# Cisco CallManager 4.1 SIP Trunk Configuration Guide

Valcom Session Initiation Protocol (SIP) VIP devices are compatible with Cisco Unified Communications Manager systems. For versions of Communications Manager that do not support SIP endpoints (such as CallManager version 4.1.3), the Valcom device may be added to the system as a SIP Trunk.
The following steps outline the typical configuration process for CallManager 4.1.3:

Media Termination Point

1. Logon to the Cisco CallManager server. The SIP Trunk feature requires a Media Termination Point (MTP) to be available. If the MTP is already configured, skip to Step 5.
    Click the **Cisco CallManager Serviceability** option on the **Application** menu.
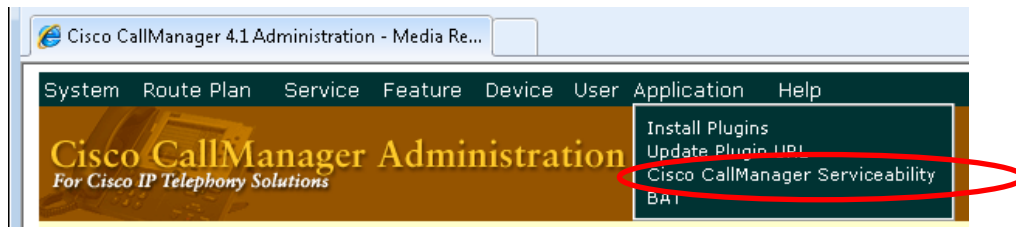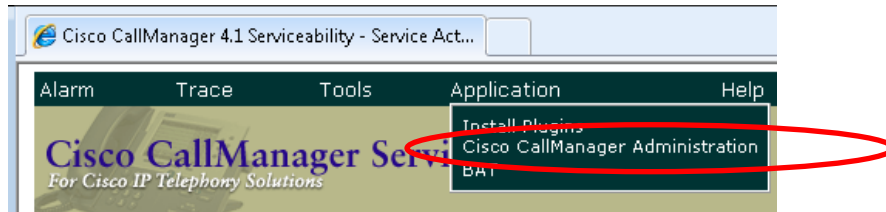

2. Click the **Service Activation** option on the **Tools** menu.
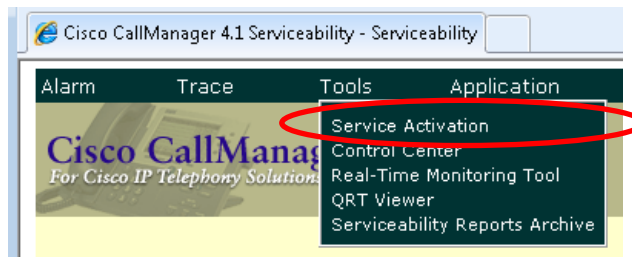
3. On the Service Activation page, confirm the **Cisco IP Voice Media Streaming App** is Activated. If it is not Activated, click the checkbox beside the service and then click Update.
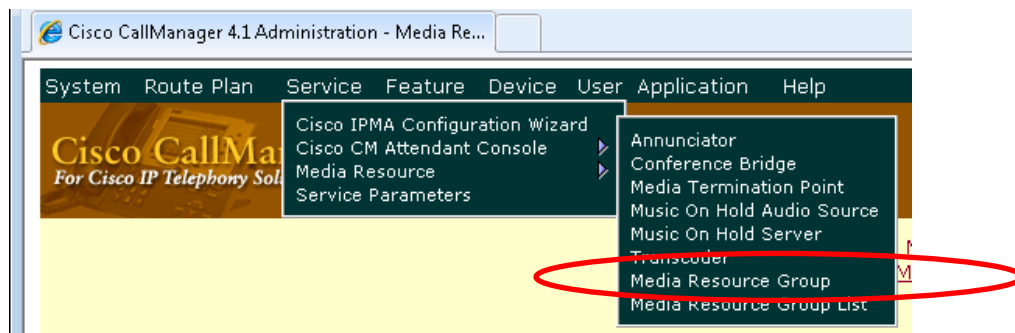
4. Return to CallManager Administration. On the **Application** menu, click the **Cisco CallManager Administration** option.

Media Resource Group

5. If a Media Resource Group with a Media Termination Point is not available for this Valcom SIP device, use the following steps to create one. Click the **Service** menu, then click **Media Resource**, then click **Media Resource Group**.

6. On the Media Resource Group page, click **Add a New Media Resource Group**. The following screen shows the Media Resource Group page with entries for this example. The Media Resource Group Name has been entered as "VIP-172L" with an appropriate Description. In the Devices for This Group section, the Media Termination Point named "MTP_CCM-4-1-3(MTP)" has been moved from the Available Media Resources list to the Selected Media Resources list. After making the appropriate entries, click the **Insert** button to create the Media Resource Group.

Media Resource Group List

7. The Media Resource Group just created must be included in a Media Resource Group List. Click the **Service** menu, then click **Media Resource**, then click **Media Resource Group List**. On the Find and List Media Resource Group Lists page, click the link to **Add a New Media Resource Group List**.

   The following screen shows the Media Resource Group page with entries for this example. The Media Resource Group List Name has been entered as "Valcom". In the Media Resource Groups for this List section, the Media Resource Group named "VIP-172L" has been moved from the Available Media Resource Groups list to the Selected Media Resource Groups list. After making the appropriate entries, click the **Insert** button to create the Media Resource Group List.

   This List is used in the SIP Trunk configuration to define the Media Termination Point for the trunk.
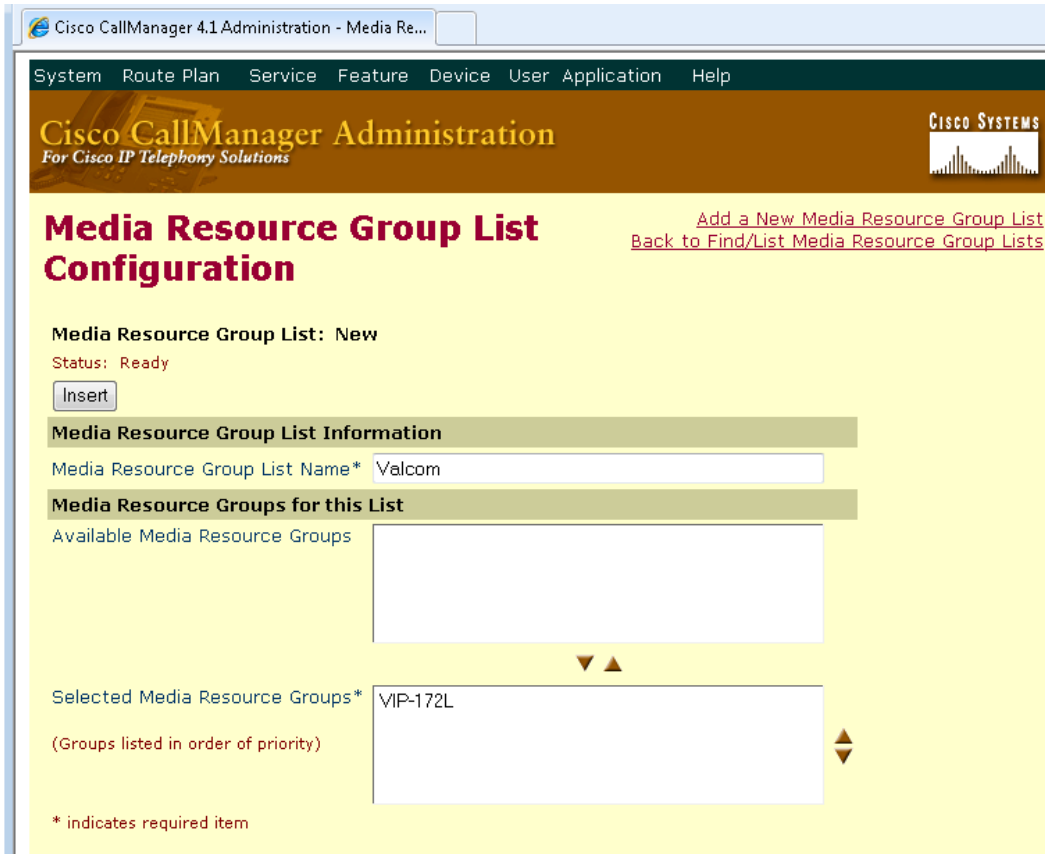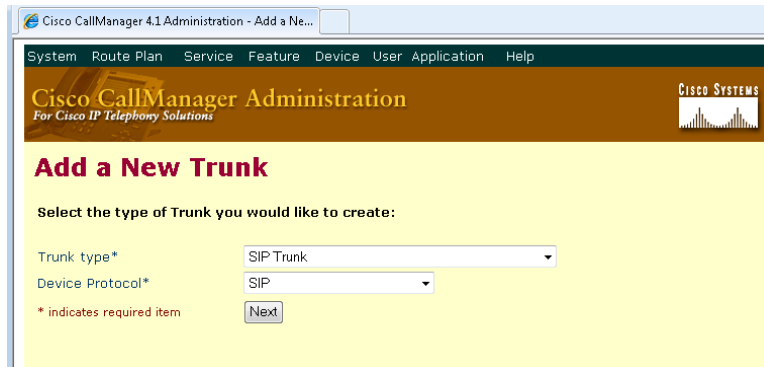
SIP Trunk

8. With the Media Termination Point determined, the SIP Trunk can be created. Click the **Trunk** option on the **Device** menu. On the Find and List Trunks page, click Add a New Trunk.

9. On the Add a New Trunk page, select **SIP Trunk** for Trunk type. For SIP Trunks, the Device Protocol option must be SIP. Make these selections, then click **Next**.

10. On the Trunk Configuration page, complete the entries appropriate for the Valcom device. Each Valcom SIP device will require a separate SIP Trunk definition. The entries marked with an asterisk are required.
    The Media Resource Group List is the list created in a previous step. Our example uses "Valcom".
    The Destination Address is the IP Address of the Valcom device.
    The **Destination** and **Incoming Ports** default to 5060, but may be different if other SIP trunks are installed. The Incoming Port must be unique for every SIP trunk. Note the values listed so that they may be entered into the Valcom device correctly.
    The **Outgoing Transport Type** must be **UDP**.
    The **Preferred Originating Codec** must be **711ulaw** for Valcom SIP devices.

    The Call Routing Information section should be completed with values appropriate to the CallManager system and dial plan in use. For this example, we are using a four-digit dial plan, and will create a Route Pattern using exactly four digits. The Significant Digits field will be selected as "All", meaning all of the digits entered at the Cisco phone set will be passed through the SIP Trunk to the Valcom device. If a different dial plan is in use, then this entry should be set to match the requirements. Other entries, such as Calling Search Space, must be determined by the administrator based on the CallManager configuration.

    After all entries are made, click the **Insert** button to create the SIP Trunk.

Cisco CallManager 4.1 Administration - Trunk Co...

System | Route Plan | Service | Feature | Device | User | Application | Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Trunk Configuration

Add a New Trunk
Back to Find/List Trunk
Dependency Records

Product: SIP Trunk
Device Protocol: SIP
Status: Ready

[Insert]

Device Information

Device Name*	VIP-172L
Description	VIP-172L Test
Device Pool*	Default
Call Classification*	OnNet
Media Resource Group List	Valcom
Location	< None >
AAR Group	< None >

☑ Media Termination Point Required

Destination Address*	192.168.96.222

☐ Destination Address is an SRV

Destination Port	5060
Incoming Port*	5060
Outgoing Transport Type*	UDP
Preferred Originating Codec*	711ulaw

Call Routing Information
Inbound Calls

Significant Digits*	All
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	

☐ Redirecting Number Delivery - Inbound

Outbound Calls

Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	

☐ Redirecting Number Delivery - Outbound

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain	(e.g., "0000FF")
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device

* indicates required item

Back to Find/List Trunk

Route Pattern

11. A Route Pattern must be entered to route calls to and from the SIP Trunk. Click the **Route Plan** menu, then click **Route/Hunt**, then click **Route Pattern**. On the Find and List Route Patterns page, click **Add a New Route Pattern.**
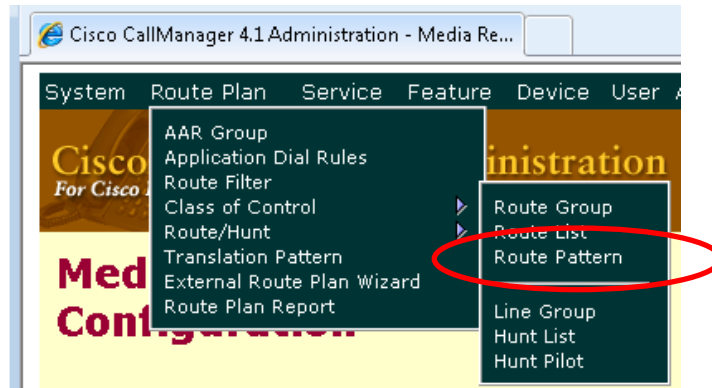
12. The Route Pattern Configuration page is used to define what phone calls are sent to the SIP Trunk. The pattern used is dependent on the configuration of the CallManager system and the device at the other end of the SIP trunk. This example uses a Valcom VIP-172L as the SIP device being "trunked" to, and the VIP-172L can have one phone number assigned to it. So, our example will use a basic four-digit pattern to route exactly one phone number to the trunk. Other Valcom devices, such as the VIP-201 or VIP-204, have multiple SIP phone numbers. The Route Pattern used with those devices will likely be different. The use of wildcard characters in the Route Pattern would allow one Pattern to work for multiple phone numbers. Each Route Pattern sends calls to a single IP address, so each Valcom IP device will require its own Route Pattern.

The Route Pattern entry will match the phone number assigned to the Valcom VIP-172L, which is "6003".
The Gateway or Route List dropdown box is changed to match the SIP Trunk created in a previous step, which we named VIP-172L.
The checkbox for Provide Outside Dial Tone was unchecked, which automatically changed the Call Classification to OnNet.

All of the Transformation settings were left at the default values for this example.

After making the appropriate entries, click the Insert button to create the Route Pattern.

Cisco CallManager 4.1 Administration - Route Pa...

System   Route Plan   Service   Feature   Device   User   Application   Help

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Route Pattern Configuration

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

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

[Insert]

Pattern Definition

Route Pattern*	6003
Partition	< None >
Description	Route to VIP-172L on SIP Trunk
Numbering Plan*	North American Numbering Plan
Route Filter	< None >
MLPP Precedence	Default
Gateway or Route List*	VIP-172L

Route Option   ● Route this pattern
                 ○ Block this pattern — Not Selected —

Call Classification*   OnNet   ☐ Allow Device Override

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

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

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

Connected Party Transformations

Connected Line ID Presentation	Default
Connected Name Presentation	Default

Called Party Transformations

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

ISDN Network-Specific Facilities Information Element

Carrier Identification Code	
Network Service Protocol	— Not Selected —

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

* indicates required item.

13. Open the Valcom VIP-102B IP Solutions Setup tool and select the Valcom SIP-enabled VIP device.

Note: The information contained in this guide is limited to configuration of the "SIP" tab in the VIP-102B IP Solutions Setup Tool for the Valcom VIP device that is to be registered to the SIP server. More information on Valcom VIP device configuration, such as IP address assignment, relay activation, etc, may be found in the VIP-102B Reference Manual. This document may be downloaded from our website at http://www.valcom.com

Required Fields: Phone Number, SIP Server, SIP Server Port, SIP Port, RTP Port, Auto Destination

In our example, the SIP Server IP address is the Cisco CallManager, "192.168.97.55", and the Register check box is <u>not</u> checked.
Phone Number is the Route Pattern in the Cisco CallManager configuration created in step 12, "6003".
SIP Server Port is the port number on which the Valcom device will transmit SIP data and must match the value of the Incoming Port on the SIP Trunk configuration in step 10.
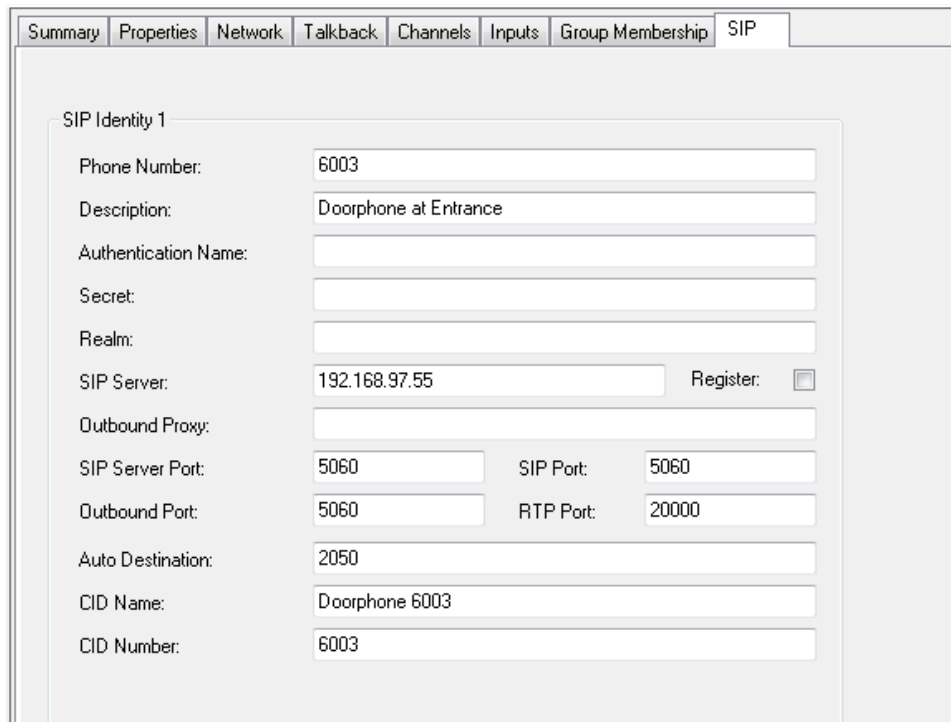SIP Port is the port number on which the Valcom VIP device is listening for SIP data and must match the Destination Port on the SIP Trunk configuration in step 10.
RTP Port is the port number on which the Valcom VIP device is set to send/receive audio packets. The default is "20000".
The Auto Destination field is the phone number to which the Valcom device will "call" when activated.
The CID Name and CID Number (Caller ID) are optional fields.

After entering the required values, select the "Update Changed Devices" button, at the upper left. When the update is complete, click reset to reboot the device. Test the configuration by making calls to and from the Valcom device.