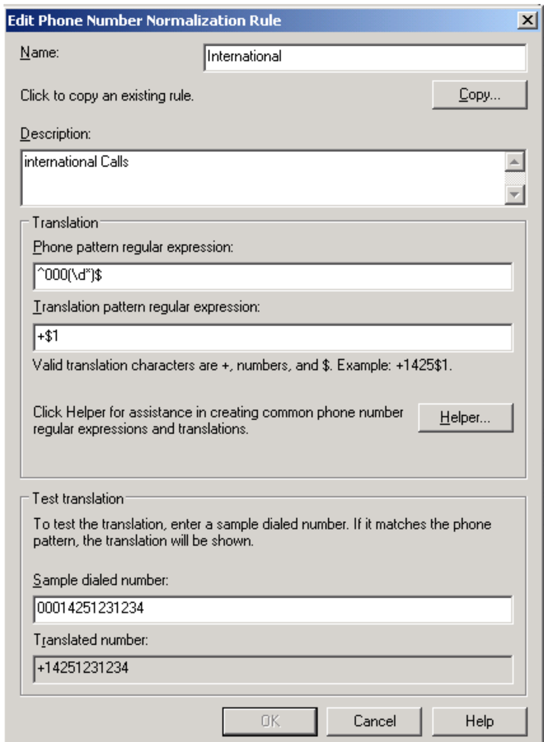


Figure C-21 Editing the International normalization rule

National

The National normalization rule transforms a dial string that starts with 00 followed by 9 digits into an RFC3966 compliant global number by removing the 00 and adding a "+33". The rule will be applied to inbound traffic to Office Communications Server from the Cisco Unified Communications Manager. The phone pattern regular expression in this example is ^00(d\{9})$, and the translation pattern regular expression is +33$1.

For calls passed by Cisco Unified Communications Manager across the SIP trunk, only the FROM field should contain dial strings that start with 00. This will be the case when the calls have been dialed by a PSTN user in France, after the Cisco Unified Communications Manager has added the prefix 0. The normalization rule will translate that FROM field to an RFC3966 global number for use by Office Communications Server, as shown in Figure C-22.

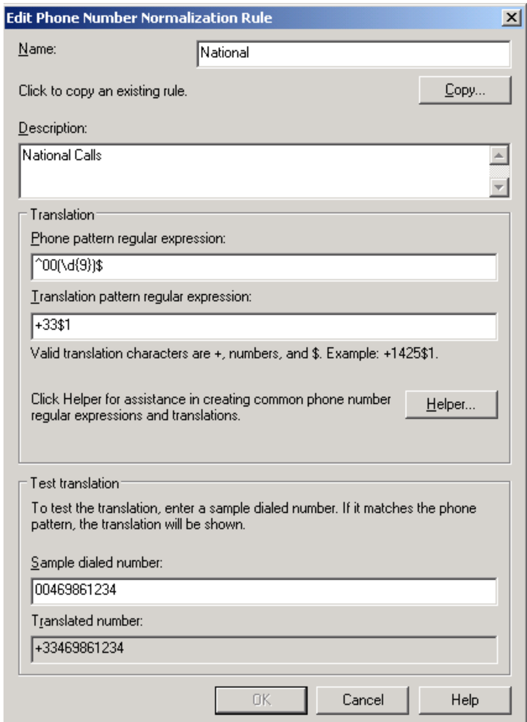


Figure C-22 Editing the National normalization rule

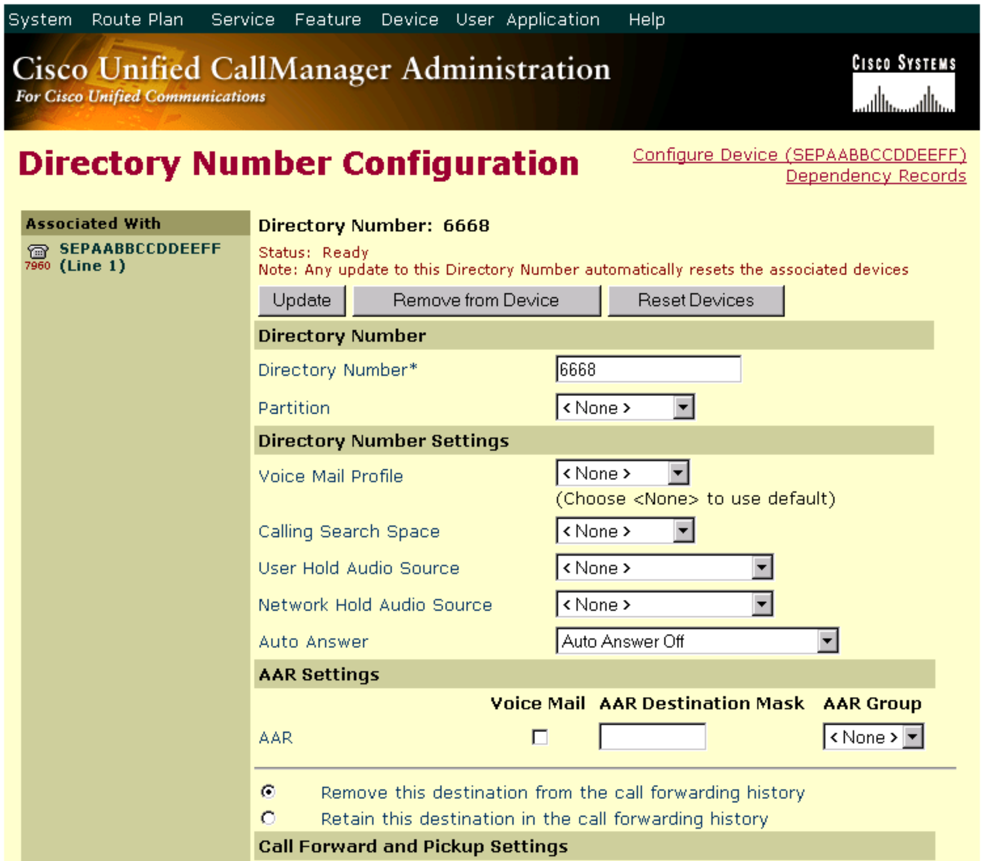
Advanced Scenario: Setting Up Support for Numbers Outside of the Range

The following steps provide a way to support migrating users from Cisco Unified Communications Manager to Office Communications Server without having to change their original extension to a new one taken from the contiguous range of numbers allocated to Office Communications Server. That is a frequent scenario for VIP or externally facing users whose phone number is broadly known outside of the organization. This can also be used as the general scenario when a dedicated range has not been allocated to Office Communications Server.

This scenario is achieved by creating a "Forward All" setting on the extension line (for example, extension 6668) to a prefixed line, for example 96668, and then by creating a new route pattern on the SIP trunk to match strings of the 9xxxx pattern, remove the 9, and redirect to extension 6668 on Office Communications Server.

Step 1: Setting Up the Forward All for the Extension

The following steps set up Forward All on the IP phone line extension 6668 to 96668. The prefix 9 will be used to reroute that extension to Office Communications Server. Note the Forwarded Call information, as shown in Figure C-23.


System Route Plan Service Feature Device User Application Help
Cisco Unified CallManager Administration
For Cisco Unified Communications
Cisco Systems
Directory Number Configuration
Configure Device (SEPAABBCCDDEEFF)
Dependency Records
Associated With
SEPAABBCCDDEEFF (Line 1)
7960
Directory Number: 6668
Status: Ready
Note: Any update to this Directory Number automatically resets the associated devices
Update
Remove from Device
Reset Devices
Directory Number
Directory Number*
6668
Partition
< None >
Directory Number Settings
Voice Mail Profile
< None >
(Choose <None> to use default)
Calling Search Space
< None >
User Hold Audio Source
< None >
Network Hold Audio Source
< None >
Auto Answer
Auto Answer Off
AAR Settings
Voice Mail
AAR Destination Mask
AAR Group
AAR
< None >
Remove this destination from the call forwarding history
Retain this destination in the call forwarding history
Call Forward and Pickup Settings

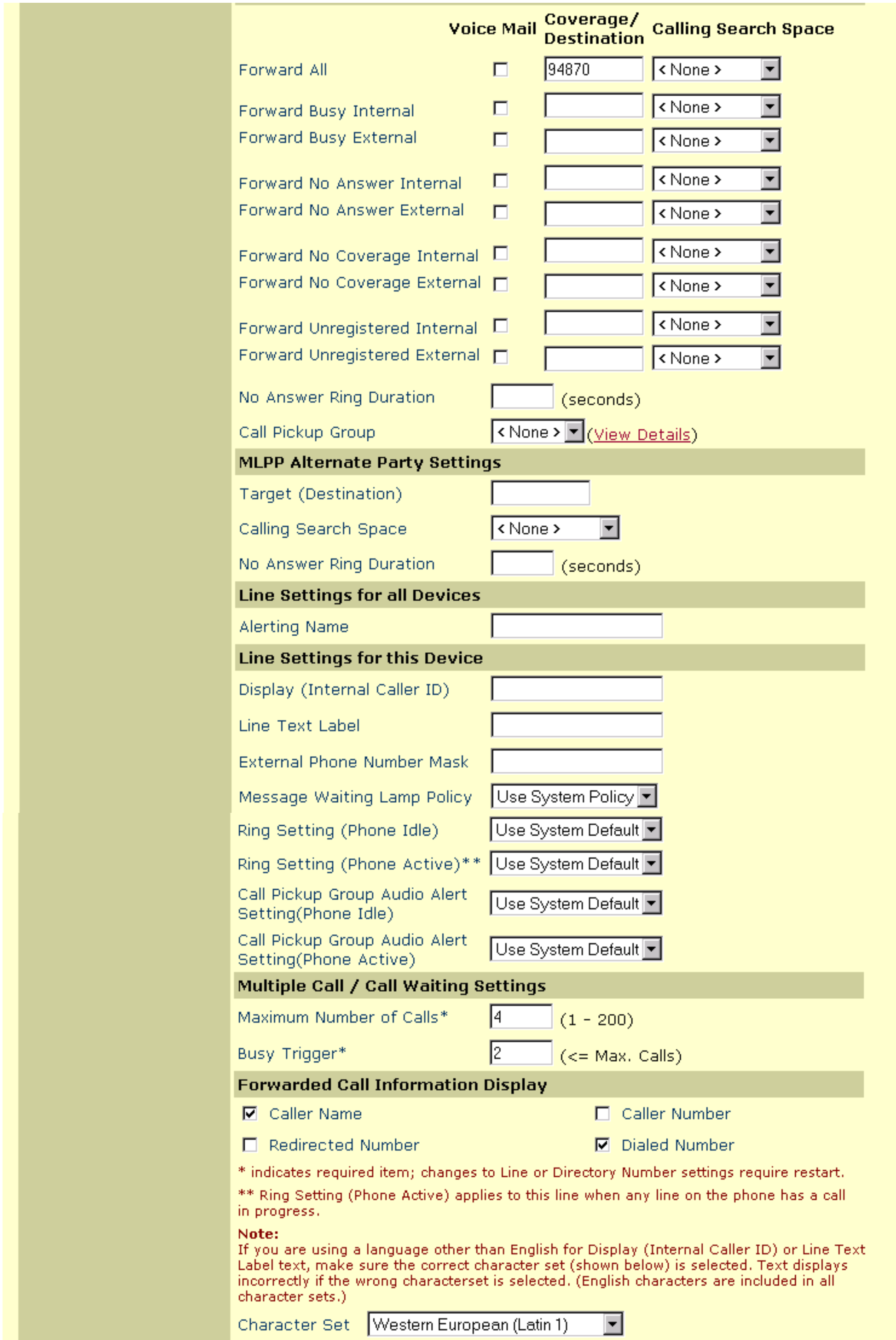
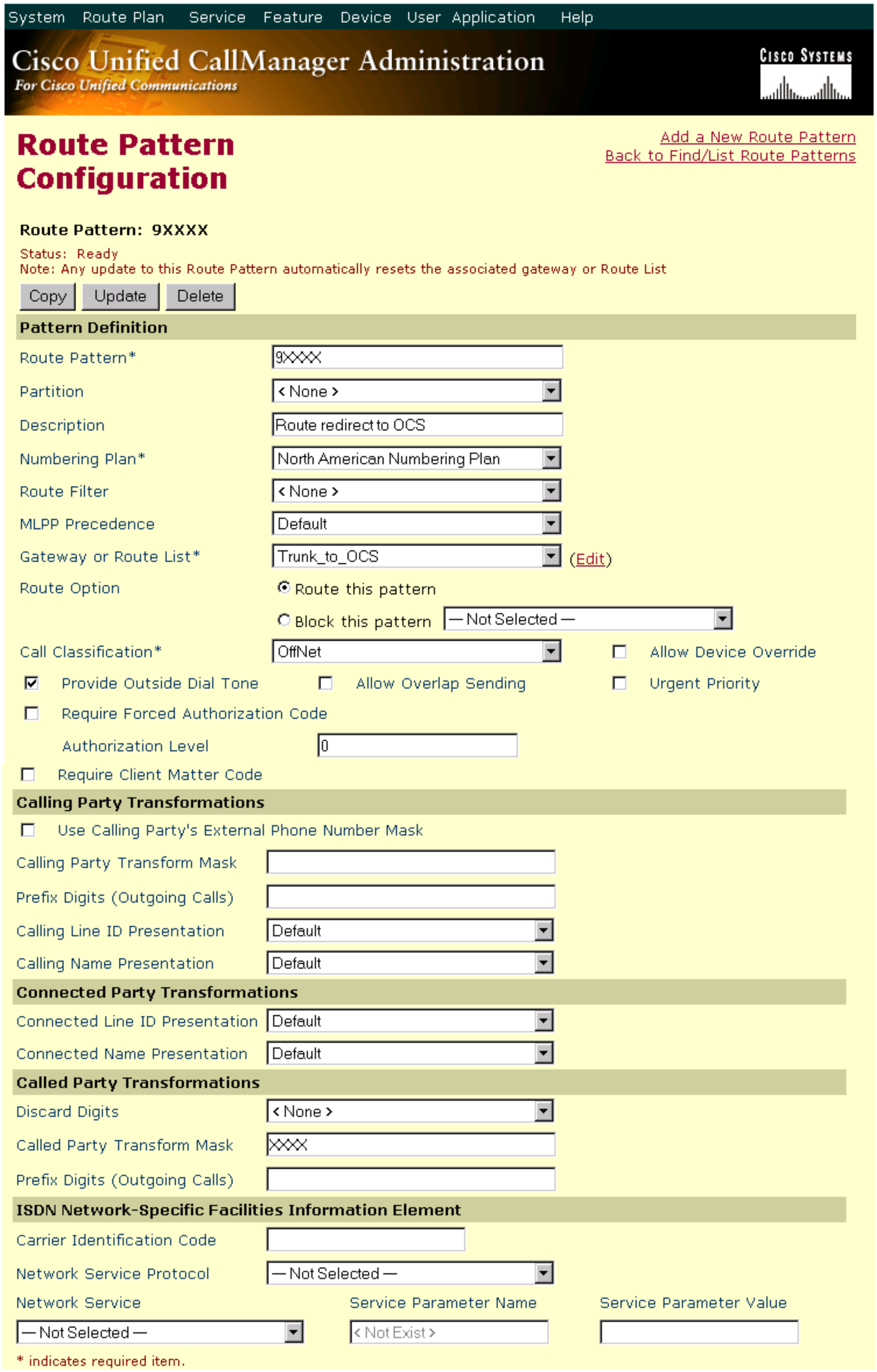


Figure C-23 Setting up Forward All

Step 2: Creating a Route Pattern Associated with the Prefix

Configure the settings as shown in Figure C-24 to create a route pattern to reroute all traffic with the 9xxxx pattern (meaning traffic that has the 9 prefix) to the SIP trunk to Office Communications Server.

The prefix will be removed (as indicated by the Called Party Transform Mask setting) and the call redirected to extension 6668 on Office Communications Server.


System Route Plan Service Feature Device User Application Help

Cisco Unified CallManager Administration
For Cisco Unified Communications
Cisco Systems

Route Pattern Configuration

Add a New Route Pattern
Back to Find/List Route Patterns

Route Pattern: 9XXXX

Status: Ready
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Copy | Update | Delete

Pattern Definition

Route Pattern*	9XXXX
Partition	< None >
Description	Route redirect to OCS
Numbering Plan*	North American Numbering Plan
Route Filter	< None >
MLPP Precedence	Default
Gateway or Route List*	Trunk_to_OCS (Edit)
Route Option	◉ Route this pattern
	○ Block this pattern — Not Selected —
Call Classification*	OffNet

☐ Allow Device Override
☑ Provide Outside Dial Tone
☐ Allow Overlap Sending
☐ Urgent Priority
☐ Require Forced Authorization Code
Authorization Level: 0
☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation	Default
Calling Name Presentation	Default

Connected Party Transformations

Connected Line ID Presentation	Default
Connected Name Presentation	Default

Called Party Transformations

Discard Digits	< None >
Called Party Transform Mask	XXXX
Prefix Digits (Outgoing Calls)	

ISDN Network-Specific Facilities Information Element

Carrier Identification Code	
Network Service Protocol	— Not Selected —

Network Service	Service Parameter Name	Service Parameter Value
— Not Selected —	< Not Exist >	

* indicates required item.

Figure C-24 Creating a route pattern

Summary of the Call Flows and Number Transformations

Figures C-25, C-26, and C-27 summarize the flow of calls for incoming calls, outgoing calls, and international calls.

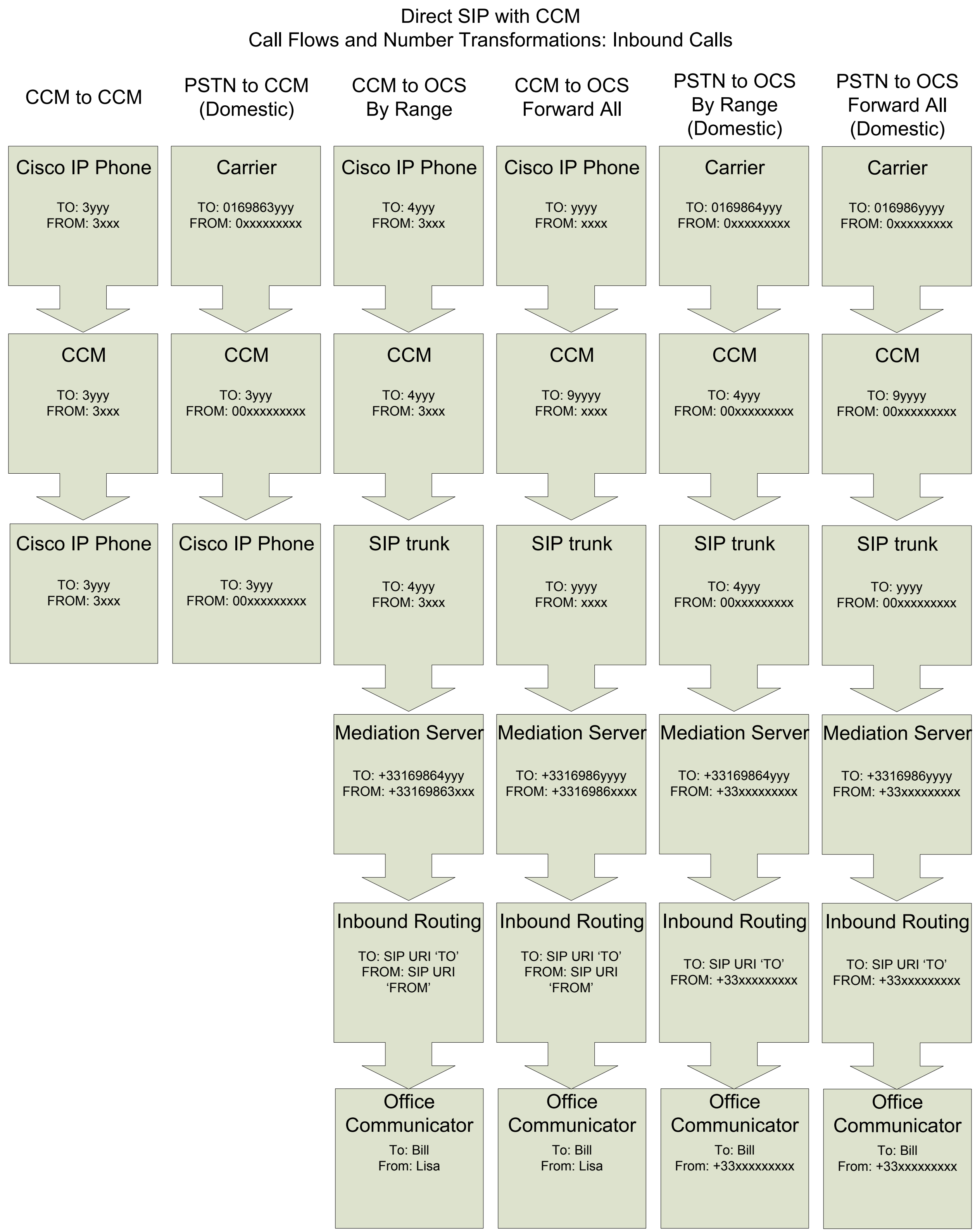


Figure C-25 Call flow for inbound calls

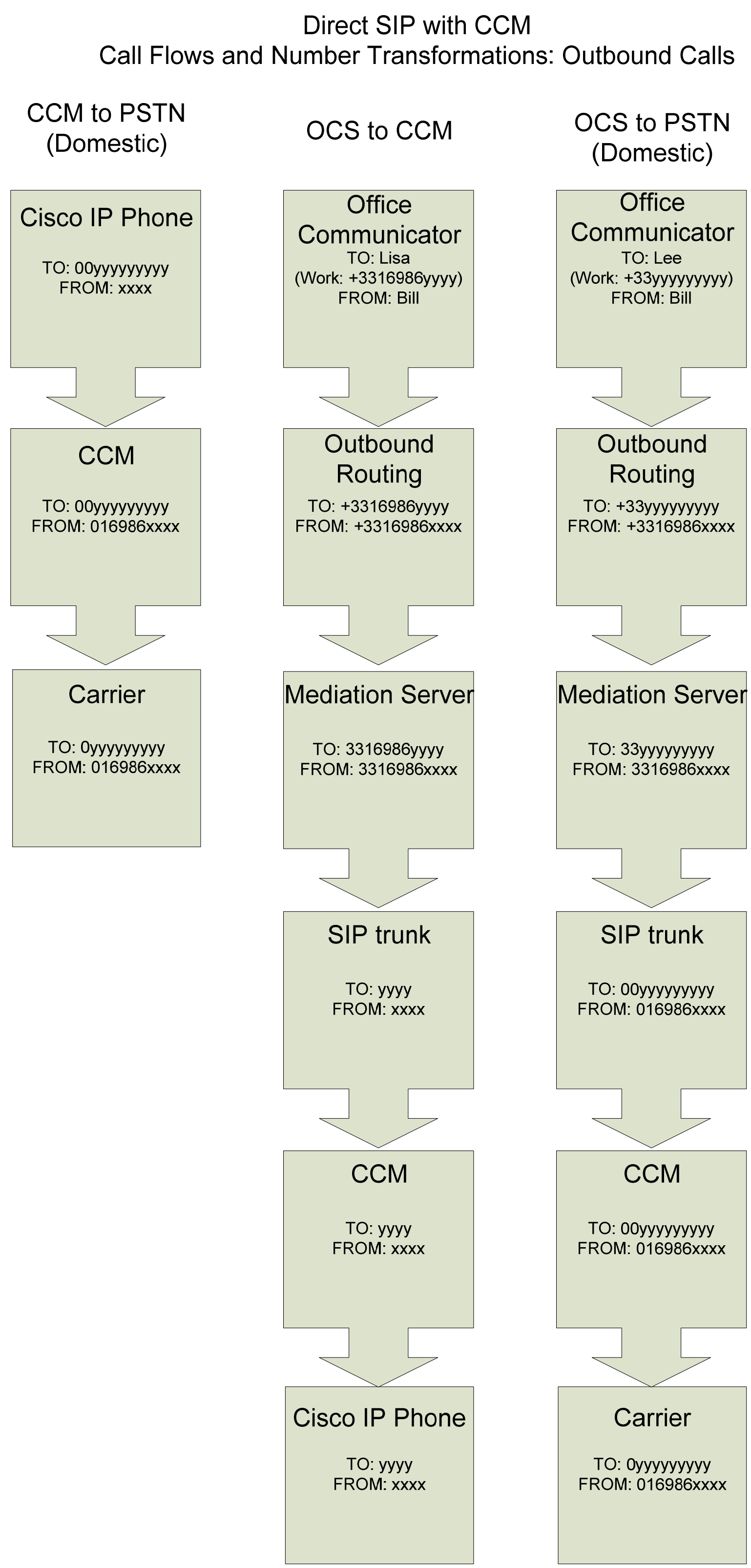


Figure C-26 Call flow for outbound calls

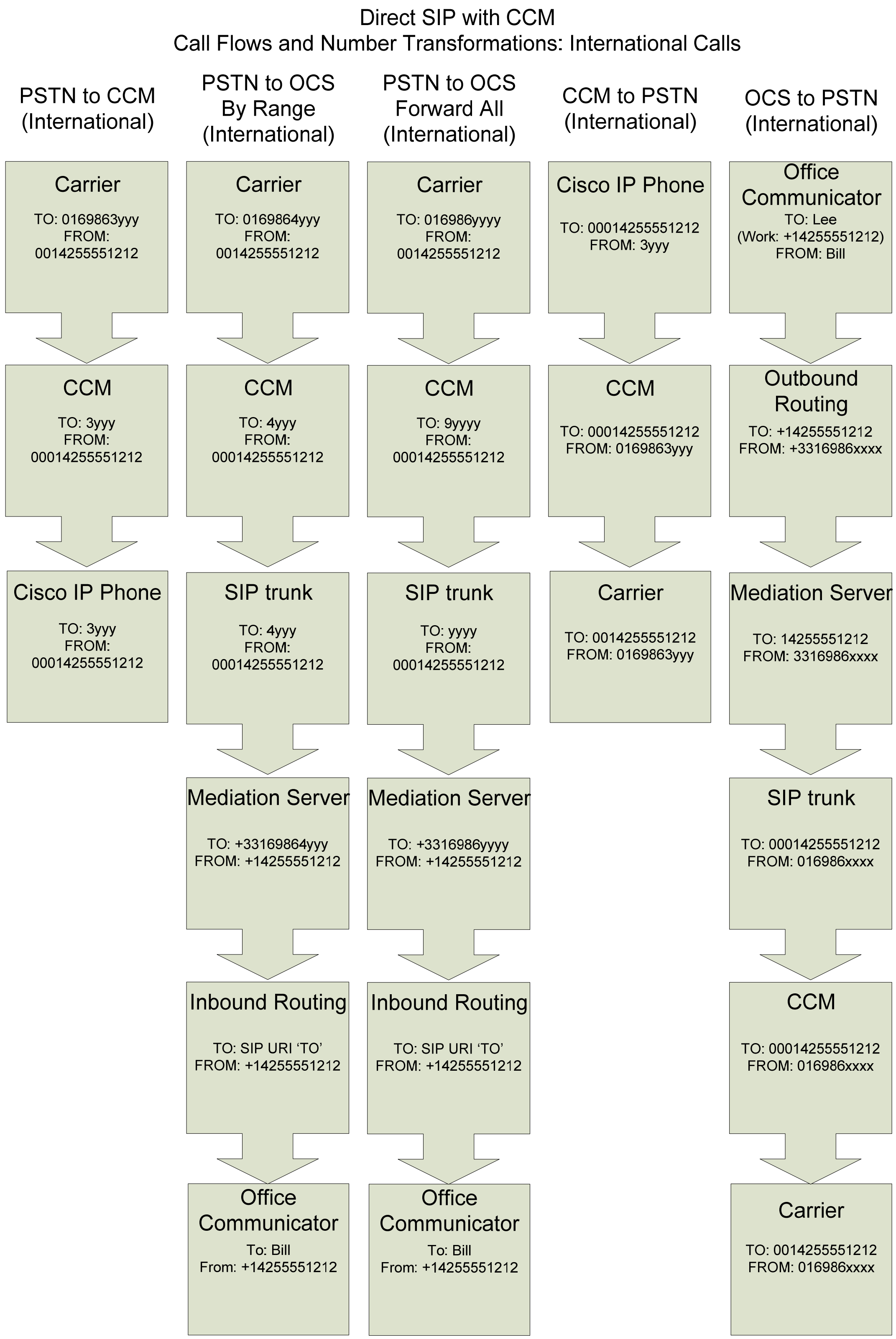


Figure C-27 Call flow for international calls