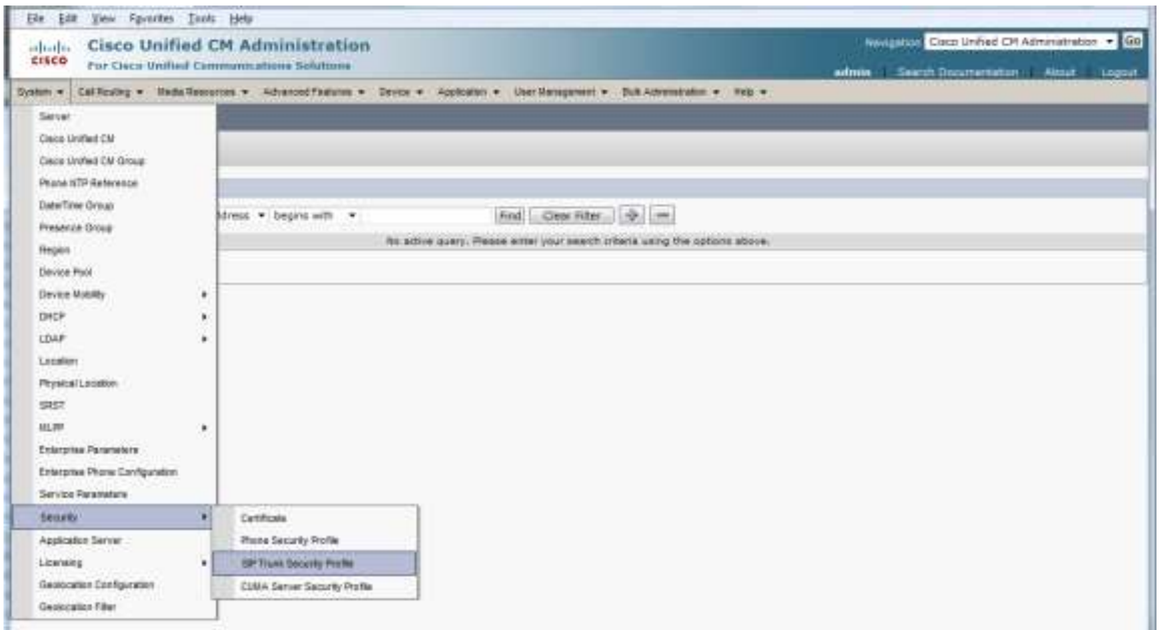
Cisco Unified Communications Manager SIP Trunk Configuration Guide

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.

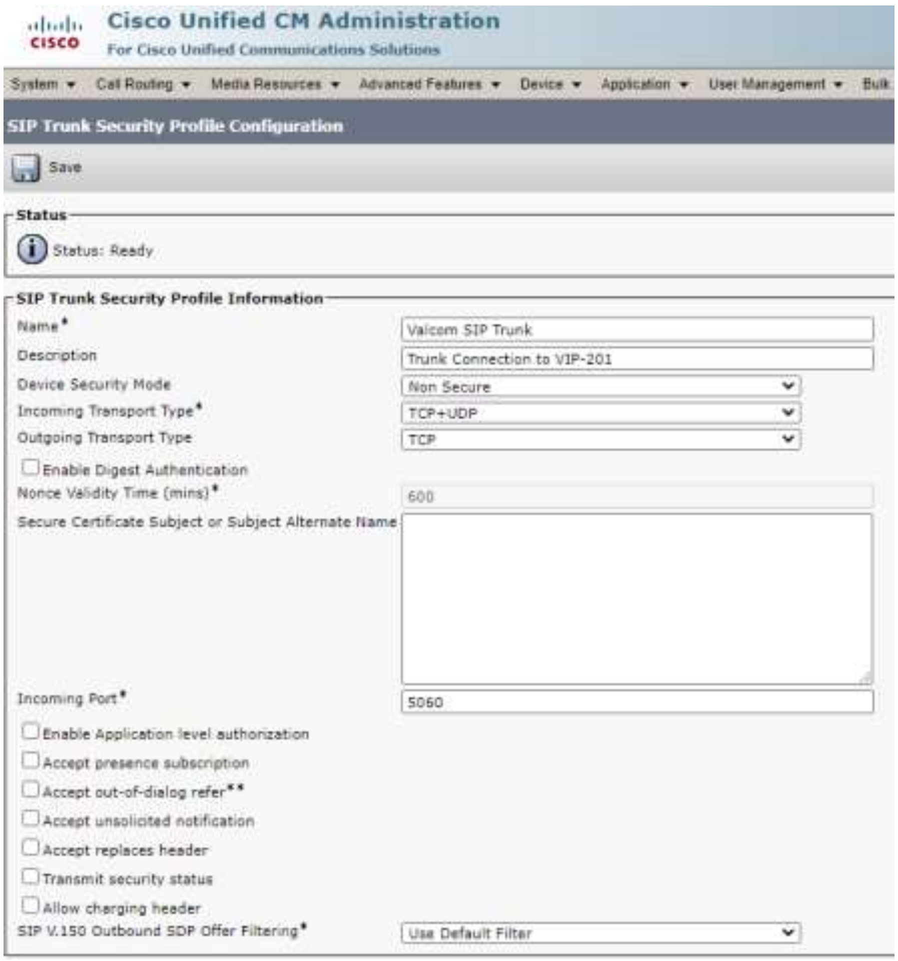
1. Navigate your web browser to the IP address of your Cisco Unified Communications Manager server and login. Go to the “System” menu, and then click “Security”, then click “SIP Trunk Security Profile”.



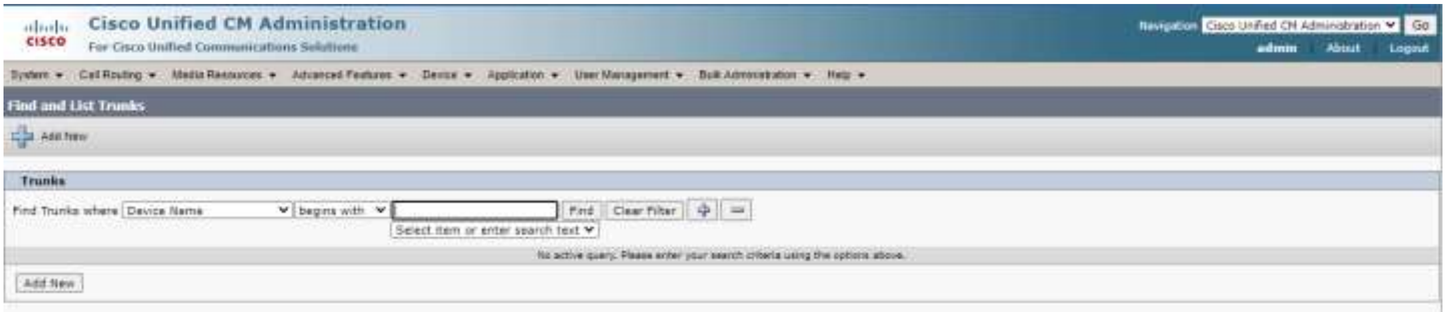
2. Click on “Add New”

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.



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



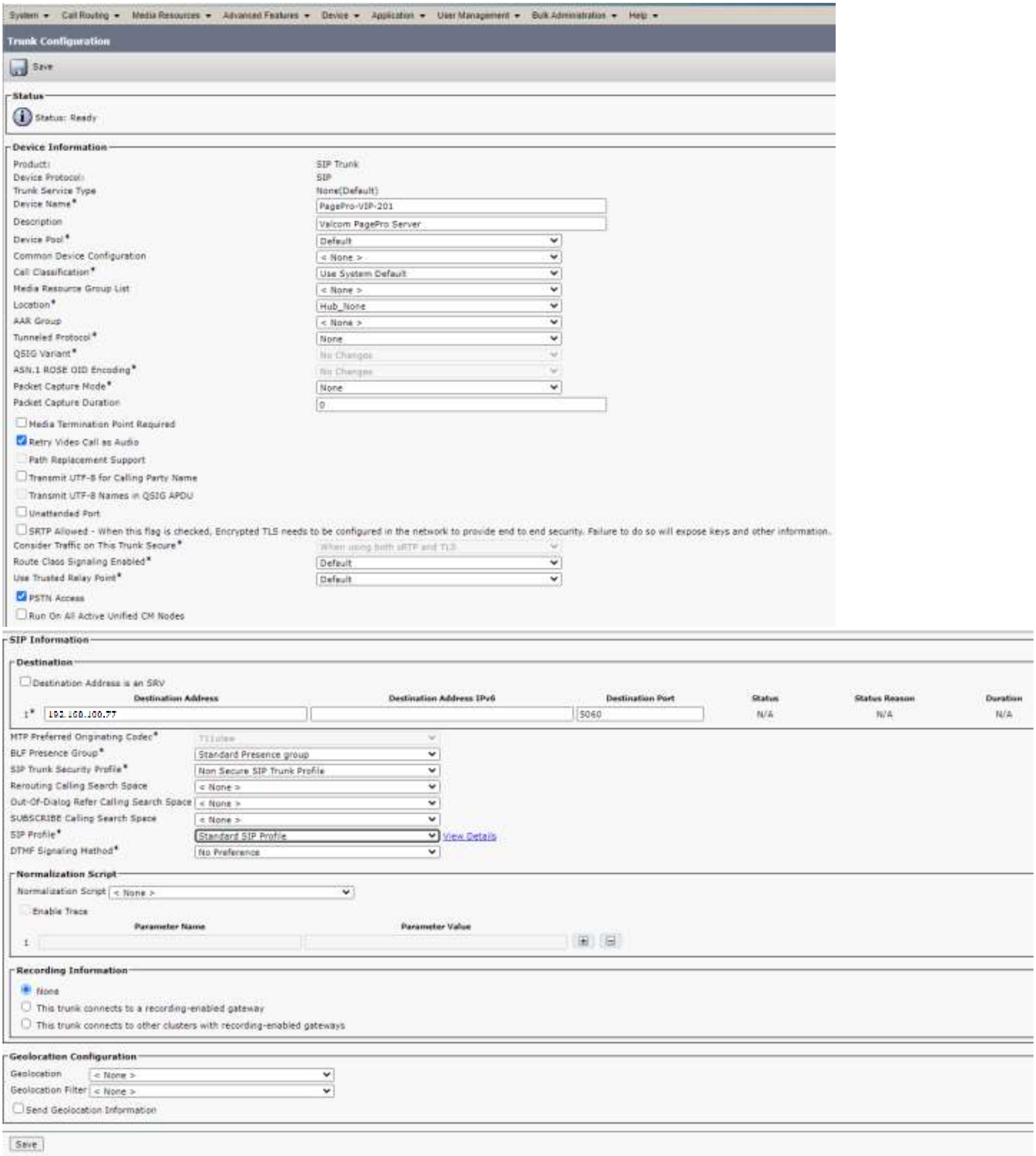
5. On the Find and List Trunks page, click on “Add New”.

6. Enter the Trunk Information.

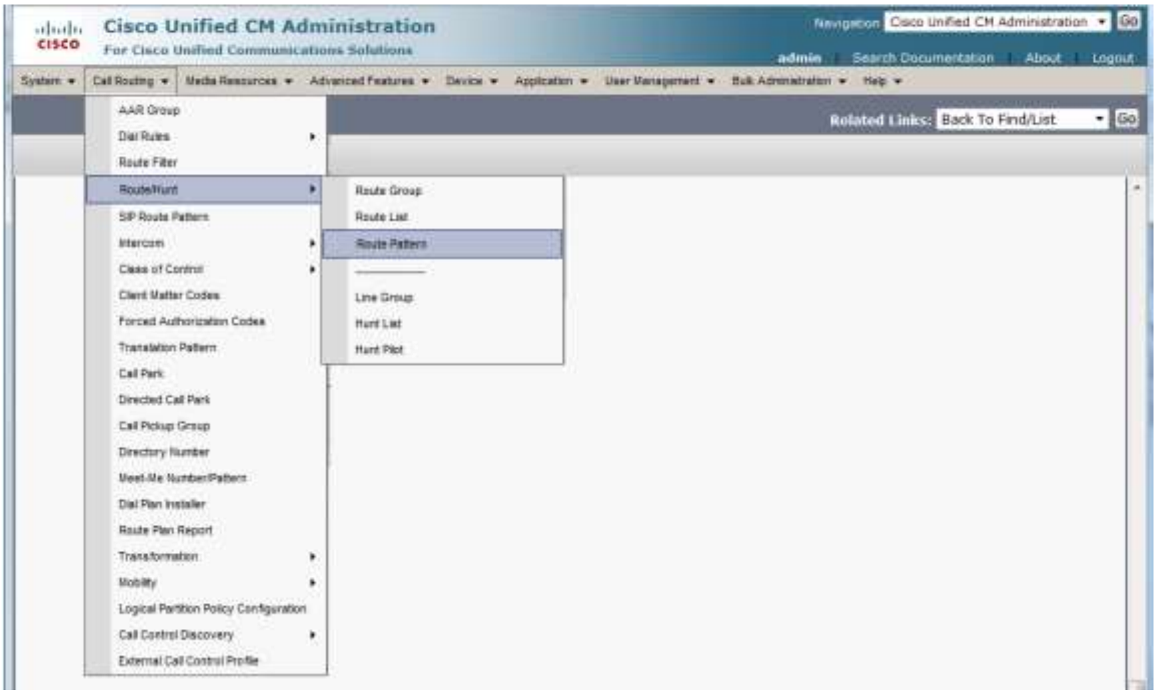
    A) Select “SIP Trunk” for Trunk Type
    B) Select “SIP” for Device Protocol
    C) Select “None” for Trunk Service Type (not all versions have this parameter)
    D) Select “Next” at the top or bottom of the screen

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

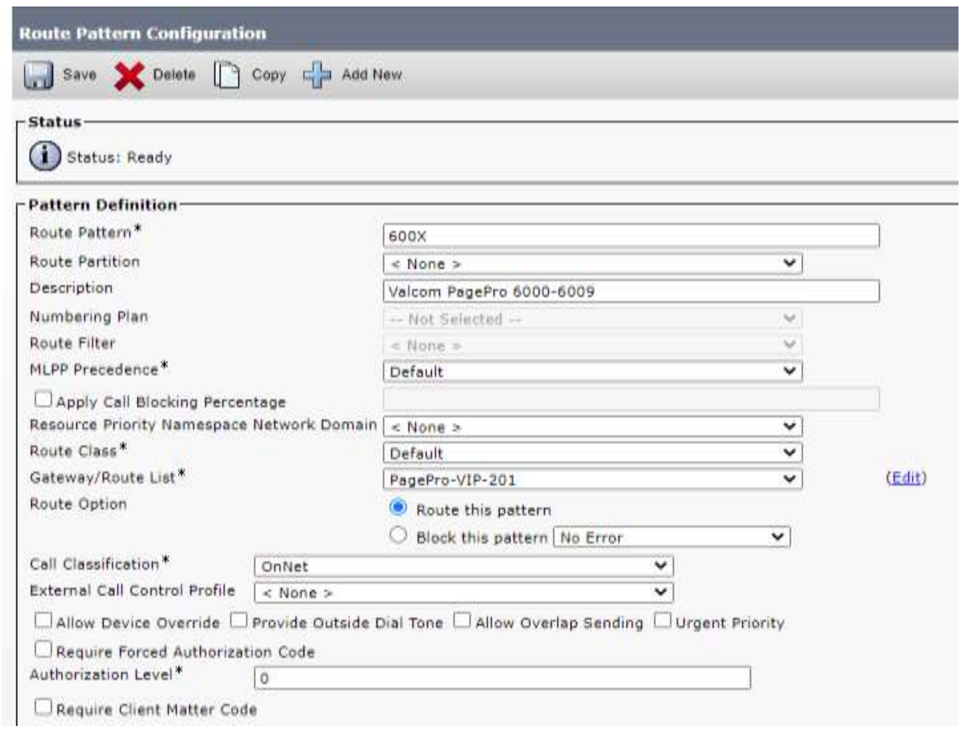

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Trunk Configuration
Save
Status
Status: Ready
Device Information
Product: SIP Trunk
Device Protocol: SIP
Trunk Service Type None(Default)
Device Name* PagePro-VIP-201
Description Valcom PagePro Server
Device Pool* Default
Common Device Configuration < None >
Call Classification* Use System Default
Media Resource Group List < None >
Location* Hub_None
AAR Group < None >
Tunneled Protocol* None
QSIG Variant* No Changes
ASN.1 ROSE OID Encoding* No Changes
Packet Capture Mode* None
Packet Capture Duration 0
Media Termination Point Required
Retry Video Call as Audio
Path Replacement Support
Transmit UTF-8 for Calling Party Name
Transmit UTF-8 Names in QSIG APDU
Unattended Port
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.
Consider Traffic on This Trunk Secure* When using both sRTP and TLS
Route Class Signaling Enabled* Default
Use Trusted Relay Point* Default
PSTN Access
Run On All Active Unified CM Nodes
SIP Information
Destination
Destination Address is an SRV
Destination Address Destination Address IPv6 Destination Port Status Status Reason Duration
1* 192.168.100.77 5060 N/A N/A N/A
MTP Preferred Originating Codec* 711ulaw
BLF Presence Group* Standard Presence group
SIP Trunk Security Profile* Non Secure SIP Trunk Profile
Rerouting Calling Search Space < None >
Out-Of-Dialog Refer Calling Search Space < None >
SUBSCRIBE Calling Search Space < None >
SIP Profile* Standard SIP Profile View Details
DTMF Signaling Method* No Preference
Normalization Script
Normalization Script < None >
Enable Trace
Parameter Name Parameter Value
1
Recording Information
None
This trunk connects to a recording-enabled gateway
This trunk connects to other clusters with recording-enabled gateways
Geolocation Configuration
Geolocation < None >
Geolocation Filter < None >
Send Geolocation Information
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”.

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”



    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.

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 http://www.valcom.com

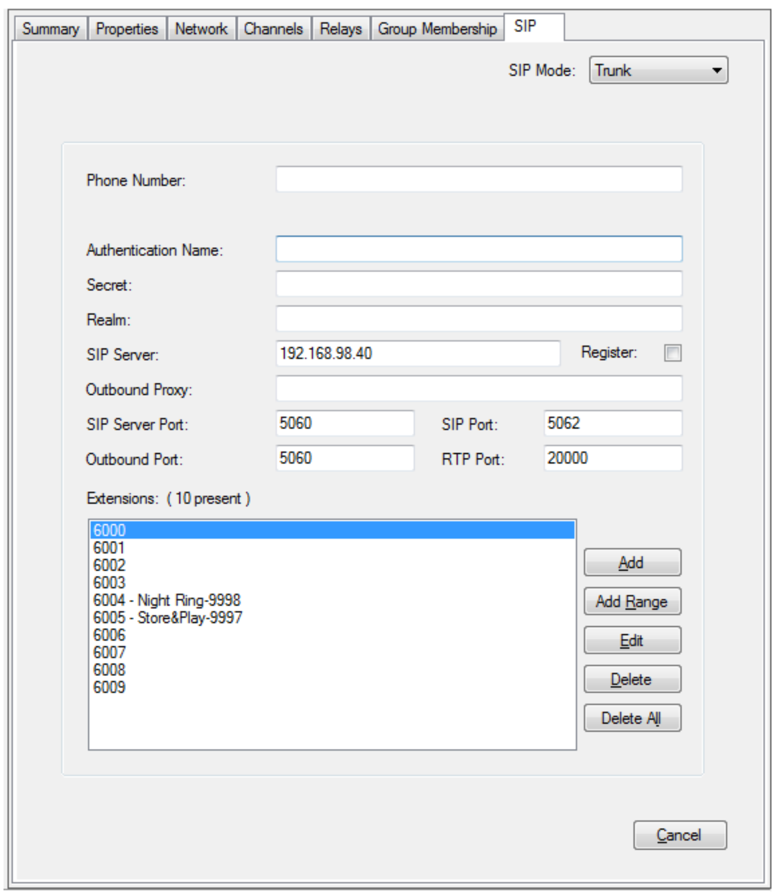
    A) On the SIP tab, set the SIP Mode to "Trunk" using the dropdown box in the upper right corner

    B) 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.

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.



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.

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.