

Optimum Business Trunking and the Cisco Manager Express Version 8.5 Configuration Guide

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1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Cisco PBX for proper operation Optimum Business Sip Trunking.

2 SIP Trunk Adaptor Set-up Instructions

These instructions describe the steps needed to configure the LAN side of the Optimum Business SIP Trunk Adaptor.

Step 1:

Log on to the Optimum Business SIP Trunk Adaptor

1. Connect a PC to port 4 of the Optimum Business SIP Trunk Adaptor, the silver device labeled Edgewater Networks, 4550 series.



2. Open a Web browser and go to IP Address <http://10.10.200.1>. A login box will appear.
3. Enter login and password and click 'OK':
Login: pbxinstall
Password: s1ptrunk



Step 2:

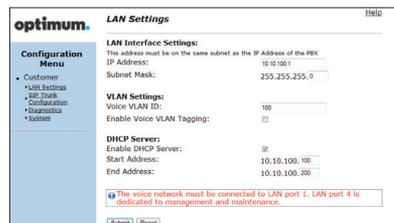
Click on the LAN Settings Link

1. Assign an IP Address to the LAN interface of the SIP Trunk Adaptor. The IP address must be on the same subnet as the IP PBX. This changes the address on port 1 of the Optimum Business SIP Trunk Adaptor.

Note: This will become your local SIP proxy IP address. No other IP addresses will be provided by Cablevision.

2. Optional: Specify a VLAN for your voice traffic. Click the 'Enable Voice VLAN Tagging' check box. The default VLAN ID is 100.

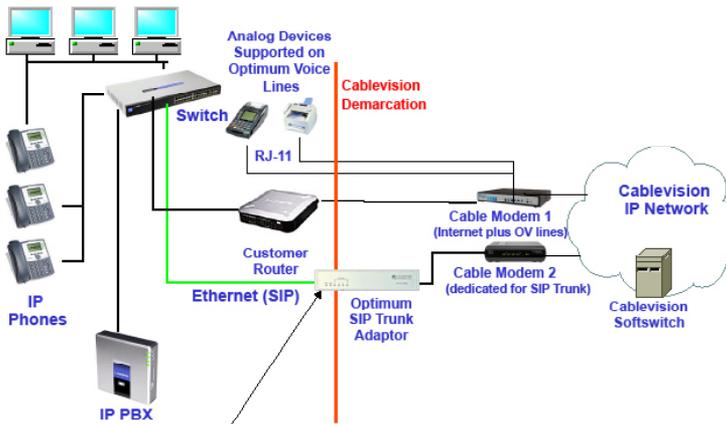
Note: VLAN 200 should not be used. It is dedicated to port 4 for management.



3. Optional: Enable the DHCP server. This will allow the SIP Trunk Adaptor to act as a DHCP server, which will provide IP addresses to the voice network, and create a dedicated voice LAN, as per diagram 2.

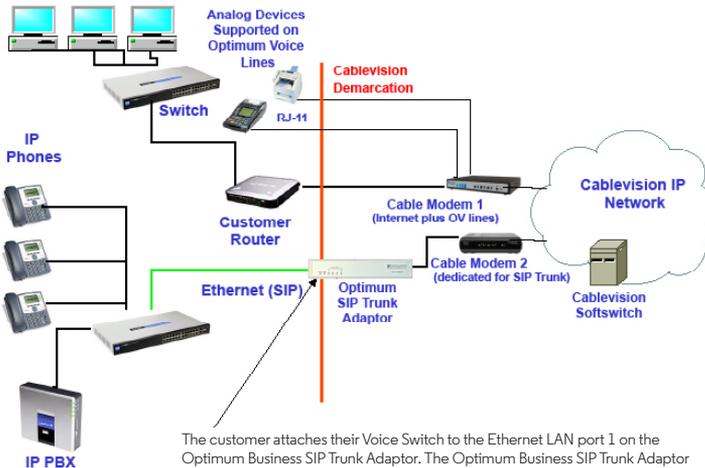
4. Click 'Submit'.

Diagram 1 SIP Trunk Adaptor for IP-PBXs Example: Single LAN Configuration



Using a connection from the customer's LAN, the SIP Trunk Adaptor's address can be a statically assigned private IP address. It may not be assigned a Public IP address.

Diagram 2 SIP Trunk Adaptor for IP-PBXs Example: Separate Voice and Data Networks Configuration

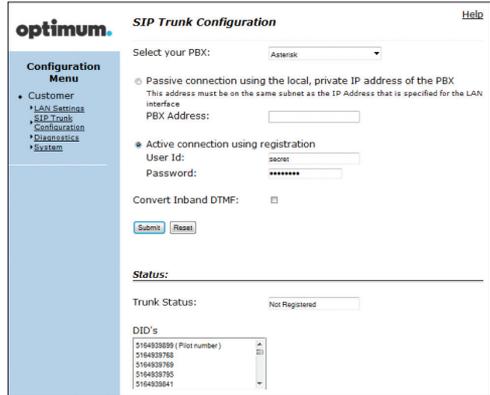


The customer attaches their Voice Switch to the Ethernet LAN port 1 on the Optimum Business SIP Trunk Adaptor. The Optimum Business SIP Trunk Adaptor can be enabled as a DHCP server to provide routing for the separate voice network.

Step 3:

Click on the SIP Trunk Configuration Link

1. Select your IP PBX make and model from the drop-down menu.
2. Specify how the IP PBX will register to the Optimum Business SIP Trunk Adaptor.
3. The Cablevision network only supports Inband DTMF. Click on the check box next to "Convert Inband DTMF" if you cannot configure your IP PBX to send out Inband DTMF. The DTMF tone duration generated by the phones and/or PBX may need to be increased from their default setting. Some phones and/or PBX have a default setting between 180ms to 200ms. This setting is too low. The recommended setting is 600ms.
4. Click 'Submit'.

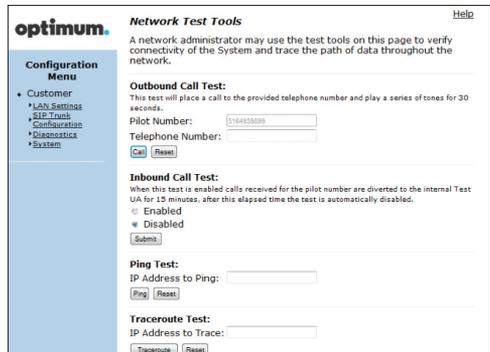


The screenshot shows the 'SIP Trunk Configuration' page. On the left is a 'Configuration Menu' with links for Customer, LAN Settings, SIP Trunk Configuration, Diagnostics, and System. The main content area includes a 'Select your PBX:' dropdown menu set to 'Asterisk'. Below this are two radio button options: 'Passive connection using the local, private IP address of the PBX interface' and 'Active connection using registration'. The 'Active connection' option is selected, showing fields for 'User ID:' (set to 'secret') and 'Password:' (masked with asterisks). There is a checkbox for 'Convert Inband DTMF:' which is currently unchecked. 'Submit' and 'Reset' buttons are present. A 'Status:' section shows 'Trunk Status:' as 'Not Registered' and a list of 'DID's' including 5164433889, 5164433763, 5164433769, 5164433765, and 5164433841.

Step 4:

Diagnostics Link

You can make a test call directly from your phone or use the test call application under the Diagnostics link.



The screenshot shows the 'Network Test Tools' page. On the left is the same 'Configuration Menu' as in Step 3. The main content area includes a paragraph explaining that a network administrator can use these tools to verify connectivity. There are three test sections: 'Outbound Call Test:' with fields for 'Pilot Number:' (5164433889) and 'Telephone Number:', and 'Call'/'Reset' buttons; 'Inbound Call Test:' with a 'Submit' button and a note that calls are diverted to an internal test URI for 15 minutes; and 'Ping Test:' with an 'IP Address to Ping:' field and 'Ping'/'Reset' buttons. The 'Traceroute Test:' section has an 'IP Address to Trace:' field and 'Traceroute'/'Reset' buttons.

Step 4 continued

Field	Description
Outbound Call Test TelephoneNumber	Specifies an outside phone number to which an outbound call will be initiated. The pilot telephone number of the SIP Trunk will be prepopulated.
Pilot Number	Displays the provisioned pilot number, which is used for outbound and inbound call tests.
Call	Initiates a call outbound to a telephone number entered or inbound to the pilot number displayed.
Inbound Call Test (radio button)	Indicates whether inbound test call will be enabled or disabled. If inbound test calls are enabled, calls made to the pilot number will be redirected to the test UA for fifteen minutes. When the pilot number is dialed, you will hear a test message play.
Submit	Enables or disables the inbound call test.
IP Address to Ping	Verifies basic connectivity to a networking device. Successful ping test results indicate that both physical and virtual path connections exist between the system and the test IP address.
Ping Button	Sends a ping to the IP address specified in the field "IP Address to Ping".
IP Address to Trace	Tracks the progress of a packet through the network. The packet can be tracked through the WAN or LAN interfaces of the adaptor.
Interface (radio button)	Indicates whether a packet will be tracked through the LAN or the WAN.
Traceroute Button	Initiates a traceroute to the specified IP address on either the LAN or the WAN.
Reset	Clears all fields and selections and allows you to enter new information. Reset applies to outbound call test, ping and traceroute.

3 Additional Set-up Information Systems


System
[Help](#)

Configuration Menu

- Customer
 - ▶ LAN Settings
 - ▶ SIP Trunk Configuration
 - ▶ Diagnostics
 - ▶ System

Software Version:
Version 11.6.14.1 -- Fri Jan 4 17:49:28 PST 2013

Hostname:
5164939899

Model:
EdgeMarc 4552

Vendor:
Cablevision

LAN Interface MAC Address:
A8:70:A5:00:D8:18

Registration Status:
The ALG feature is registered. View [license key](#).

System Date:
02/29/2016 15:03:40 UTC

Change Password:

- [pbxinstall](#)

Field	Description
Pbxinstall Link	Select to change the default password for the pbxinstall login ID. Only the password can be changed. The login ID cannot be changed.

Password


Set Password
[Help](#)

Change the GUI password by filling in the fields below. The password must be between 6 and 8 characters in length.

Configuration Menu

- Customer
 - ▶ LAN Settings
 - ▶ SIP Trunk Configuration
 - ▶ Diagnostics
 - ▶ System

Username:

Current Password:

New Password:

Confirm Password:

Field	Description
Username	Specifies the username for which the password can be changed.
Current Password	Specifies the current password.
New Password	Specifies the new password.
Confirm Password	Confirms the new password.
Submit	Applies the settings configured on this page.
Reset	Clears all fields and selections and allows you to enter new information.

4 International Calling

Optimum Voice Business Trunking offers an optional International Calling Service for direct-dialed calls made from the Customer's business or from any phone via the Optimum Voice International Calling remote access number to destinations outside of the United States, Puerto Rico, Canada and the U.S. Virgin Islands at per minute rates. The Customer must login to the Optimum Business Account Center and activate the service on the Optimum Business Trunk Pilot telephone number to activate the service and manage the monthly International spending limit for the account.

Activating International calling on the Pilot TN will enable International calling for all Direct Inward Dial (DIDs) telephone numbers as well. Blocking International calling for one or more DIDs is managed by the customer directly from the PBX phone system configuration. To minimum the exposure to fraudulent calling, It is recommended to limit International calling capability to those DID's that require it and set up an account spending limit that reflects what is necessary to run the business.

It is the Customer (and/or the Customer Agent's) responsibility to properly secure the customer's PBX to prevent the PBX from being compromised and fraudulent calls from being made by unauthorized (internal or external) users. If fraudulent calls are detected, Cablevision reserves the right to disable International Calling until the PBX is properly secured by the customer.

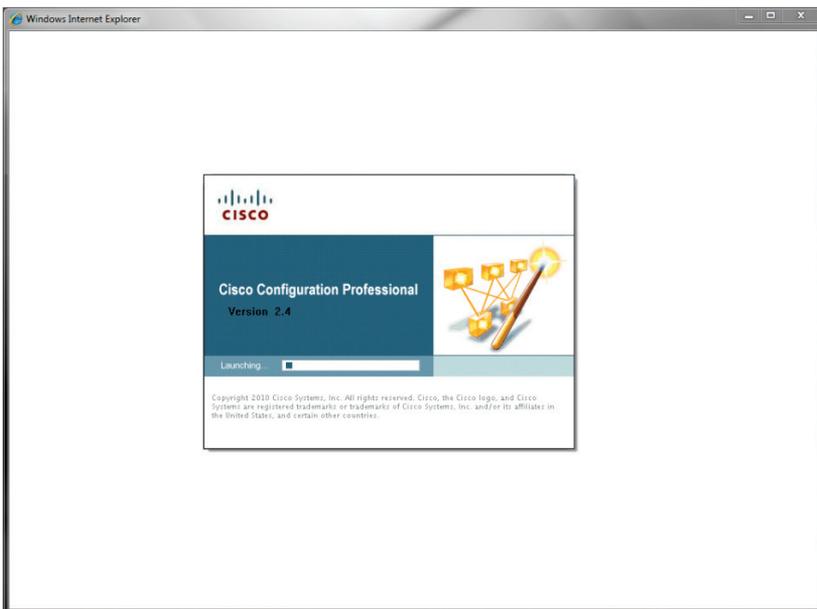
5 PBX Configuration

The steps on the next pages describe the minimum configuration required to enable the CME PBX to use Optimum Business SIP Trunking for inbound and outbound calling. Please refer to the Cisco CME product documentation for more information on other advanced PBX features.

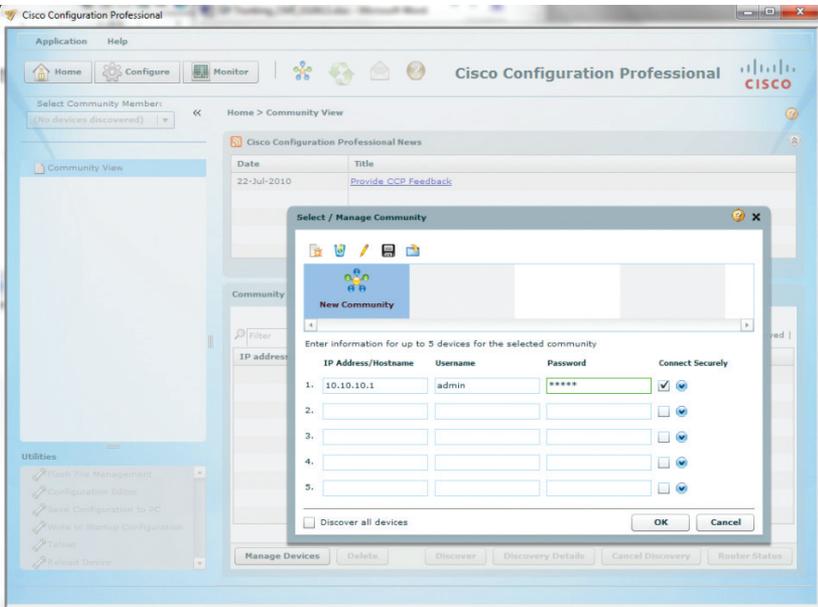
The configuration described here assumes that the PBX is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on Cisco 2901, IOS version 15.1(3)T (c2900-universalk9-mz.SPA.151-3.T.bin) with built-in CME v8.5.

The Cisco 2901 came with two Ethernet ports, GE 0/0 and GE 0/1, with GE 0/0 defaults to an IP address of 10.10.10.1/248. One of these two Ethernet ports and the Optimum Business SIP Trunk Adaptor's LAN port 1 should be in the same LAN segment. The GE 0/0 port and the Optimum Business SIP Trunk Adaptor's LAN port 1 have been assigned with IP addresses of 10.10.10.1/248 and 10.10.10.2/248 respectively. The GE 0/1 port has been assigned with an IP address of 192.168.0.1/24 and it will be used as the PBX's LAN port for communicating with the IP phones, as well as the DHCP server for the IP phones.

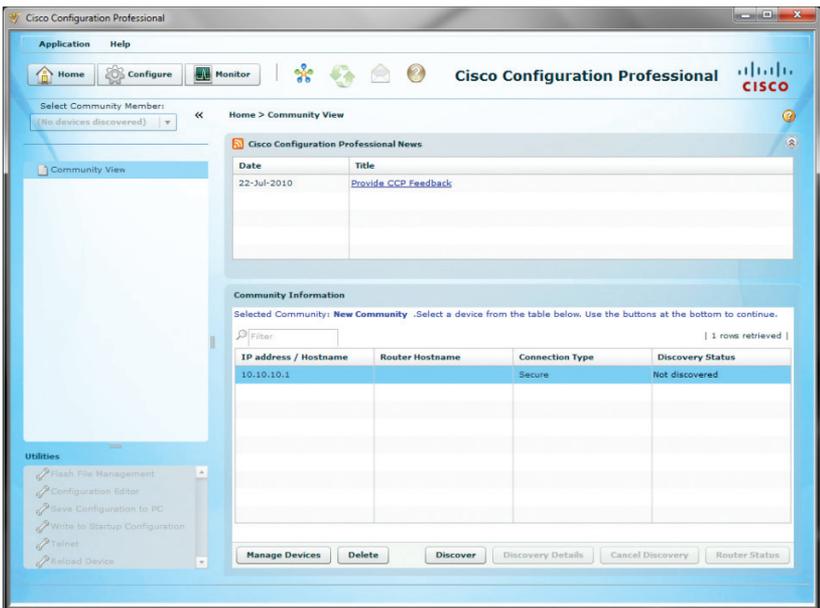
Before you can configure the CME PBX for SIP trunk services, you need to download and install the Cisco Configuration Professional (CCP) GUI on a Windows PC. As of this writing, CME 8.5 and CCP 2.4 are the latest software available. If you have a problem getting CCP 2.4 to work on your Windows PC, check the release notes (i.e.: certain versions of the Java Runtime Environment may require certain settings in order for CCP 2.4 to work). After CCP 2.4 has been successfully installed, connect the Windows PC to the 10.10.10.0/248 LAN segment and start the CCP GUI.



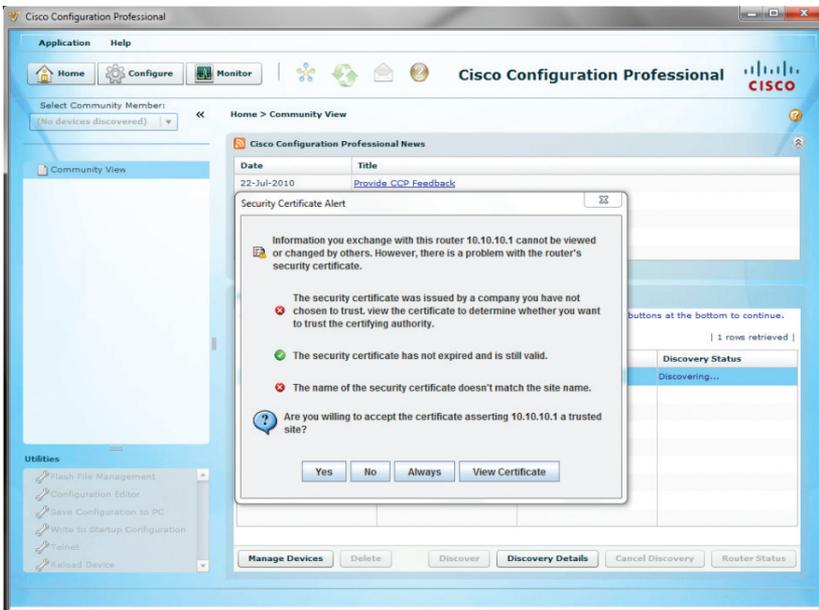
1. From the **“Select/Manage Community”** screen, enter the CME PBX info in the first entry of the devices to be discovered.
 - a. Enter the PBX's IP address in the **“IP Address/Hostname”** field.
 - b. Enter the username in the **“Username”** field.
 - c. Enter the password in the **“Password”** field.
 - d. Check the **checkbox** next to the password field.
 - e. Click the **“OK”** button.



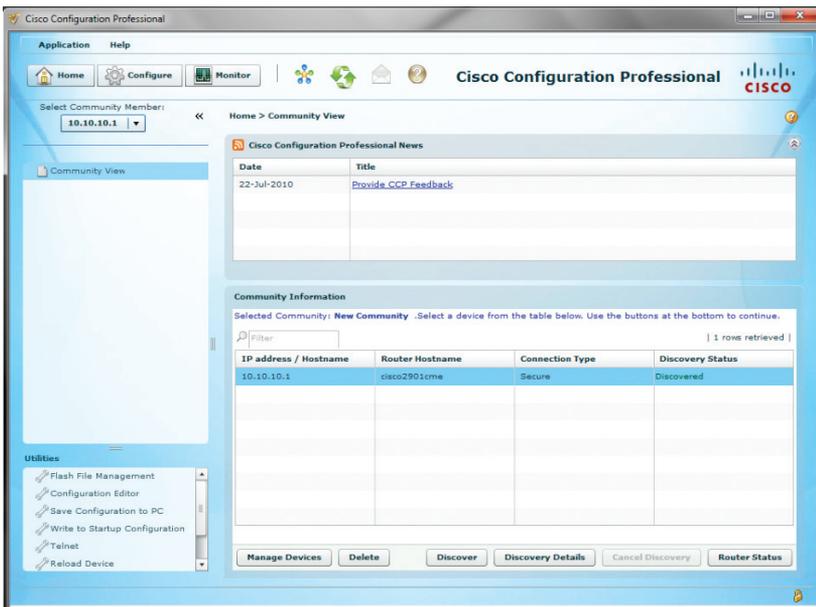
2. Click the **“Discover”** button.



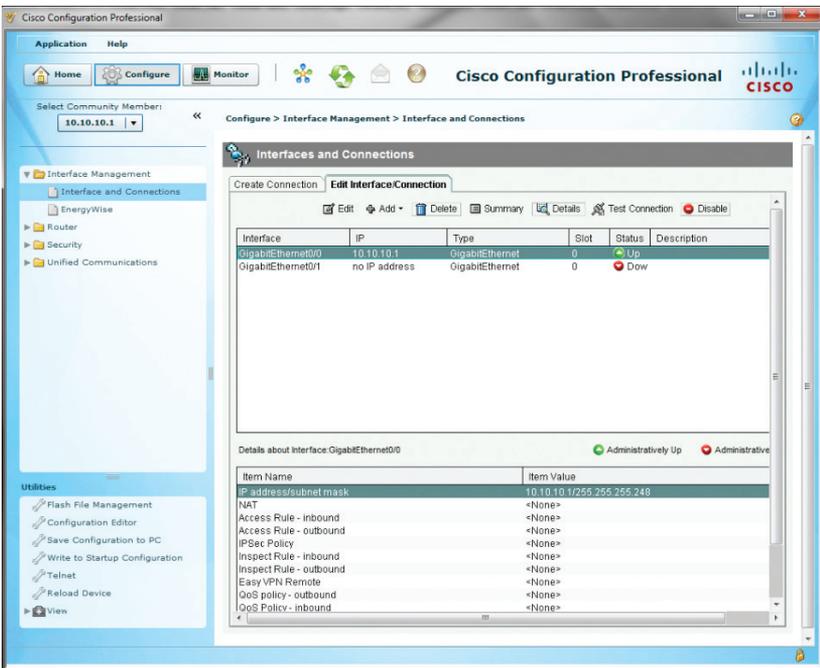
3. Click the **“Yes”** button.



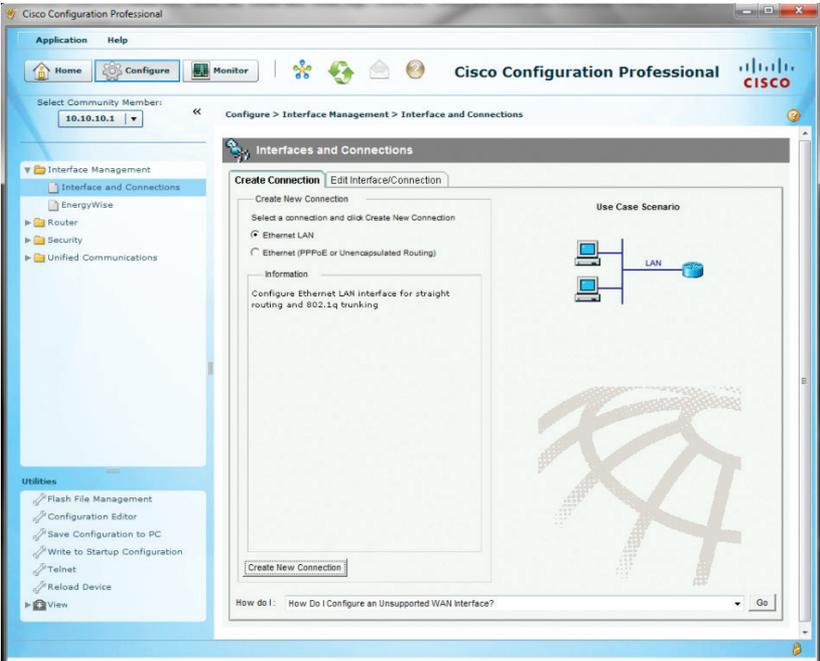
4. CCP has discovered the CME PBX. Click the **“Configure”** button to start configuring the PBX.



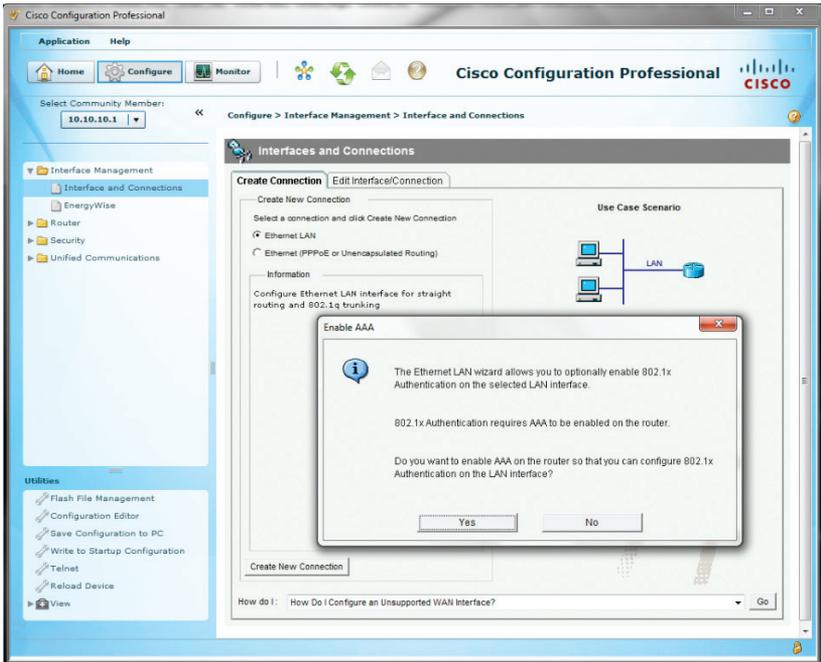
5. Navigate to **“Configure” > “Interface Management” > “Interface and Connections”** to set up the GE 0/1 Ethernet port needed for the PBX to communicate with the IP phones.
 - a. Select the **“Edit Interface/Connection”** tab to make sure the GE 0/1 port is not in use.



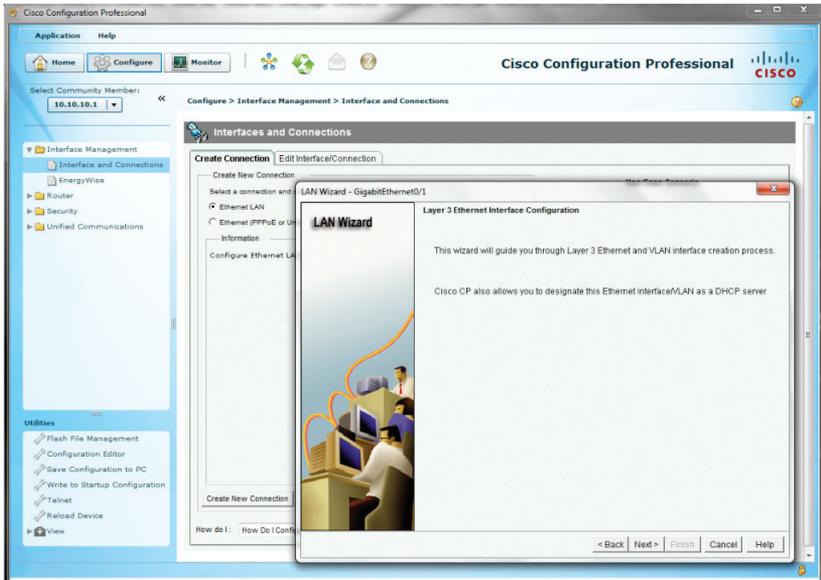
6. Select the **“Create Connection”** tab to set up the GE 0/1 Ethernet port, select the radio button for **“Ethernet LAN”** and then click the **“Create New Connection”** button.



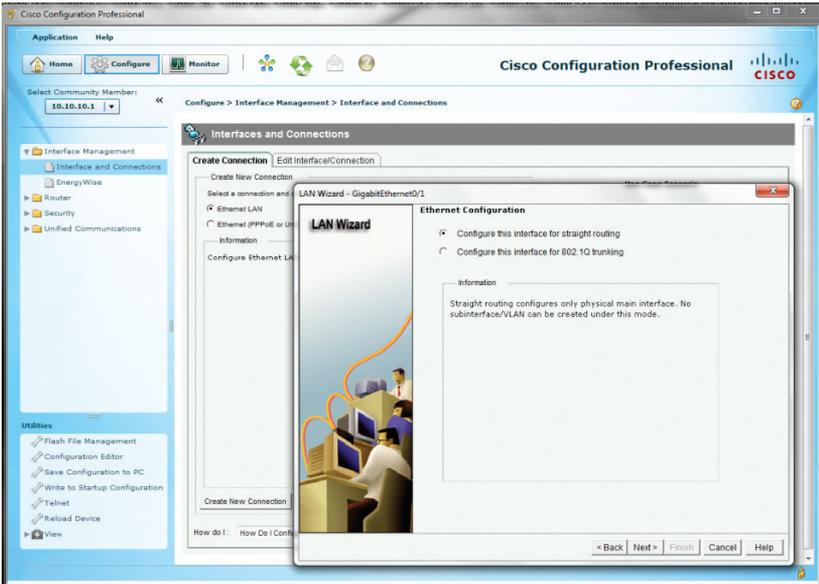
7. Click the **“No”** button from the **“Enable AAA”** screen.



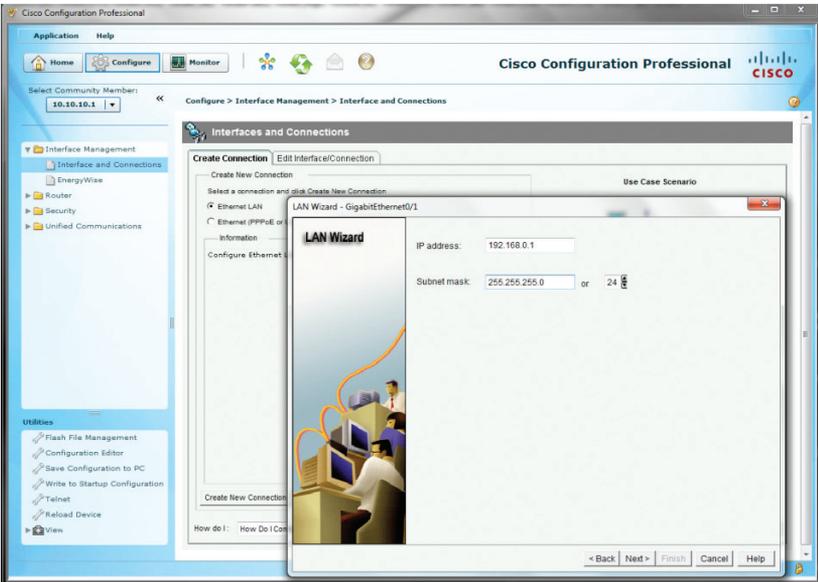
8. Click the **“Next >”** button.



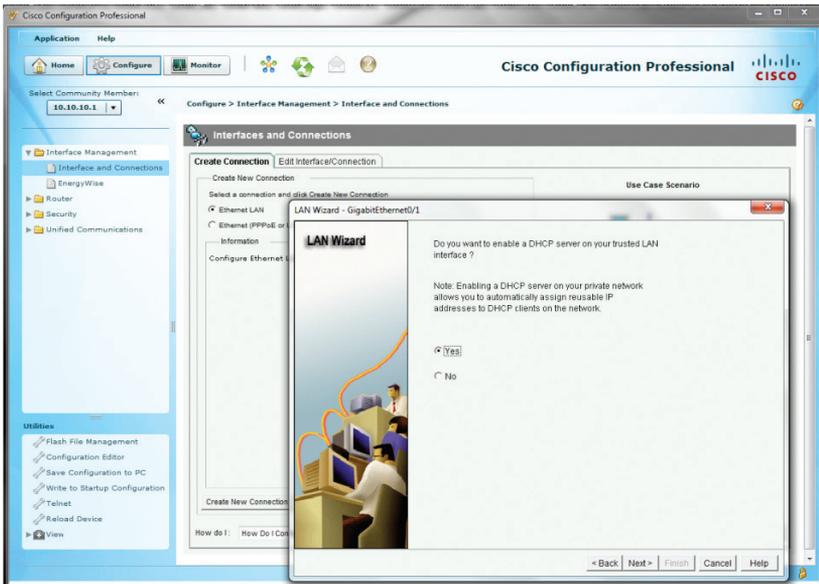
9. Click the **“Next >”** button.



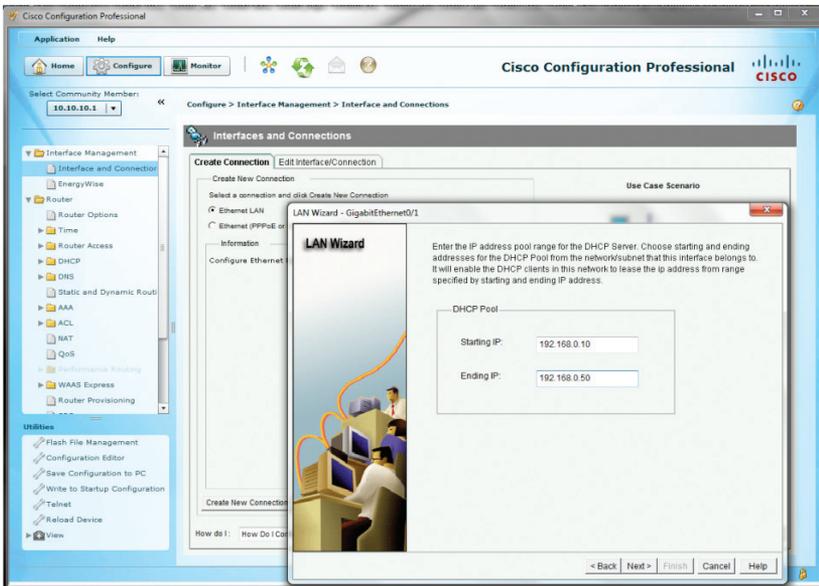
10. Enter the IP address for the GE 0/1 port in the **“IP address”** field, enter the subnet mask in the **“Subnet mask”** field and then click the **“Next >”** button.



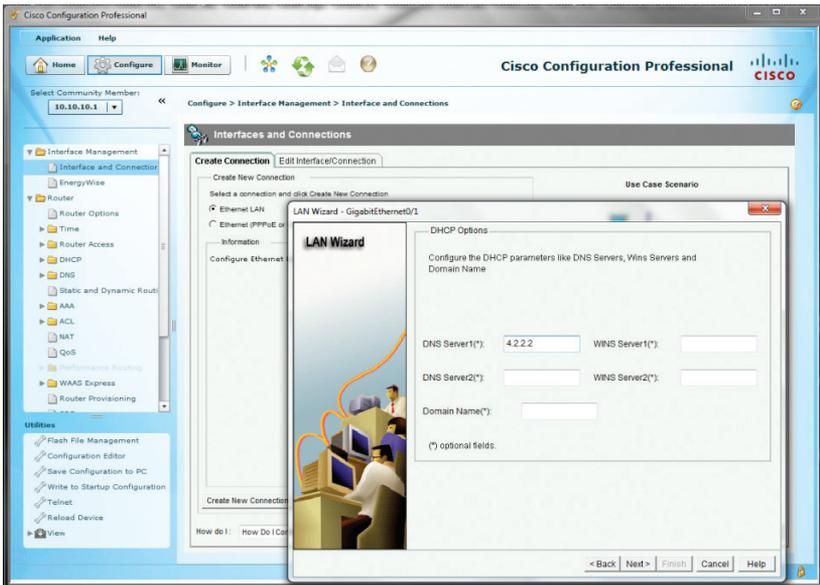
11. Select the radio button for **“Yes”** to enable the DHCP server and then click the **“Next >”** button.



- Enter the starting IP address in the **“Starting IP”** field, enter the ending IP address in the **“Ending IP”** field and then click the **“Next >”** button.



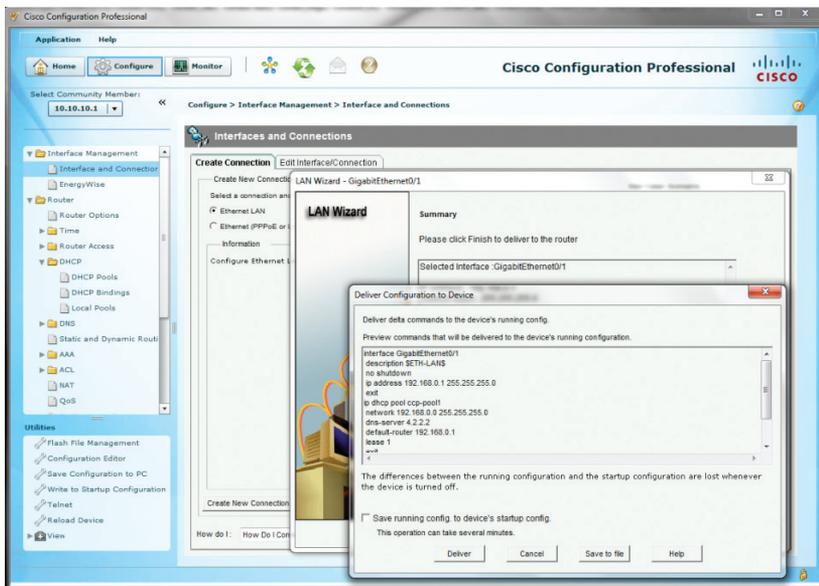
- Enter a valid IP address for the DNS server in the **“DNS Server1”** field, leave other fields as default and then click the **“Next >”** button.



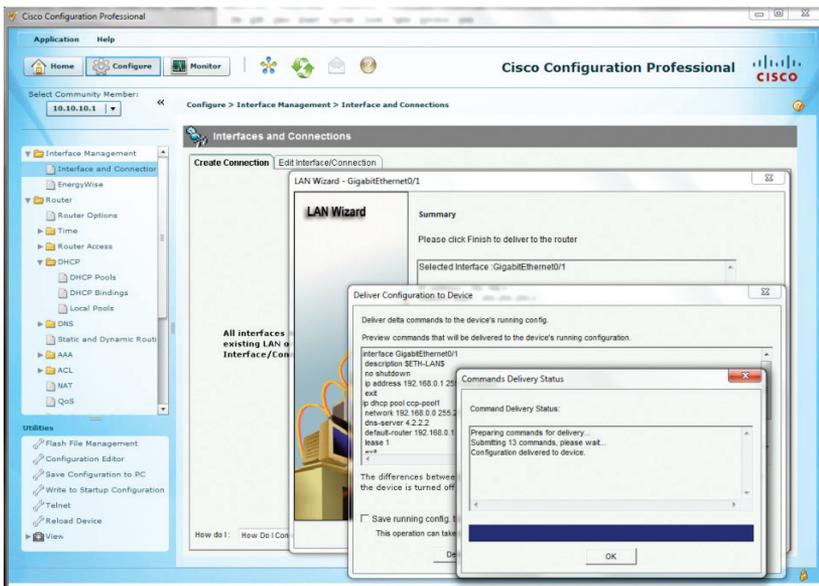
14. Click the **“Finish”** button.



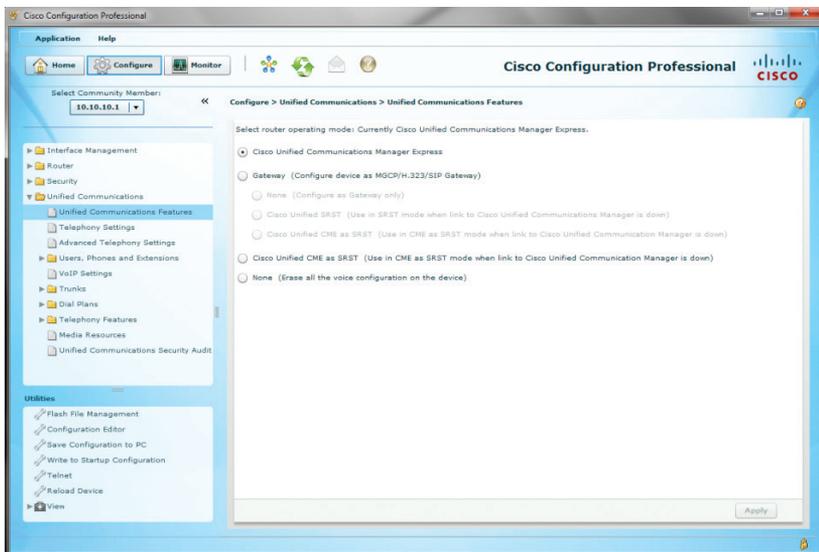
15. Click the **“Deliver”** button.



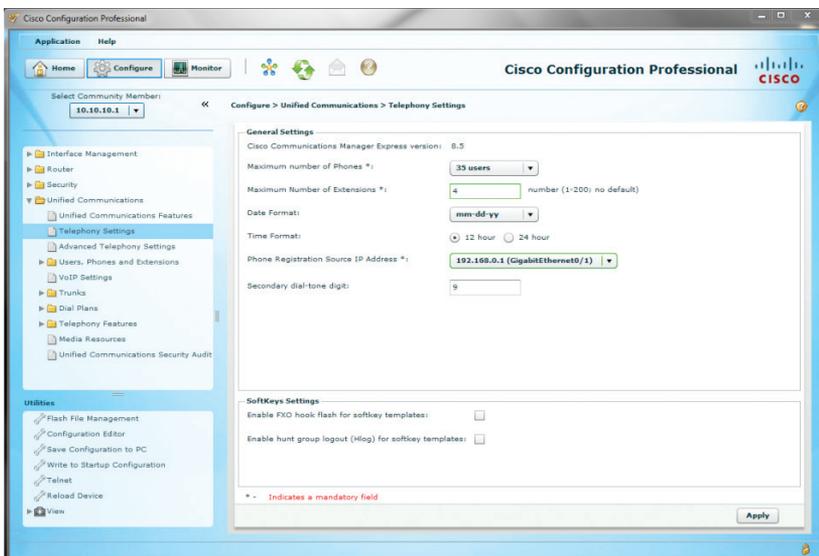
16. Click the **“OK”** button.



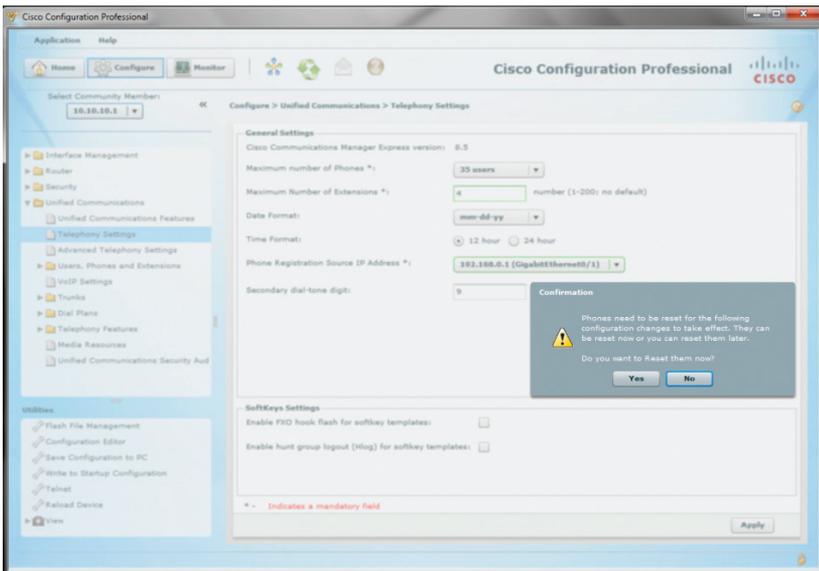
17. Navigate to **“Configure”>“Unified Communications”>“ Unified Communications Features”** to make sure the radio button for **“Cisco Unified Communications Manager Express”** is selected. If not, select it and then click the **“Apply”** button.



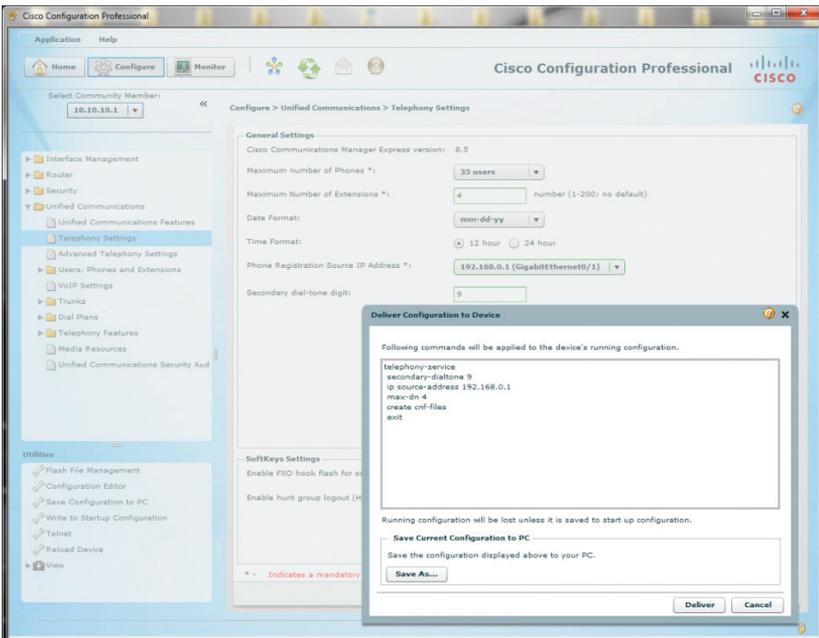
18. Navigate to **“Configure”>“Unified Communications”>“Telephony Settings”** to configure the general settings.
 - a. Select **“35 users”** from the drop-down list of the **“Maximum number of Phones”** field.
 - b. Enter the number of extensions needed in the **“Maximum Number of Extensions”** field.
 - c. Select the date format in the **“Date Format”** field.
 - d. Select the radio button for the **“Time Format”** field.
 - e. Select the IP address for the GE 0/1 port from the drop-down list of the **“Phone Registration Source IP Address”** field.
 - f. Enter **“9”** in the **“Secondary dial-tone digit”** field, leave other fields as default.
 - g. Click the **“Apply”** button.



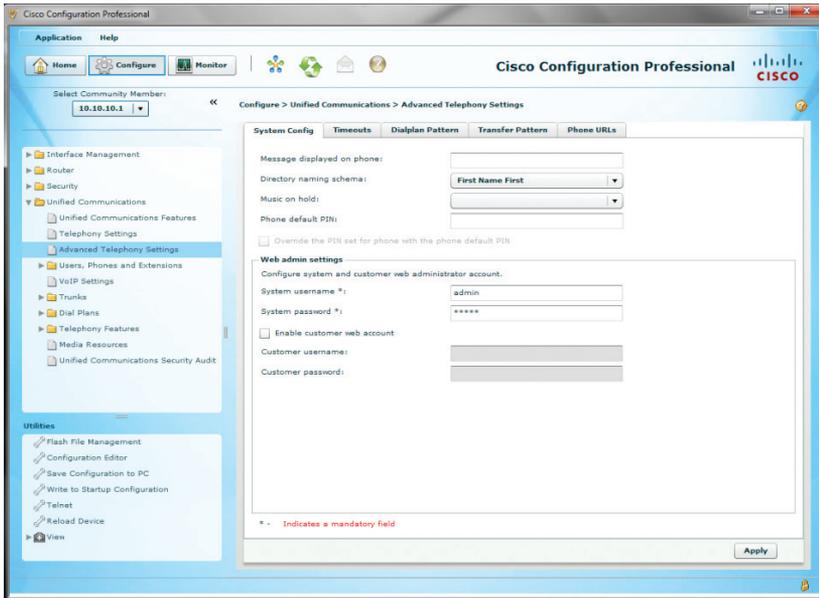
19. Click the **“No”** button.



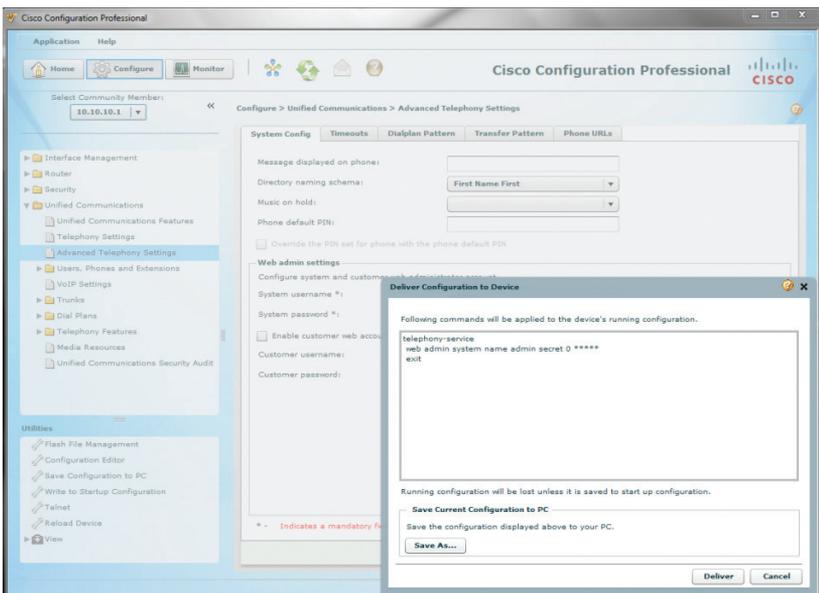
20. Click the **“Deliver”** button.



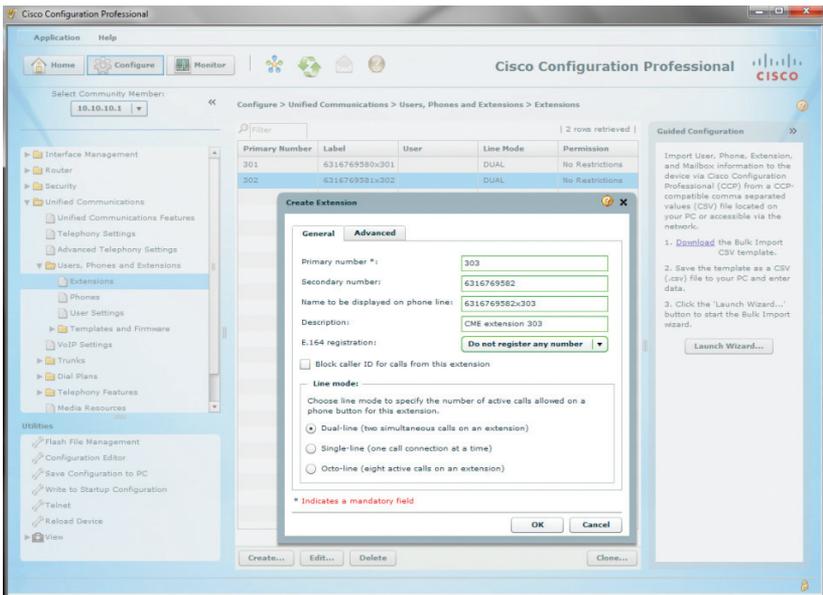
21. Navigate to **“Configure”>”Unified Communications”>”Advanced Telephony Settings”** to configure the Web admin settings.
 - a. Select the **“System Config”** tab.
 - b. Enter the administrator’s username in the **“System Username”** field.
 - c. Enter the password in the **“System password”** field.
 - d. Leave other fields as default and then click the **“Apply”** button.



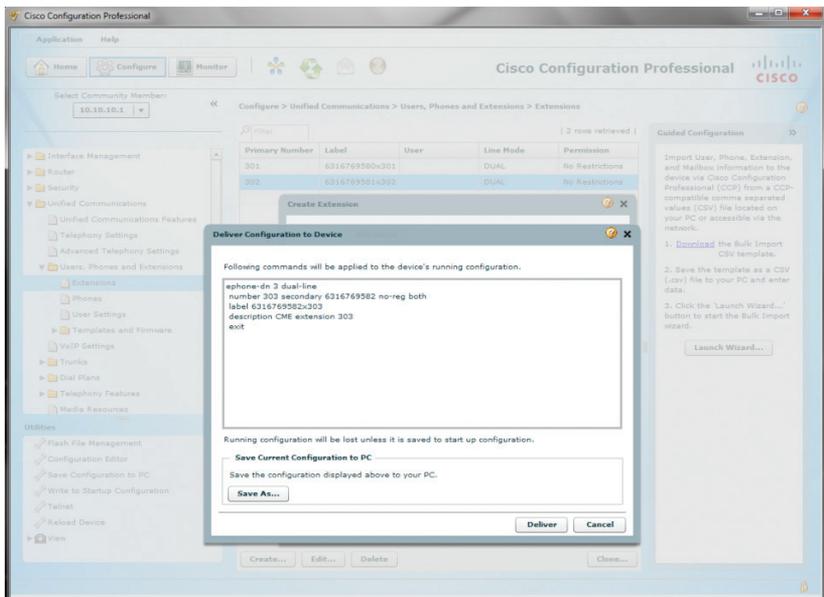
22. Click the **“Deliver”** button.



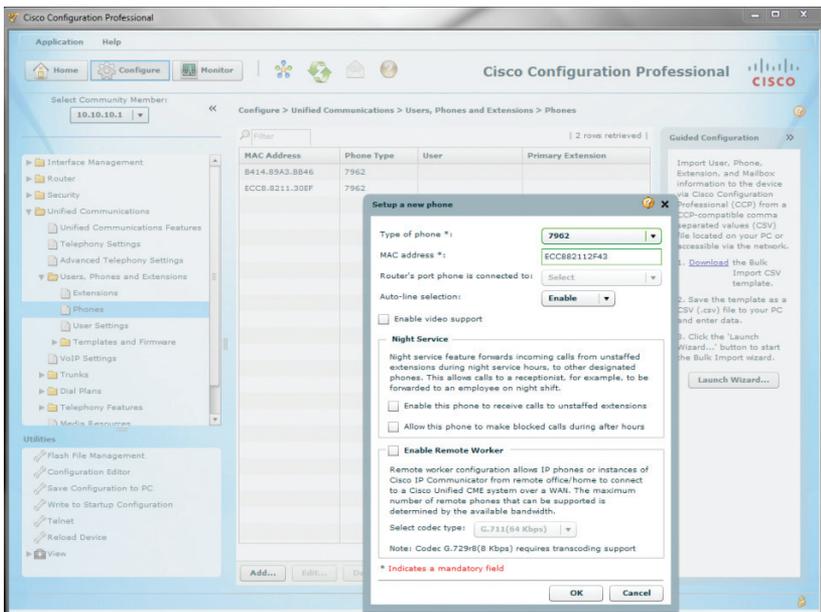
23. Navigate to **“Configure”>“Unified Communications”>“ Users, Phones and Extensions”>“Extensions”** to create extensions for routing inbound calls from the SIP trunk.
 - a. Select the **“General”** tab to create a new extension.
 - b. Enter a new extension in the **“Primary number”** field.
 - c. Enter a DID assigned for routing inbound calls to this extension in the **“Secondary number”** field.
 - d. Enter a descriptive name for displaying on the IP phone in the **“Name to be displayed on phone line”** field.
 - e. Enter a description in the **“Description”** field.
 - f. Select **“Do not register any number”** in the **“E.164 registration”** field.
 - g. Leave other fields as default and then click the **“OK”** button.



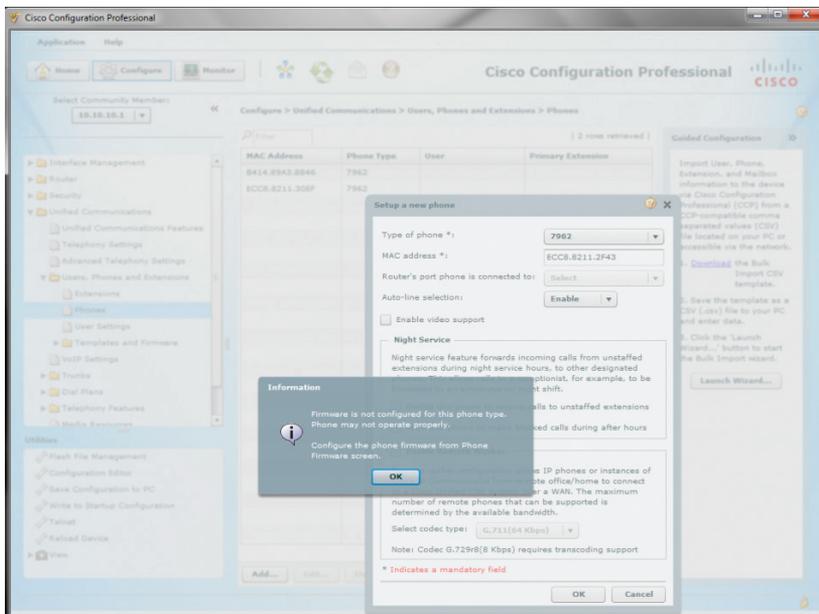
24. Click the **“Deliver”** button.



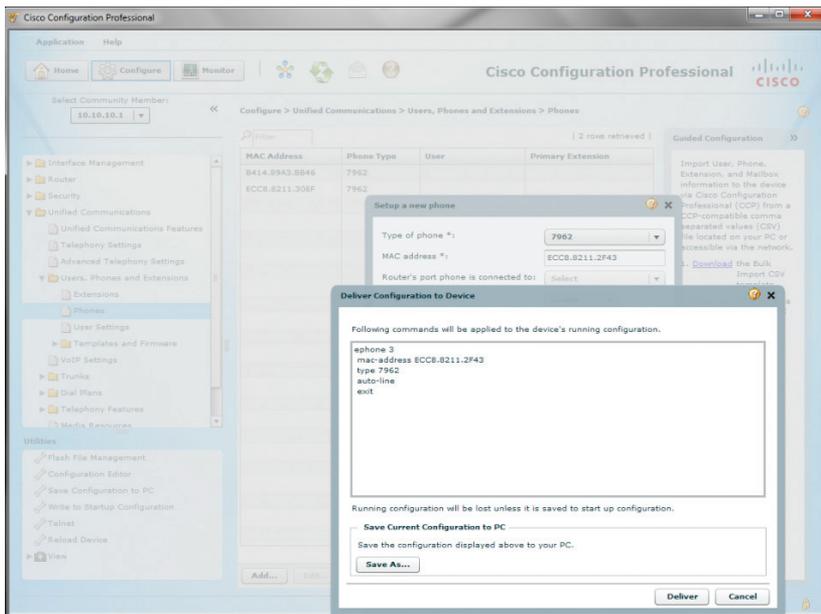
25. Navigate to **“Configure”>“Unified Communications”>“Users, Phones and Extensions”>“Phones”** to add IP phones.
 - a. Click the **“Add”** button and from the **“Setup a new phone”** screen:
 - b. Select **“7962”** from the drop-down list of the **“Type of phone”** field.
 - c. Enter the IP phone’s MAC address in the **“MAC address”** field.
 - d. Leave other fields as default and then click the **“OK”** button.



26. Click the **“OK”** button.

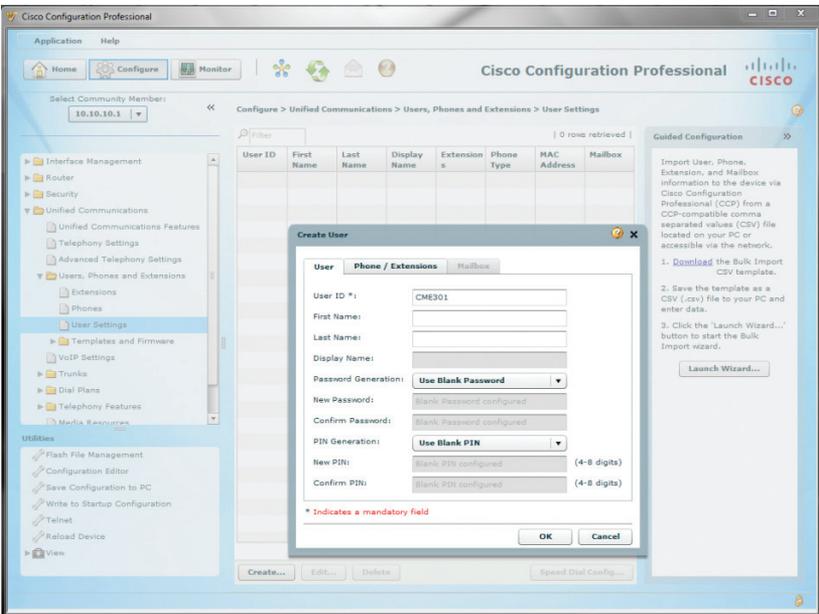


27. Click the **“Deliver”** button.



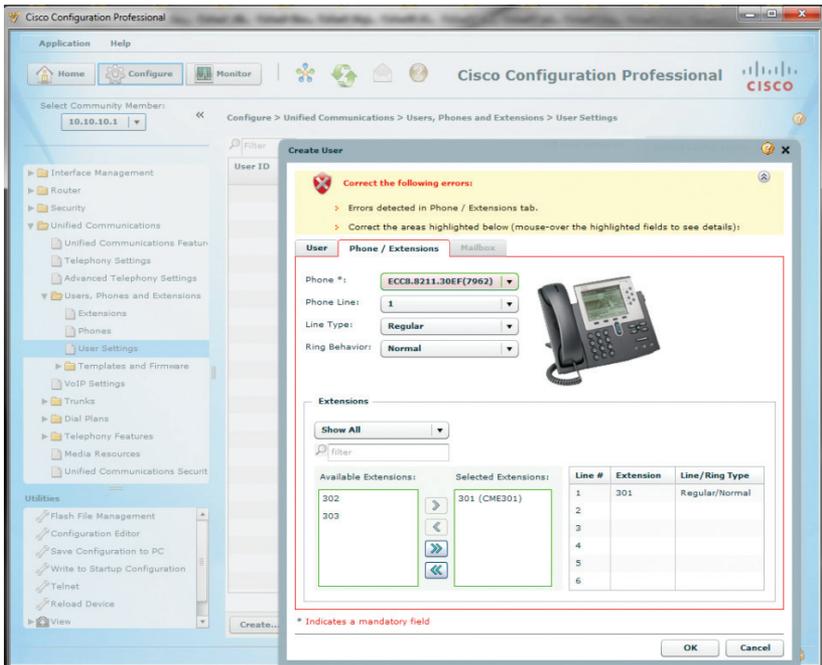
28. Navigate to **“Configure”>“Unified Communications”>“Users, Phones and Extensions”>“User Settings”** to assign IP phones and extensions to users.

- a. Click the **“Create”** button and select the **“User”** tab from the **“Create User”** screen.
- b. Enter a user ID in the **“User ID”** field.
- c. Leave other fields as default and then click the **“OK”** button.

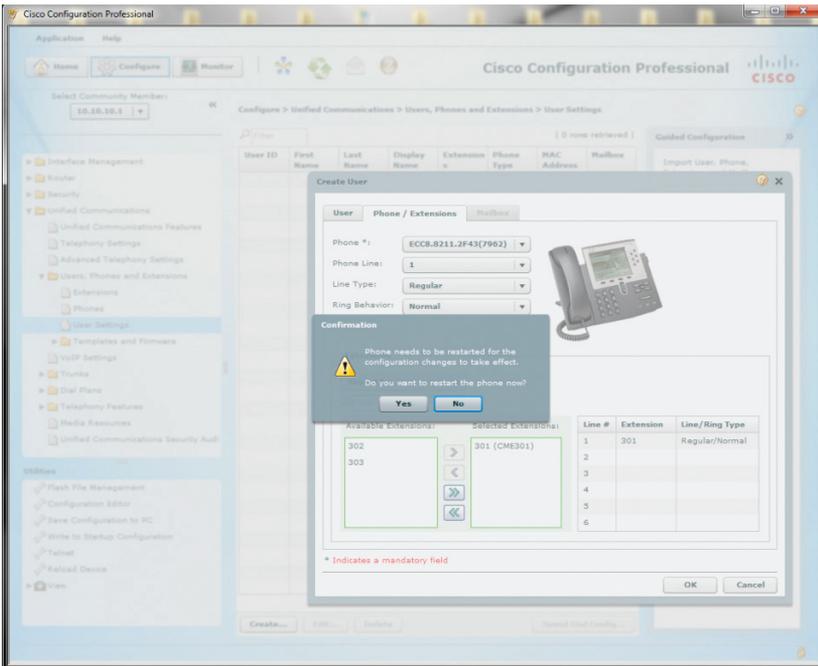


29. Select the **“Phone/Extensions”** tab.

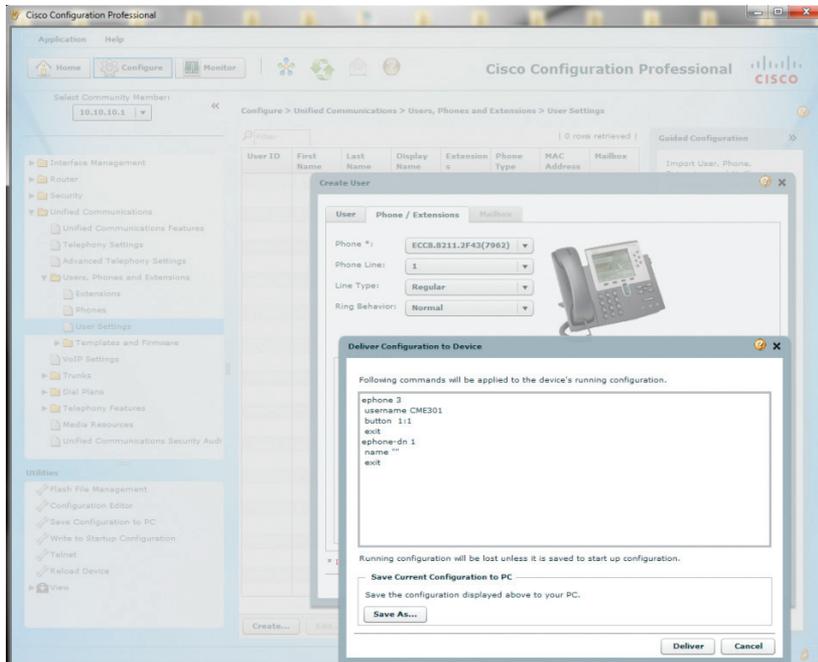
- Assign the phone to the user by selecting the MAC address of the IP phone from the drop-down list of the **“Phone”** field.
- Select **“1”** from the drop-down list of the **“Phone Line”** field.
- Assign the extension to the user by selecting the extension from the **“Available Extensions”** box and then click the **“>”** button.
- Leave other fields as default and then click the **“OK”** button.



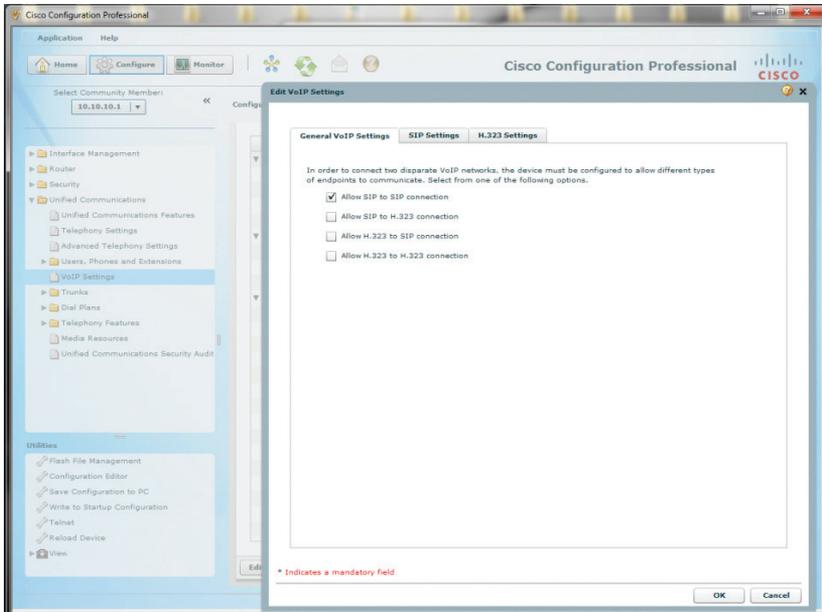
30. Click the **“No”** button.



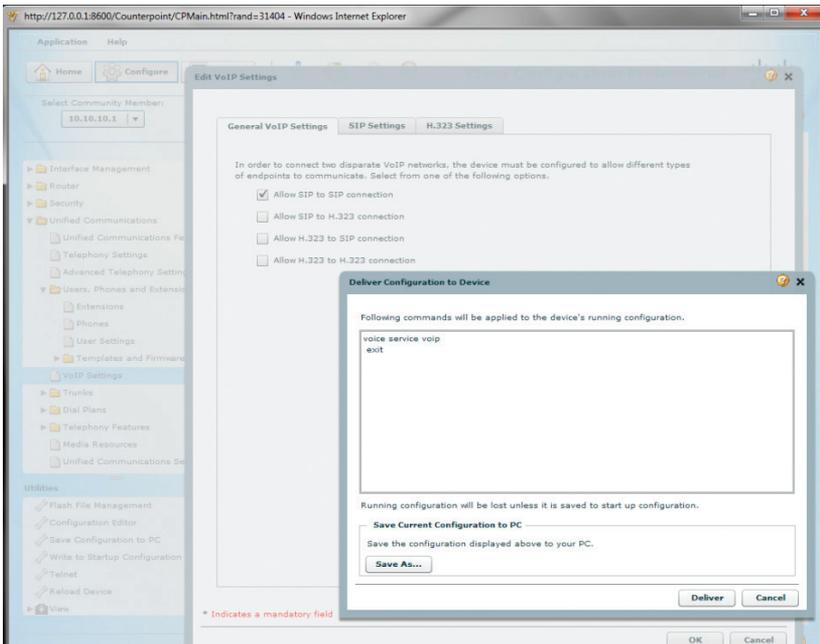
31. Click the **“Deliver”** button.



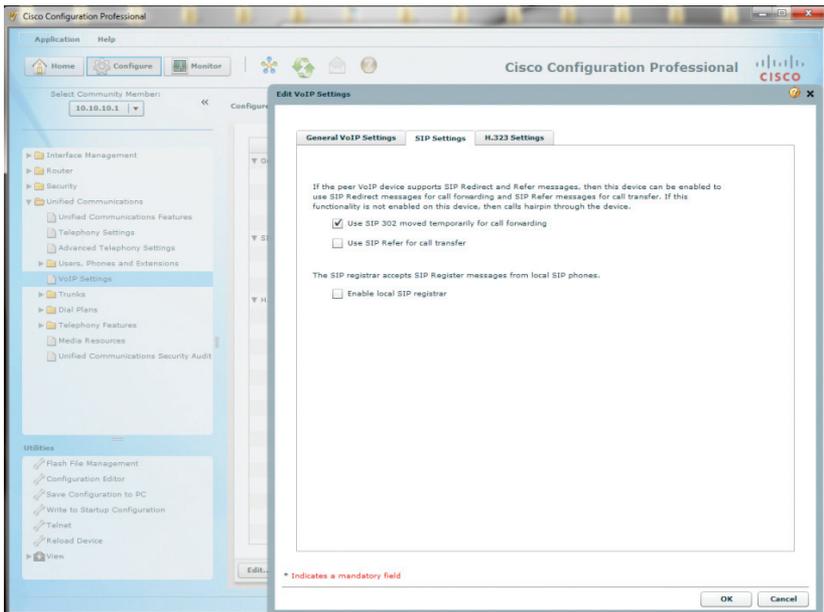
32. Navigate to **“Configure”>“Unified Communications”>“VoIP Settings”** and then click the **“Edit”** button to configure the VoIP parameters.
- a. Select the **“General VoIP Settings”** tab, check the checkbox for **“Allow SIP to SIP connection”**, leave the other fields as default and then click the **“OK”** button.



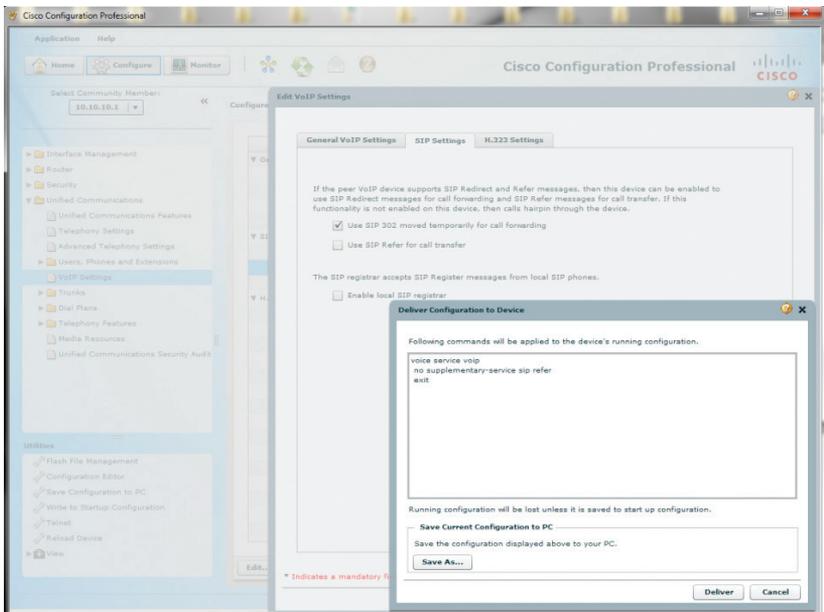
33. Click the **“Deliver”** button.



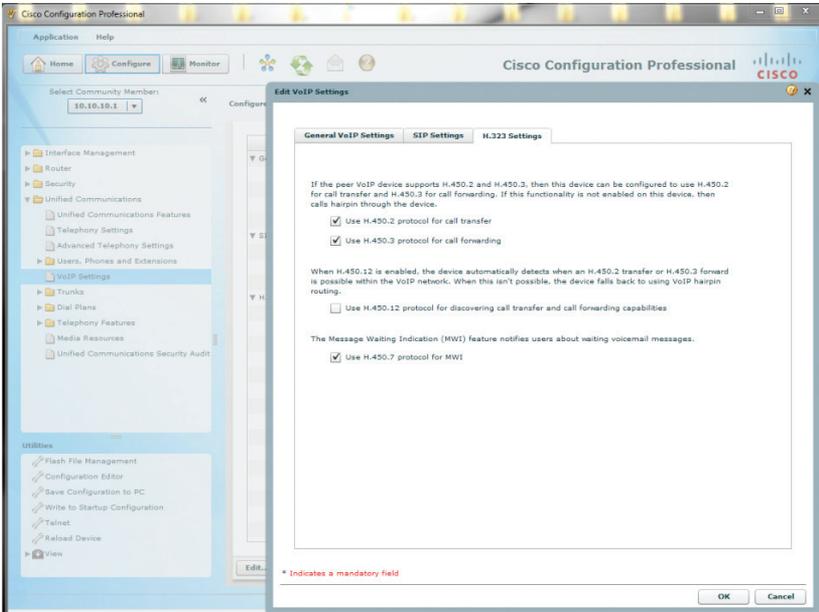
34. Select the **“SIP Settings”** tab, check the checkbox for **“Use SIP 302 moved temporarily for call forwarding”**, uncheck the checkbox for **“Use SIP Refer for call transfer”**, leave the other fields as default and then click the **“OK”** button.



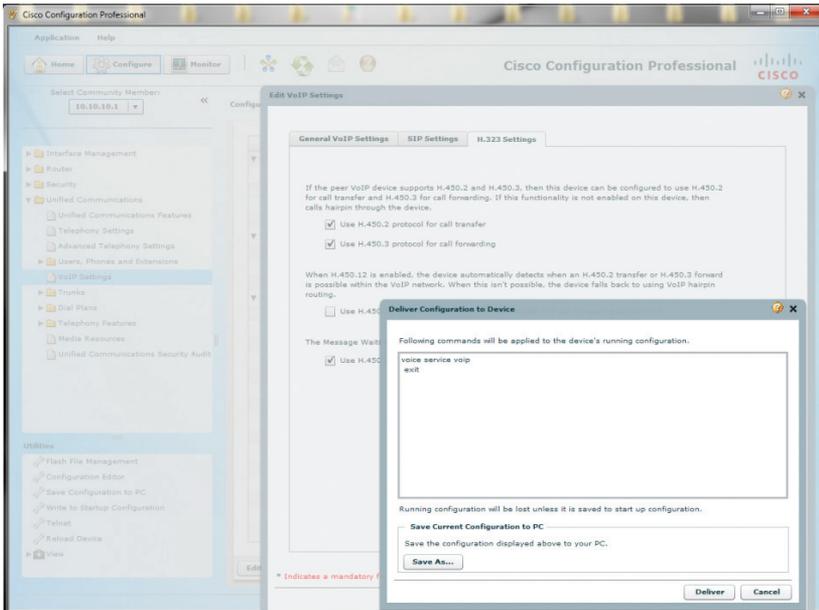
35. Click the **“Deliver”** button.



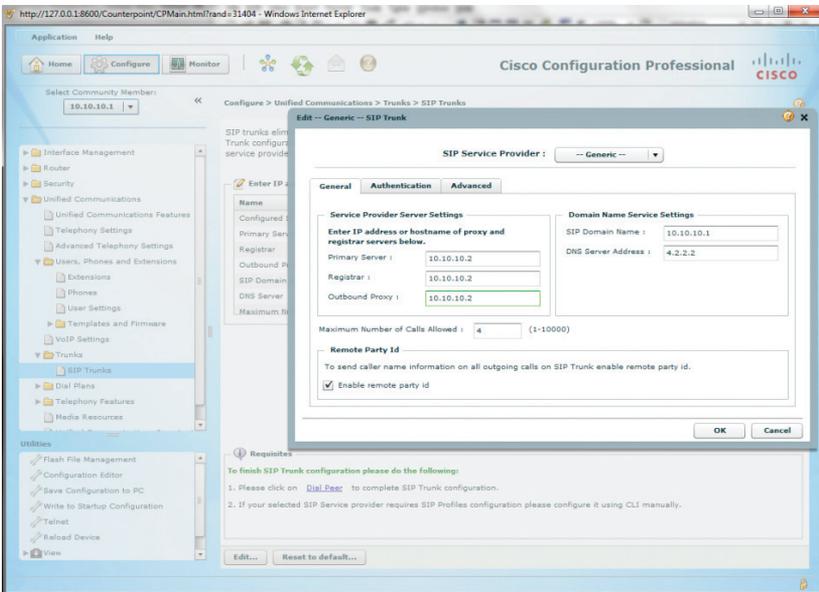
36. Select the **“H.323 Settings”** tab,
a. check the checkboxes for **“Use H.450.2 protocol for call transfer”**,
“Use H.450.3 protocol for call forwarding” and **“Use H.450.7 protocol for call MWI”**
b. leave other fields as default and then click the **“OK”** button.



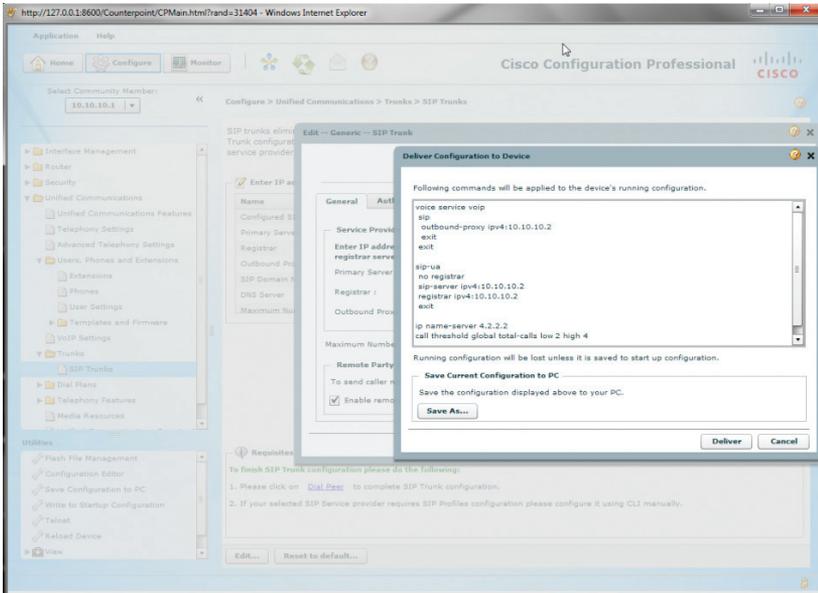
37. Click the **“Deliver”** button.



38. Navigate to **“Configure”>“Unified Communications”>“Trunks”>“SIP Trunks”** to configure Optimum Business SIP Trunk Adaptor as the SIP service provider.
- Click the **“Edit”** button, select **“-- Generic --”** from the drop-down list of the **“SIP Service Provider”** field and select the **“General”** tab.
 - Enter the **“Primary Server”** IP address. This is the IP address that was assigned to the Optimum SIP Trunk Adaptor’s LAN interface. This address was configured in Step 2 in the Optimum Business SIP Trunk Set-Up Guide.
 - Enter the **“Registrar”** IP Address if you are configuring the PBX for SIP registration mode. This is the IP address that was assigned to the Optimum SIP Trunk Adaptor’s LAN interface. This address was configured in Step 2 in the Optimum Business SIP Trunk Set-Up Guide. Leave this field blank if you are configuring the PBX for static IP mode.
 - Enter the **“Outbound Proxy”** IP address. This is the IP address that was assigned to the Optimum SIP Trunk Adaptor’s LAN interface. This address was configured in Step 2 in the Optimum Business SIP Trunk Set-Up Guide.
 - Enter the PBX’s GE 0/0 port’s IP address in the **“SIP Domain Name”** field.
 - Enter a valid DNS IP address in the **“DNS Service Address”** field.
 - Enter the maximum number of concurrent calls allowed in the **“Maximum Number of Calls Allowed”** field.
 - Click the **“OK”** button.

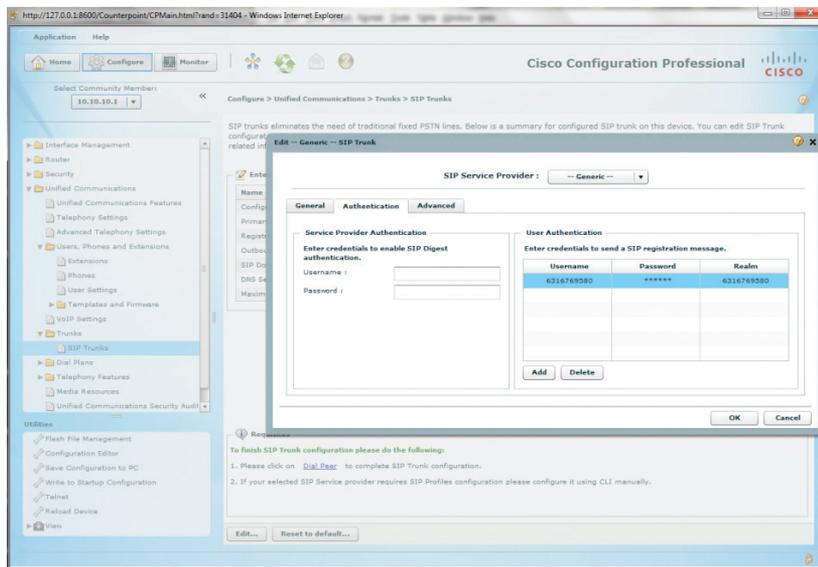


39. Click the **“Deliver”** button.

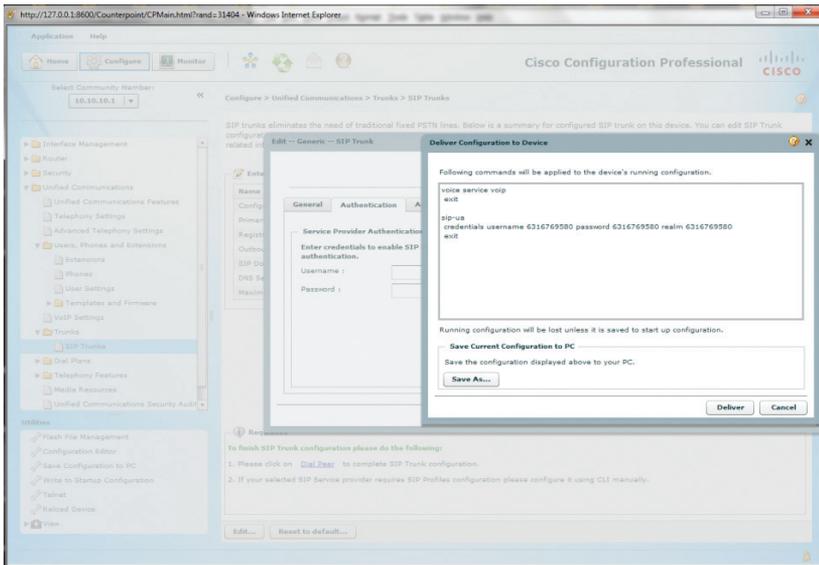


Note: This step is needed only if you are configuring the PBX for SIP registration mode.

40. Click the **“Edit”** button, select **“-- Generic --”** from the drop-down list of the **“SIP Service Provider”** field and select the **“Authentication”** tab.
 - a. Click the **“Add”** button to set up the PBX to send SIP registration for the pilot DID. Note that the username and password must match what has been configured on the Optimum SIP Trunk Adaptor. This is step 3 in the Optimum Business Sip Trunk Set-Up Guide.
 - b. Enter the username in the **“Username”** field.
 - c. Enter the password in the **“Password”** field.
 - d. Enter the realm (if needed) in the **“Realm”** field.
 - e. Click the **“OK”** button.



41. Click the **“Deliver”** button.

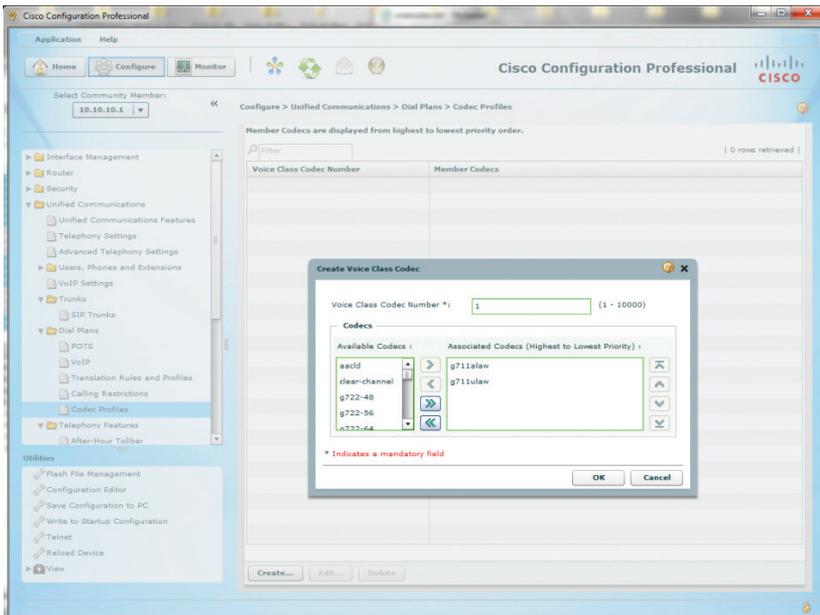


42. Navigate to **“Configure”>“View”>“Running Configuration”** to see if the DHCP server on GE 0/1 interface (IP address of 192.168.0.1/24 in this example) is set up to send **“option 150”** (TFTP server IP address) to the DHCP clients when the IP phones are restarted. If **“option 150”** is not in the **“ip dhcp pool ccp-pool1”** section, the IP phones would not be able to retrieve the files created for the IP phones by the PBX. To take care of this problem, enter the following CLI commands from the console:

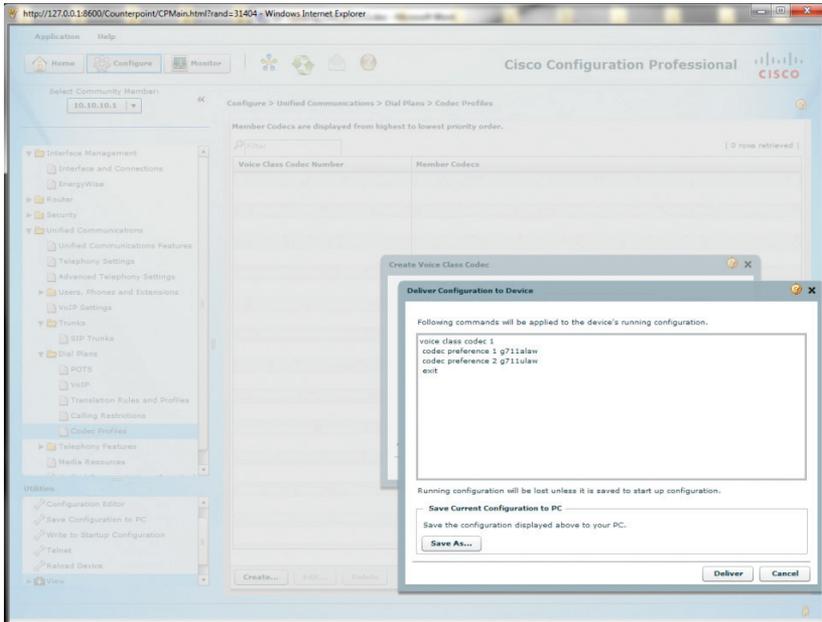
```
cisco2901cme#configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
cisco2901cme(config)#ip dhcp pool ccp-pool1
cisco2901cme(dhcp-config)#option 150 ip 192.168.0.1
cisco2901cme(dhcp-config)#exit
cisco2901cme(config)#exit
cisco2901cme#
```



43. Navigate to **“Configure”>“Unified Communications”>“Dial Plans”>“Codec Profiles”** to group **“g711alaw”** and **“g711ulaw”** into the same codec class.
 - a. Click the **“Create”** button and from the **“Create Voice Class Codec”** screen:
 - b. Enter **“1”** in the **“Voice Class Codec Number”** field.
 - c. Select the **“g711alaw”** codec from the **“Available Codecs”** list and then click the **“>”** button.
 - d. Select the **“g711ulaw”** codec from the **“Available Codecs”** list and then click the **“>”** button.
 - e. Click the **“OK”** button.

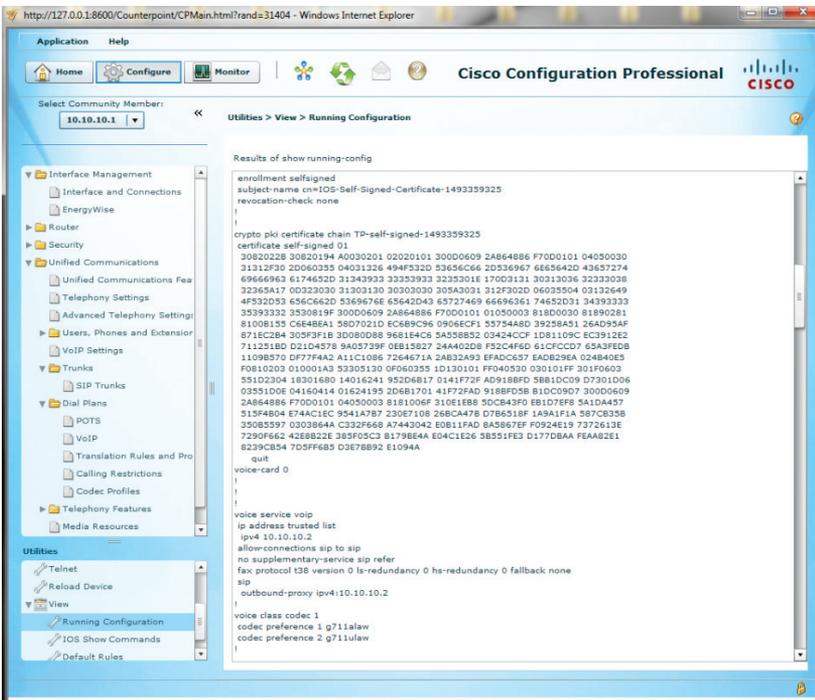


44. Click the **“Deliver”** button.

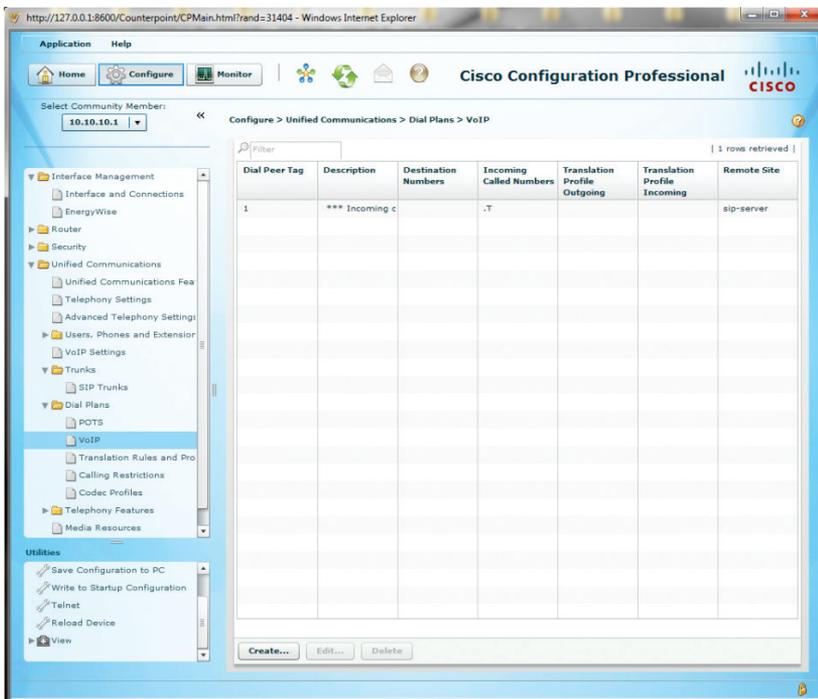


45. Navigate to **“Configure”>“View”>“Running Configuration”** to see if EdgeMarc’s IP address is in the **“ip address trusted list”** under the **“voice service voip”** section. If it is not there, inbound SIP calls from EdgeMarc will be rejected by the PBX with a SIP **“403/Forbidden”** response. To take care of this problem, enter the following CLI commands from the console:

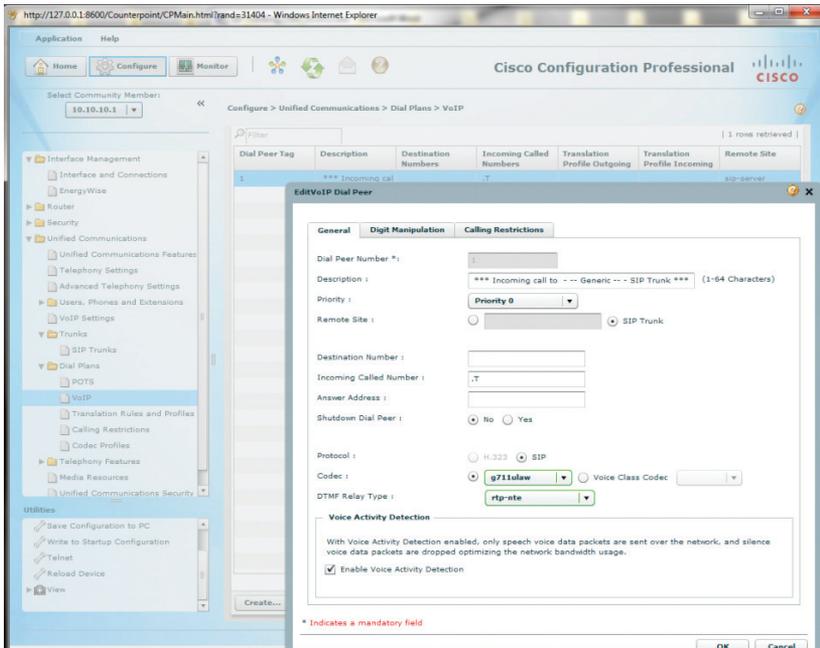
```
cisco2901cme#configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
cisco2901cme(config)#voice service voip
cisco2901cme(conf-voi-serv)#ip address trusted list
cisco2901cme(cfg-iptrust-list)#ipv4 10.10.10.2
cisco2901cme(cfg-iptrust-list)#exit
cisco2901cme(conf-voi-serv)#exit
cisco2901cme(config)#exit
cisco2901cme#
```



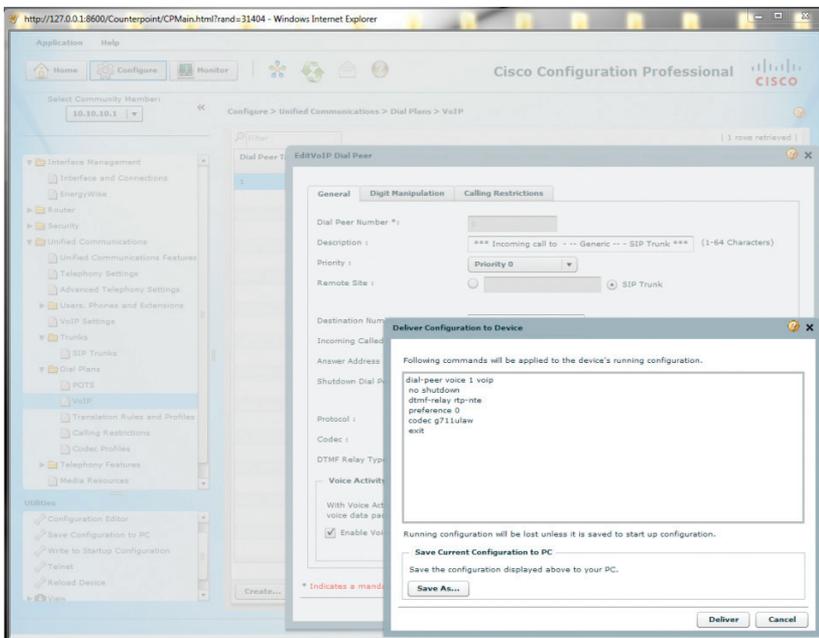
46. Navigate to **“Configure”>“Unified Communications”>“Dial Plans”>“VoIP”** to complete SIP trunk configuration for Dial Peer. Note that the first Dial Peer entry was automatically created for incoming calls by the PBX when SIP trunk was configured.



47. Select the Dial Peer number 1 entry, click the **“Edit”** button and then select the **“General”** tab to change some parameters and leave Dial Peer Number 1 for incoming calls only.
 - a. Select **“g711ulaw”** from the drop-down list of the **“Codec”** field for now as selection for **“Voice Class Codec”** is not yet available.
 - b. Select **“rtp-nte”** from the drop-down list of the **“DTMF Relay Type”** field. This allows the PBX to relay outbound DTMF with RFC-2833 DTMF. Note that, for inbound DTMF, the PBX will recognize RFC-2833 DTMF only.
 - c. Leave other fields as default and then click the **“OK”** button.

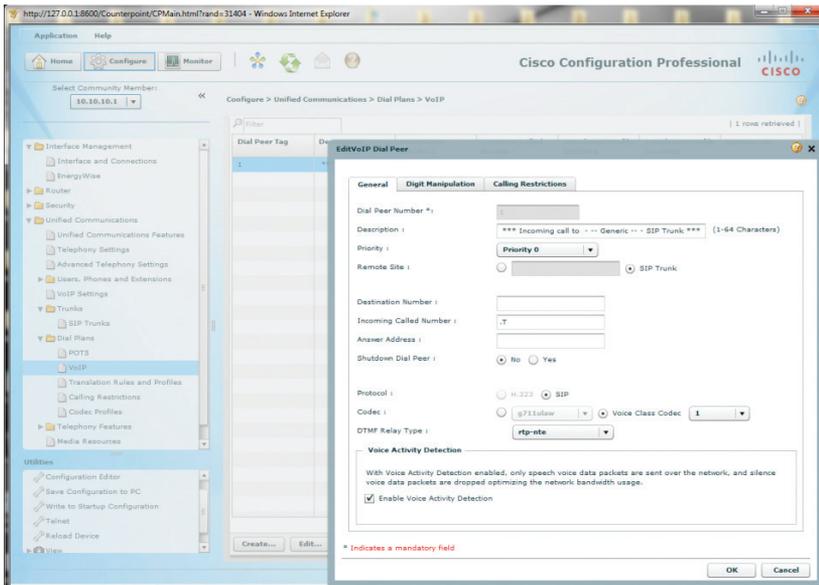


48. Click the **“Deliver”** button.

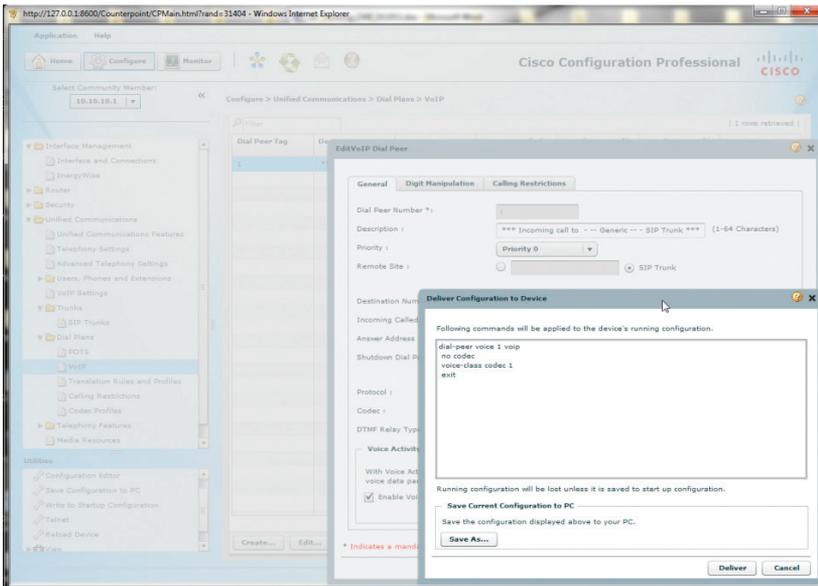


49. Select the **Dial Peer number 1** entry

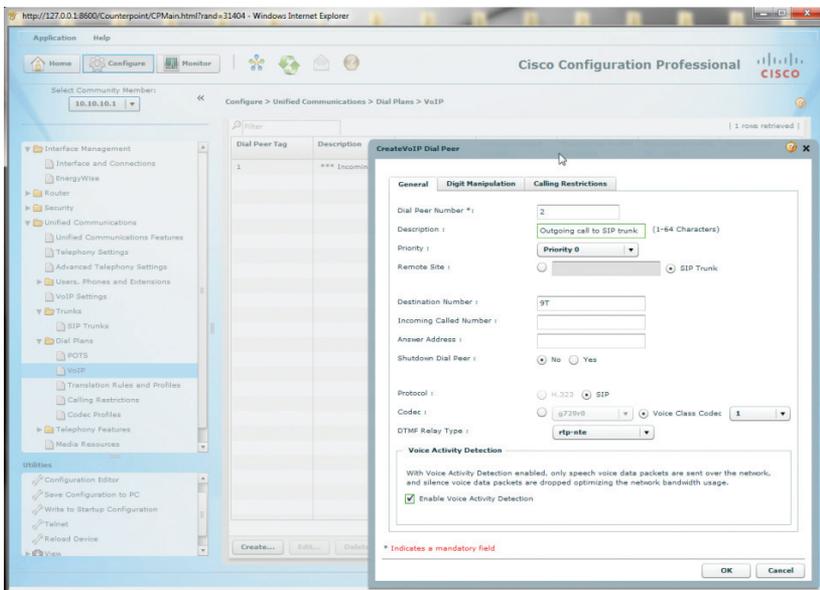
- a. Click the **“Edit”** button and then select the **“General”** tab again to choose the Voice Class Codec that includes both **“g711ulaw”** and **“g711alaw”**.
- b. Select the radio button for the **“Voice Class Codec”**.
- c. Select **“1”** from the drop-down list of the **“Voice Class Codec”** field and then click the **“OK”** button.



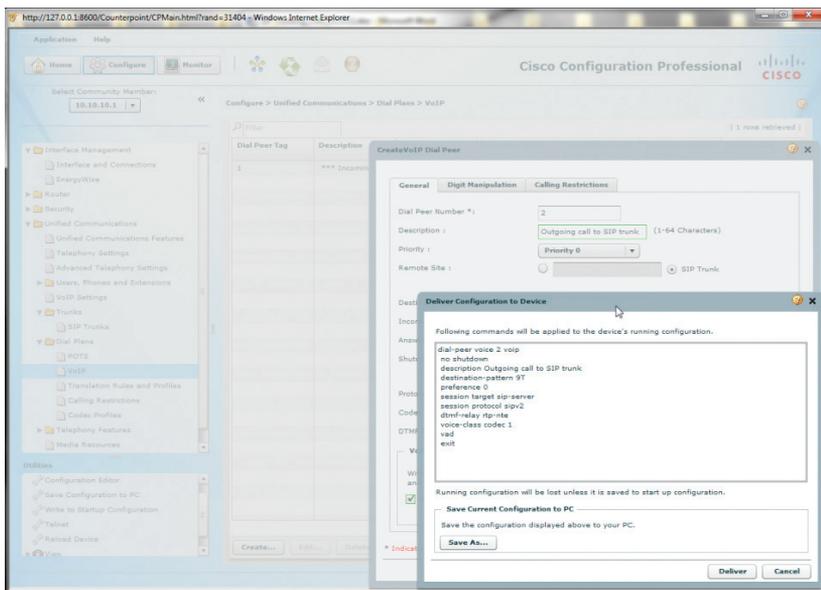
50. Click the **“Deliver”** button.



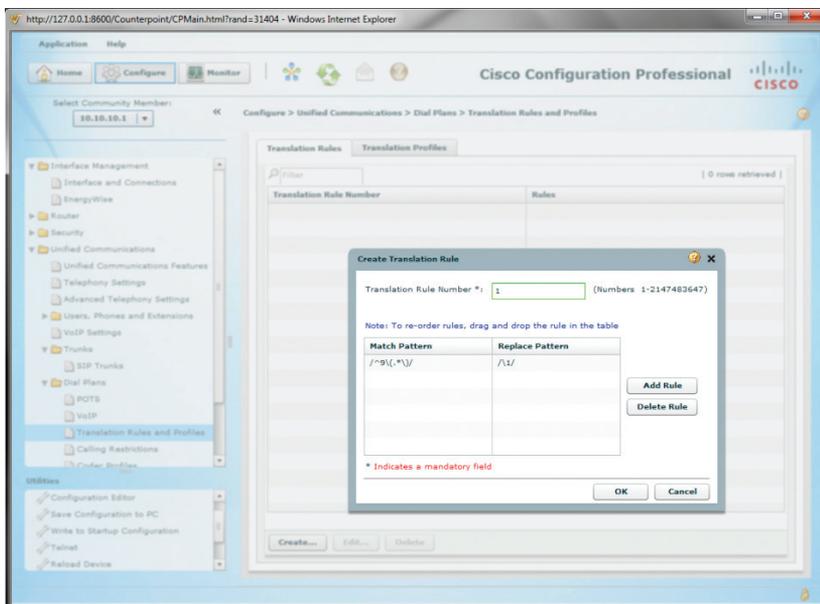
51. Navigate to **“Configure”>“Unified Communications”>“Dial Plans”>“VoIP”** to create a second Dial Peer for SIP trunk for outgoing calls only.
- Click the **“Create”** button and then select the **“General”** tab.
 - Enter a description in the **“Description”** field.
 - Enter **“9T”** (9 followed any numbers) in the **“Destination Number”** field.
 - Select the radio button for **“Voice Class Codec”** for the **“Codec”** field.
 - Select **“1”** from the drop-down list of the **“Voice Class Codec”** field.
 - Select **“rtp-nte”** from the drop-down list of the **“DTMF Relay Type”** field. This allows the PBX to relay outbound DTMF with RFC-2833 DTMF. Note that, for inbound DTMF, the PBX will recognize RFC-2833 DTMF only.
 - Leave other fields as default and then click the **“OK”** button.



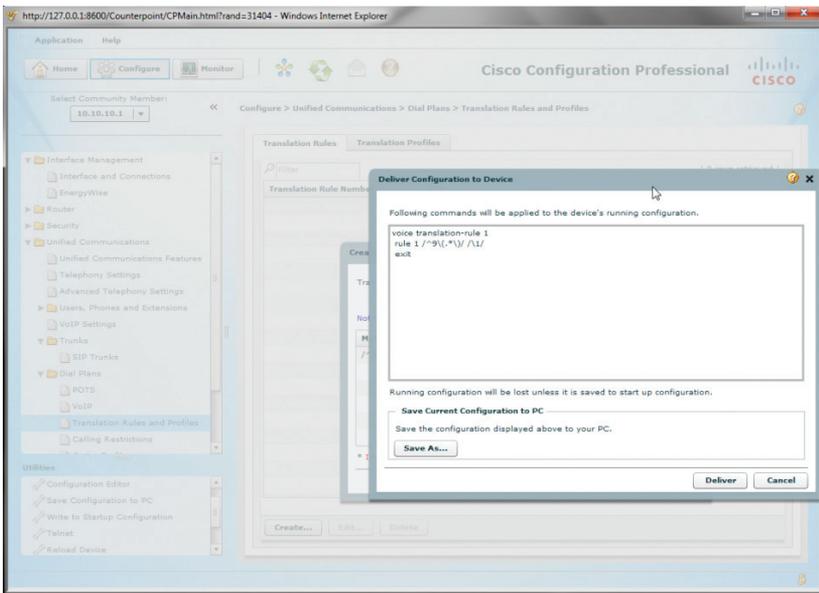
52. Click the **“Deliver”** button.



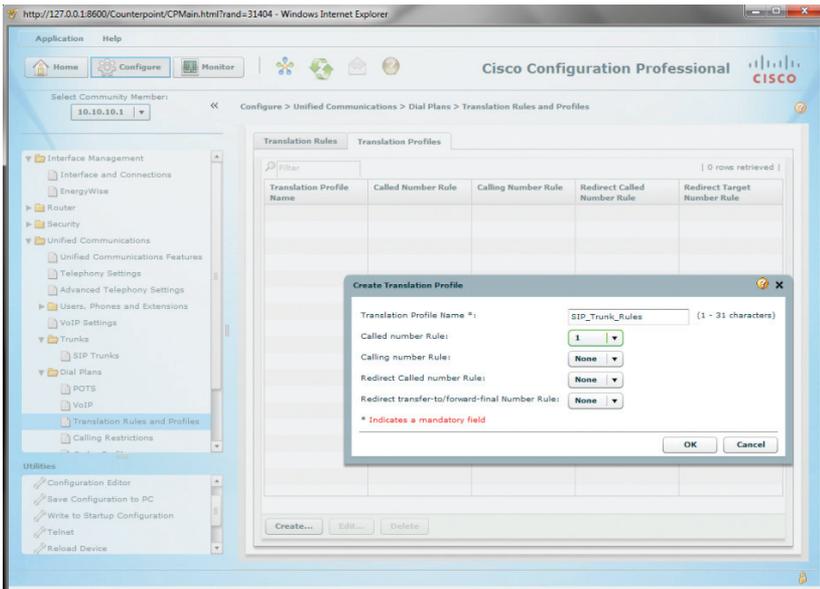
53. Navigate to **“Configure”>“Unified Communications”>“Dial Plans”>“Translation Rules and Profiles”** to create an outbound rule for removing SIP trunk access code before dialing.
 - a. Select the **“Translation Rules”** tab and click the **“Create”** button.
 - b. Enter **“1”** in the **“Translation Rule Number”** field.
 - c. Click the **“Add Rule”** button.
 - d. Enter **“/^\9(.*)/”** in the **“Match Pattern”** field to match a string of numbers starting with a **“9”**.
 - e. Enter **“/\1/”** in the **“Replace Pattern”** field to replace the matched numbers with the same numbers without the leading **“9”** digit.
 - f. Click the **“OK”** button.



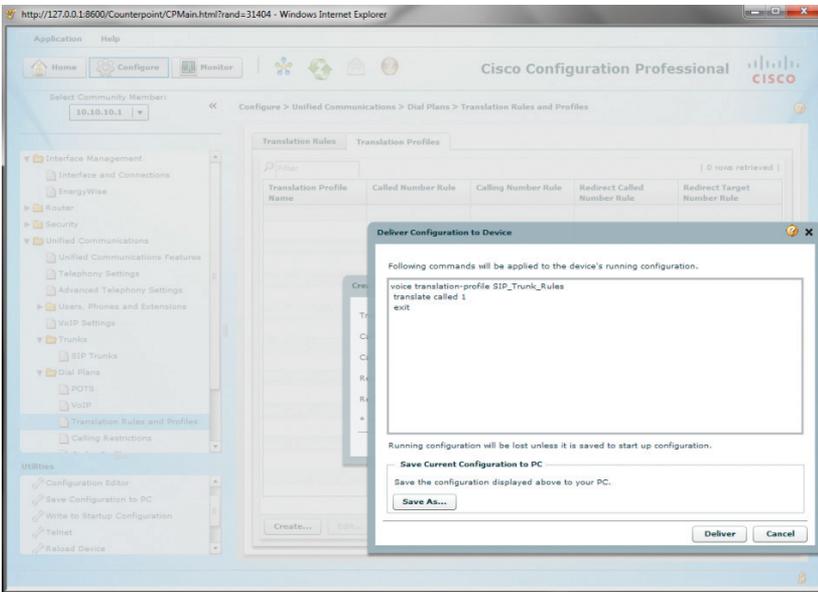
54. Click the **“Deliver”** button.



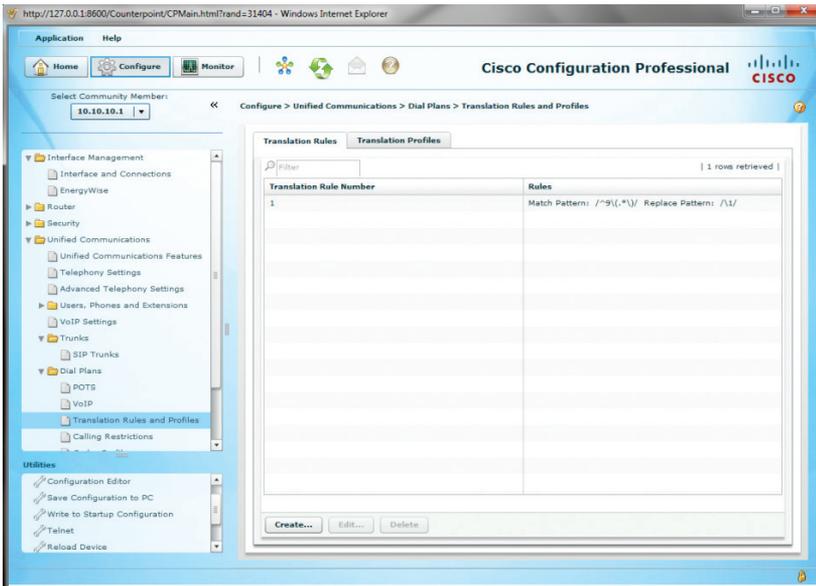
55. Navigate to **“Configure”>“Unified Communications”>“Dial Plans”>“Translation Rules and Profiles”** to create a Translation Profile to hold the Translation Rule for the called number.
 - a. Select the **“Translation Profiles”** tab and click the **“Create”** button.
 - b. Enter a descriptive name in the **“Translation Profile Name”** field.
 - c. Select **“1”** (Translation Rule #1) in the **“Called number Rule”** field.
 - d. Leave other fields as default and then click the **“OK”** button.



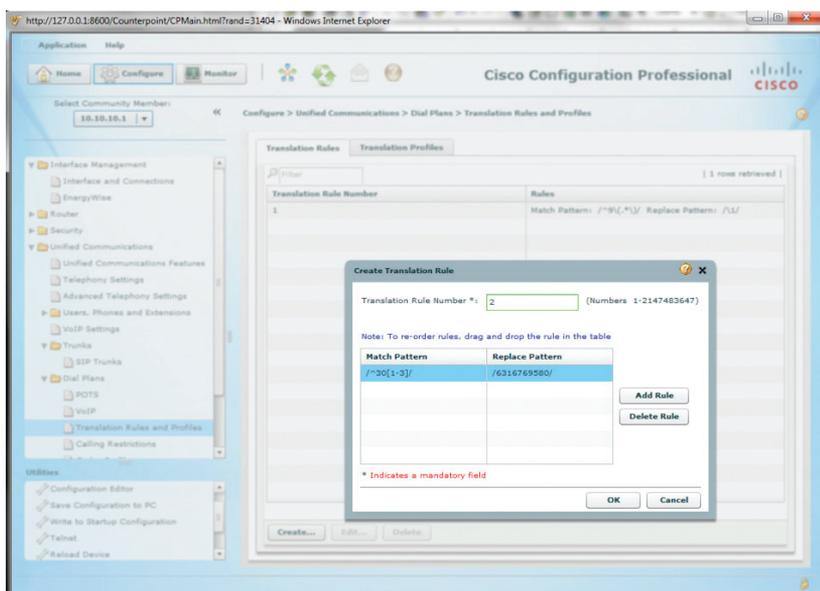
56. Click the **“Deliver”** button.



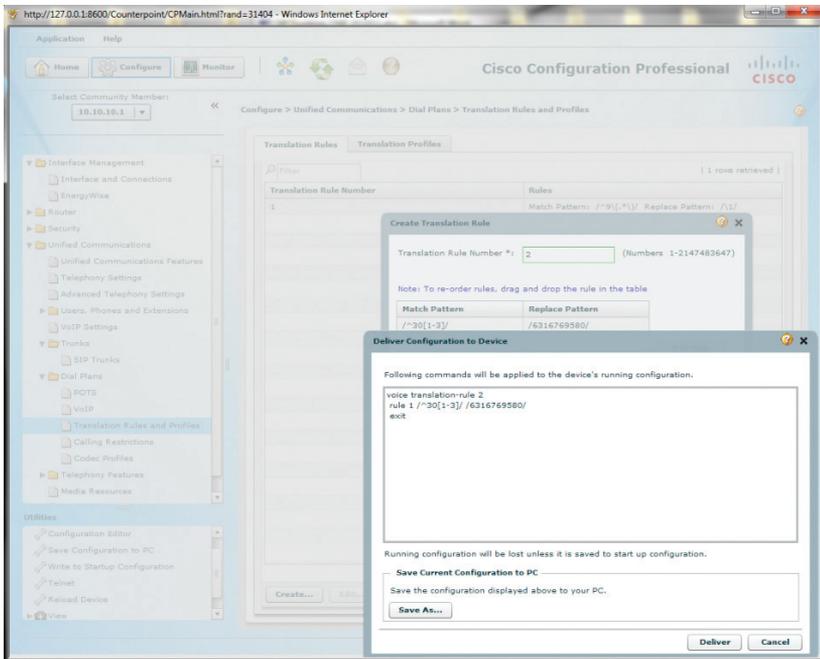
57. Navigate to **“Configure”>“Unified Communications”>“Dial Plans”>“Translation Rules and Profiles”** to create outbound rules for configuring caller ID.



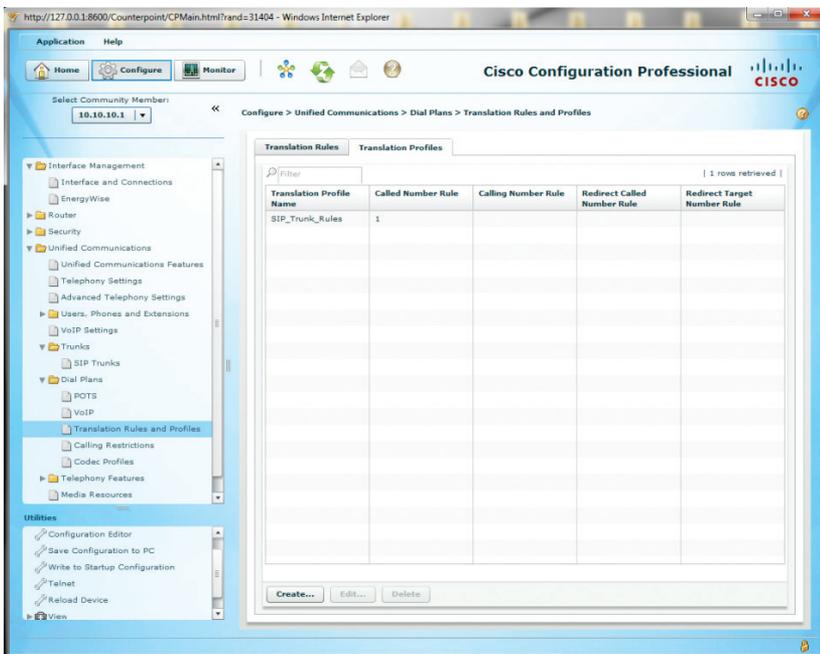
- a. Select the **“Translation Rules”** tab and click the **“Create”** button.
- b. Enter **“2”** in the **“Translation Rule Number”** field.
- c. Click the **“Add Rule”** button.
- d. Enter **“/∧30[1-3]/”** in the **“Match Pattern”** field to match all the extensions starting with a **“3”**, **“301”**, **“302”** and **“303”**.
- e. Enter **“/6316769580/”** in the **“Replace Pattern”** field to replace each extension with the pilot DID. Note that this example is for SIP registration mode where only the pilot DID can be used as the caller ID. For static IP mode where each extension may use its assigned DID as the caller ID, different translation rules will be needed for mapping all the extensions with different caller ID. For example, to map extension **“303”** with caller ID of **“6316769582”**, set the **“Match Pattern”** to **“/∧303/”** and set the **“Replace Pattern”** to **“/6316769582/”**.
- f. Click the **“OK”** button.



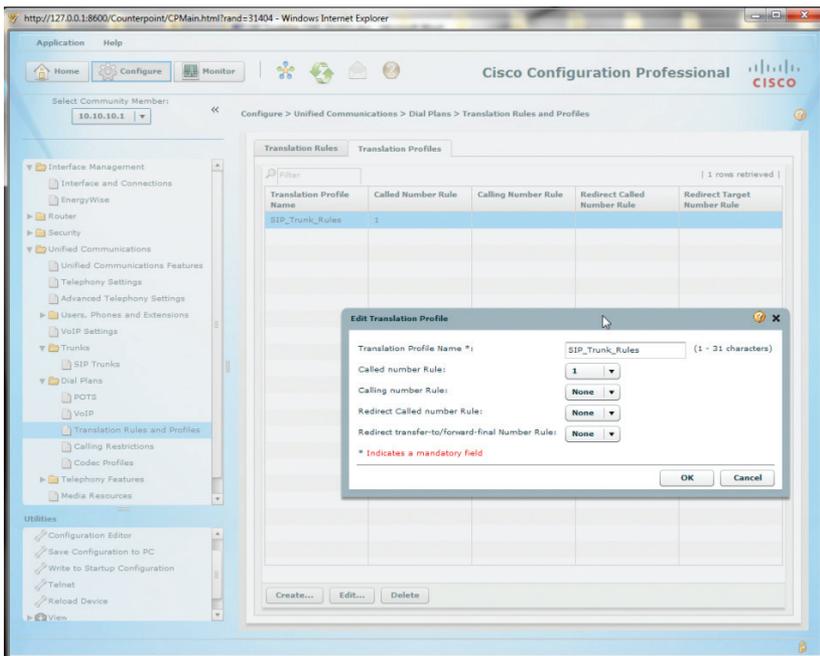
58. Click the **“Deliver”** button.



59. Navigate to **“Configure”>“Unified Communications”>“Dial Plans”>“Translation Rules and Profiles”** and select the **“Translation Profiles”** tab to add Translation Rule #2 for the calling number to the same Translation Profile.

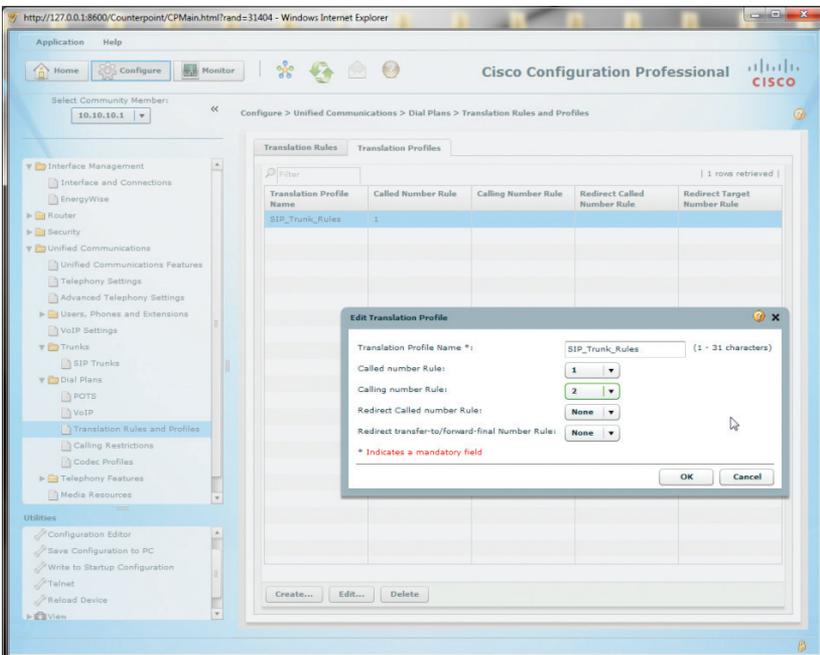


a. Select the Translation Profile and then click the **“Edit”** button.

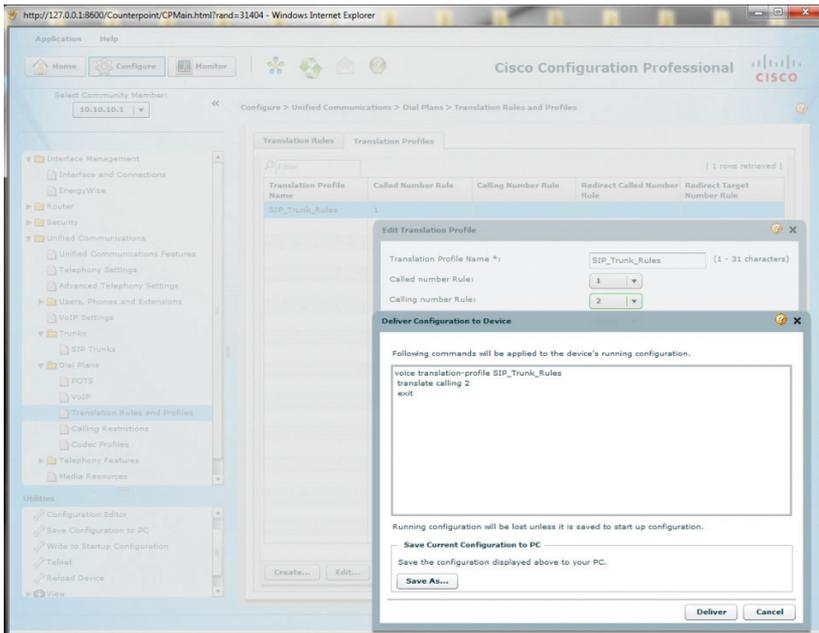


b. Select **“2”** (Translation Rule #2) in the **“Calling number Rule”** field.

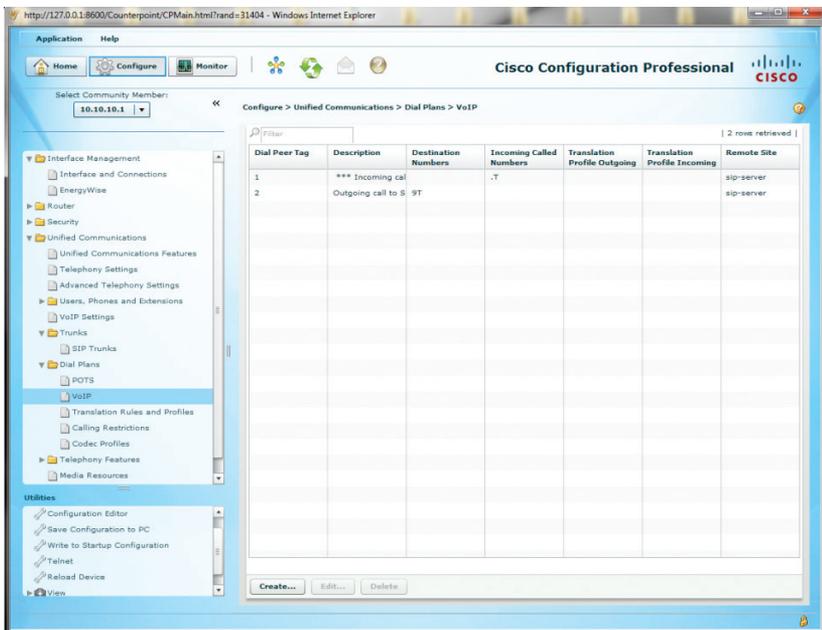
c. Click the **“OK”** button.



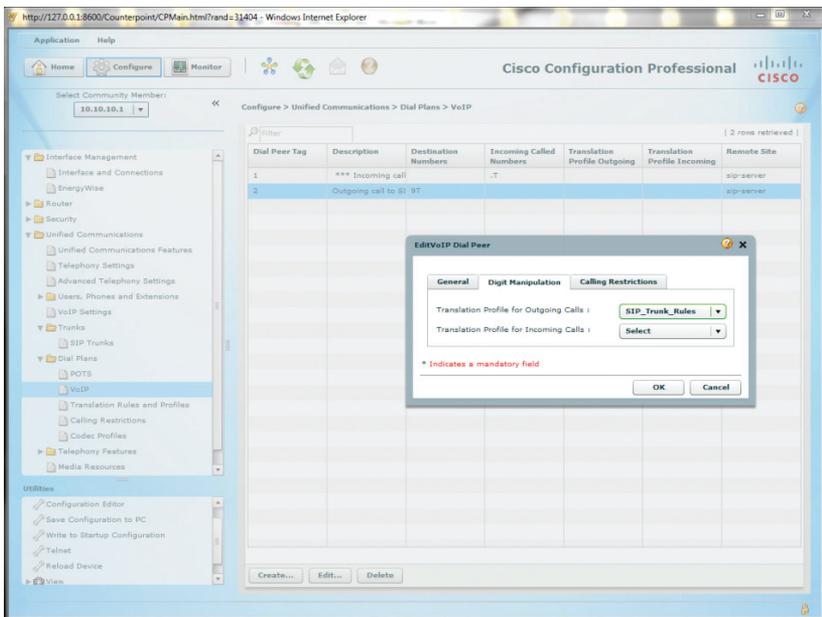
60. Click the **“Deliver”** button.



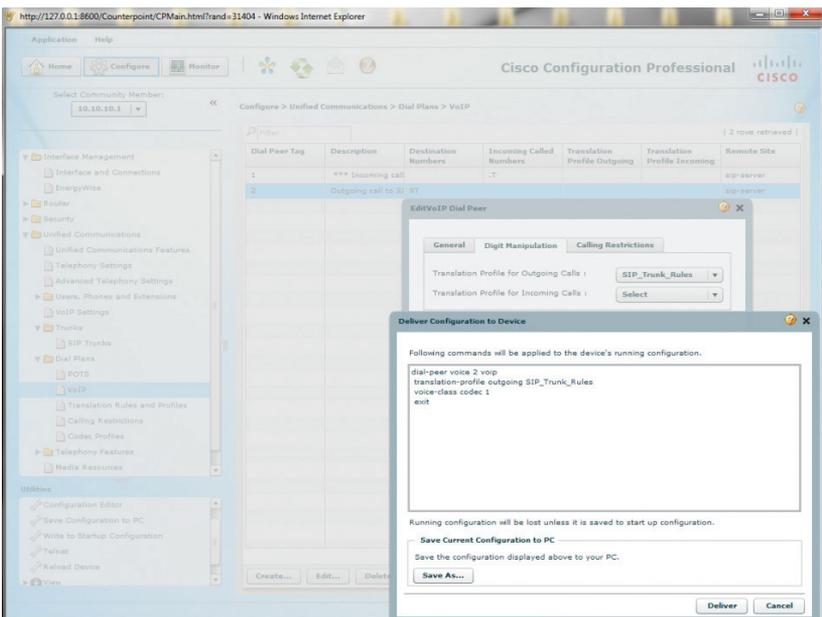
61. Navigate to **“Configure”>“Unified Communications”>“Dial Plans”>“VoIP”** to apply the Translation Profile and its Translation Rules to the outbound calls.



- Select the Dial Peer #2 (“**Outgoing calls to SIP trunk**”) and then click the “**Edit**” button.
- Select the “**Digit Manipulation**” tab from the “**EditVoIP Dial Peer**” screen.
- Select the translation profile (“**SIP_Trunk_Rules**”) from the drop-down list of the “**Translation Profile for Outgoing Calls**” field.
- Click the “**OK**” button.

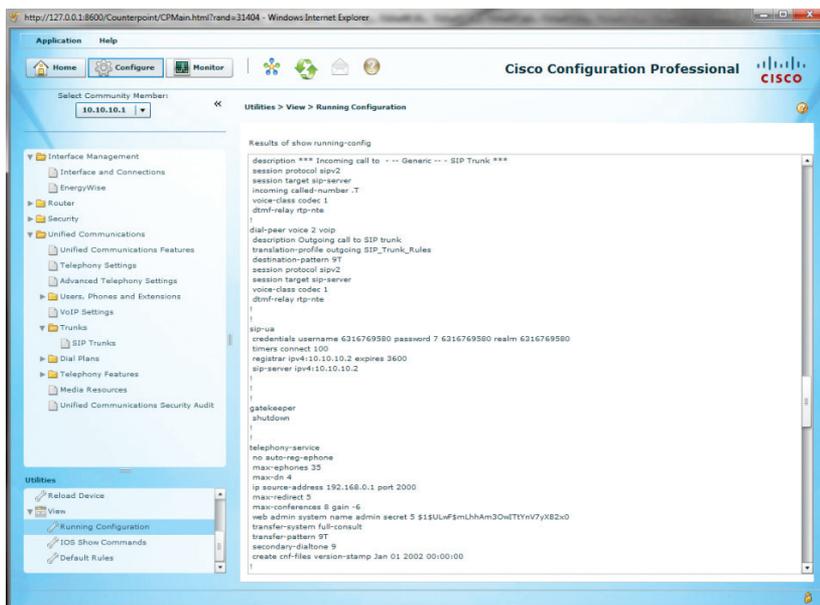


- Click the “**Deliver**” button.



63. Navigate to “Configure”>“View”>“Running Configuration” to see if the “transfer-pattern 9T” and “transfer-system full-consult” commands are set up under the “telephony-service” section. If these commands are not there, you cannot dial the “9” access code to transfer a LAN-to-WAN call to another WAN phone. To take care of this problem, enter the following CLI commands from the console:

```
cisco2901cme#configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
cisco2901cme(config)#telephony-service
cisco2901cme(config-telephony)#transfer-pattern 9T
cisco2901cme(config-telephony)#transfer-system full-consult
cisco2901cme(config-telephony)#exit
cisco2901cme(config)#exit
cisco2901cme#
```



Note: The Cablevision network only supports inband DTMF tones. The Cisco CME PBX only supports out-of-band DTMF tones. In order for the Cisco CME PBX to operate correctly with the Cablevision network, the Optimum SIP Trunk Adaptor must be configured to convert out-of-band DTMF tones sent by the Cisco CME PBX to inband DTMF tone.

64. To enable this conversion, log into the Optimum SIP Trunk Adaptor using the login and password specified in the Optimum SIP Trunk Adaptor Set-Up Guide.
- On the SIP Trunk Configuration page, you must check the **“Convert Inband DTMF”** checkbox.
 - Click the **“Submit”** button to update this setting.



SIP Trunk Configuration [Help](#)

Select your PBX:

Passive connection using the local, private IP address of the PBX
This address must be on the same subnet as the IP Address that is specified for the LAN interface
PBX Address:

Active connection using registration
User Id:
Password:

Convert Inband DTMF:

Status:

Trunk Status:

DID's

5164939899 (Pilot number)
5164939768
5164939769
5164939795
5164939841