

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

helpdesk@relianceglobalcall.
com

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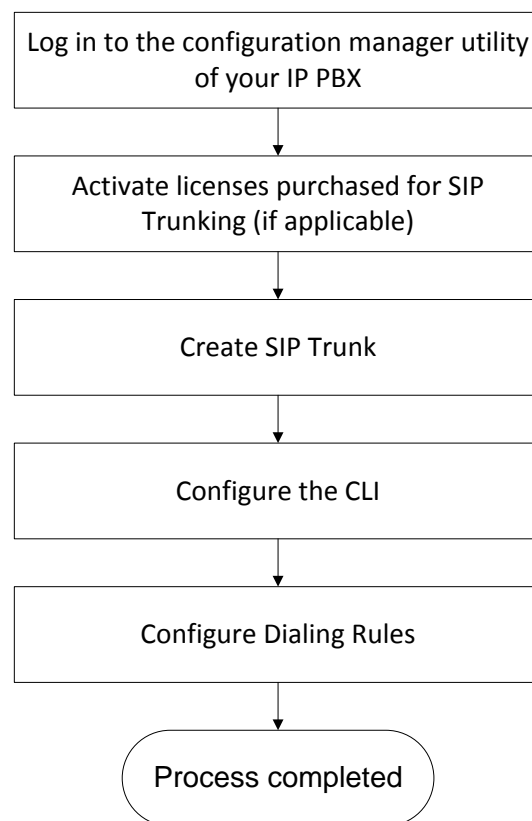
Prerequisites:

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:

Flow Chart for SIP Trunking Configuration



Detail SIP Trunking Configuration:

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

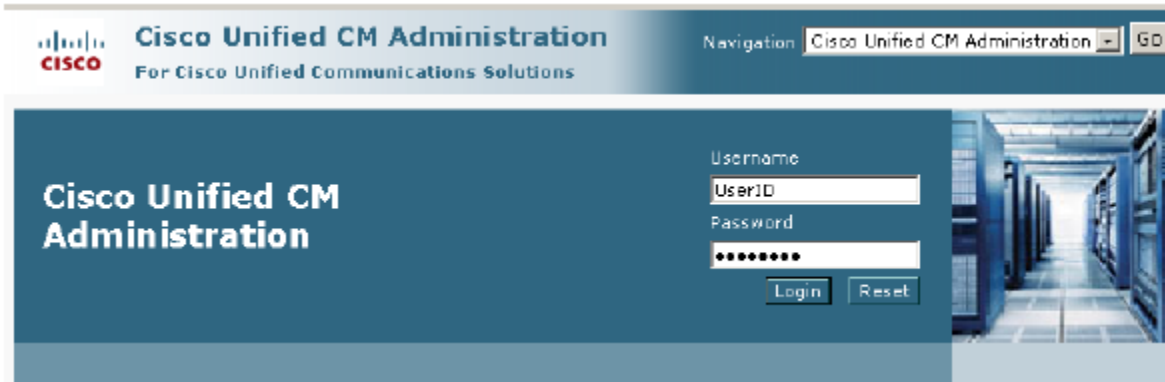


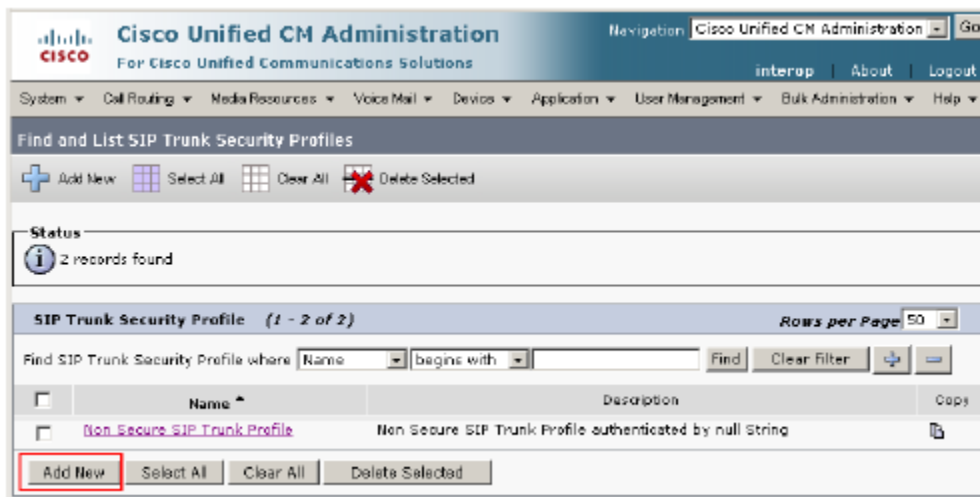
Figure 1: Login Screen for CISCO UCM

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

SIP Trunk Security Profile Configuration Related Links

Save Delete Copy Reset Add New

Status
Status: Ready

SIP Trunk Security Profile Information

Name *	Reliance
Description	SIP Connection to Reliance
Device Security Mode	Non Secure
Incoming Transport Type *	UDP
Outgoing Transport Type	UDP
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins) *	600
K.509 Subject Name	
Incoming Port *	5060
<input type="checkbox"/> Enable Application Level Authorization	
<input type="checkbox"/> Accept Presence Subscription	
<input type="checkbox"/> Accept Out-of-Dialog REFER	
<input type="checkbox"/> Accept Unsolicited Notification	
<input type="checkbox"/> Accept Replaces Header	
<input type="checkbox"/> Transmit Security Status	

Save Delete Copy Reset Add New

Step 3b: Creation of Trunk Group

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

Cisco Unified CM Administration Navigation: Cisco Unified CM Administration Go

For Cisco Unified Communications Solutions interop | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Find and List Trunks

+ Add New

Trunks

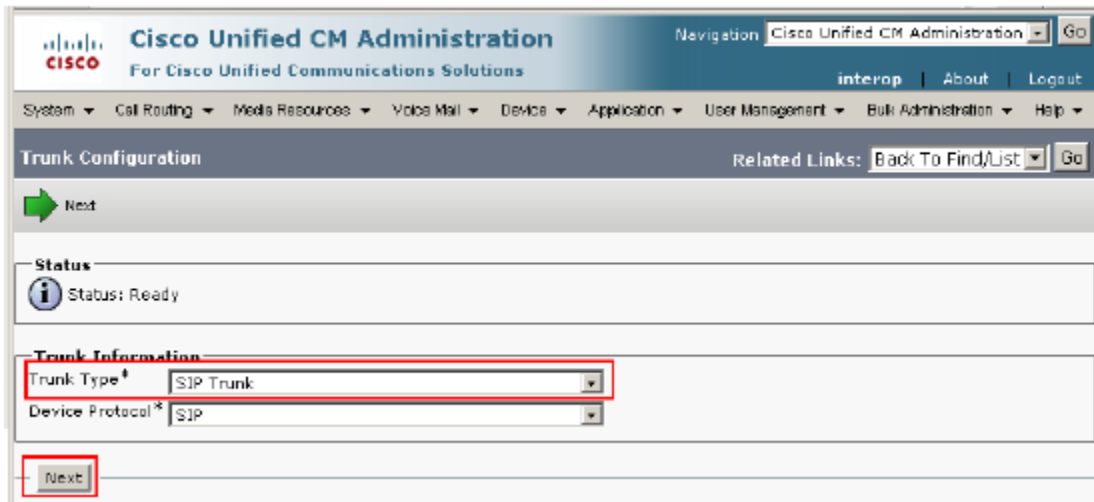
Find Trunks where Device Name begins with Find Clear Filter

Select item or enter search text

No active query. Please enter your search criteria using the options above.

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

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and it includes navigation menus for System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, and Bulk Admin. Below the navigation is a toolbar with Save, Delete, Reset, and Add New buttons. The main content area is divided into sections: Status (Ready), Device Information, and Common Device Configuration. The Device Information section is highlighted with a red box and contains the following fields:

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	Reliance_SIP_Trunk
Description	SIP Trunk to Reliance
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Use Trusted Relay Point*	Default

Outbound Calls

Called Party Transformation CSS < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection* Originator

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Caller ID DN

Caller Name

Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address 80.77.12.74

Destination Address is an SRV

Destination Port* 5060

MTP Preferred Originating Codec* 711ulaw

Presence Group* Standard Presence group

SIP Trunk Security Profile* Reliance

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Standard SIP Profile

DTMF Signaling Method* RFC 2833

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	<p>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.)</p> <p>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</p>

	<p>window:</p> <ul style="list-style-type: none"> • 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. <p>The external phone number mask can contain up to 24 digits.</p>
Calling Party Transform Mask	<p>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.</p> <p>The calling party transform mask can contain up to 50 digits.</p>
Prefix Digits (Outgoing Calls)	<p>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.</p>

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

The screenshot shows the 'Find and List Route Patterns' interface in Cisco Unified CM Administration. At the top, there are navigation tabs for 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. Below this is a search bar for 'Find Route Patterns where' with a dropdown set to 'Pattern' and a filter set to 'begins with'. A table below the search bar has columns for 'Pattern', 'Description', 'Partition', 'Route Filter', 'Associated Device', and 'Copy'. At the bottom of the table area, the 'Add New' button is highlighted with a red box.

The screenshot shows the 'Route Pattern Configuration' page. The 'Route Pattern' field is set to '011', the 'Description' is 'To Reliance', and the 'Gateway/Route List' is 'Reliance_SIP_Trunk'. These three fields are highlighted with red boxes. Other visible fields include 'Route Partition' (set to '< None >'), 'Numbering Plan' (set to '-- Not Selected --'), 'Route Filter' (set to '< None >'), 'MLPP Precedence*' (set to 'Default'), 'Resource Priority Namespace Network Domain' (set to '< None >'), 'Route Option' (set to 'Route this pattern'), 'Call Classification*' (set to 'OffNet'), and 'Authorization Level*' (set to '0').

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