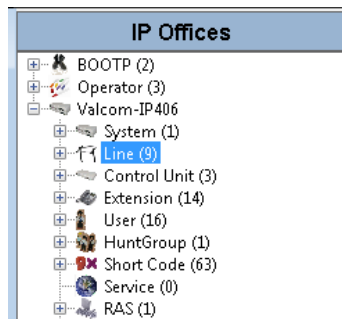



Valcom Session Initiation Protocol (SIP) VIP devices are compatible with SIP-enabled versions of Avaya IP Office (5.0 and higher). The Valcom device can be added to the IP Office system as a SIP Trunk. Valcom devices with multiple SIP channels may be configured in a one-to-many arrangement, where one Avaya SIP Trunk Channel can service multiple SIP identities on a single Valcom device. Call routing will be determined using Short Codes. Each Avaya SIP Trunk Channel instance allows one concurrent call over the IP connection.

These instructions cover the minimum configuration necessary for a Valcom SIP-enabled device. Refer to Avaya documentation for details on SIP services and entering SIP Trunk Channel License information. Other settings may also be necessary, depending on your specific system environment.

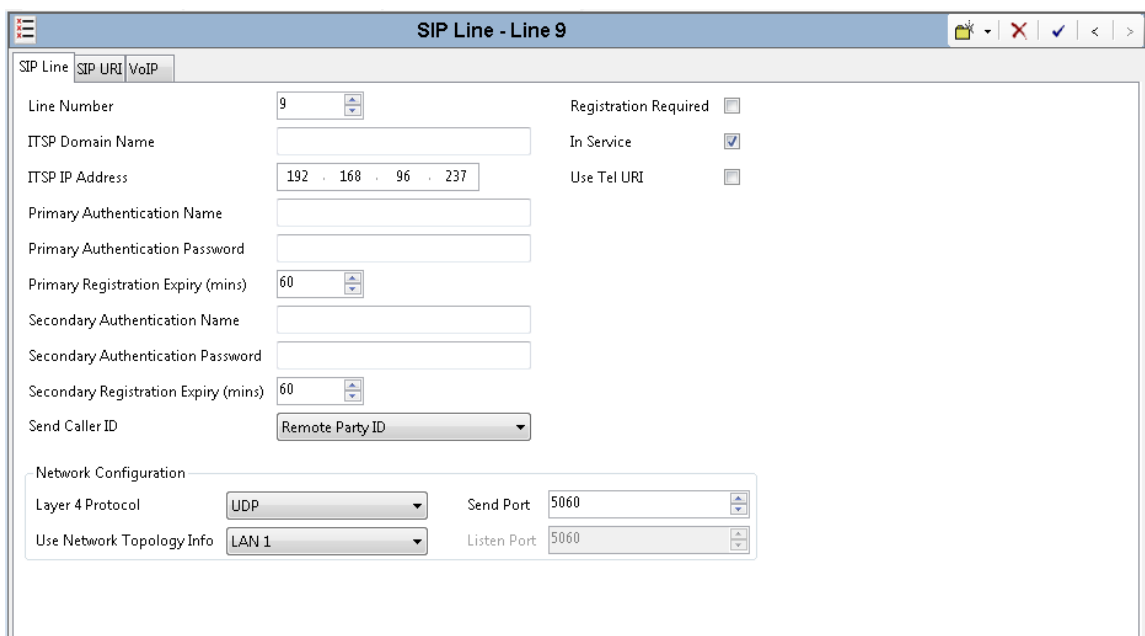
The following steps outline the typical configuration process:

1. Open the Avaya IP Office Manager application and connect to the IP Office server where the new SIP trunk will be created. In the Navigation pane, click on the Line category.



2. Click the New icon  in the upper-right corner of the Details pane, and choose SIP Line to open a new Details pane. In the Line Number field, accept the default or enter a new number. In the ITSP IP Address field, enter the IP address of the Valcom device that will be the destination for calls on this trunk. Set the Layer 4 Protocol selection to UDP.

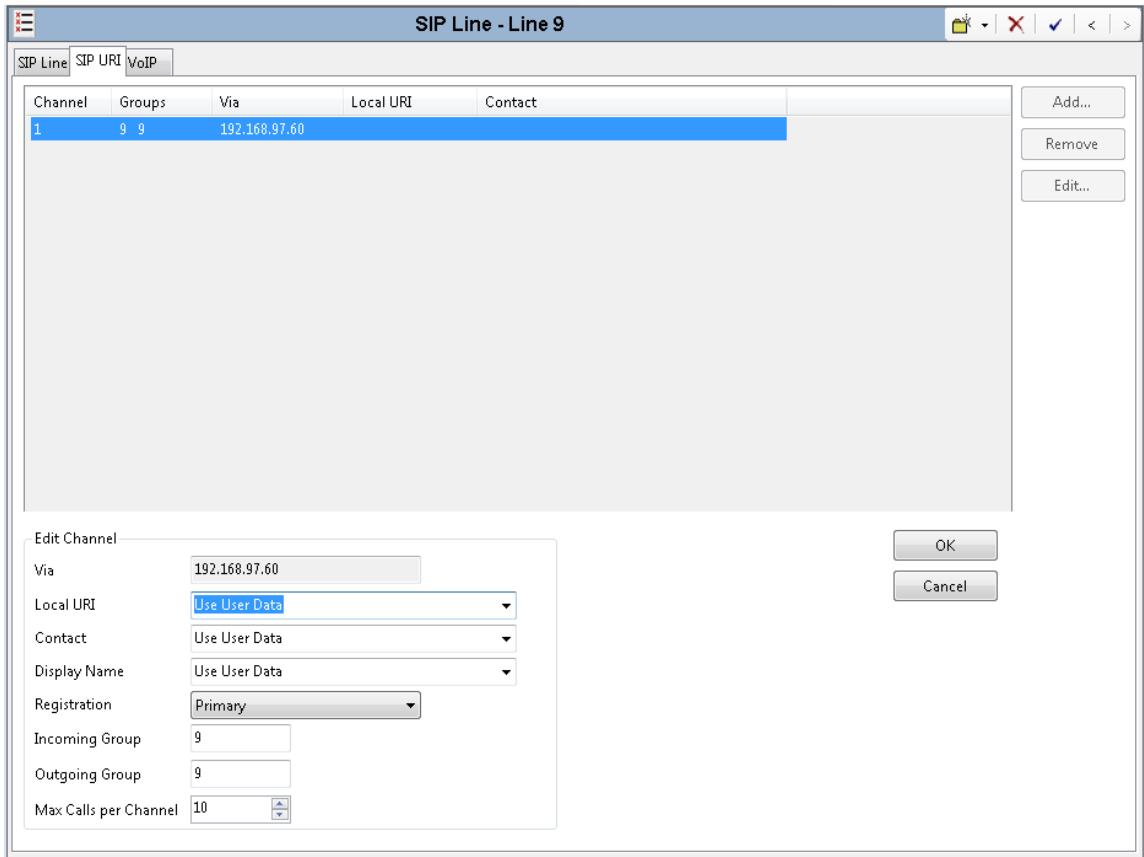
For this example, SIP Line 9 is being created and will send calls to IP address 192.168.96.237. The protocol will be UDP and will connect to port 5060 on the Valcom device.



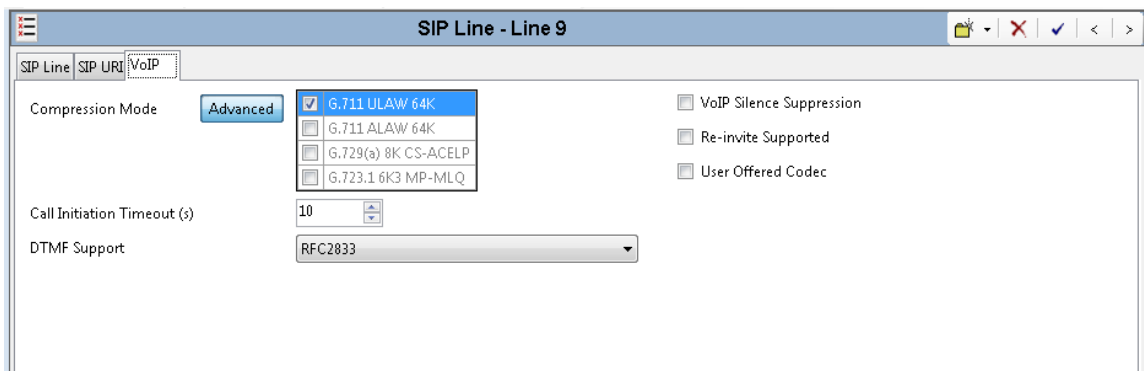
The screenshot shows the configuration window for a SIP Line. The title bar reads "SIP Line - Line 9". There are three tabs: "SIP Line", "SIP URI", and "VoIP". The "SIP Line" tab is active. The configuration fields are as follows:

Line Number	9	Registration Required	<input type="checkbox"/>
ITSP Domain Name		In Service	<input checked="" type="checkbox"/>
ITSP IP Address	192 . 168 . 96 . 237	Use Tel URI	<input type="checkbox"/>
Primary Authentication Name			
Primary Authentication Password			
Primary Registration Expiry (mins)	60		
Secondary Authentication Name			
Secondary Authentication Password			
Secondary Registration Expiry (mins)	60		
Send Caller ID	Remote Party ID		
Network Configuration			
Layer 4 Protocol	UDP	Send Port	5060
Use Network Topology Info	LAN 1	Listen Port	5060

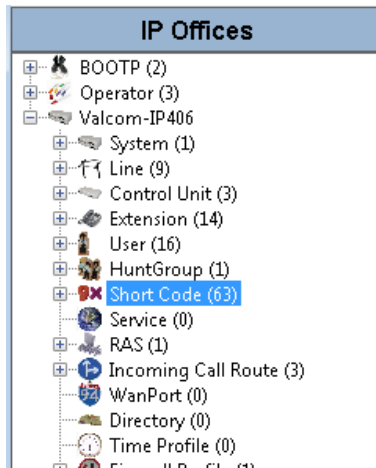
- Click the SIP URI tab on the Details pane. Click the Add button to create a new channel. In the section at the bottom of the window, enter the SIP Line number (just created in the previous step) for both the Incoming Group and Outgoing Group fields. In this example, we enter "9". For this example, the Local URI, Contact and Display Name have also been set to Use User Data. Click the OK button when the entries have been made.




- Click the VoIP tab on the Details pane. In the Compression Mode list, click Advanced, then choose the "G.711 ULAW 64K" option. For DTMF Support, select "RFC2833". Then click the OK button at the bottom of the window.



- In the Navigation pane, click on the Short Code category. For this example, the Valcom VIP-201 Paging Server is being configured as the trunk endpoint. The VIP-201 has 8 SIP identities (phone numbers), which will be configured as extensions 5801 through 5808. A Short Code will be defined to route calls to those destination numbers over the SIP Line (trunk) created in the previous steps.

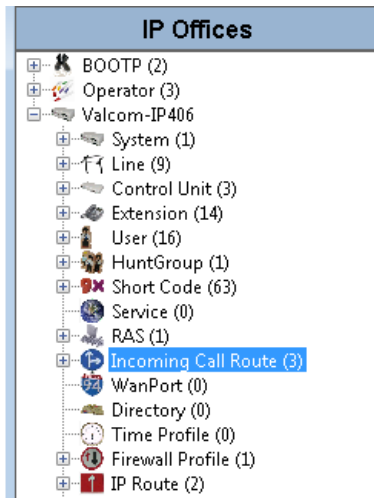


- Click the New icon  in the upper right corner of the Details pane to create a new Short Code. In the Details pane, enter a code that will match the dial code used in your system. For our example, the Valcom VIP-201 will be configured with phone numbers 5801 – 5808, so the Code will be entered as "580X;". Select "Dial" for the Feature. The Telephone Number to send will be "580N"@192.168.96.237", and the Line Group ID used will be the SIP Line created in previous steps (9, in this example).

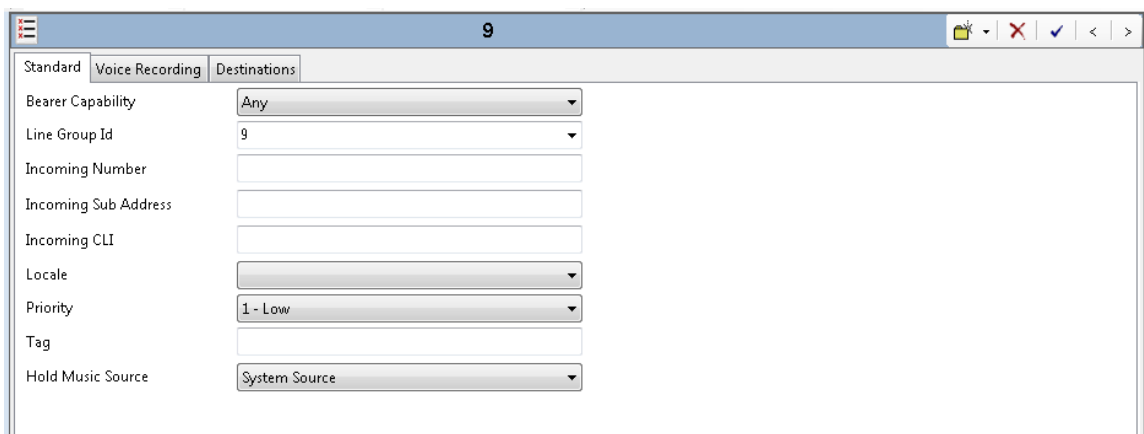
The screenshot shows the configuration details for a Short Code. The title bar reads "580X;; Dial". The form contains the following fields:

Code	580X;
Feature	Dial
Telephone Number	580N"@192.168.96.237"
Line Group Id	9
Locale	
Force Account Code	<input type="checkbox"/>

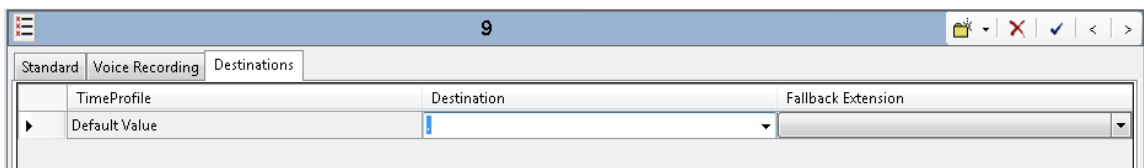
7. In the Navigation pane, click on the Incoming Call Route category.



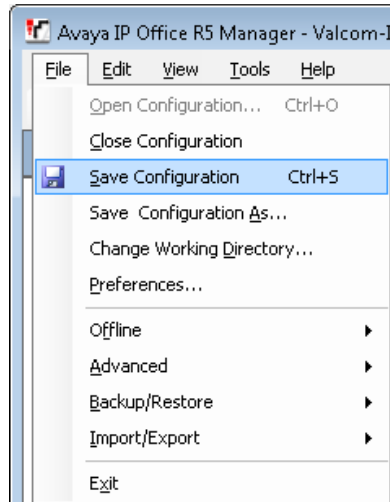
8. On the Standard tab, select the Line Group Id to match the SIP Line created in previous steps (9, in our example).



9. Click the Destinations tab, then enter "." in the Destination field. Entering the "." period character allows IP Office to route incoming calls to any extension. When finished, click the OK button at the bottom of the window.



10. After all of the Avaya settings have been entered in the Manager application, the configuration changes must be submitted to the IP Office server. Click the File menu, then Save Configuration. This will initiate the process to send the configuration to the server to make them effective. Follow the prompts to complete this process. A reboot of the server may be necessary.

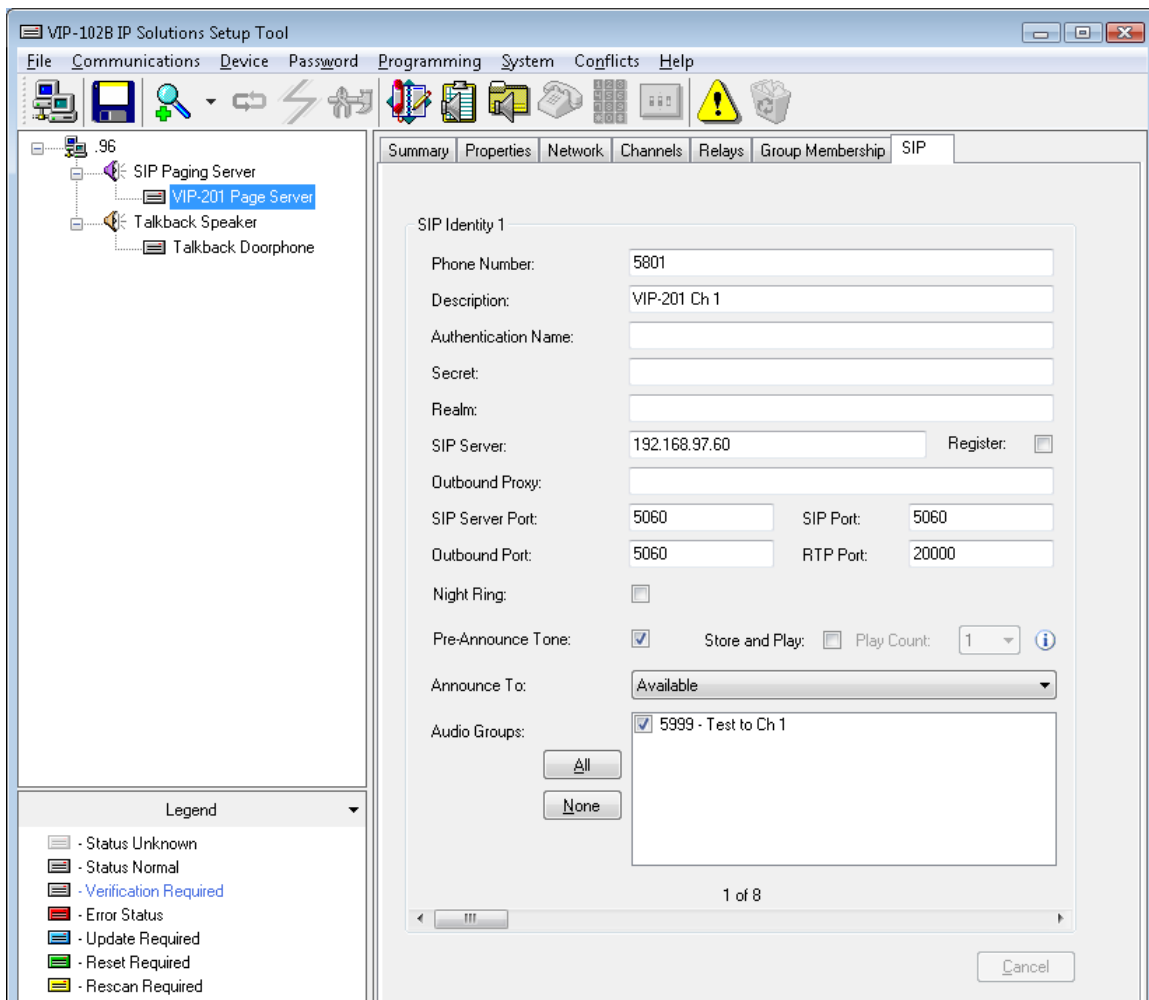


11. The Valcom SIP device must now be programmed with the settings used on the IP Office server, using the VIP-102B IP Solutions Setup Tool.

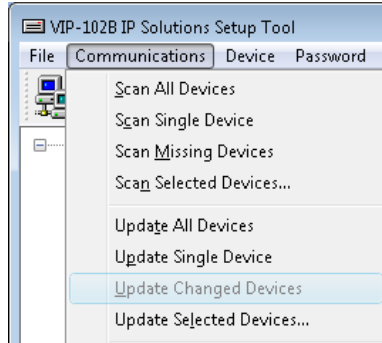
Note: The information contained in this guide is limited to configuration of the “SIP” tab in the VIP-102B IP Solutions Setup Tool for the Valcom device. More information on Valcom VIP device configuration, such as IP address assignment, relay activation, etc, may be found in the VIP-102B Reference Manual. 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

In our example, the Valcom VIP-201 Page Server is the device that will be connected via SIP trunk to the IP Office server. The VIP-201 has 8 SIP Phone Numbers available. The Phone Number shown for this example (5801) is within the range of numbers IP Office will route to the SIP Line. The other 7 SIP Identities would be similarly configured with Phone Numbers that will be routed by IP Office to the same SIP Line (5802, 5803, etc). SIP Server is the address of the Avaya IP Office, “192.168.97.60”, and the Register box is not checked. SIP Server Port is the port number on which the Avaya IP Office SIP server is listening for SIP data. SIP Port is the port number on which the Valcom device is listening for SIP data. By default, both SIP ports are set to “5060”. RTP Port is the port number on which the Valcom device will send/receive audio packets. Default is “20000”.



- When the Valcom device configuration is complete, select the “Update Changed Devices” option from the Communications menu. The new configuration will be copied to the Valcom device, and you will be prompted to reset the device for the changes to become effective. Allow the device to reset before continuing.



- With both the IP Office configuration and the Valcom device configuration completed, the success of the configuration may be checked using the Avaya System Status application. Expand the Trunks section, then click the Line corresponding to the SIP Line that was created previously. The Trunk Summary will be displayed as shown in the example below.

IP Office System Status

Help Snapshot LogOff Exit About

- System
- Alarms (13)
- Extensions (12)
- Trunks (9)
 - Line: 1
 - Line: 2
 - Line: 3
 - Line: 4
 - Line: 5
 - Line: 6
 - Line: 7
 - Line: 8
 - Line: 9
- Active Calls
- Resources
- Voicemail
- IP Networking

SIP Trunk Summary

Peer Domain Name: sip://192.168.96.237
 Gateway Address: 192.168.96.237
 Line Number: 9
 Number of Administered Channels: 10
 Number of Channels in Use: 1
 Administered Compression: G711Mu
 Silence Suppression: Off
 SIP Trunk Channel Licences: 1 0%
 SIP Trunk Channel Licences in Use: 0

SIP Device Features:

Channel Number	URI Gro. Ref	Call State	Current State	Time in State	Remote RTP Address	Codec	Connection Type	Caller ID or Dialed Digit on Call	Other Party on Call	Direction of Call	Round Trip Delay	Receive Jitter	Receive Pa Loss Fracti	Transmit Jitter	Transmit Pa Loss Fracti
1	1 5	Connected	00:00:49		192.168.96...	G71...	VCM		Extn 6002, Ext 6002	Outgoing	0ms	0ms	0%		
2		Idle	01:59:32												
3		Idle	01:59:32												
4		Idle	01:59:32												
5		Idle	01:59:32												
6		Idle	01:59:32												
7		Idle	01:59:32												
8		Idle	01:59:32												
9		Idle	01:59:32												
10		Idle	01:59:32												

Trace Trace All Pause Ping Call Details Print... Save As...

5:04:45 PM Online