

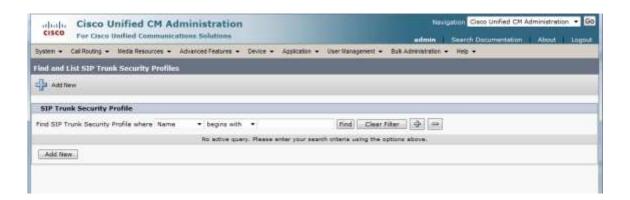
Cisco Unified Communications Manager SIP Trunk Configuration Guide

Valcom PagePro SIP (Session Initiation Protocol) Paging Servers, models VIP-201 and VIP-204, are compatible with Cisco Unified Communications Manager as either a Third-party SIP Device (Basic or Advanced) or as a SIP Trunk. This configuration guide provides information for configuring the PagePro server as a trunk endpoint. The screen illustrations are based on a Cisco Unified Communications Manager server version 12.5, but the instructions are similar for other versions. The following steps outline the typical configuration process.

1. Navigate your web browser to the IP address of your Cisco Unified Communications Manager server and login. Go to the "System" menu, and then click "Security", then click "SIP Trunk Security Profile".

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2. Click on "Add New"





- 3. On the SIP Trunk Security Profile Configuration screen, enter the appropriate values for the trunk. For Valcom devices, the Outgoing Transport Type must be UDP. The Incoming Port defaults to 5060, but can be changed. If it is changed here, then it must also be changed to the same value in the Valcom device configuration. Required information:
 - A) Enter "Name*" (ex. Valcom SIP Trunk)
 - B) For "Device Security Mode", select "Non Secure" from the dropdown list
 - C) For "Incoming Transport Type*", select "TCP+UDP" from the dropdown list
 - D) For "Outgoing Transport Type*" select "UDP" from the dropdown list
 - E) The "Incoming Port*" should be left at the default of 5060, unless it is also changed in the Valcom device.
 - F) Click the "Save" button when all fields have been entered.

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SIP Trunk Security Profile Configuration	
Save	
- Status	
(i) Status: Ready	
SIP Trunk Security Profile Information	
Name*	Valcom SIP Trunk
Description	Trunk Connection to VIP-201
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP V
Outgoing Transport Type	(TCP 👻
Enable Digest Authentication Nonce Validity Time (mins)*	600
Secure Certificate Subject or Subject Alternate N	lame
Incoming Port*	5060
Enable Application level authorization	
Accept presence subscription	
Accept out-of-dialog refer**	
Accept unsolicited notification	
Accept replaces header	
C) Transmit security status	
CAllow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter 👻

4. The Trunk Security Profile just created will be used when creating the Trunk. Go to the "Device" menu, then click on "Trunk".

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Add New	

- 5. On the Find and List Trunks page, click on "Add New".
- 6. Enter the Trunk Information.
 - A) Select "SIP Trunk" for Trunk Type
 - B) Select "SIP" for Device Protocol
 - C) Select "None" for Trunk Service Type (not all versions have this parameter)
 - D) Select "Next" at the top or bottom of the screen

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Trunk Configuration			12	Related Links: Back To Find/List • Go
Next				
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- Trunk Information	SIP Trunk			
Device Protocol*	SIP			
Trunk Service Type*	None(Default)	•		
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(i) * indicates req	aired item.			

- 7. On the Trunk Configuration page, enter the specific device information. Configuration items on the web page marked with an asterisk (*) are required entries. Complete those items with values appropriate for your site, particularly for items such as Device Pool, Calling Search Space, Location, etc. The device-specific items are mostly in the SIP Information area near the bottom of the screen.
 - A) In the Destination Address field, enter the IP address and Port assigned to the Valcom PagePro server
 - B) Confirm the Codec selection is "711ulaw"
 - C) For SIP Trunk Security Profile, select the profile created in Step 3 (example: Valcom SIP Trunk)
 - D) Select "Standard SIP Profile" for the SIP Profile
 - E) DTMF Signaling Method should be "RFC 2833"
 - F) Click the "Save" button when all items have been entered

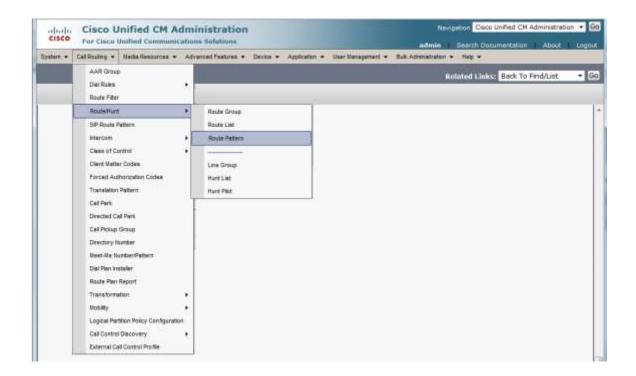


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System - Call Routing - Media Resource	es - Advanced Features	- Device - Ap	oplication - User Management -	Bulk Administration +	Help -			
Trunk Configuration								
Save		_			_	_		
Status Status: Ready								
Device Information								
Producti		SIP Trunk						
Device Protocols Trunk Service Type		SIP None(Default)						
Device Name*		PagePro-VIP-2	201					
Description		Valcom PageP						
Device Pool*		Default	19.941141					
Common Device Configuration		< None >		~				
Call Classification*		Use System C	efault.	~				
Hedia Resource Group List		< None >		*				
Location		Hub_None		*				
AAR Group		< None >		~				
Tunneled Protocol*		None		Y				
Q5EG Variant*		his Changes		. v				
ASN.1 ROSE OID Encoding		Top Charges		(W)				
Packet Cepture Node*		None		*				
Packet Capture Duration		0						
Hedia Termination Point Required								
Retry Video Call as Audio								
Path Replacement Support								
Transmit UTF-8 for Calling Party Nam								
Transmit UTF-8 Names in QSIG APDU	1							
Unattended Port								
SRTP Allowed - When this flag is cher	cked, Encrypted TLS need			to end security. Failure to	do so will expose key	s and other informat	bon.	
Consider Traffic on This Trunk Secure*			offs aRTP and TLB					
Route Class Signaling Enabled* Use Trusted Relay Point*		Default		~				
Contract of the second second		Default						
PSTN Access								
Run On All Active Unified CM Nodes								
-SIP Information								
Destination								
Destination Address is an SRV								
Destination A	ddress	04	Destination Address IPv6	Desti	nation Port	Status	Status Reason	Duration
1* 192.168.100.77				\$060		N/A	N/A	N/A
MTP Preferred Originating Codec*	TILINE		Ψ.					
BLF Presence Group*	Star-dard Presence grou	ø	~					
SIP Trunk Security Profile*	Non Secure SIP Trunk P	rofile						
Rerouting Calling Search Space	< None >		*					
Out-Of-Dialog Refer Calling Search Space	Statistical Statistics		*					
SUBSCRIBE Calling Search Space	< None >							
53P Profile* DTMF Signaling Hathod*	Standard SIP Profile		View Details					
Drive signaling Hatrion	No Preference							
Normalization Script								
Normalization Script < None >								
Enable Trace								
Parameter Na	ime		Parameter Value	1000 0001				
1				(H) (H)				
CRecording Information								
Tipne								
This trunk connects to a recording-	enabled gateway							
O This trunk connects to other cluster		gateways						
1		Terraria and						
Geolocation Configuration								
Geolocation (< None >		~						
Geolocation Filter < None >		*						
Send Geolocation Information								
Annestrations								
Save								



8. With the Trunk device created, a Route Pattern will need to be created to send calls to the trunk. Go to the "Call Routing" menu, then click on "Route/Hunt", then click on "Route Pattern"



9. On the Find and List Route Patterns screen, click on "Add New".

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Route Patterns			
ind Route Patterns where Pattern	+ begins with +	find Clear filter 🌵 🚥	
	No active query. Please enter y	our search criteria using the options above.	
Add New			



10. Configuration items on the Route Pattern Configuration web page marked with an asterisk (*) are required entries. Complete those items with values appropriate for your site, particularly for items such as Device Pool, Calling Search Space, Location, etc.

Complete the following steps:

- A) Enter a Route Pattern appropriate for the Directory Numbers that will be assigned to the Valcom PagePro. For our example "600X" will be used to route all numbers from 6000 to 6009 to the Valcom trunk.
- B) For the Gateway/Route List, select the Valcom trunk device that was created in Step 7. Our example device is named "PagePro-VIP-201".
- C) Confirm the Route Option selection is "Route this pattern"
- D) Call Classification should be "OnNet"

🔜 Save 💥 Delete [Copy 👍 Add			
🚺 Status: Ready			
Pattern Definition			
Route Pattern*	600X		
Route Partition	< None >	~	
Description	Valcom PagePro 6000-6009		
Numbering Plan	Not Selected	¥	
Route Filter	< None >	~	
MLPP Precedence*	Default	v	
Apply Call Blocking Percentage			
Resource Priority Namespace Network Doma	n < None >	~	
Route Class*	Default	~	
Gateway/Route List*	PagePro-VIP-201	*	(Edit
Route Option	Route this pattern		
	O Block this pattern No Error	~	
Call Classification* OnNet	*		
External Call Control Profile < None >	~		
Allow Device Override Provide Outside	Dial Tone 🗌 Allow Overlap Sending 🔲 Urgent P	riority	
Require Forced Authorization Code			

- E) Provide Outside Dial Tone options should be unchecked
- F) Select "Save" at the top or bottom of the screen.

This completes the configuration on the Cisco Communications Manager.



B)

11. Open the VIP-102B IP Solutions Setup Tool interface for the Valcom PagePro device.

Note: The information presented here is limited to configuration of the "SIP" tab in the VIP-102B IP Solutions Setup Tool for the Valcom PagePro 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 <u>http://www.valcom.com</u>

A) On the SIP tab, set the SIP Mode to "Trunk" using the dropdown box in the upper right corner

			-						4
	Scatun Name	£	-						4
Secret									4
Ready			-						
SP Se	rven:		- Prinary	Server 192.168.98	20		Part 5010		
+	Register.		Backup 1	1.446-0.446.044			5060		
t	INS SRV:		Beckup 2				5060		
		120	Backup 3	1			5060		
input V				0	Output Volume Output Volume	5060	-	- 10	0
input V Outboo	tokame and Proxy:	feat	3600	0	Outbound Port.	-			3
input V Outboo	hilame and Proxy: Nive Timer (pec	celt	3600	- 0		-			0
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input V Clubou Keep A SIP Po Beginn	tolume and Proxy: Yive Timer (sec if:		5060	0	Outbound Port.	-			
input V Clubou Keep A SIP Po Beginn	hitene and Proxy: Hive Timer (sec rt: ang RTP Port:		5060	0	Outbound Port.	-			
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input V Clubou Keep A SIP Po Beginn	hitene and Proxy: Hive Timer (sec rt: eng RTP Port		5060		Outbound Port.	-		Add Add Rarge	

For this example, the trunk connection is not using authentication, so the Authentication Name or Secret fields do not need to be completed. At a minimum, required entries are the SIP Server Port and at least one Extension (phone number). The SIP Server Port must match the Destination Port in the SIP Information for the trunk created in Step 7. The SIP Server field can be used to document which SIP server this PagePro is connected with.

- C) For our example, we are using a range of Extensions (phone numbers). These can be entered in one operation using the "Add Range" button. Clicking that button brings up the dialog box shown below.
- D) In the Lower Extension field, enter the beginning phone number (6000 for our example)
- E) In the Upper Extension field, enter the ending phone number (6009 for our example)
- F) Other options in this window may be selected if desired. Any other options, such as Night Ring or an Audio Group assignment, will be applied to all of the Extensions in the range.



G) Click the OK button to create the range of Extensions and return to the main SIP window.

Lower Edension:	6000
Upper Extension	eooal
Night Heng	10
Pre-Announce Tone:	9 Son and Play El Hay Caust 1 + 0
Announce To	Available •
Audo Genues	1000

H) After entering the range, the main SIP window will display the Extensions this PagePro server will recognize. Other extensions may be added at any time. Extensions do not have to be in a contiguous range.

Summary	Properties	Network	Channels	Relays	Group	Membership	SIP	
							SIP Mode	e: Trunk
	Phone Numb	er:						
	Authenticatio	n Name:						
	Secret:							
	Realm: SIP Server:		192.	168.98.40)			Register:
	Outbound Pr	oxy:						
	SIP Server P	ort:	5060			SIP Port:	50	62
	Outbound Po	ort:	5060			RTP Port	: 20	000
	Extensions:	(10 presen	t)					_
	6000 6001 6002 6003 6004 - Night 6005 - Store 6006 6007 6008 6009	t Ring-9998 &Play-9997	377					Add Add Range Edit Delete All
								<u>C</u> ancel



Cisco Unified Communications Manager SIP Trunk Configuration Guide

- I) To associate an Extension to a paging group or to set a feature such as Night Ring on an Extension, highlight the individual Extension to be modified, then click the Edit button. This will display the dialog box shown below. Functions such as Night Ring or Store and Play can be enabled by checking the box for that option. Paging Groups that have been created in the system will be displayed in the Audio Groups box. To associate this SIP Extension with an audio group, click the checkbox beside the desired group number.
- J) Click the OK button when done.

g SIP Extension	Press Parente
Extension	6002
Description:	1 N
Night Hong	2
Pre-Announce Tone	2 Severit Rep 10 Paylouts (1 + 0)
Announce Ta	Available +
Auto Genzes 	22 5825 - 00 Test
	QK Garcel

When all configuration options have been set as desired, update the PagePro device configuration using the update commands from the Communications menu of the VIP-102B IP Solutions tool.

To verify operation, use a telephone on the Communications Manager to call one of the phone numbers assigned to the Valcom PagePro server and confirm the call is connected.