SIP Trunking Configuration guide for Reliance Global Call

The Reliance Global Call SIP Trunking solution enables enterprise customers to directly dial international numbers from their office desk phones, without having to dial any intermediary access number making International calling a seamless one step process with reduced call set up time. This document describes the configuration steps to create a SIP Trunk to the Reliance Global Call platform.

Reliance Global Call

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11/11/2011

Prerequisites:

1. SIP capable IP PBX (or Traditional PBX with SIP capable Media Gateway)
2. Licenses for SIP Trunking (if applicable)
3. Access to PBX configuration utility and configuration rights
4. Port on PBX open to public internet
5. Firewall in network should not block UDP and SIP packets
6. PBX support for SIP INFO Method

The process of creation of SIP Trunks varies from PBX to PBX. We have tried to identify the basic commons steps that are involved.

Flow Chart for SIP Trunking Configuration:

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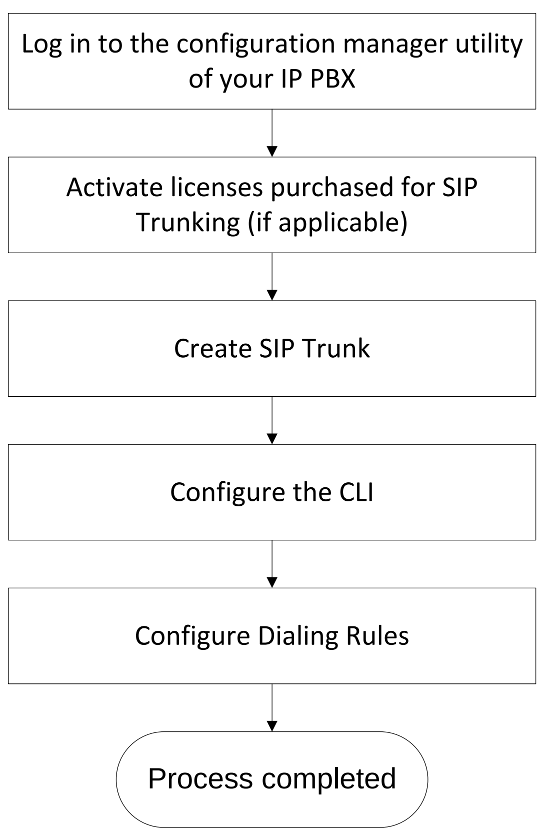


Detail SIP Trunking Configuration:

Step 1: Log in to the configuration manager utility of your IP PBX

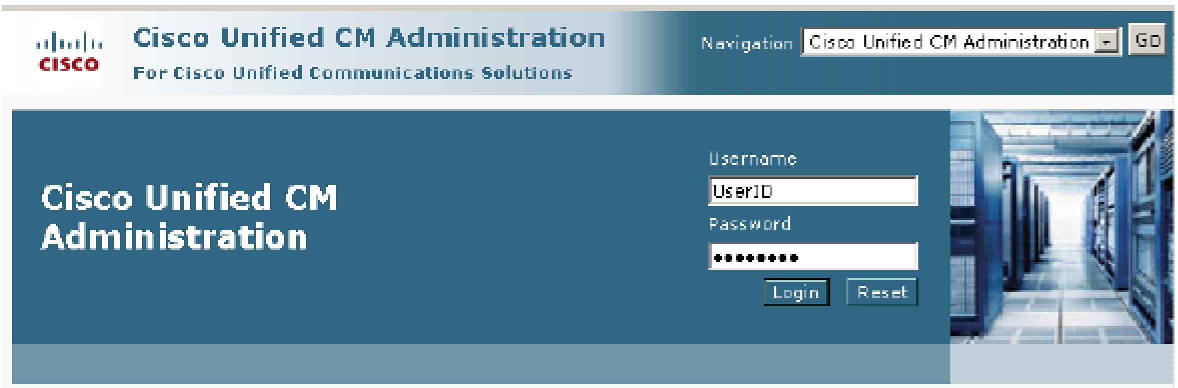


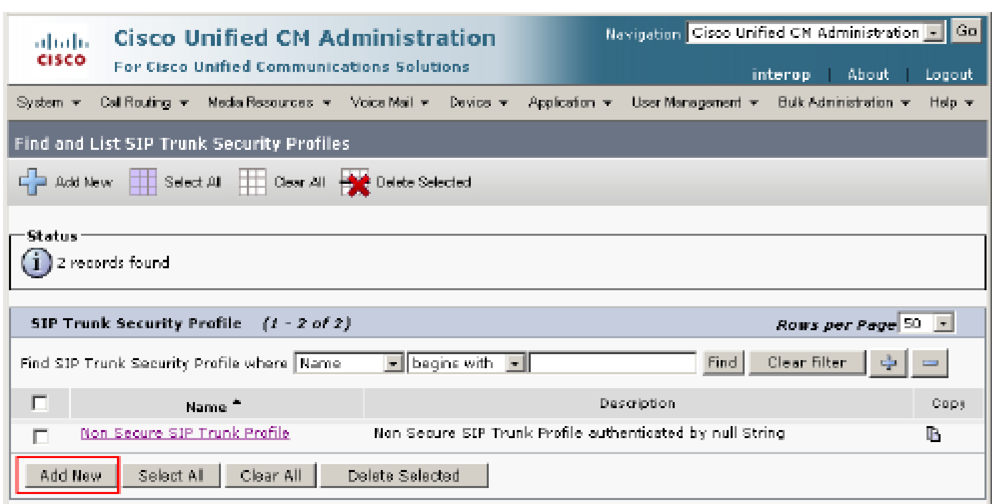
Figure 1: Login Screen for CISCO UCM

Step 2: Activate licenses purchased for SIP Trunking (if applicable)

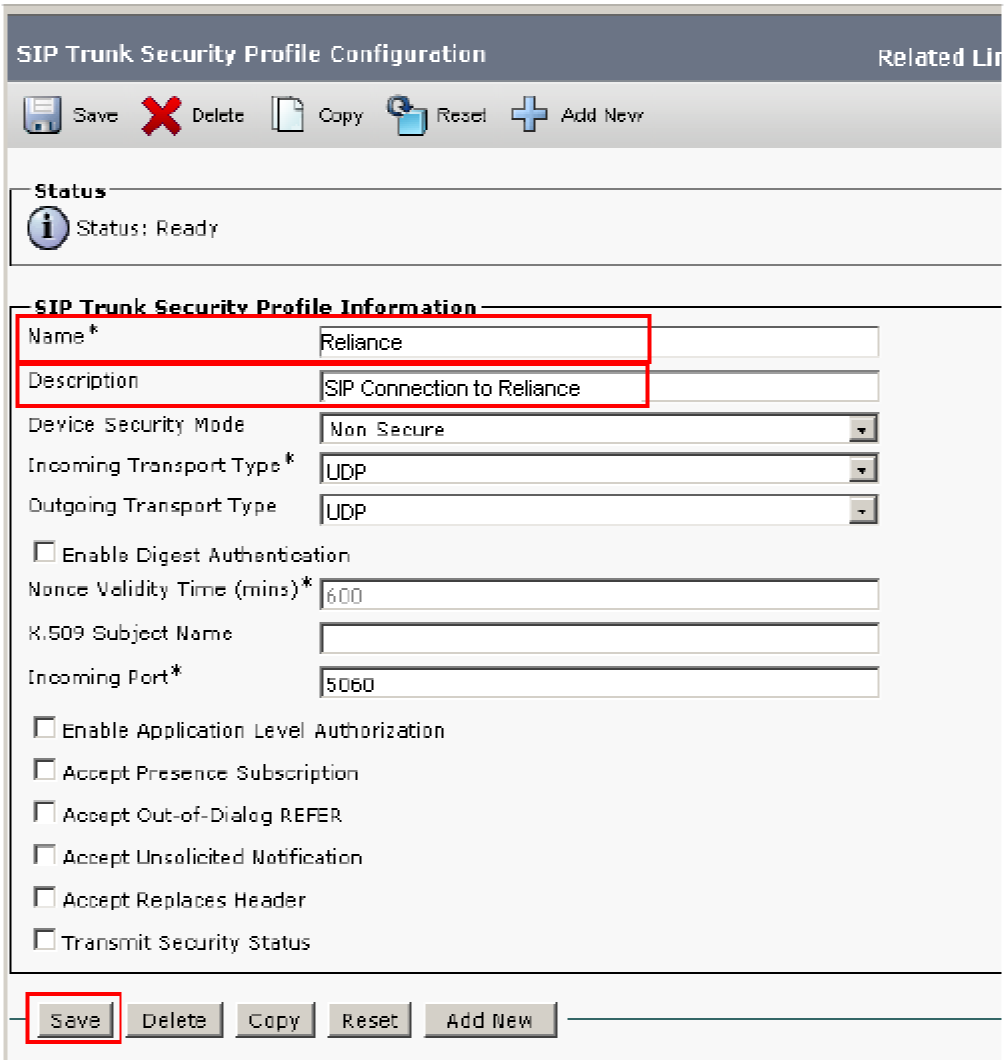
Step 3: Creation of SIP Trunk

Step 3a: Creation of Security Profile (Non Mandatory)

Select: System-> Security Profile - > SIP Trunk Security Profile from top menu and then click "Add New"

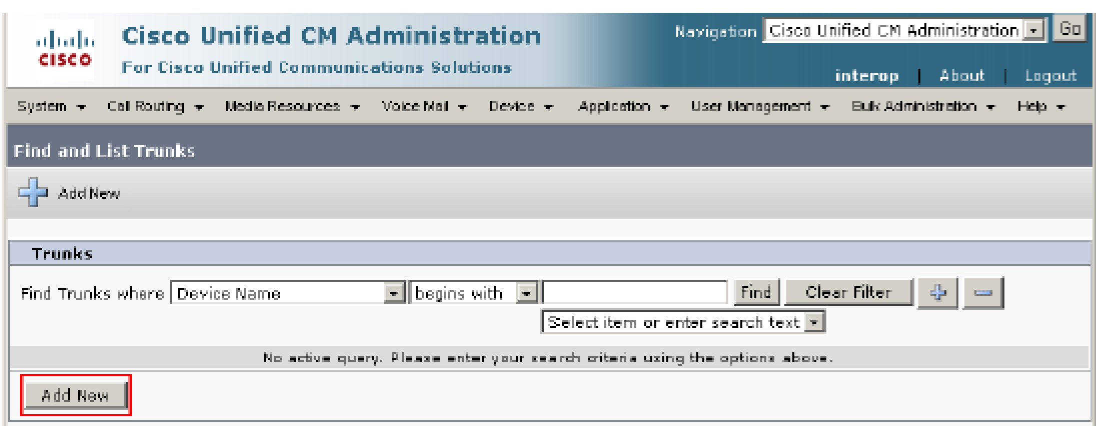


Make sure that you name the security profile appropriately

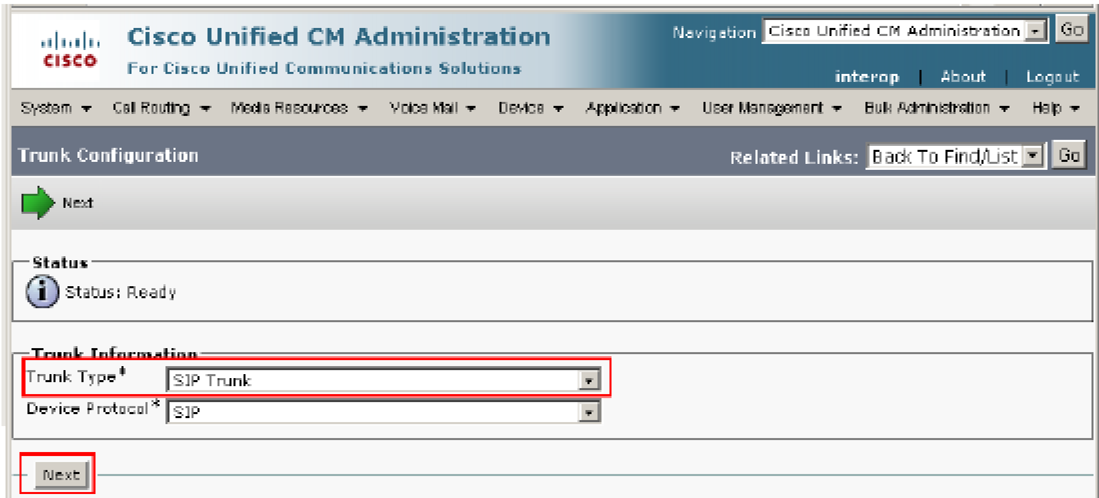


Step 3b: Creation of Trunk Group

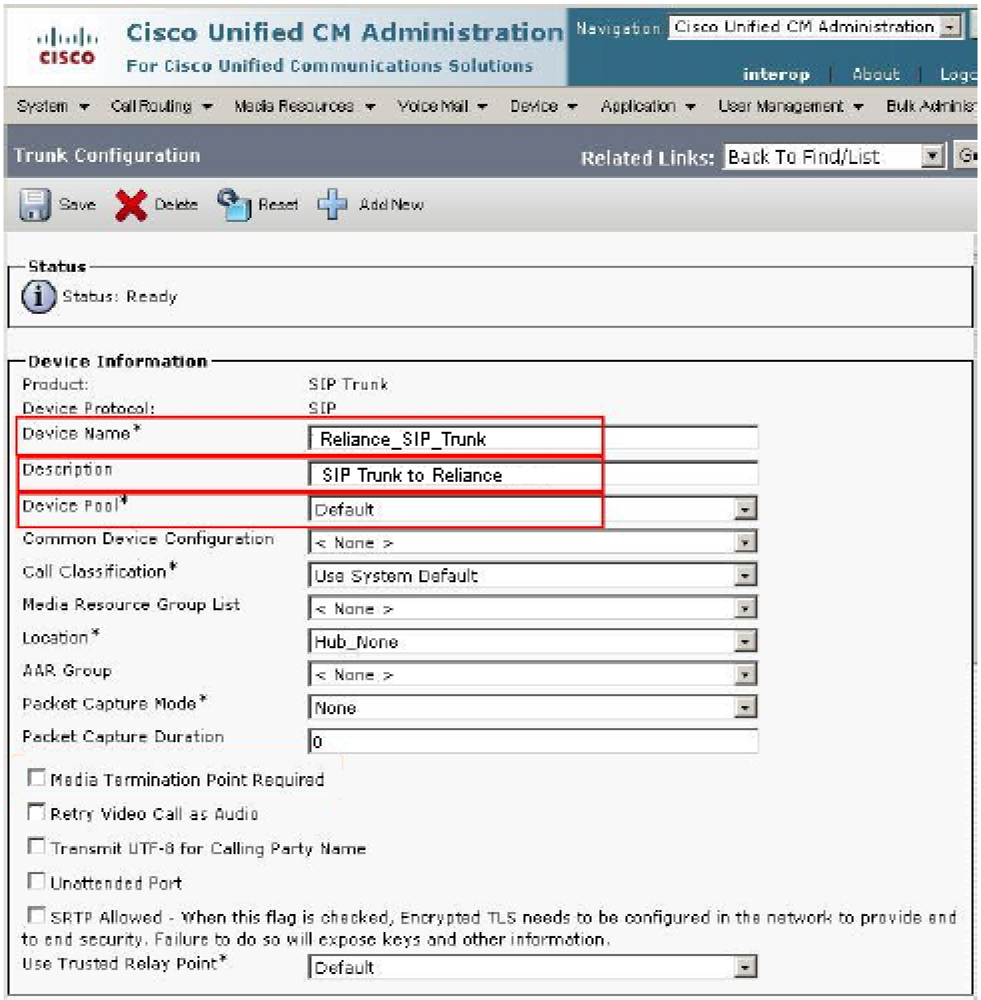
Select: Device -> Trunk from the top menu and click “Add New”

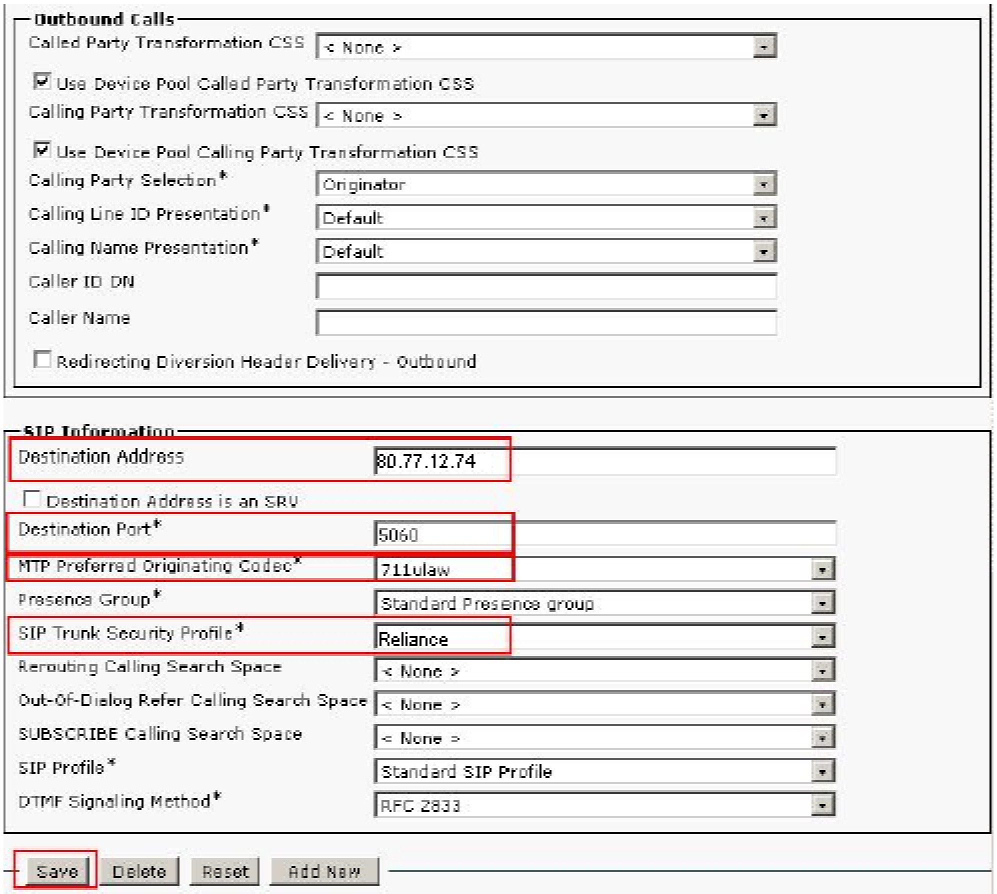


Select SIP Trunk as the Trunk Type and the Device Protocol field will automatically be change to SIP. Click Next to continue.



Step 3c: Enter the appropriate information for the SIP Trunk





Reliance Point of Presence	Destination IP on Reliance Network
London, UK Hong Kong Los Angeles, USA New York, USA	Please contact Reliance Account Manager for IP address detail

Step 4: CLI Configuration

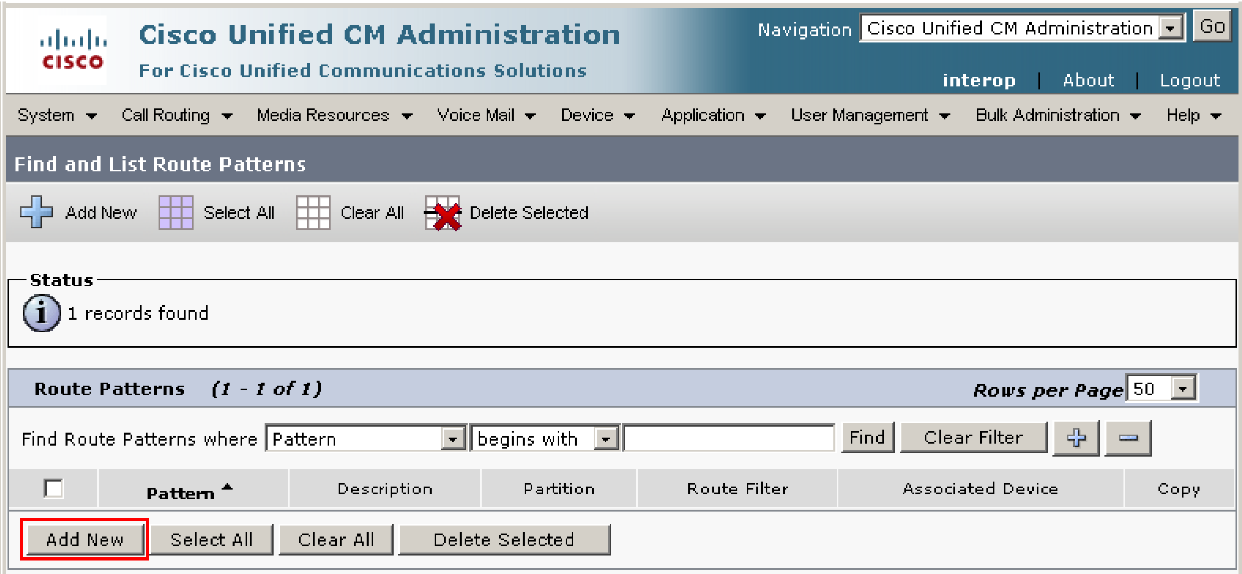
The CLI configuration guide on CISCO PBX is shown below. The fields and definition would change for different PBX platforms

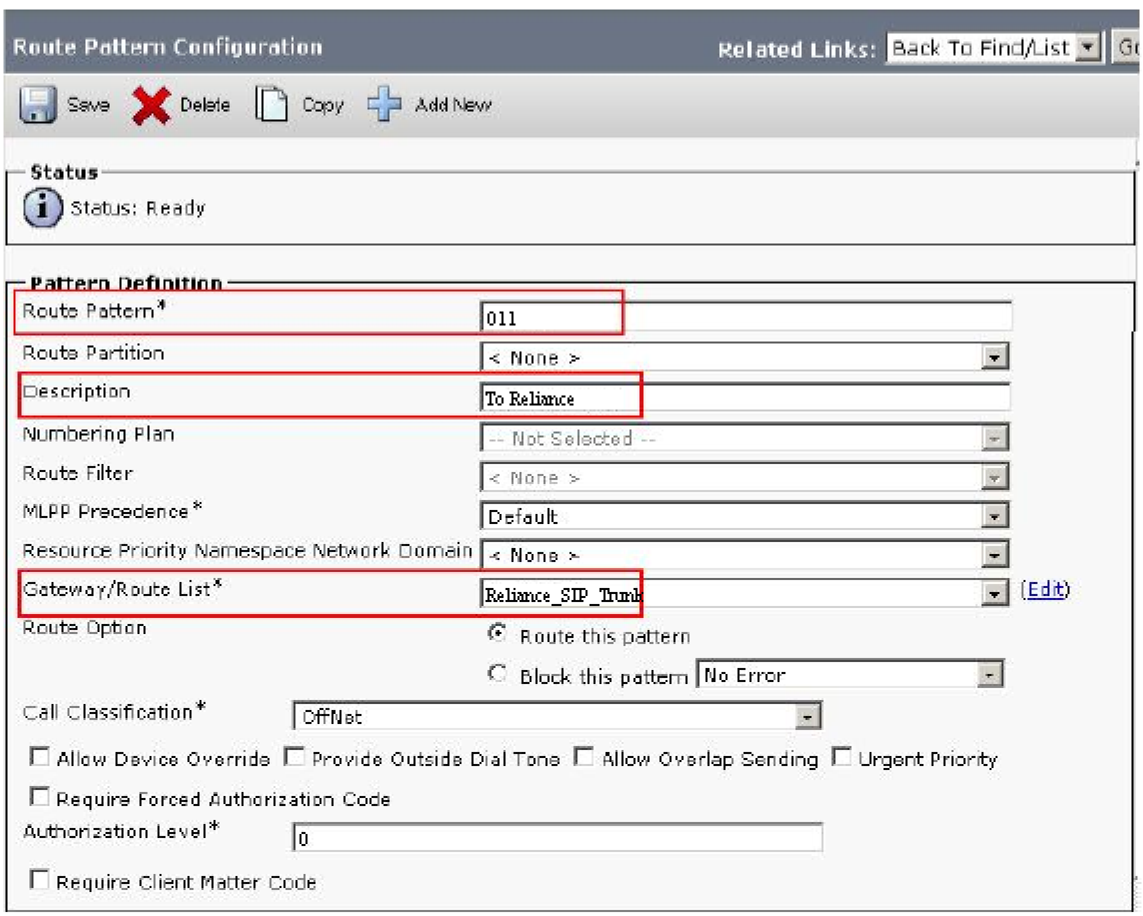
Field Name	Description
Use Calling Party's External Phone Number Mask	This field determines whether the full, external phone number is used for calling line identification (CLID) on outgoing calls. (Configure the external number by using the Directory Number Configuration window.) You can set the following Calling Party Transformations settings for the route group by clicking the members in the Route List Details panel of the Route List Configuration

	window: • Default: This setting indicates that the route group does not govern the calling party external phone number and calling party transform masks. If a calling party external phone number mask or transform mask is chosen for the route pattern, calls that are routed through this route group use those masks. • Off: This setting indicates that the calling party external phone number is not used for CLID. If no transform mask is entered for this route group, calls that are routed through this group do not get associated with a CLID. • On: This setting indicates that the calling party full, external number is used for CLID. The external phone number mask can contain up to 24 digits.
Calling Party Transform Mask	This field specifies the calling party transform mask for all calls that are routed through this route group. Valid values for this field range from 0 through 9, the wildcard character X, and the characters * and #. You can also leave this field blank. If it is blank and the preceding field is set to Off, this means that no calling party number is available for CLID. The calling party transform mask can contain up to 50 digits.
Prefix Digits (Outgoing Calls)	This field contains a prefix digit or a set of Prefix Digits (Outgoing Calls) that are appended to the calling party number on all calls that are routed through this route group. Valid values for this field range from 0 through 9, the characters * and #, and blank. Prefix Digits (Outgoing Calls) can contain up to 50 digits on route patterns or up to 24 digits on DNs.

Step 5: Dialing Rules Configuration

Select Call Routing -> Route/Hunt-> Route Pattern then click Add New to add a new route pattern for calls beginning with 011 which is the access code for international dialing in the USA





Note –

a. Call Termination happens via Cisco CM – Media termination option is required
b. End Users firewall should allow SBC IP
c. On Cisco CM 011 needs to be stripped off in its PBX setup – if customer uses 011 Dialing pattern for international calls