



SMART Designs

Cisco Unified Communications



Multi-Systems Cisco BE3000 Solution

Application Note

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Cisco SMART Designs

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Multi-Systems Cisco BE3000 Solution Application Note

This application note provides validated and optimized configuration steps for interconnecting multiple Cisco Business Edition 3000 (BE3000) systems to accomplish a distributed call control Unified Communications solution. To learn about basic deployment of BE3000 as a standalone PBX, visit the SMART Design website at the following URL: <http://www.cisco.com/go/partner/smartdesigns>.

This application note is verified using Cisco BE3000 software version 8.6.4. Other versions of BE3000 might require different configuration steps. If required to upgrade Cisco BE3000 software, complete the following steps.

Procedure

- Step 1** Visit <http://www.cisco.com/go/be3000>.
 - Step 2** Select the **Models** tab.
 - Step 3** Click the **Cisco Business Edition 3000** link to visit the BE3000 product page.
 - Step 4** On the product page, click the **Download Software** link at the right side under the Support section.
-

This document is written for voice professionals who participate in the design of converged IP networks. It is assumed that readers have a basic knowledge of IP networking and security. The underlying network infrastructure is assumed to be based on the one of the Cisco SMART Designs Network Foundation designs, either Cisco Secure Network Foundation (SNF) or Cisco Small Business Network Foundation (SBNF).

The design guide and configuration guides of SNF or SBNF describe the network foundation and the configurations of the various devices used, with the following key elements:

- LAN—Connects voice endpoints and voice application servers
- WAN—Provides connectivity to the Internet
- Virtual private network (VPN)—Provides secure connectivity between business locations
- Quality of service (QoS)—Ensures real-time traffic priority across the network
- Network security—Ensures the protection of sensitive communications between business locations



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To learn more about SNF and SBNF architectures, see to the design guide and configuration guides at SMART Designs website (<http://www.cisco.com/go/partner/smartdesigns>).

Overview

This application note pertains to all business locations defined by SMART Designs: main office, remote office, home office, and mobile worker. The BE3000 GUI screens use the concept of *sites* to configure the BE3000 system. These sites correspond to the SMART Designs business locations as shown in [Table 1](#).

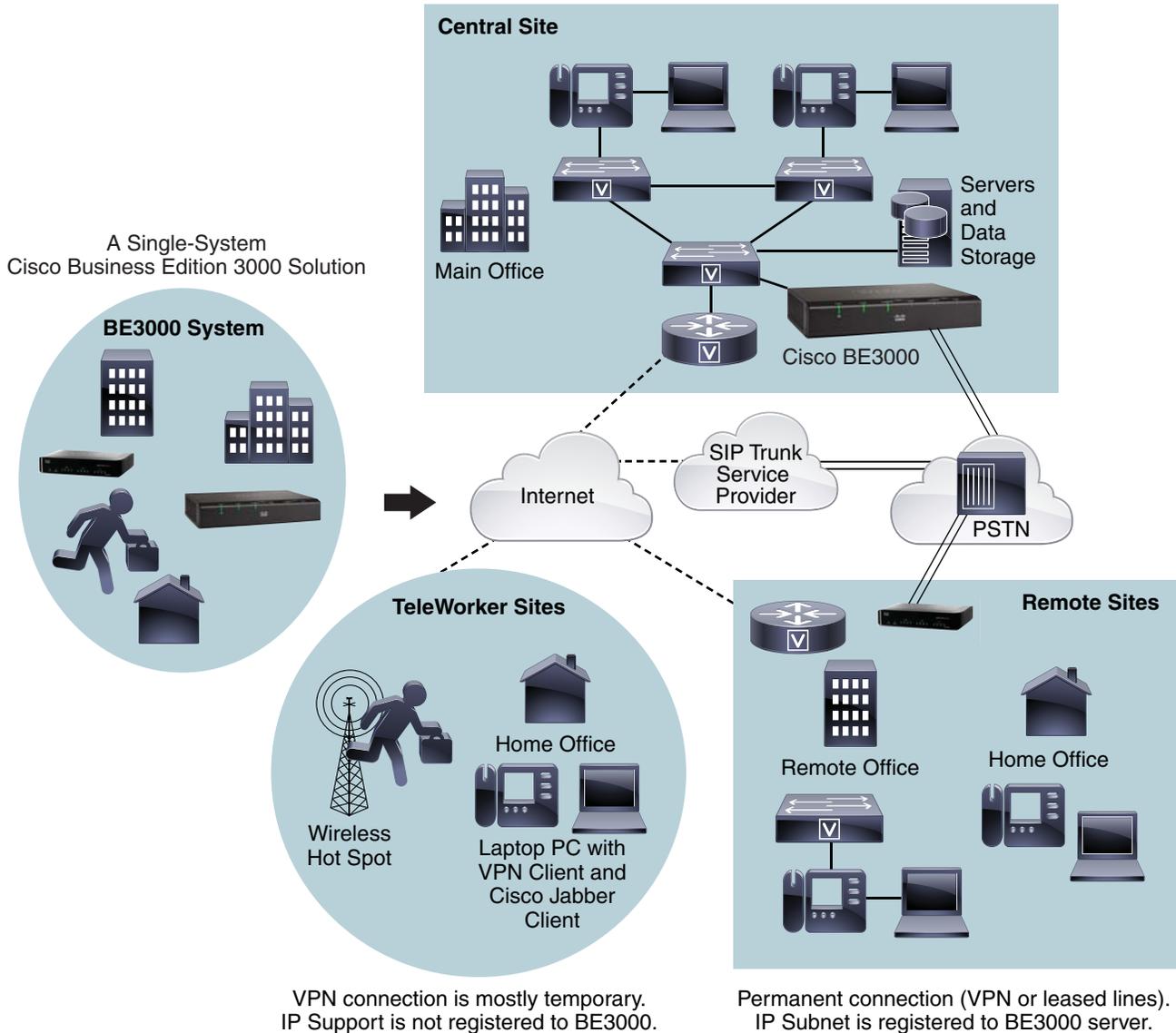
Table 1 SMART Designs Business Locations

Site As defined in BE3000 Administrative Interface	Business Location As defined by SMARTDesigns	Description
Central site	Main office	The main business location where most employees work, and where most of the networking equipment (including BE 3000) and PSTN connections having the main business phone numbers are deployed.
Remote site	Remote office	Connected to the main office (central site) via permanent VPN. Each remote office registers with the BE3000. Using a gateway registered to BE3000, these sites can optionally have a PSTN connection for an additional local business phone number.
	Home office	Connected to the main office (central site) via permanent VPN. Extending the office voice network to a home office is optional. If the voice network is extended, the home office registers with the BE3000.
Teleworker site	Mobile worker	Establishes temporary VPN connections from public places such as hotels or airports. Cisco Jabber client registered to BE3000 is used for voice calls.
	Home office	Establishes temporary VPN connections from employee residence. IP phones and/or a Jabber client registered to BE3000 are used for voice calls.

Note that both the remote sites and teleworker sites are optional. Typically, sites are geographically distant locations with different area codes. However, in some cases when a group of users is using a separate PSTN gateway, the sites could also be logical, such as different buildings, floors, or sections of a building.

For the sake of naming a solution with multiple BE3000 servers, Cisco defines a collaboration solution founded on a single Cisco BE3000 server as a *single-system BE3000 solution*, which may consist of multiple sites of various types, as shown in [Figure 1](#).

Figure 1 Single-System BE3000 Solution with Multiple Sites



Need for Multi-Systems BE3000 Solution

In some cases, when a remote office location is distant and does not have reliable data connectivity, it is better to install a separate BE3000 server at that branch office. In such cases, more than one BE3000 systems are required. Business owners need to connect these systems in such a way that employees in different locations are able to call each other without dialing via the PSTN, and using an abbreviated dial plan. This solution provides the following benefits:

- Each location with BE3000 still has telephony service even when connectivity with the main office is lost.
- Employees can dial internal numbers to reach employees in another business location.
- Employees at each business location can dial PSTN calls via the local PSTN connection.

This solution provides only basic connectivity between the autonomous BE3000 systems. The limitations of multi-systems BE3000 solutions are as follows:

- Each BE3000 system has its own auto-attendant (AA) and voicemail (VM) system.
- Each BE3000 system has its own directory, and AA can perform dial-by-name only for local users.
- The site feature transparency does not work.
- PSTN calls must be routed from trunks registered on the local BE 3000.
- The busy lamp field (BLF) does not work across two BE3000 systems.
- Call park, call pickup, pickup groups, and hunt lists do not work across two BE3000 systems.

**Note**

To achieve the same length dial plan across the combined solution, all BE3000 systems must have an extension range with same length, such as 3-digits or 4-digits.

Session Initiation Protocol

Session Initiation Protocol (SIP) is an application-layer protocol, which means it requires the services of transport layer protocols TCP or UDP. SIP uses the format of the most popular protocols of the Internet, HTTP and SMTP, and thus has quickly become the protocol of choice. SIP is used to control, as a signaling protocol, and to create, modify, and terminate multimedia sessions among one or more participants, such as audio sessions among one or more IP phones.

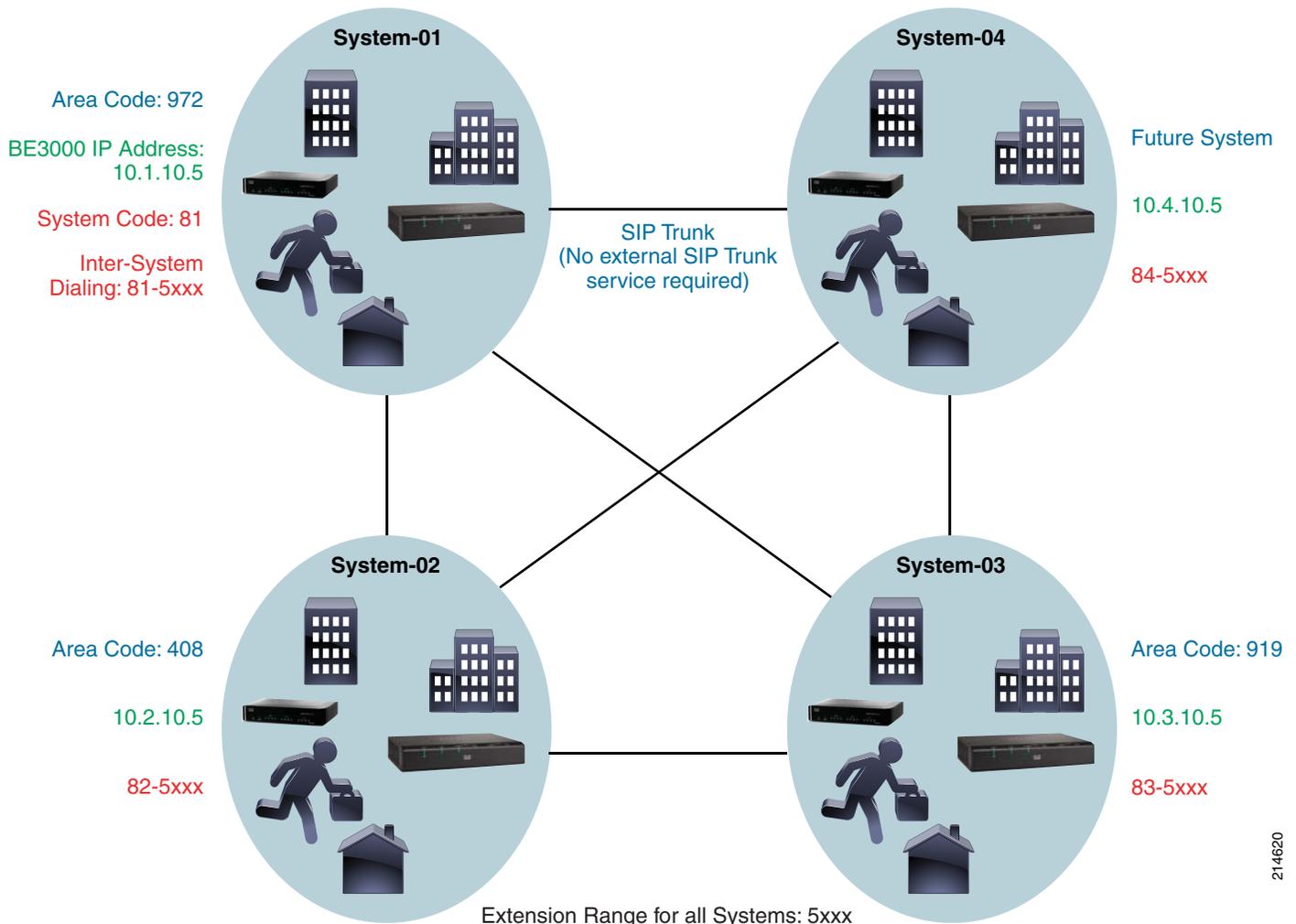
As shown in [Figure 1](#), SIP trunk services are now available in addition to traditional circuit-switched PSTN services. Customers can use their Internet connection to use a commercial SIP trunk service. SIP proxy devices and servers reside in the SIP service provider's data center. SIP communication happens between SIP proxy/servers and clients, such as SIP gateways, and IP phones at customer premises.

**Note**

The solution described in this document uses a private SIP trunk between two BE3000 systems. The customer does not need to subscribe to a SIP trunk service.

This document describes how to connect multiple BE3000 systems; specifically, three BE3000 systems located in different business locations are connected. All three solutions are identical and use the same extension ranges. Each system is assigned a 2-digit system code, such as 81, 82, 83, and so on, for connecting a fourth system in the future. These single-system BE3000 solutions are connected using a system-to-system SIP trunk and abbreviated dial plan, as shown in [Figure 2](#).

Figure 2 Multi-Systems BE3000 Solution



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**Note**

There is no technical limit to the number of BE3000s in a multi-system solution. However, because of the number of SIP trunks, it becomes difficult to manage when you try to go beyond four BE3000 servers. Cisco strongly recommends using the Cisco Business Edition 6000 Series if you want to connect more than four autonomous locations.

WAN Connection Quality

To achieve good voice call quality on inter-system SIP trunks over a VPN, Cisco strongly recommends having a ping roundtrip time of less than 150 milliseconds. Less than 25 milliseconds roundtrip time provides a very high quality call. A roundtrip delay of near 150 milliseconds provides a voice quality similar to a trans-oceanic international call.

Customers should have a premium class Internet service with a large amount of bi-directional speed, 768 kbps or more. This bandwidth requirement can be much higher, depending on the traffic profile over the VPN.

Call Admission Control (CAC) limits the number of calls on the WAN to match the VoIP bandwidth reserved by QoS. The amount of calls that can pass across a WAN link depends on the amount of bandwidth available and the codec used. Low Latency Queuing (LLQ) is configured on routers to reserve a specific amount of bandwidth for voice calls; for example, for six simultaneous G711 VoIP calls, a dedicated bandwidth of 512 kbps is required. To learn more details of CAC, QoS, and bandwidth requirements, see the UC and SNF design guides at the Cisco SMART Designs website (<http://www.cisco.com/go/partner/smartdesigns>).

Provisioning Sequence

This application note assumes that the existing Cisco BE3000 solution is operational and is deployed according to the *Cisco Business Edition 3000 (Single-Site Deployment) Implementation Guide* available at the following website: <http://www.cisco.com/go/smartdesigns/uc>.

The following is the validated sequence for connecting multiple BE3000 systems:

1. Verify extension range of every BE3000 system
2. Provision system-to-system SIP trunks
3. Provision connection groups for every SIP trunk and PSTN connection
4. Edit the dial plan of every BE3000 system
5. (Optional) Edit the SIP trunk to show the system code for incoming call

Each of the above steps is explained in detail in the following sections.

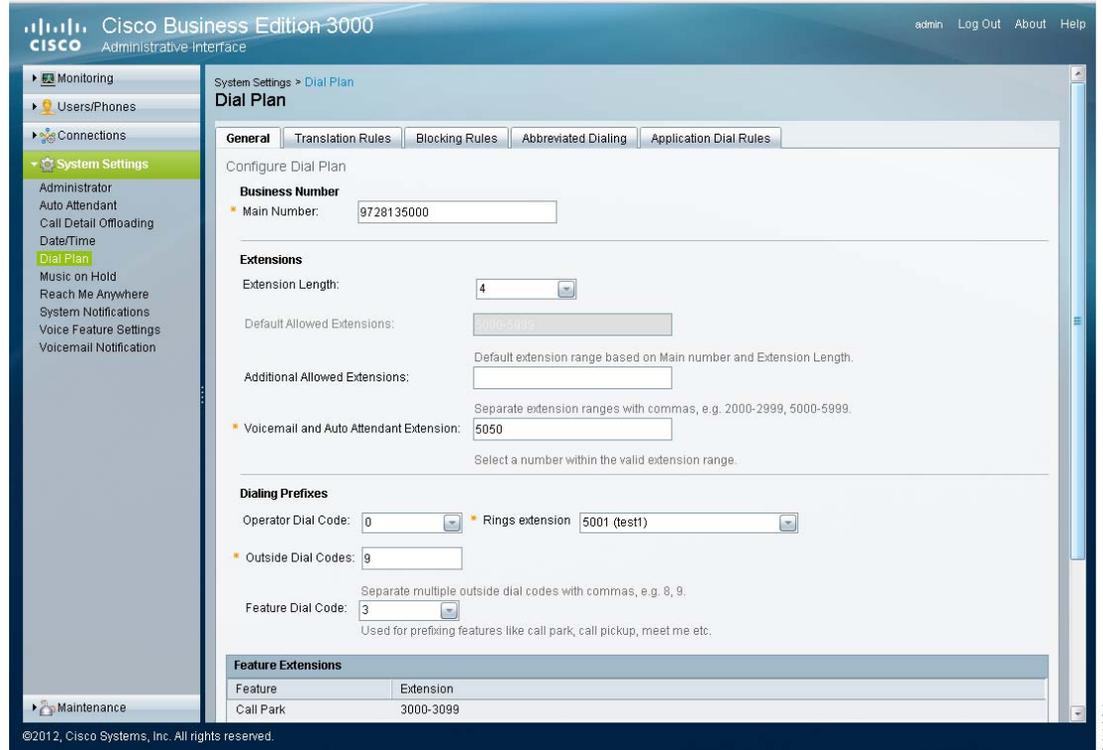
Verifying Extension Range

To connect multiple BE3000 systems with the same length dial plan across the business collaboration solution, all BE3000 systems must have same length of extension range. In this application note, it is also assumed that all BE3000 systems are identical and using a 4-digit dial plan with a personal extension range of 5xxx and a default feature extension range of 3xxx.

To verify the extension range on all BE3000 servers, complete the following steps.

Procedure

- Step 1** Log in to Cisco BE3000 Administrative utility.
- Step 2** Select **System Settings** > **Dial Plan** in the Administrative Interface of Cisco Business Edition 3000.
- Step 3** Verify the extension length and extension ranges, in addition to the business main number and other information, as shown in [Figure 3](#).

Figure 3 Verifying Extension Ranges of all BE3000 Systems

- Step 4** If any system is found to have a different length than the others, re-install the BE3000 with the correct dial plan accordingly before provisioning system-to-system SIP trunks, as described in next section. Extension length in BE3000 cannot be changed by simply re-configuring; the entire system must be re-installed from the beginning, and the digit length set in the first-time setup wizard. For details on configuring the BE3000 solution, see the Implementation Guide and Administration Guide listed in the [“For More Information” section on page 21](#).

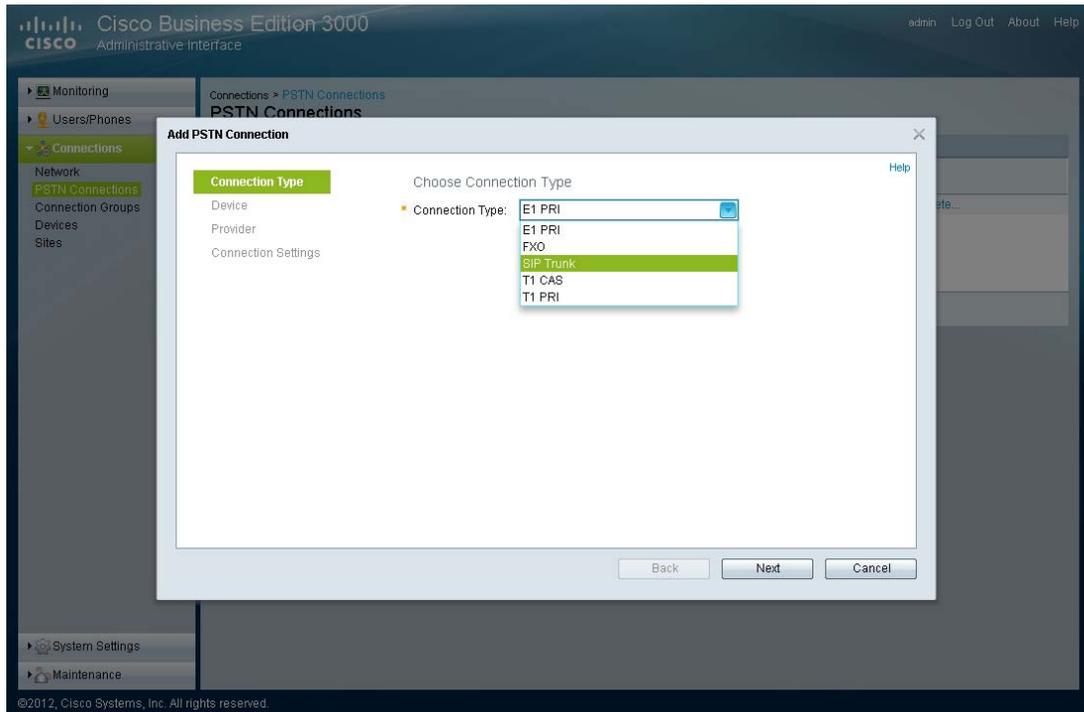
Provisioning System-to-System SIP Trunks

To provision a SIP trunk on the Cisco BE3000, complete the following steps.

Procedure

- Step 1** Log in to Cisco BE3000 Administrative utility.
- Step 2** Select **Connections > PSTN Connections**, and click **Add PSTN Connection....**
- The Add PSTN Connection window appears, as shown in [Figure 4](#).
- Step 3** From the Connection Type pull-down selection list, select **SIP Trunk**, as shown in [Figure 4](#), and then click **Next** to add the device for the SIP trunk.

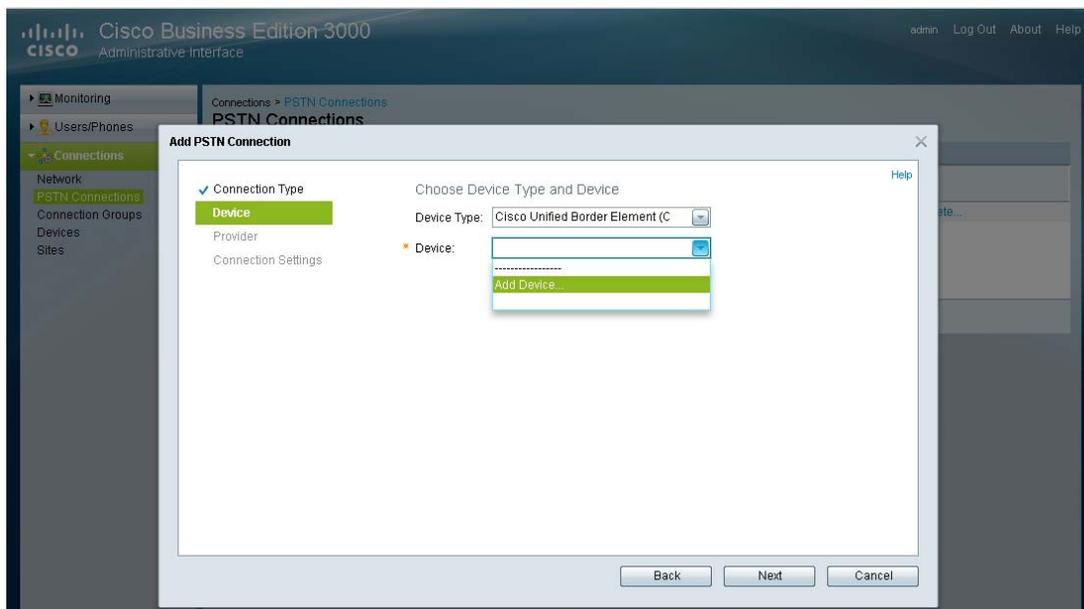
Figure 4 Adding a SIP Trunk under the PSTN Connections Section



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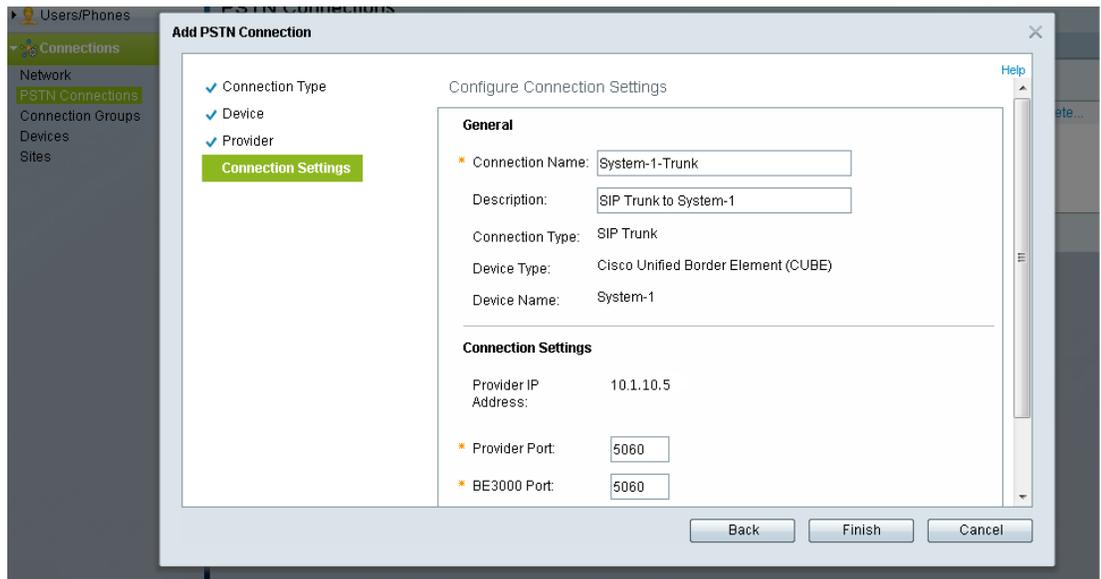
- Step 4** The next configuration section is Device. Although the destination of the SIP trunk is another BE3000 system, select the Device Type as **Cisco Unified Border Element**, and for Device, select **Add Device**, as shown in [Figure 5](#).
- Step 5** Click **Next**.

Figure 5 Adding a Device for the SIP Trunk



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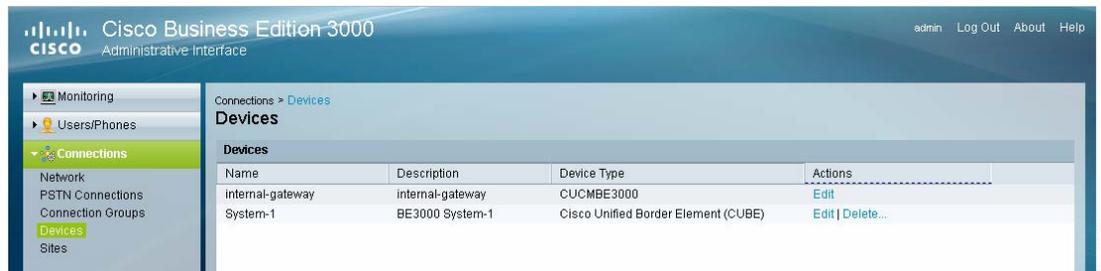
Figure 7 *Configuring Connection Settings with Name and Description of SIP Trunk*



This completes the provisioning of one SIP trunk to a remote Cisco BE3000 system.

- Step 11** Optionally, you may want to verify that the remote BE3000 is listed in the Device section as a Cisco Unified Border Element (CUBE) by clicking **Connections > Devices**, as shown in [Figure 8](#).

Figure 8 *Verifying that the Remote BE3000 is Listed as a Device*



You have to provision SIP trunks on all BE3000 systems toward other remaining BE3000 systems. For example, this document uses three BE3000 systems; therefore, every BE3000 system is provisioned with two SIP trunks toward the other two BE3000 systems. [Figure 9](#) shows two SIP trunks to two remote BE3000 systems.

Figure 9 Listing of SIP Trunks and External PSTN Connection

In the next section, you provision the connection groups for SIP trunks and the PSTN connection.

Provisioning Connection Groups

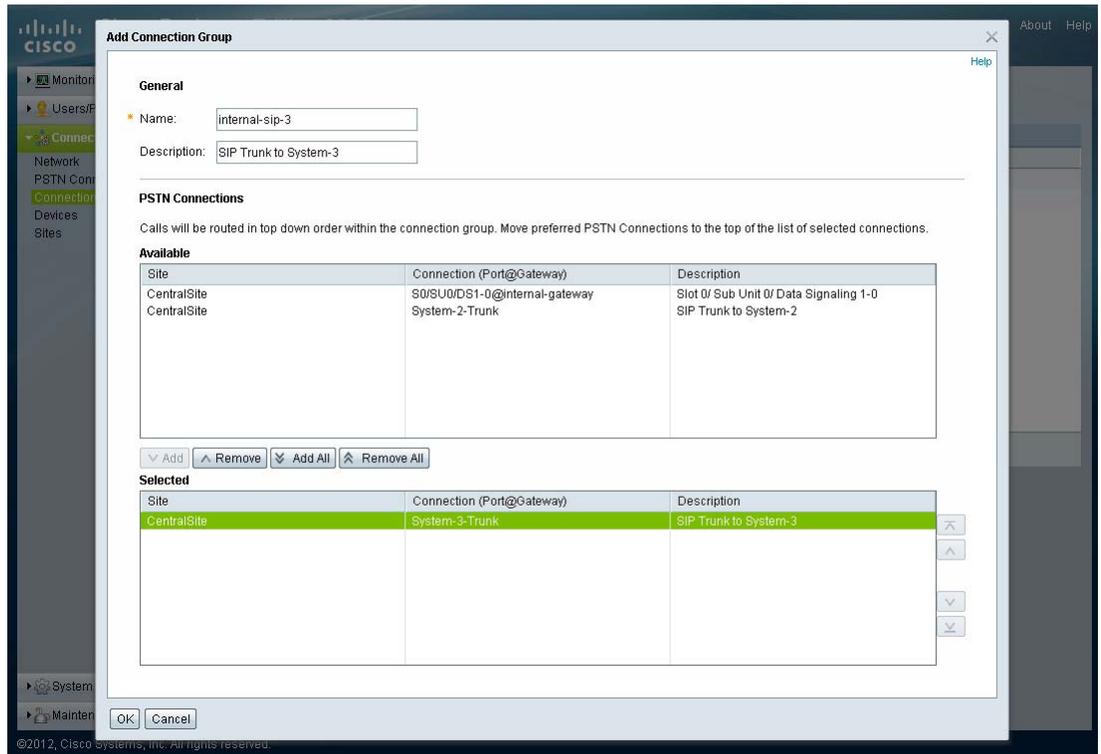
Connection groups are collections of one or multiple trunks (SIP, PRI, analog, and so on) provisioned on the BE3000 server. The default connection group of BE3000 is “All Connections”, consisting of all the trunks configured on BE3000. A connection group is selected for a specific dial pattern of a called number, and calls are routed in top-down order of the trunks in the group. One default connection group is sufficient until all external calls of a BE3000 system are destined to PSTN. However, the multi-system BE3000 solution requires separate a connection group for every other system, as well as for external PSTN calls.

To configure a connection group, complete the following steps.

Procedure

- Step 1** Select **Connections > Connection Groups**, and click **Add Connection Group...**
The Add Connection Groups window appears, as shown in [Figure 10](#).
- Step 2** Type a meaningful Name and Description for the connection group for the desired remote BE3000 system. In the Available section, select the SIP trunk configured for this remote BE3000 system, and click **Add** to move the SIP trunk in the Selected section, as shown in [Figure 10](#). Click **OK** to complete the configuration and create the connection group.

Figure 10 Adding a Connection Group for a Remote BE3000 System



You must configure connection groups for the other remote BE3000 systems as well.



Note

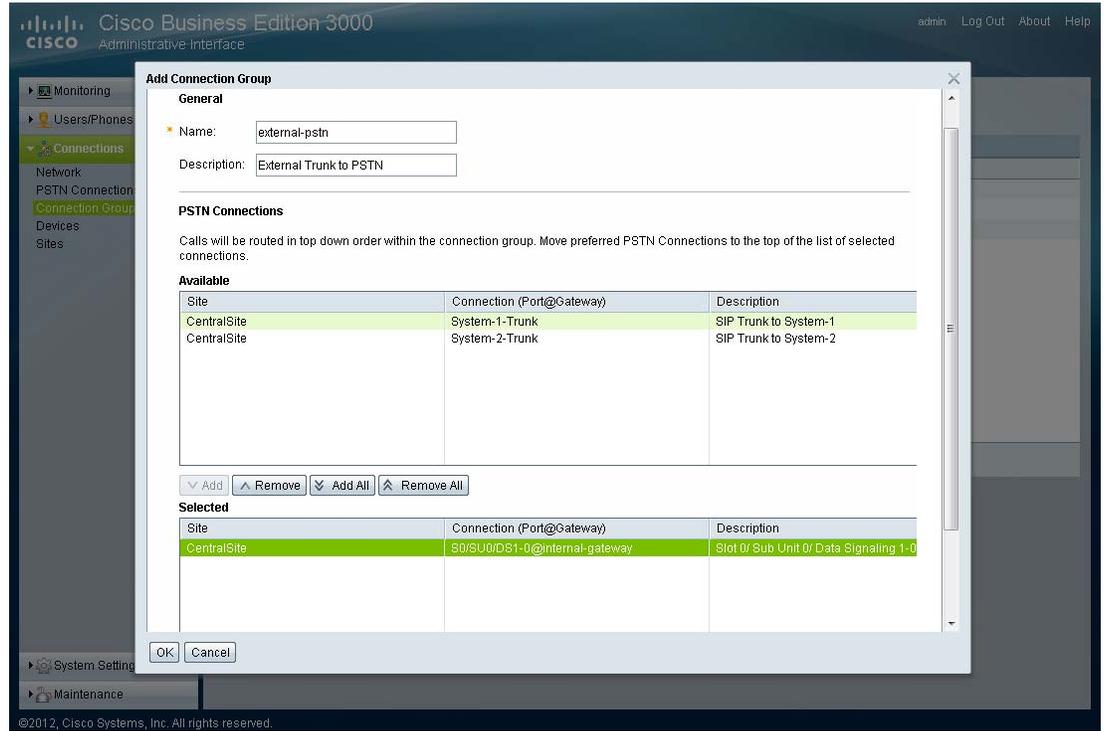
As a best practice, Cisco strongly recommends a separate connection group for external calls destined to PSTN. See the next step.

Step 3

Configure a connection group for external calls destined to the PSTN by selecting all external PSTN trunks, as shown in [Figure 11](#).

In this example, a built-in PRI on the internal gateway is shown; however, if available, select external SIP and analog FXO trunks also.

Figure 11 Adding a Connection Group for External PSTN Calls



Step 4 Select **Connections > Connection Groups** and verify that you have configured connection groups for both internal (to all remote BE3000 systems) and for external PSTN calls, as shown in [Figure 12](#).

Figure 12 Verifying the Connection Groups for Both Internal and External Calls



In the next section, you will edit the dial plan to select the best connection group for all types of dial patterns.

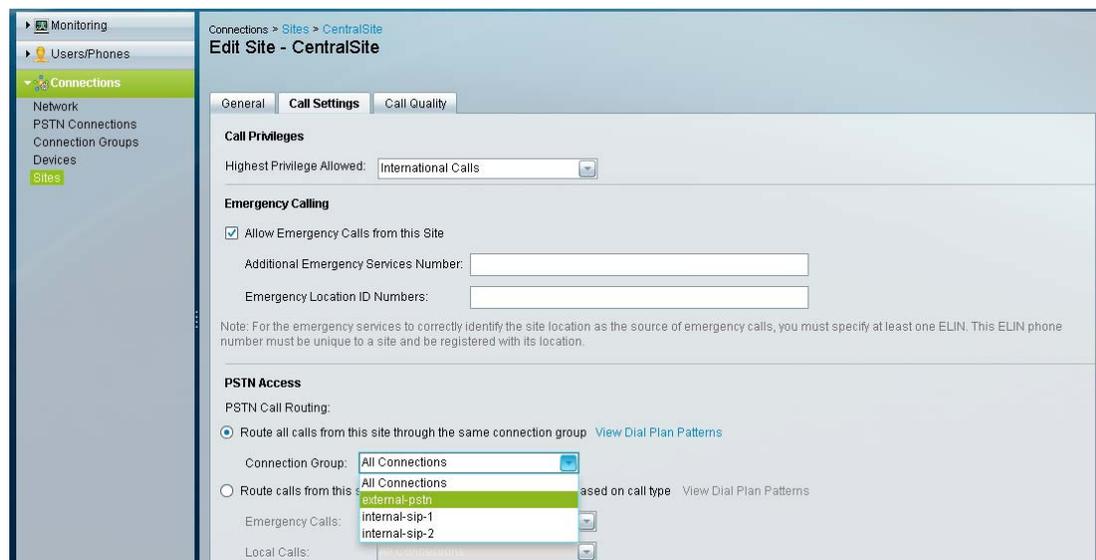
Editing the Dial Plan

This solution requires using an open dial plan, which has some limitations and is complex to configure. For more details, see the BE3000 dial plan whitepaper, which appears during the configuration. This is a very important step in provisioning the multi-system BE3000 solution. To make sure that the dial pattern of every call goes through the correct SIP trunk by selecting the correct connection group, you need to carefully edit the dial plan of the BE3000 as shown in the following steps.

Procedure

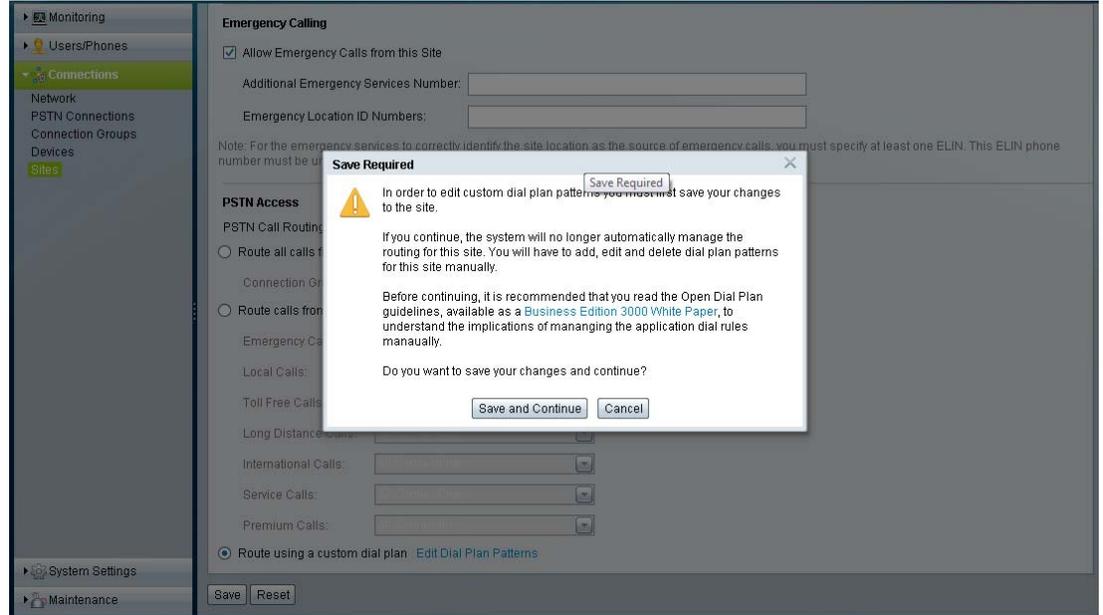
- Step 1** Change the default connection group from **All Connections** to the connection group created for external PSTN calls in the previous section. Select **Connections > Sites** to edit settings for *CentralSite*. Then select the **Call Settings** tab to edit the dial plan settings, as shown in [Figure 13](#).
- Step 2** In the PSTN Access section, leave the default first radio button “Route all calls from this site through the same connection group” selected for PSTN Call Routing. Then select the newly created connection group for external PSTN calls, as shown in [Figure 13](#).
- Step 3** Click **Save**, and continue through the warning windows.

Figure 13 Changing the Default Connection Group from All Connections to New Connection Group for PSTN Calls



Applying this change of connection groups for all calls takes a few minutes to populate the database in the server. Wait for the process to complete before moving to the next step.

- Step 4** To make sure you see the new connection group for external PSTN calls as the default connection group to route all calls out from the site, again select **Connections > Sites**. You can also confirm it by clicking the **View Dial Plan Patterns** link. Make sure you see the new connection group selected in previous step (*RL-external-pstn* for the example in this document), and NOT the *RL-All Connections*.
- Step 5** If verified, select the last radio button **Route using custom dial plan** and click **Save**, followed by **Save and Continue** in the Save Required pop-up window, as shown in [Figure 14](#).

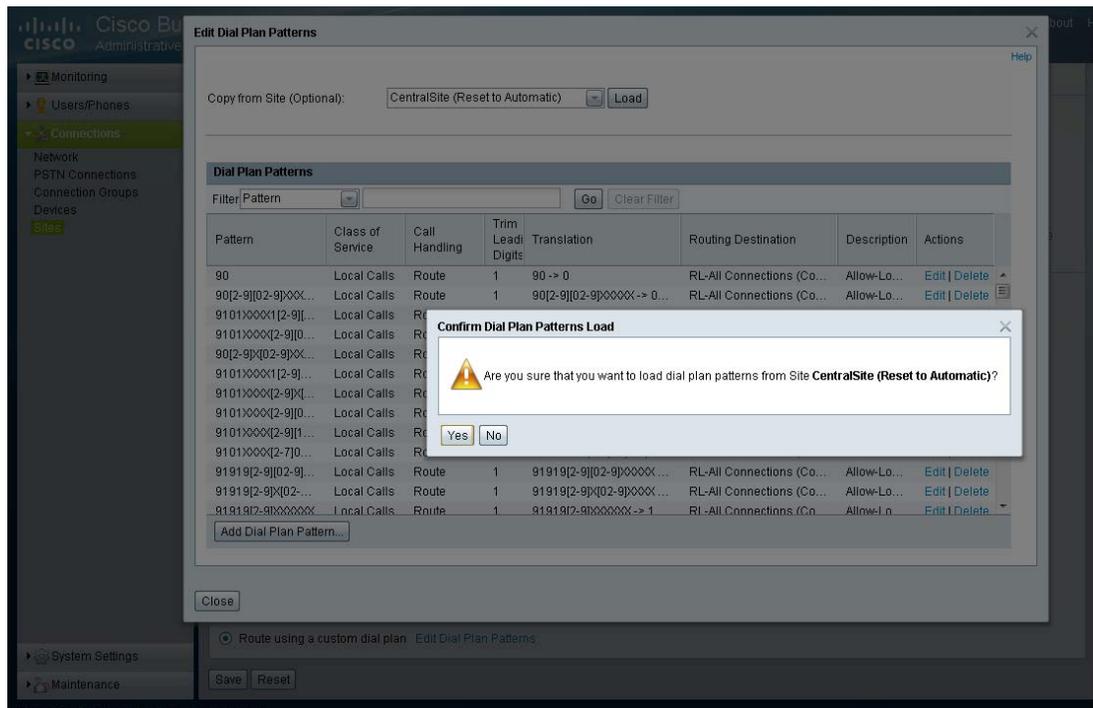
Figure 14 Changing Call Routing to Use Custom Dial Plan

Wait again for the server to save the new settings before moving to the very important next step.

Step 6 Click **Edit Dial Plan Pattern**.

A pop-up configuration window appears, where you observe that the incorrect connection group *RL-All Connections* is listed as the Routing Destination for all patterns. On the top of this pop-up configuration window, for Copy from Site (Optional), select the default **CentralSite (Reset to Automatic)** and click **Load**. Click **Yes** in the **Confirm Dial Plan Pattern Load** warning window, as shown in [Figure 15](#).

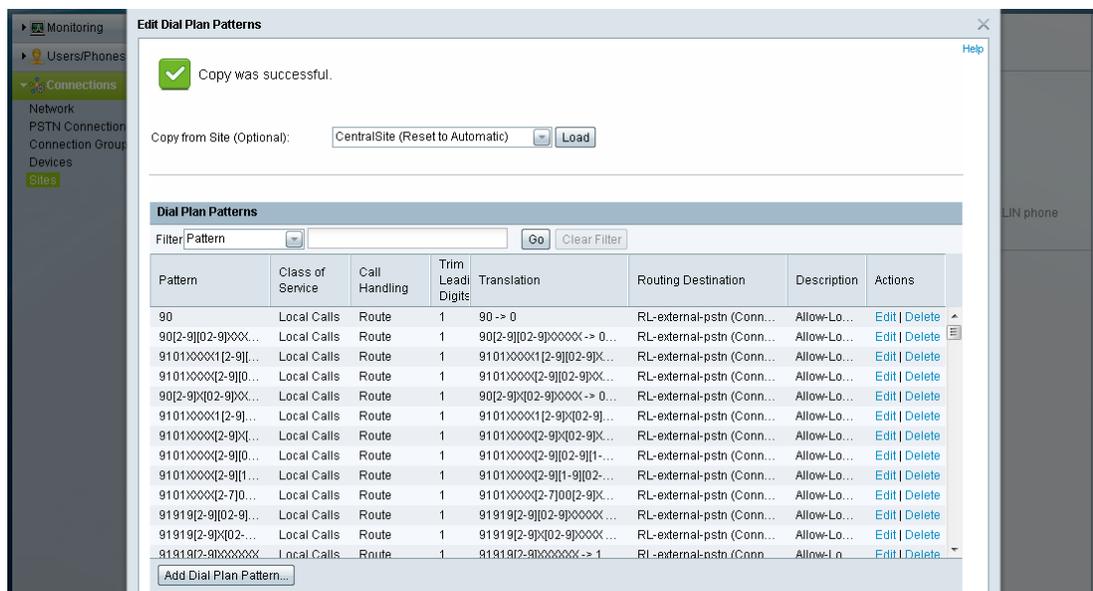
Figure 15 *Editing Dial Plan Patterns after Loading the Previously Configured Patterns from CentralSite*



You see a progress bar until the server copies the site patterns. It may take several minutes

- Step 7** Verify that after the copying process is successful, you see the new connection group (*RL-external-pstn* in this example) as the routing destination, as shown in [Figure 16](#).

Figure 16 *Copying the Dial Plan from the Site with Correct PSTN Connection Group*



Step 8 Click **Add Dial Plan Pattern**.

The Add Dial Plan Pattern pop-up configuration window appears.

Step 9 In the Pattern field, type the destination site's dial pattern, which is a combination of the system code (such as 81, 82, 83, and so on) and the user extensions range (such as 5XXX, 4XXX, 1XXX, and so on).

This example uses 815XXX as the pattern.

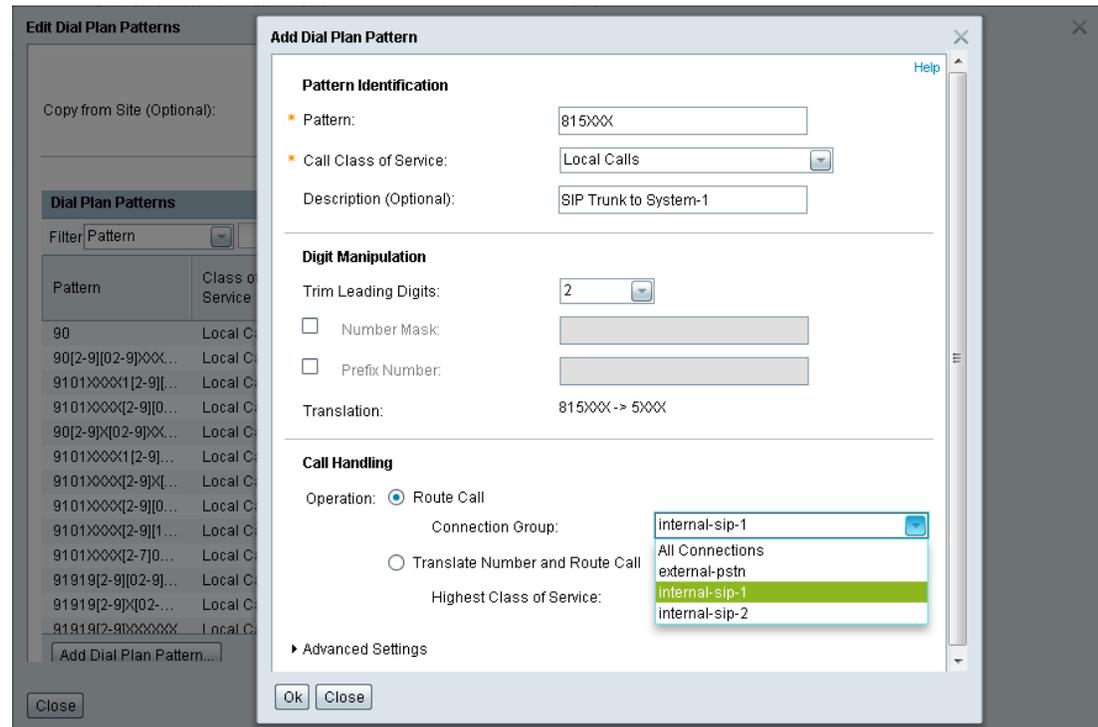
Step 10 Select **Call Class of Service** as Local Calls.**Step 11** In the Digit Manipulation section, select **2** for Trim Leading Digits.

Notice that the dynamic translation appears on screen as 815XXX → 5XXX.

Step 12 In the Call Handling section, select the **Route Call** radio button, and select the appropriate **Connection Group**.

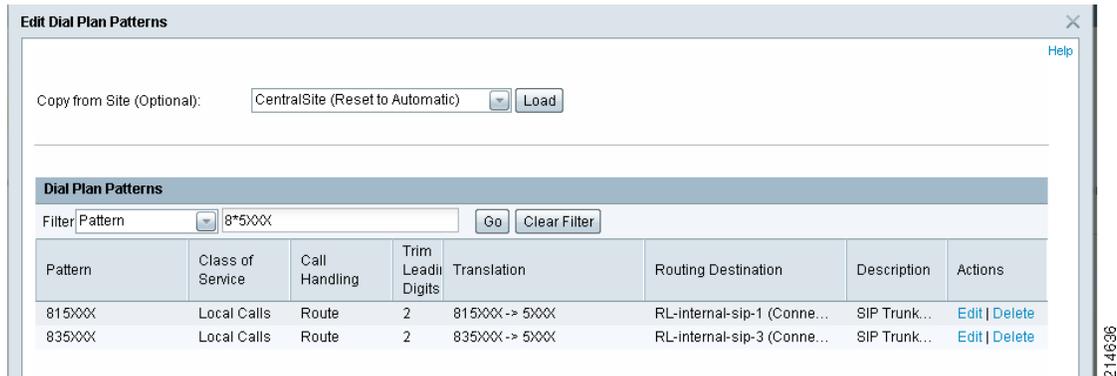
In this example, 81 is site code for System-1, for which the connection group *external-sip-1* was created; therefore, in this example, *internal-sip-1* is selected as the connection group, as shown in Figure 17.

Figure 17 Adding a Dial Plan Pattern for a Remote BE3000 System



After this configuration is applied, any call with the dial pattern 815XXX is redirected to the connection group *internal-sip-1*, after trimming the first two digits, which are the system code for BE3000 System-1. The translated dial pattern 5XXX is then received across the SIP trunk by BE3000 System-1, which dials the user extension in the extension range 5XXX.

Step 13 Verify that dial plan patterns exist for both remote BE3000 systems. You can search these patterns (such as 815XXX, 825XXX, 835XXX, and so on) by searching a regular expression (such as 8*5XXX), and clicking the **Go** button, as shown in Figure 18.

Figure 18 Searching and Listing the Dial Plan Patterns for Remote BE3000 Systems**Note**

There can be more than one dial plan pattern for one remote BE3000 system (such as 815XXX, 813XXX, 814XXX, and so on) if it has more than one extension range for users. This example assumes only one user extension range.

Step 14

You must repeat this configuration on all BE3000 systems to create SIP trunks, connection groups, and add dial plan patterns for other BE3000 systems. This then completes the configuration of the multi-system BE3000 solution.

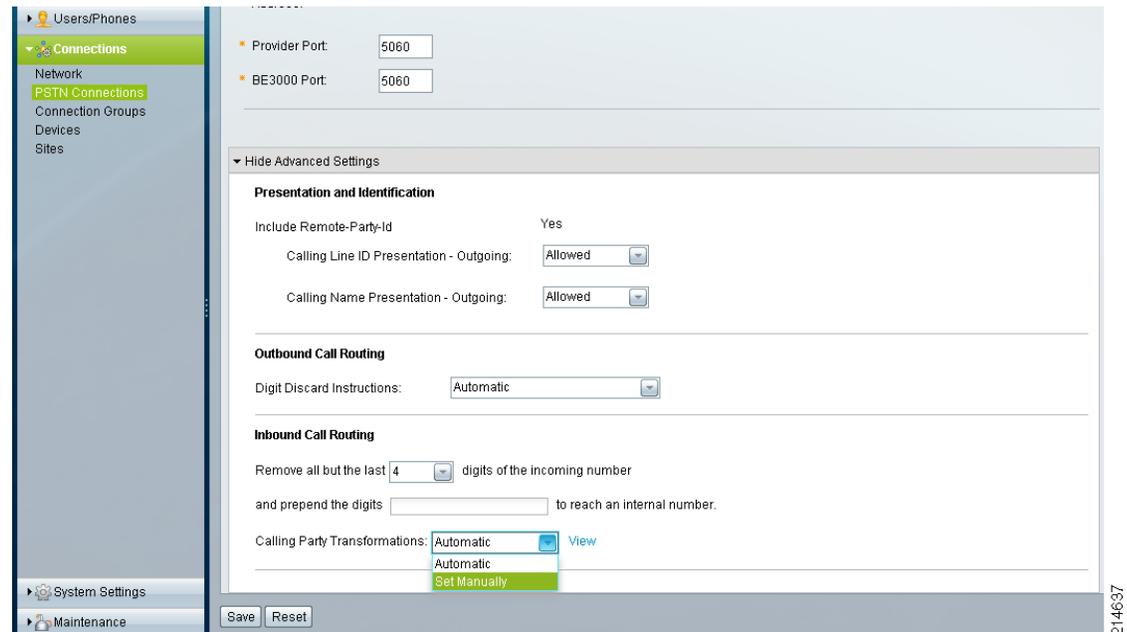
The next section is optional, but it adds the convenience of callback by adding a system code in the caller-id. Because as of version 8.6.4, the caller name is not displayed for inter-system calls, the called party can recognize that it is an internal call coming from a remote business location by seeing the system code in the caller-id field.

Caller-ID Enhancement (Optional)

This is an optional configuration for enhancing the caller-id of inter-system calls, so that the called party can recognize the call as internal, and is also able to call back without manually dialing the number. Complete the following steps.

Procedure

- Step 1** Select **Connections > PSTN Connections**, identify the SIP trunk to the remote BE3000 system, and click **Edit** listed under the Action column. (See [Figure 9](#).)
- Step 2** In the Inbound Call Routing section, leave the first two settings as default, but change the Calling Party Transformations setting from default Automatic to **Set Manually**, as shown in [Figure 19](#).
After selecting Set Manually, the link name changes from View to Edit.
- Step 3** Click **Save** and continue through all the warning windows.

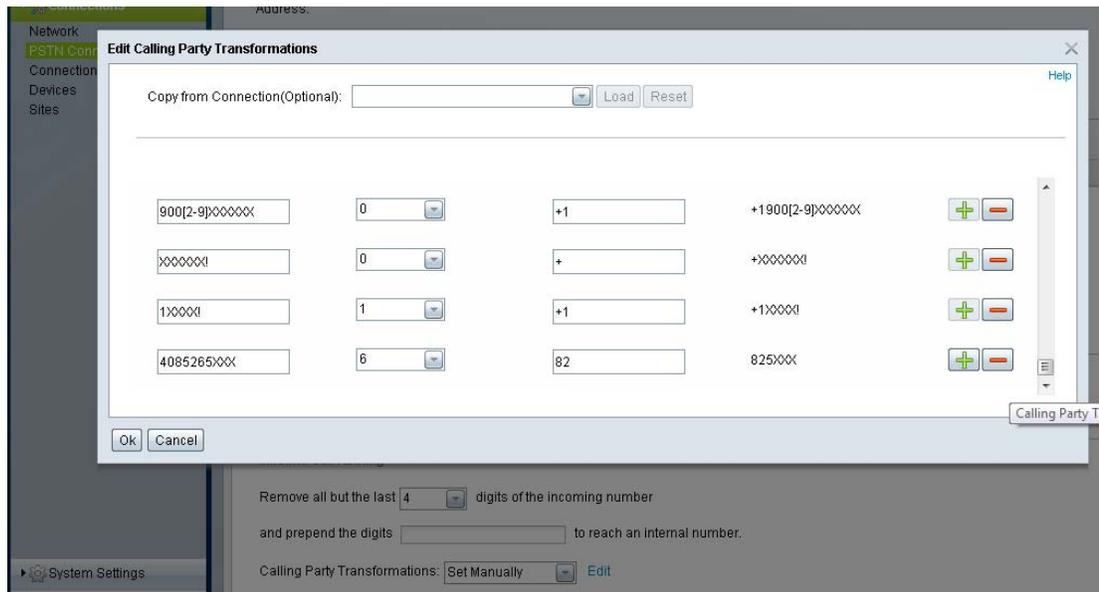
Figure 19 Changing the Inbound Calling Party Transformation to Set Manually

After selecting Set Manually, the link name changes from View to Edit. Compare [Figure 19](#) and [Figure 20](#).

- Step 4** Click **Edit**, and continue through all warnings.
- The Edit Calling Party Transformations pop-up configuration window appears.
- Step 5** Scroll down to the list of Transformations, and click the “+” sign to add new transformation.
- Step 6** In the Inbound Calling Number field, type the full 10-digit main number of remote BE3000 system on this trunk by replacing last four digits as extension range (for example, 4085265XXX, 9728135XXX, or 9194765XXX as used in this document; see [Figure 2](#), and [Figure 3](#)).
- Step 7** In the Remove First Digits field, type **6** to remove all but last four digits, which identify the user extensions of the remote BE3000 system.
- Step 8** In the Prefix with Digits field, type the system code of the remote BE3000 system (this example uses 82 for the remote BE3000 system that has area code 408).

The Displayed on Phone (Caller ID) field is automatically changed to the dial plan pattern of the remote BE3000 system (such as 815XXX, 825XXX, 835XXX, and so on), as shown in [Figure 20](#).

Figure 20 *Editing Calling Party Transformation to Include System Code in Caller-ID*



- Step 9** Click **Ok** to apply the configuration and close the window.
- Step 10** Click **Save** in the Edit PSTN Connection Setting window.
- Step 11** Repeat this configuration on all inter-system SIP trunks on all BE3000 systems, so that all users in this multi-system BE3000 solution have a similar experience.

Always use a diagram such as [Figure 2](#) with actual area codes, full 10-digit main business numbers, extension ranges, and assigned system codes.

Testing the Inter-System SIP Trunk

To perform basic testing and verification of the inter-system SIP trunk between multiple Cisco BE3000 systems, complete the following steps.

Procedure

- Step 1** Make an inter-system call to a user extension in one BE3000 system, from an IP phone in the another BE3000 system. Perform call hold, call transfer, and AdHoc conferencing.
- Step 2** From an IP phone on a remote BE3000 system, call the AA/VoiceMail number with the inter-system dial plan pattern. The call should be answered by the auto-attendant.
- Step 3** Verify the Caller-ID for inter-system calls. Make sure it is a 6-digit number with a 2-digit system code, and that the called party can call back without manually dialing the digits.

- Step 4** Make sure that external calls via the PSTN trunk on all BE3000 systems are not affected by this configuration. Place inbound and outbound external calls across all PSTN trunks. Perform call hold, call transfer, and conferencing. Test that the main business number is redirected accordingly to AA, and that AA is able to redirect calls to extensions.
-

For More Information

- Cisco SMART Designs for SMB UC Solutions (including the Implementation Guide for BE3000)—<http://www.cisco.com/go/smartdesigns/uc>
- Administration Guide for Cisco Business Edition 3000 version 8.6.4—http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe3k/8_6_4/system/BE3K_BK_C30C50F6_00_cucm-be3000-admin-guide.pdf
- Cisco Small Business support community—<http://www.cisco.com/go/smallbizsupport>

