Avaya Solution & Interoperability Test Lab

Application Notes for Valcom VE6023 Telephone Page Server with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Valcom VE6023 Telephone Page Server to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The VE6023Telephone Page Server extends the functionality of an IP phone system allowing it to integrate seamlessly with an overhead paging system. The VE6023 allows pages from other Valcom devices to play on Avaya 9600 series IP Deskphones using H.323firmware, essentially turning those phones in to additional IP speakers.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for the Valcom VE6023 Telephone Page Server to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Valcom VE6023 Telephone Page Server provides a bridge between Valcom IP Mass Notification systems and Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

The VE6023 retransmits Valcom IP paging audio to Avaya 9600 series IP Deskphones H323 firmware sets, and thus requires additional equipment to be the source of the paging audio. The Valcom VIP-201 PagePro IP is used during compliance test as the source of the paging audio.

2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test plan included feature and serviceability test cases.

The feature testing covered Avaya Phones registering to PUSH servers, maintenance of list of phones on VE6023, basic pages, simultaneous pages, display verification, media shuffling, and audio codec negotiation. Various SIP access numbers for the Valcom VIP-201 PagePro IP device were dialed to test connections to the proper speakers and Avaya H.323 telephone groups.

The serviceability testing focused on verifying the ability of the Valcom VE6023 Telephone Page Server to recover from adverse conditions, such as disconnecting and reconnecting the Ethernet cable to the device, rebooting Communication Manager, and rebooting Session Manager.

2.2. Test Results

All feature and serviceability test cases were completed successfully. Valcom VE6023 successfully interoperates with Communication Manager and Session Manager.

2.3. Support

Technical support for Valcom can be obtained through the following:

- **Phone:** (800) VALCOM1
- **Email:** support@valcom.com

3. Reference Configuration

The VE6023 allows pages from Valcom IP Mass Communication devices to play on Avaya IP desk phones, essentially turning those phones into additional IP speakers. For compliance testing, page groups have been pre-programmed into a VIP-201 PagePro IP, which will be replicated to the VE6023 as part of the configuration process. The priority assigned to the pre-programmed groups is:

- Groups 100, 200, 201 = Low
- Groups 300, 301 = High

The provided Valcom IP Speaker is assigned to page groups 201 and 301.
Several Avaya 9600 Series IP Deskphones with H.323 firmware will be required for testing.
During compliance testing, the following configuration was used:

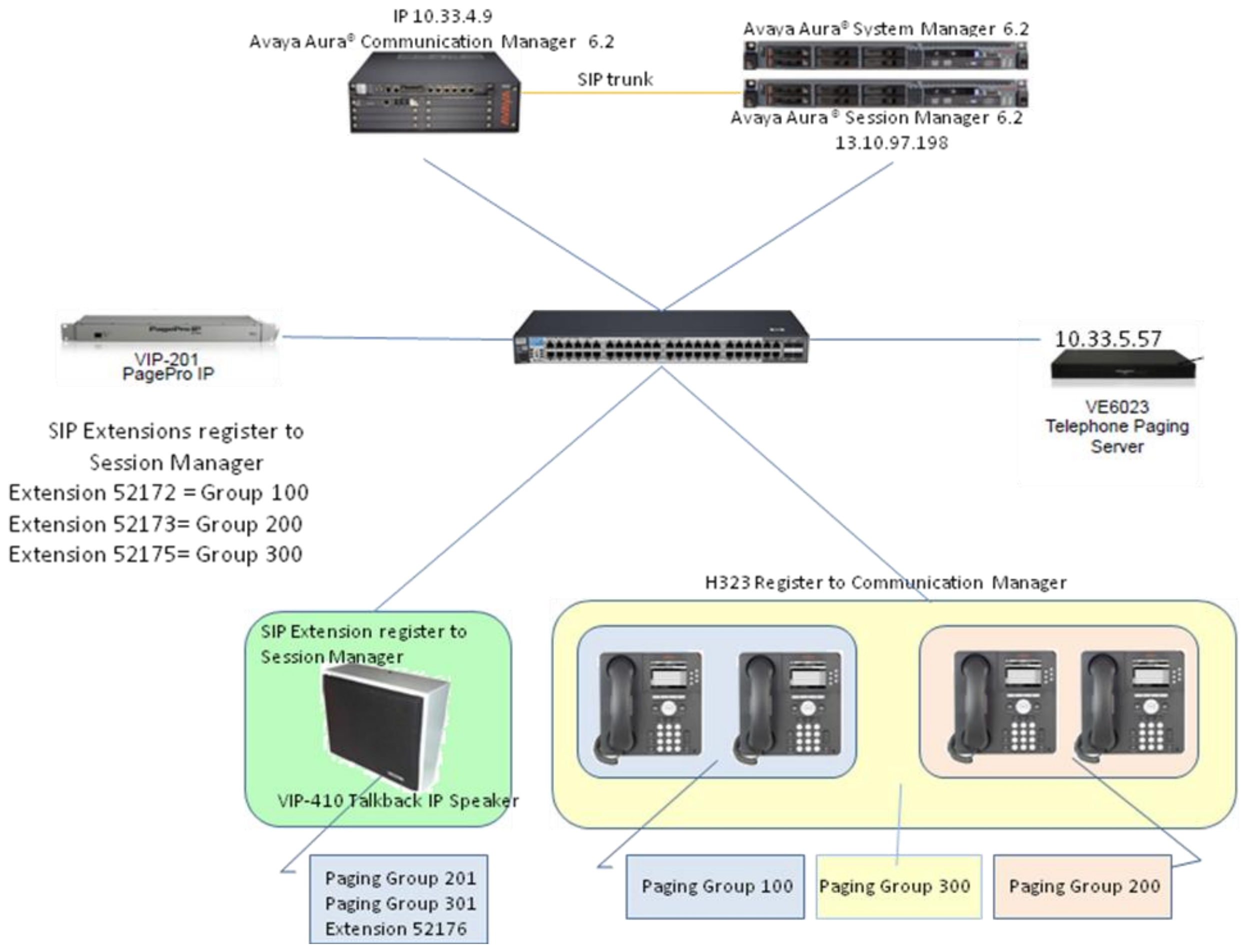


Figure 1: Valcom Telephone Page Server with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Release/Version
Avaya S8300 Server with a Avaya G450 Media Gateway	Avaya Aura® Communication Manager 6.2
Avaya S8800 Server	Avaya Aura® System Manager 6.2
Avaya S8800 Server	Avaya Aura® Session Manager 6.2
Avaya 9611 IP Deskphone Avaya 9608, 9630 IP Deskphones	96x1-IPT-H323-R6_2_2_09-071012 96xx-IPT-H323-R3_1_5-092612
Valcom VIP-201 PagePro IP	2.19.0 Startup Rev 1.42
Valcom VIP-410 Talkback IP Speakers	2.20.0 Startup Rev 1.17
VE6023 Valcom Telephone Page Server	4.0.3-673eb92 Platform Rev 2.11
VIP-102B Tool	4.1.0.0

5. Configure Avaya Aura® Communication Manager

The detailed administration of basic connectivity between Communication Manager and Session Manager is not the focus of these Application Notes and will not be described. For administration of basic connectivity between Communication Manager and Session Manager, refer to the appropriate documentation listed in **Section 10**. The G.711MU codec was configured on Communication Manager. This section provides the procedures for the following:

- Verify Communication Manager License.
- Administer H.323 Station.
- Configure PUSH interface in the 46xxsetting file.
- Configure SNMP Agent.

5.1. Verify Communication Manager License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command to verify that there is sufficient capacity for SIP stations by comparing the **Maximum Off-PBX Telephones - OPS** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the number of access numbers required for the Valcom Telephone Page Server device.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V16                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                               USED
                                Platform Maximum Ports: 65000 90
                                      Maximum Stations: 41000 24
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 41000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 18
                    Maximum Off-PBX Telephones - PBFMC: 41000 0
                    Maximum Off-PBX Telephones - PVFMC: 41000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   1
        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer H.323 Station

To add a new H.323 station, enter **add station <t>** where **t** is an available extension and configure the following:

- **Extension** : Verify the extension is shown correctly.
- **Type** : Select the type for this extension.
- **Security Code** : Enter the code used when user logs on to device.

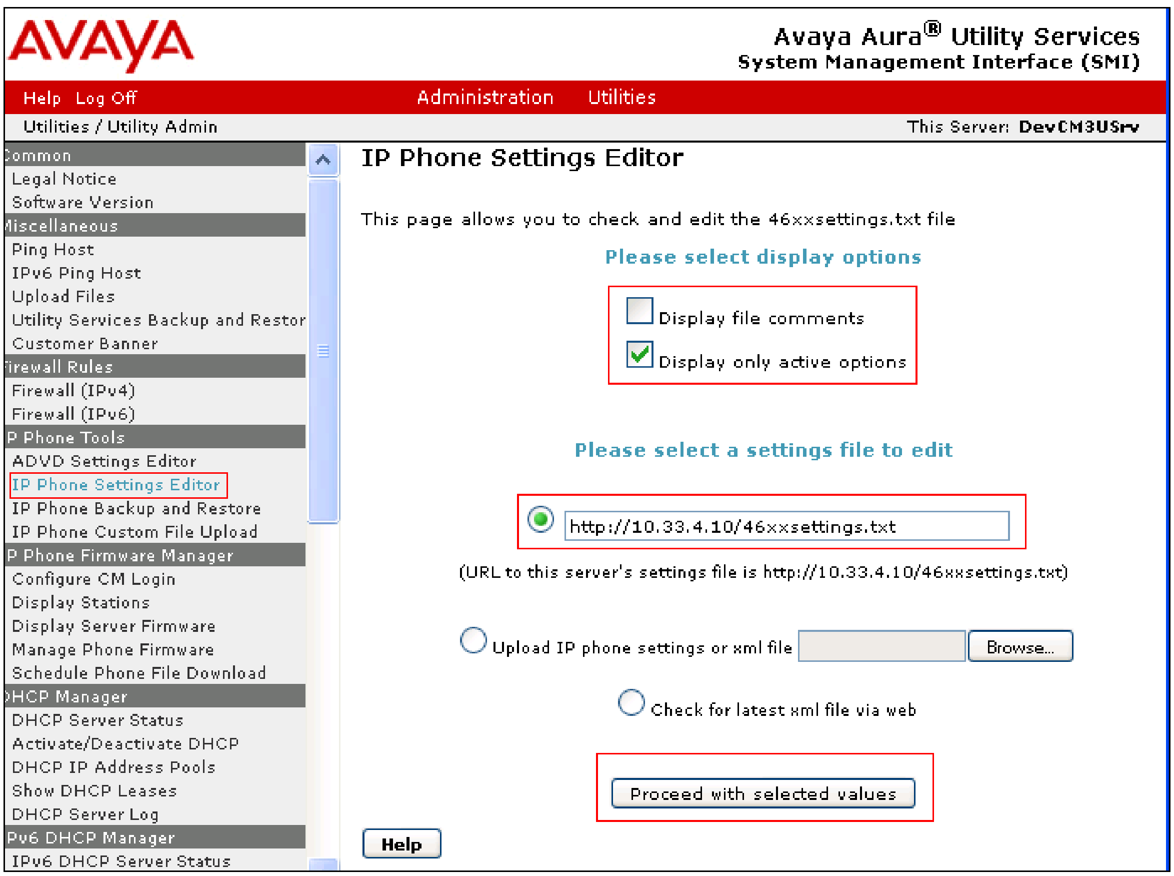
```
add station 52155                                          Page   1 of   5
                                     STATION

Extension: 52155                         Lock Messages? n             BCC: M
     Type: 9650                          Security Code: *              TN: 1
     Port: S00000                      Coverage Path 1: 2             COR: 1
     Name: Nam nam                     Coverage Path 2:               COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 52155
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english                  Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? y

                                                IP Video Softphone? y
                              Short/Prefixed Registration Allowed: default

                                               Customizable Labels?
```

5.3. Configure PUSH interface in the 46xxsetting file

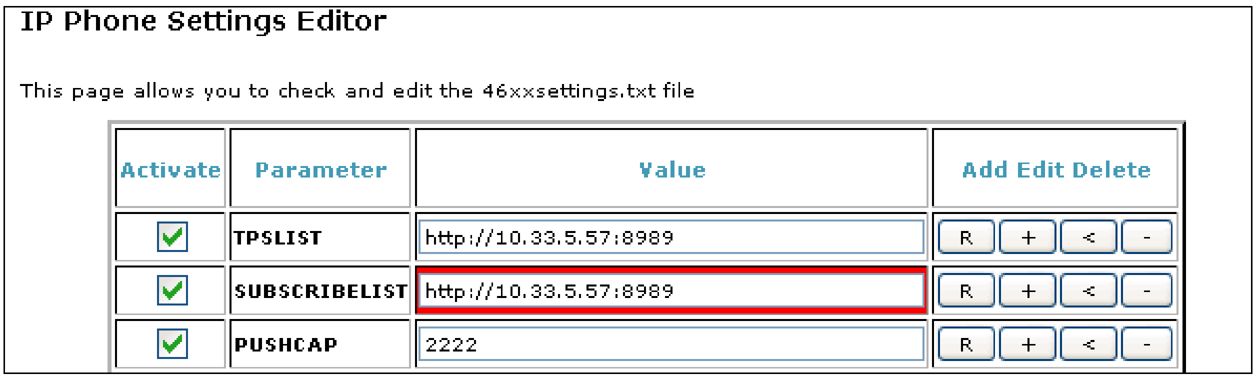
The VE6023 uses Avaya's push feature to stream audio to IP deskphones. To authorize the VE6023 as a Trusted Push Server, the URL from the VE6023 must be added to the 46xxsettings.txt file. The URL is composed of the IP address of the VE6023 server and the port configured in the VE6023 setup. The default port is 8989, but it can be changed. The URL will be in the form of “http://<*ipaddress*>:8989”.

This section describes steps to configure the PUSH interface in the 46xxsetting file on the Utility Server.

- Navigate to the Avaya Utility Server User Page Interface and log in.
- Click on “Utility Admin” from the “Utilities” menu.
- Click on IP Phone Settings Editor on the left under IP Phone Tools.
- Disable “Display File Comments” and enable “Display only active options.” Click on the “Proceed With Selected Values” button to continue.



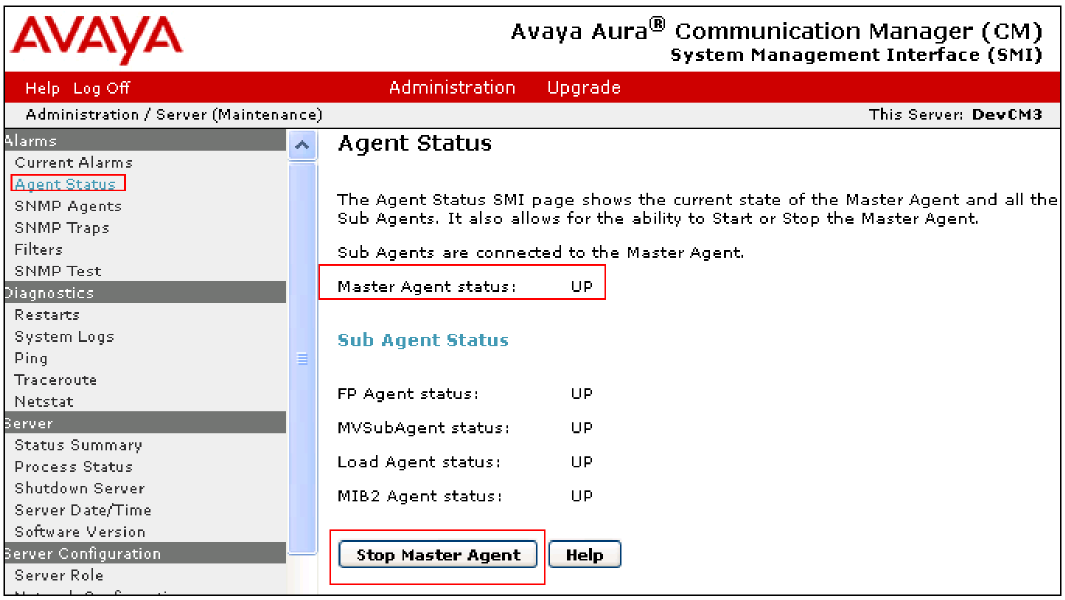
- Find the text box labeled **TPSLIST**. If the text box already contains a value, add a comma and append the URL of the VE6023 Push Server. If the text box is empty, just add the URL of the VE6023 Push Server.
- Find the text box labeled **SUBSCRIBELIST**. If the text box already contains a value, add a comma and append the URL of the VE6023 Push Server. If the text box is empty, just add the URL of the VE6023 Push Server.
- Find the text box labeled **PUSHCAP** and type in 2222.
- Click the "Save New settings File" button at the bottom of this page.
- Apply the new settings by clicking "Save 46xxsettings.txt File to this server".
- If the 46xxsettings.txt file contains separate sections for different phone models, then this procedure should be followed for each section.
- Log out and reboot the Avaya IP Deskphones required for paging.



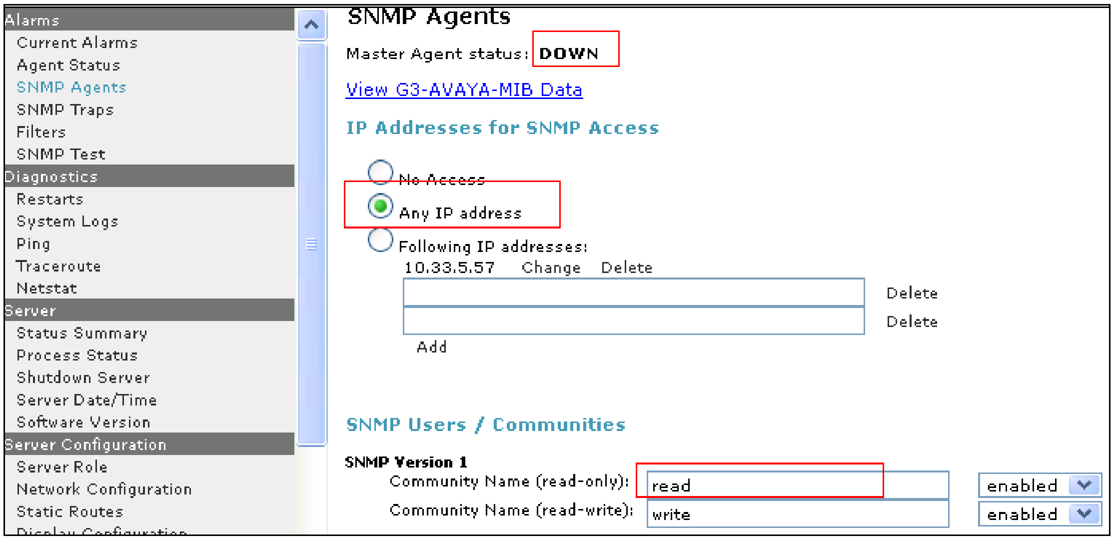
5.4. Configure SNMP Agent.

The VE6023 uses SNMP to monitor the Communication Manager for updated information on the Avaya IP deskphones. This section describes the steps to configure a SNMP Agent.

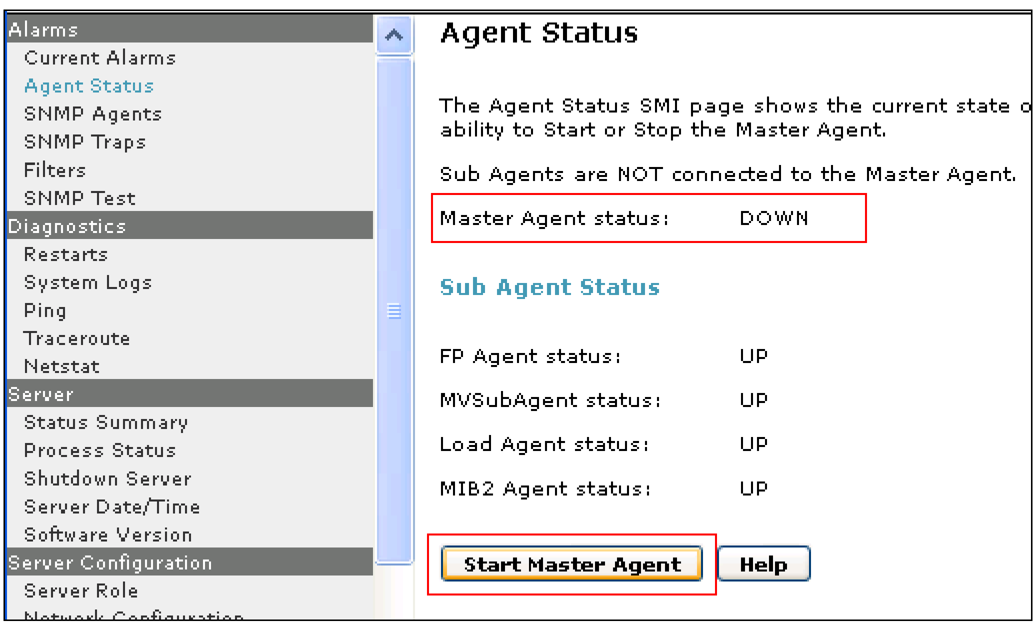
- Navigate to the Avaya Aura® Communication Manager web page and log in.
- Click on “Server (Maintenance)” from the “Administration” menu.
- Click on “Agent Status” under the “Alarms” section.
- Ensure that the Master Agent status is stopped. If it is active, stop it now.



- Click on “SNMP Agents” under the “Alarms” section.
- Select “Any IP address”.
- Set a community string for SNMP Version 1, and select “enabled” from the pull down menu, then click submit (not shown). This community string will be needed by the VE6023.



- Go back to “Agent Status” under the “Alarms” section and start the agent.



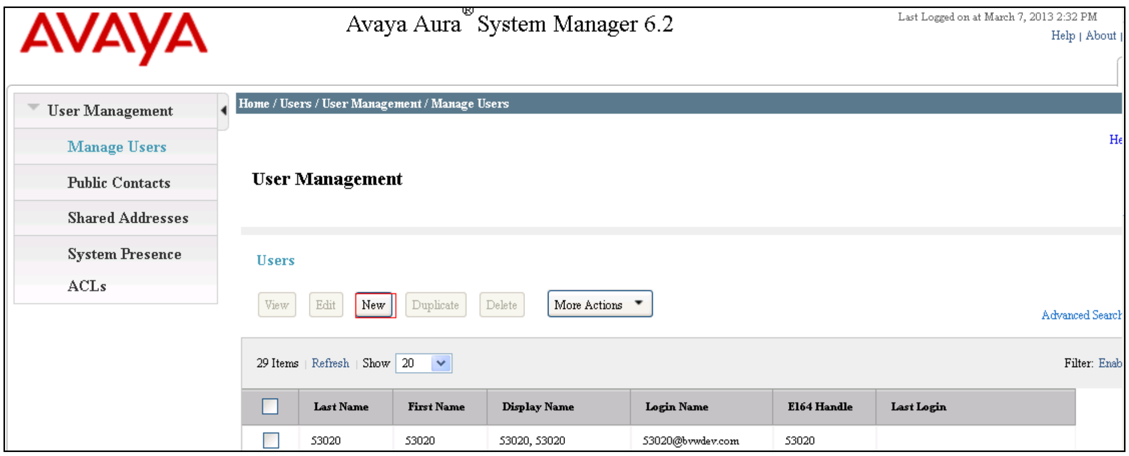
6. Configure Avaya Aura® Session Manager

It is assumed that Session Manager is configured and operational. This section only provides the procedures for configuring the SIP User for the Valcom VIP-201 PagePro IP and IP Speaker VIP-410 on Session Manager. This is included for completeness of the compliance testing documentation, but is not required for operation of the VE6023. A Valcom IP paging source is required, but may be any of the Valcom IP paging sources.

Configuration of Session Manager is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager, using the URL "https://<ip-address>/SMGR", where "<ip-address>" is the IP address of System Manager. Log in using the appropriate credentials.

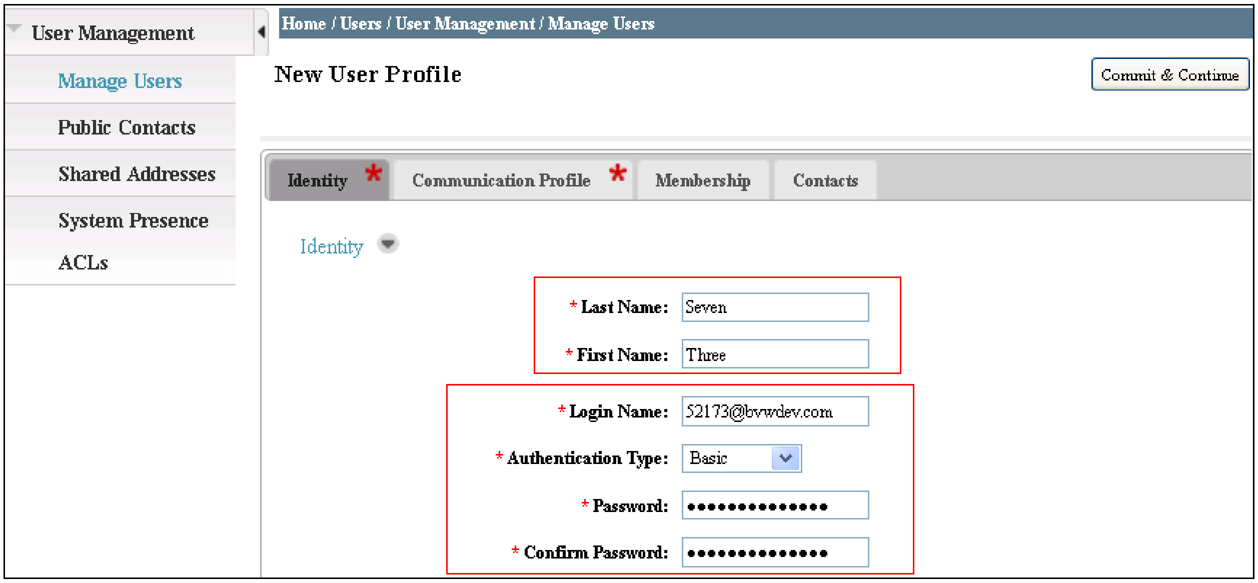
6.1. Administer User

From the menu in the left pane, navigate to **Users → Manage Users**. Select the **New** button from the right pane.



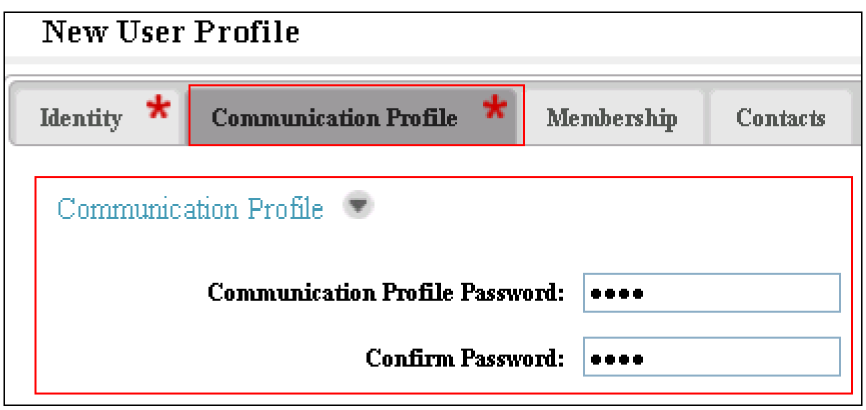
Enter the following values for the specified fields, and retain the default values in the remaining fields.

- **Last** : Enter the last name of the user.
- **First:** Enter the first name of the user.
- **Login Name**: Enter the unique system login given to the user. It takes the form of *username@domain* (e.g., "52173@bvwdev.com") and it is used to create the user's primary handle.
- **Authentication Type:** Select "Basic".
- **Password:** Enter the password used to log into System Manager.



Click on the Communication Profile tab and enter the following information for **Communication Profile** section:

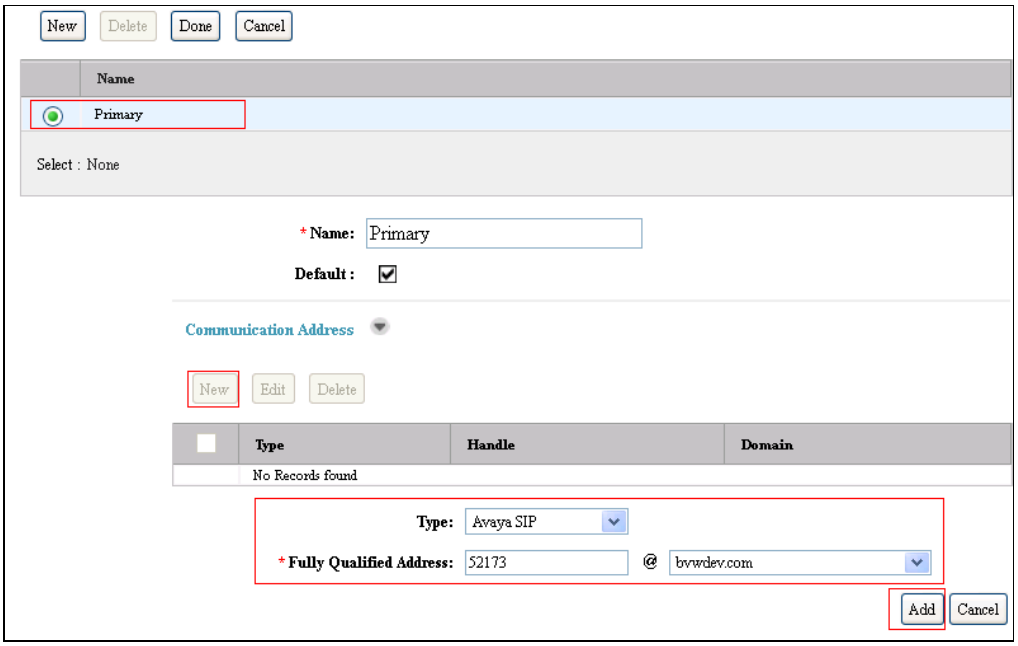
- **Communication Profile Password**: Enter a password. This password will be used in **Section 7.4.**



Under ***Communication Address***, click on **New** button:

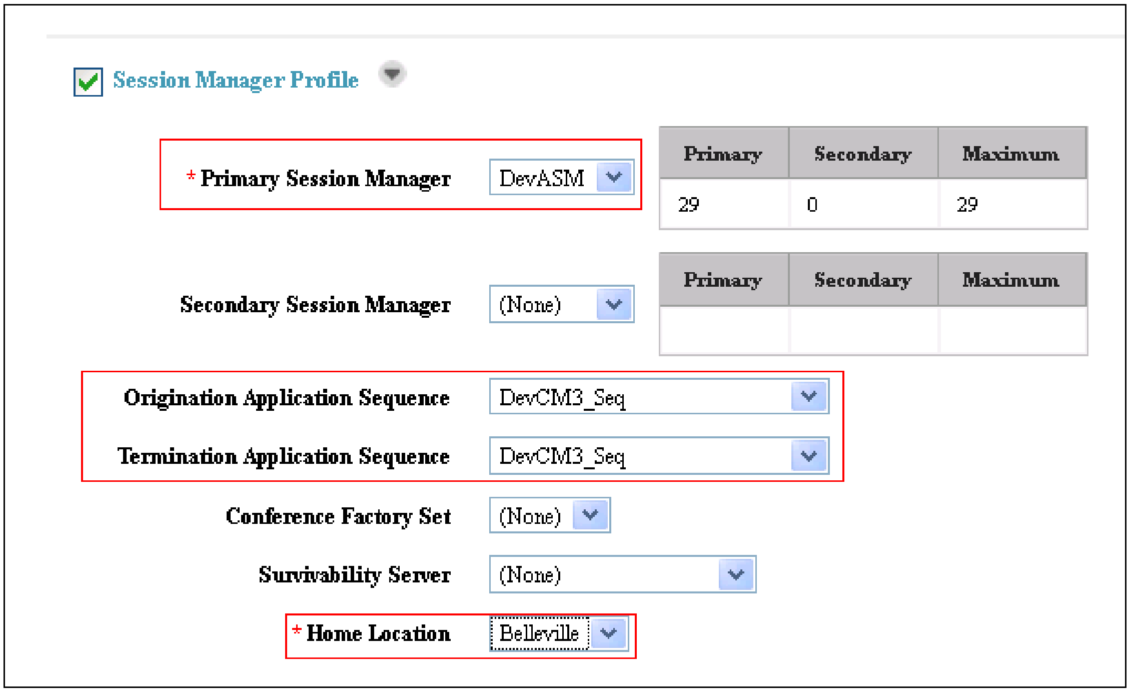
- **Type**: Select "Avaya SIP".
- **Fully Qualified Address**: Enter the extension and select the appropriate domain for the user.

Click the **Add** button.



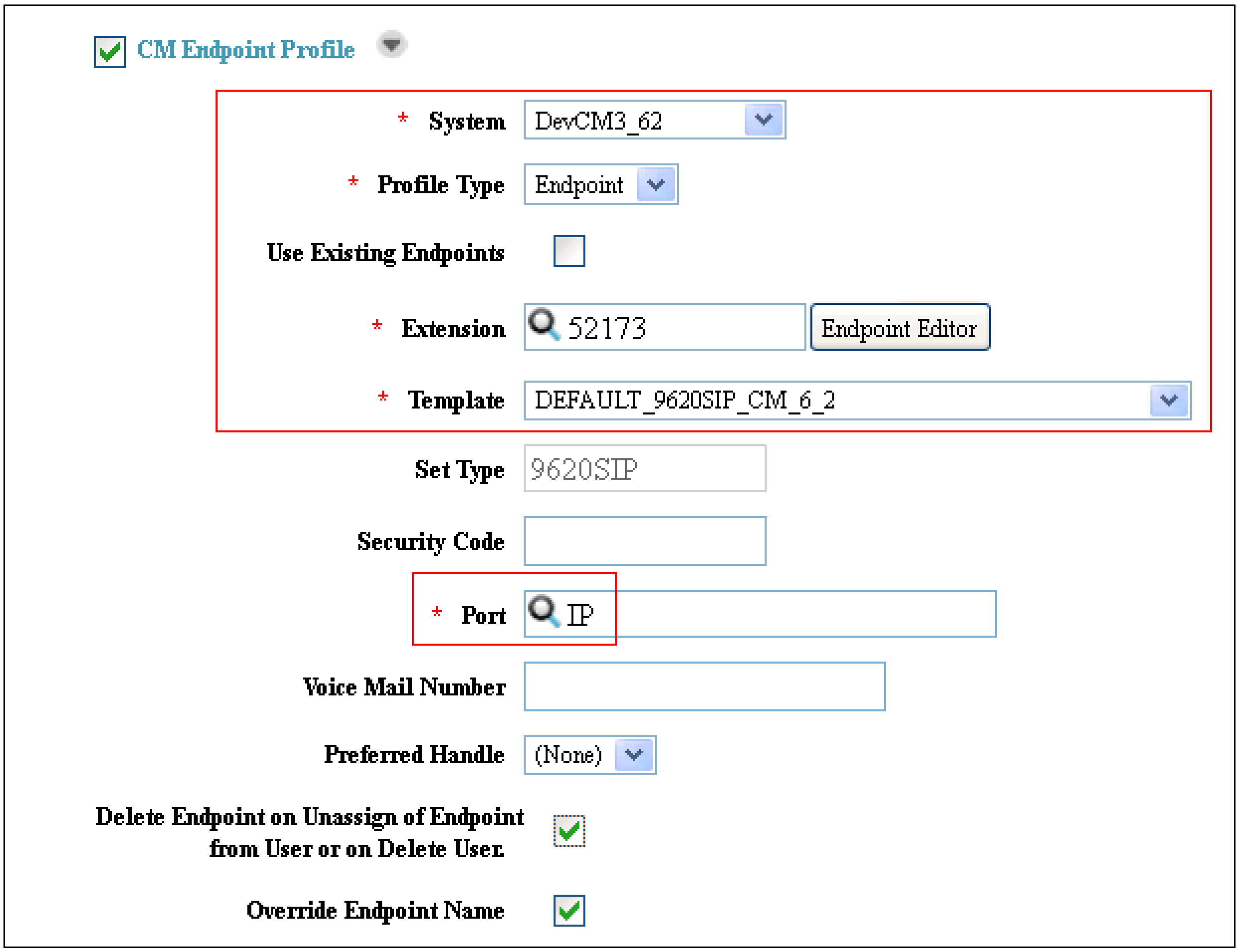
Under ***Session Manager*** section:

- **Primary Session Manager** Select the Session Manager instance that should be used as the home server for the currently displayed Communication Profile.
- **Origination Application Sequence** Select an Application Sequence that will be invoked when calls are routed *from* this user.
- **Termination Application Sequence** Select an Application Sequence that will be invoked when calls are routed *to* this user.
- **Home Location** Select the Home Location of this user.



Under ***CM Endpoint Profile***, enter the following information:

- **System**: Select the Communication Manager on which the endpoint exists.
- **Profile Type**: Select Endpoint.
- **Extension**: Enter the extension for this user.
- **Template**: Select template for the SIP user. During the compliance test, 9620SIP_CM_6_2 was used.
- **Port**: The IP Port field is automatically filled in.



Click the **Commit** button. Repeat the procedures in this section to add more SIP Users.

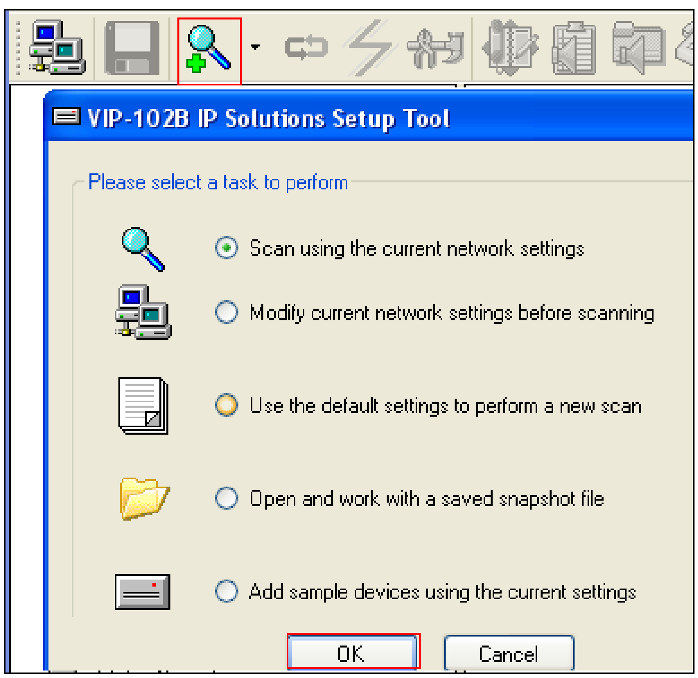
7. Configure Valcom devices

This section provides the procedures for configuring Valcom devices used during compliance test such as VIP-201 PagePro IP, IP Speaker VIP-410 and VE6023 Telephone Page Server. The information shown is the minimum for configuring the Valcom device. Complete configuration details may be found in the Valcom documentation listed in **Section 10**. The procedures include the following areas:

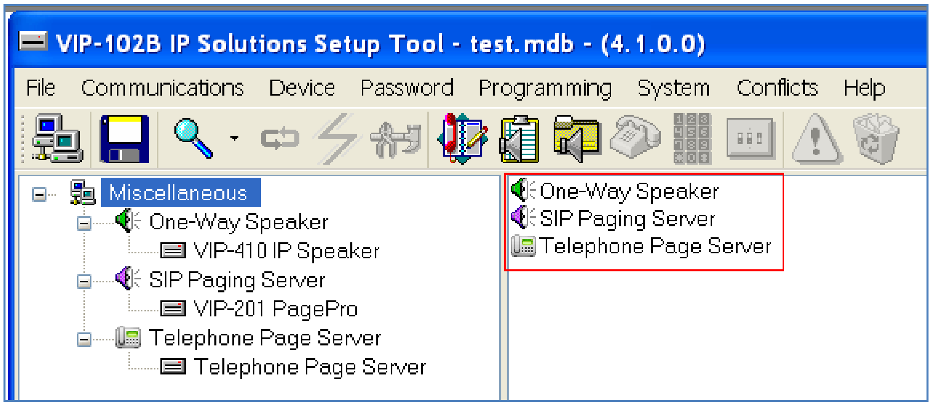
- Launch setup tool
- Administer properties
- Administer network
- Administer group membership
- Administer SIP
- Administer Telephone Page Server

7.1. Launch Setup Tool

From a PC running the Valcom VIP-102B IP Solutions Setup Tool application, select **Start → All Programs → Valcom IP Solutions → VIP-102B IP Solutions Setup Tool**. The **VIP-102B IP Solutions Setup Tool** screen is displayed. Retain the default values and click **OK** to scan for Valcom devices.

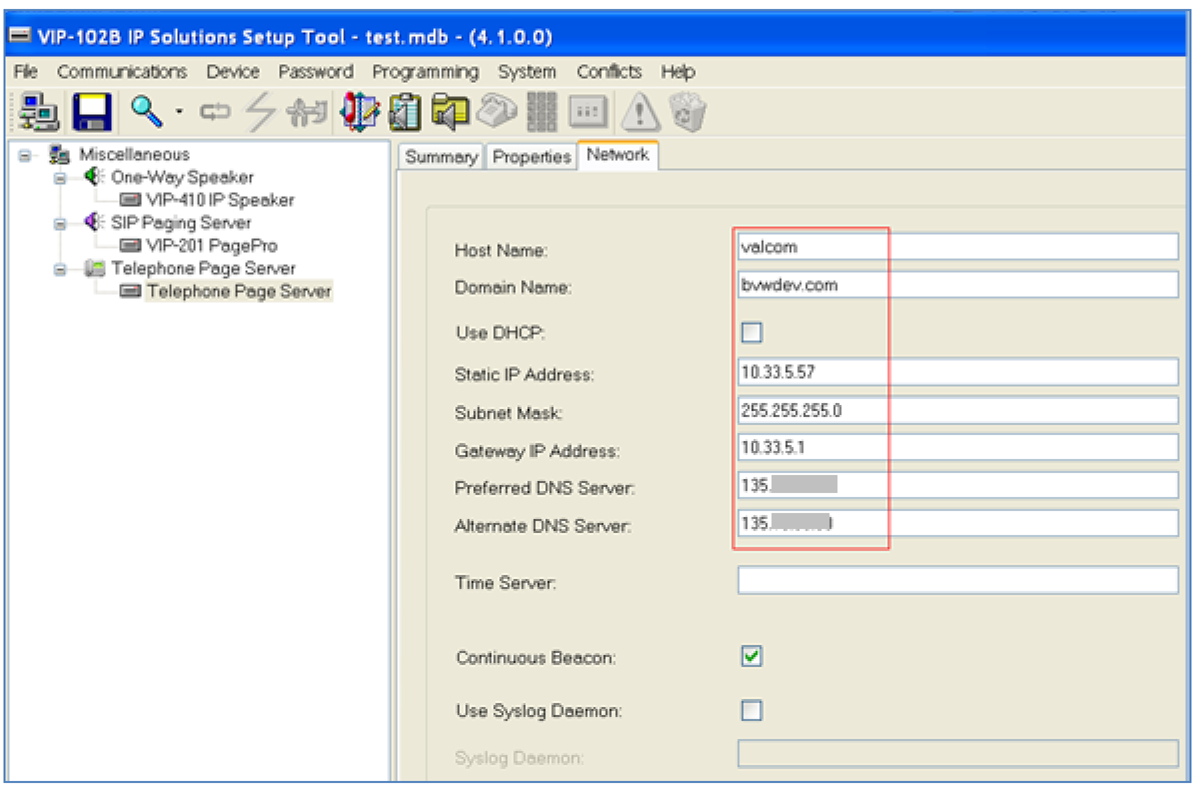


At the conclusion of the scan, the **VIP-102B IP Solutions Setup Tool** screen is updated with the discovered Valcom devices as shown below:



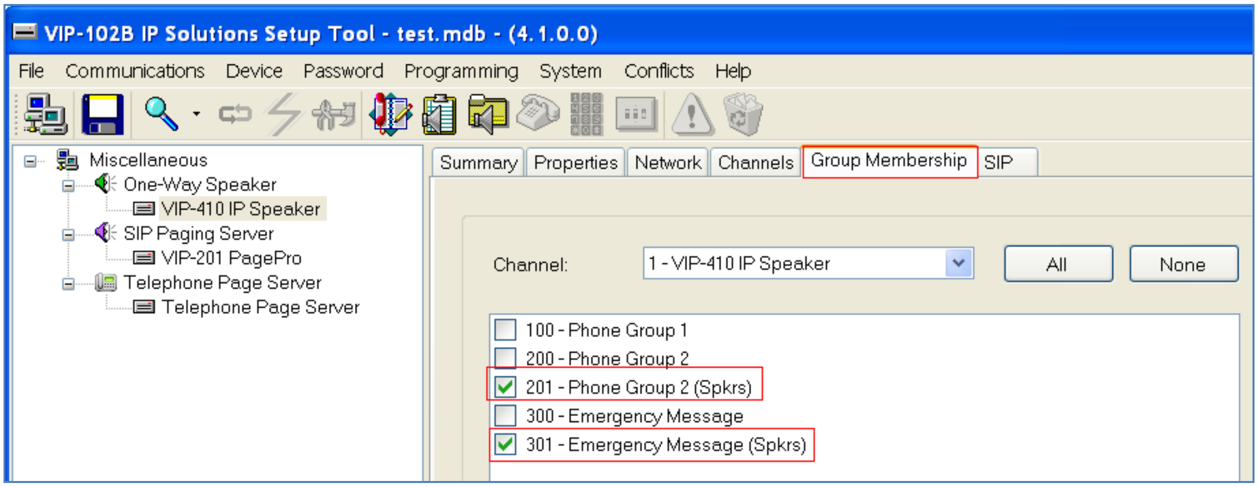
7.2. Administer Network

Select device and select the **Network** tab and enter the appropriate values. During compliance testing, a **Static IP Address**, **Subnet Mask**, and **Gateway IP Address** were populated for the network configuration. The default values in the remaining fields were retained. Below is the screenshot of the Network tab of the Telephone Page Server.

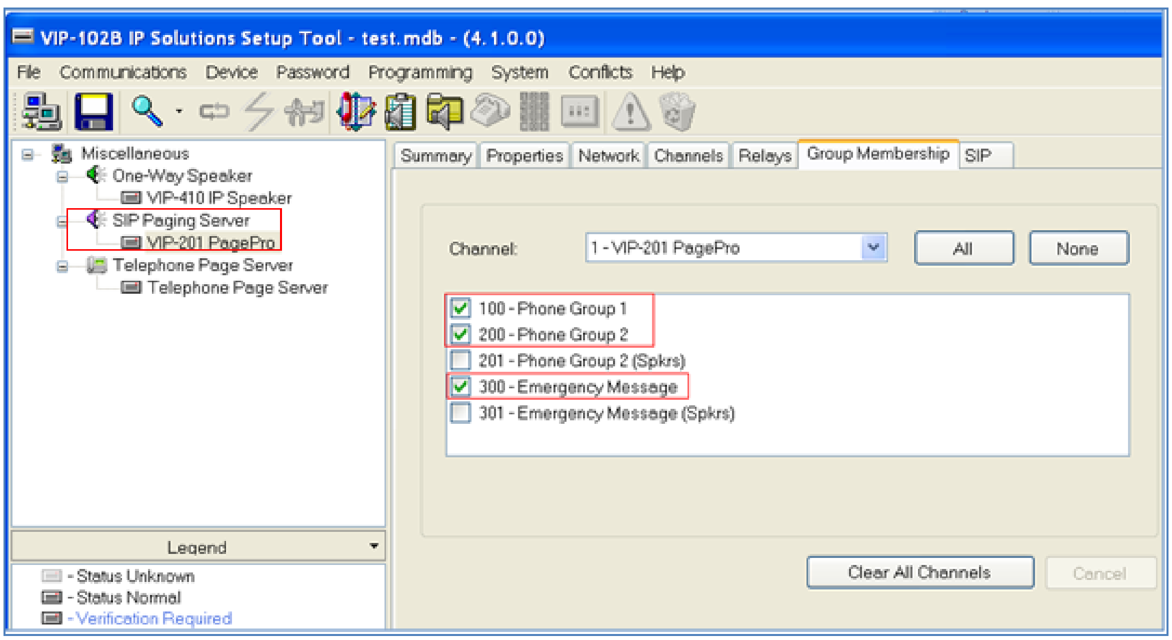


7.3. Administer Group Membership

Select the **Group Membership** tab. Follow the appropriate documentation in **Section 10** to create the applicable groups. Following is the groups assigned to **VIP-410 IP Speaker**.



The following figure shows the list of groups assigned to **VIP-201 PagePro**:



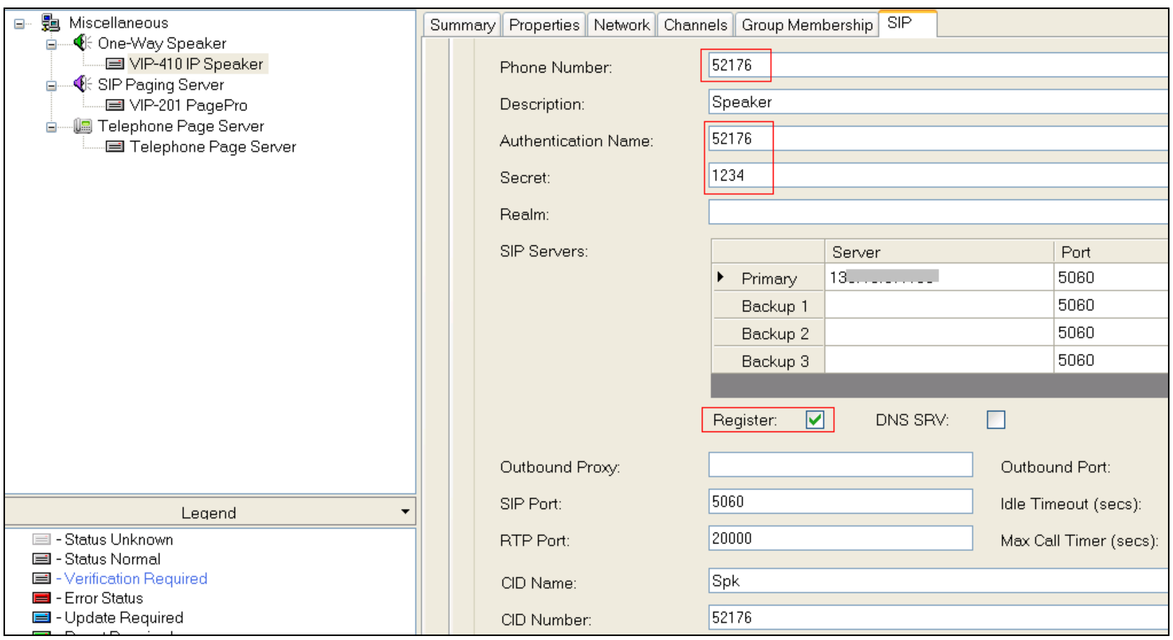
7.4. Administer SIP

SIP configuration was required for the VIP-201 PagePro IP and VIP -410 IP Speaker that were used as part of the compliance testing. The steps noted here are for completeness of the compliance testing documentation. These steps are not required for the VE6023 Telephone Page Server.

7.4.1. Administer SIP for VIP-410 IP Speaker

Select the **SIP** tab of the **VIP-410 IP Speaker**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Phone Number:** Enter the **Extension** from **Section 0**.
- **Authentication Name:** Enter the **Extension** from **Section** Error! Reference source not found..
- **Secret:** Enter the **Communication Profile Password** from **Section 0**.
- **SIP Server:** Enter the IP address of Session Manager.
- **Register:** Check this field.

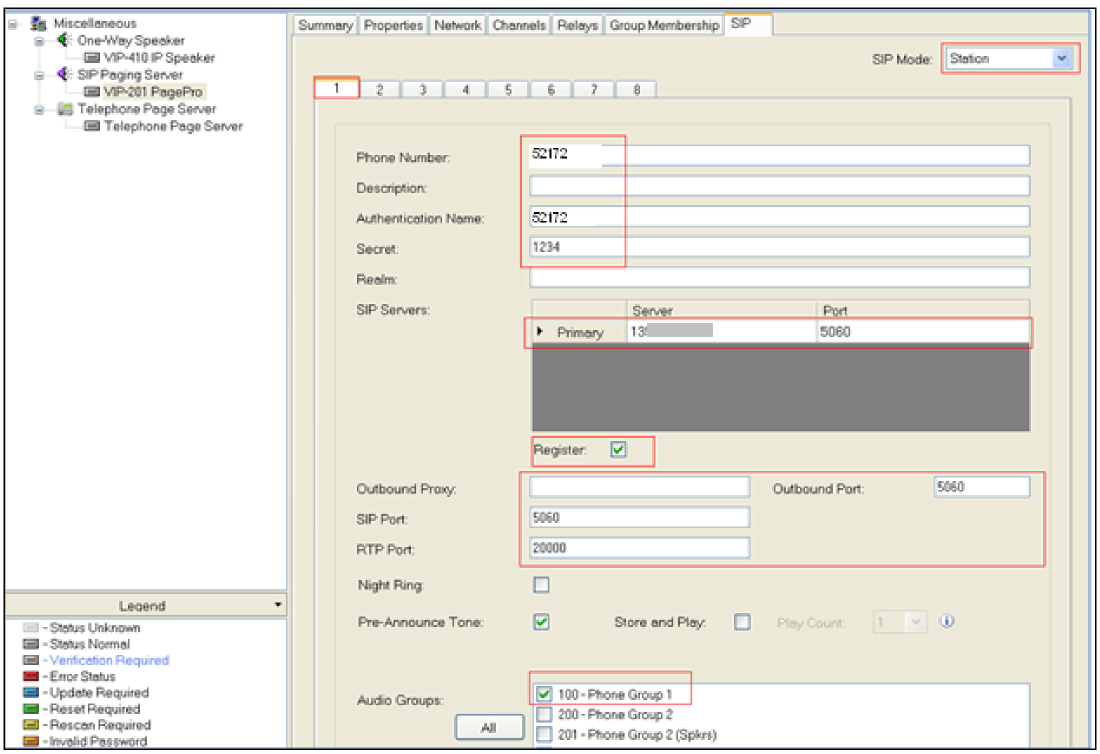


7.4.2. Administer SIP for VIP-201 PagePro IP

Select the **SIP** tab of the **VIP-201 PagePro IP**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Phone Number:** Enter the **Extension** from **Section 0**.
- **Authentication Name:** Enter the **Extension** from **Section 0**.
- **Secret:** Enter the **Communication Profile Password** from **Section 0**.
- **SIP Server:** Enter the IP address of **Session Manager**.
- **Register:** Check this field.
- **Pre-Announce Tone:** Checked this field.
- **Audio Group:** Check the selected group for SIP extension.

The following is an example showing Extension 52172 is assigned to Group 1 on PagePro IP.



Select tab **2** to administer **SIP Identity 2** (not shown), and use the credentials for the second SIP user from **Section 0**. Repeat this section to administer all SIP identities. During compliance test, only three extensions were configured on PagePro IP.

Click on the **Update Changed Devices** icon above. The **Reset Required** dialog box will appear as shown below. Click **Yes** to reset the updated devices (not shown).

7.5. Administer Valcom VE6023 Telephone Page Server

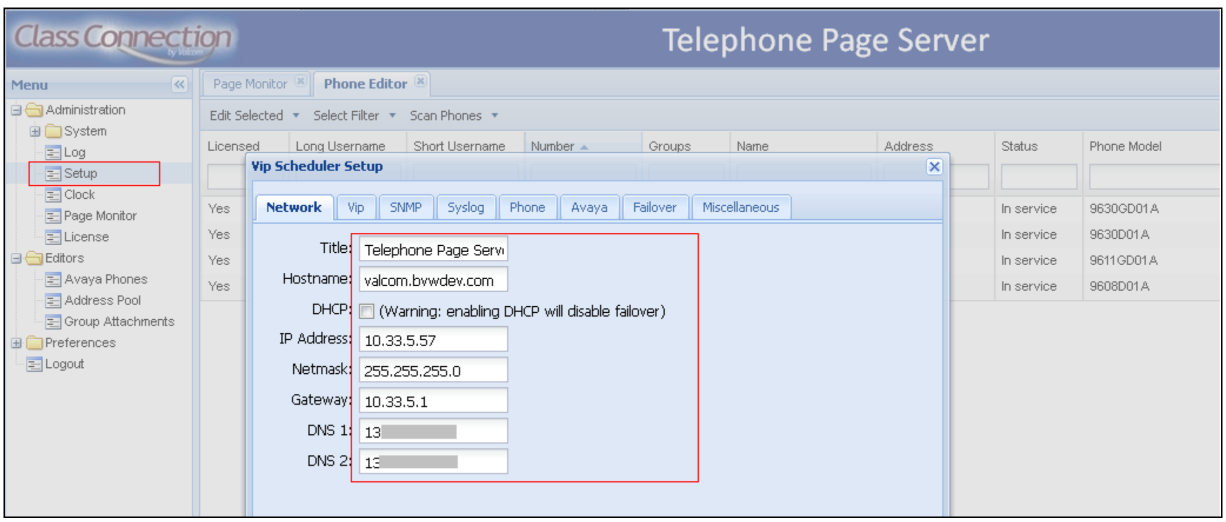
This section describes the steps to configure VE6023 through the web page. The procedure includes the following areas:

- Configure Network Parameters.
- Configure Avaya Phones.
- Configure Address Pool.
- Configure Group Attachment.

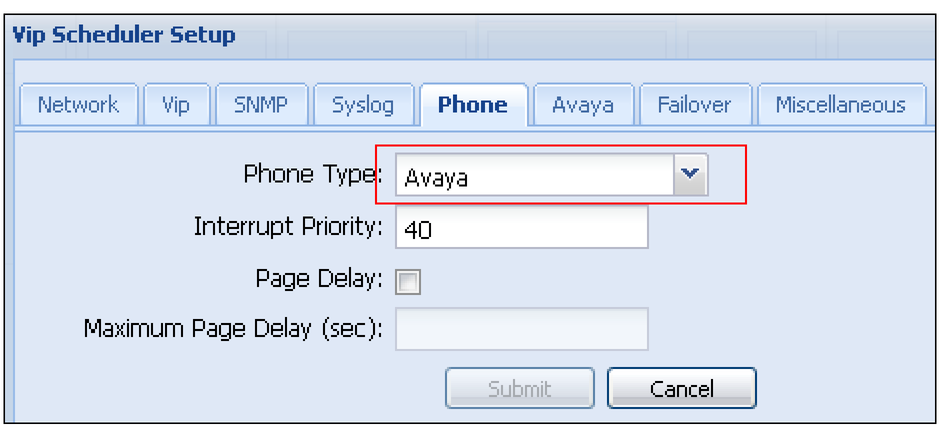
To access the web interface, log in to the system by entering its IP address in to a web browser, see **Section 7.2** for the IP address of VE6023. The default username is 'admin' and the default password is '4cc3ss'. It is recommended that the default password be changed.

7.5.1. Configure Network Parameters

Access the **Setup** window by clicking the **Setup** item on the left-hand navigation panel. The **Setup** window consists of several tabs. Following is the Network tab of VE6023 during the compliance test.



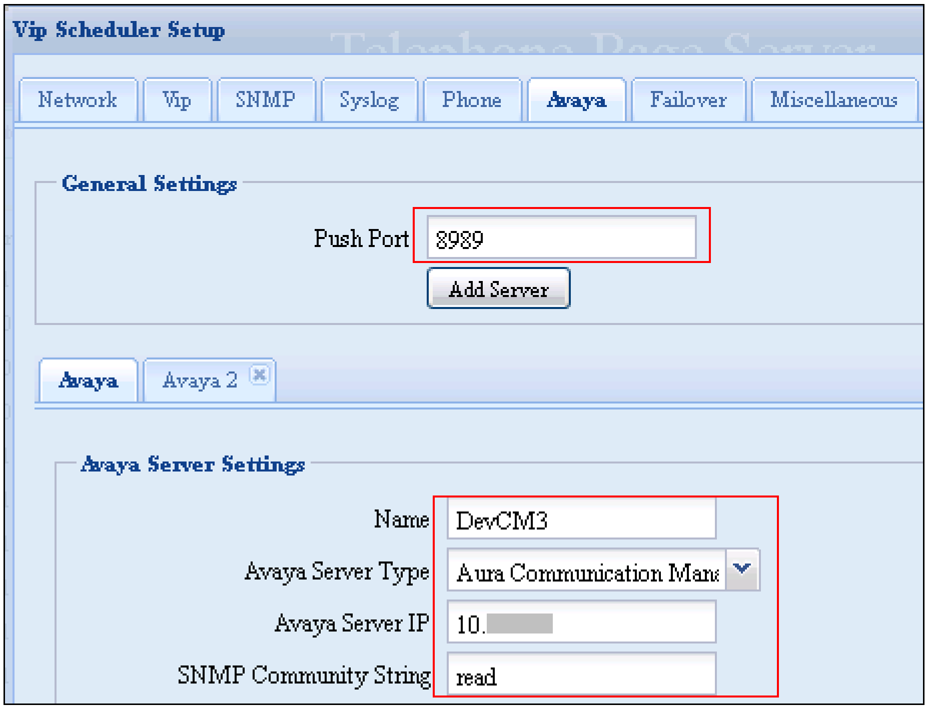
Select **Phone** tab, select **Avaya** for **Phone Type**. Click **Submit** to save changes.



Select **Avaya** tab:

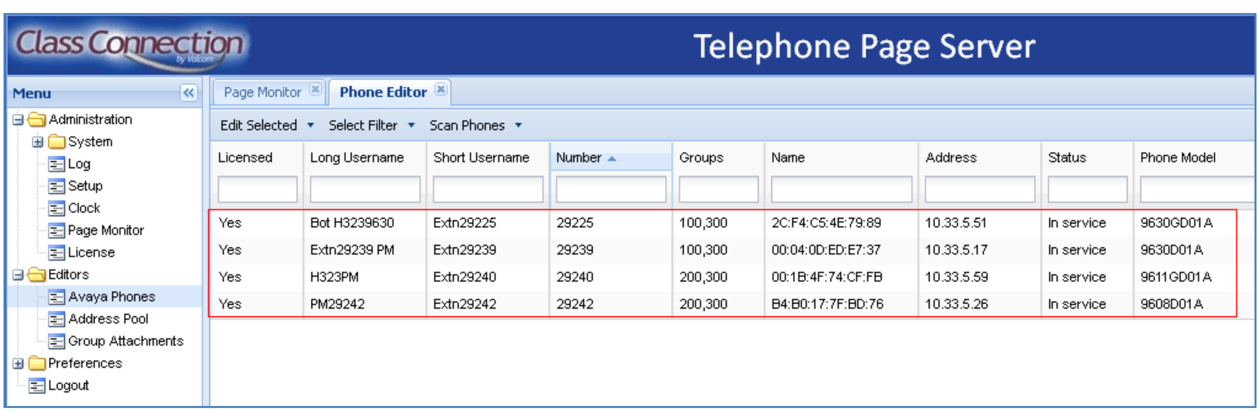
- **Push Port**: **8989** was used during the compliance test.
- **Name**: Enter a descriptive name.
- **Avaya Server IP**: Enter the IP address of Communication Manager.
- **SNMP Community String**: Enter the string created in **Section 5.3**.

Click **Submit** to save changes.

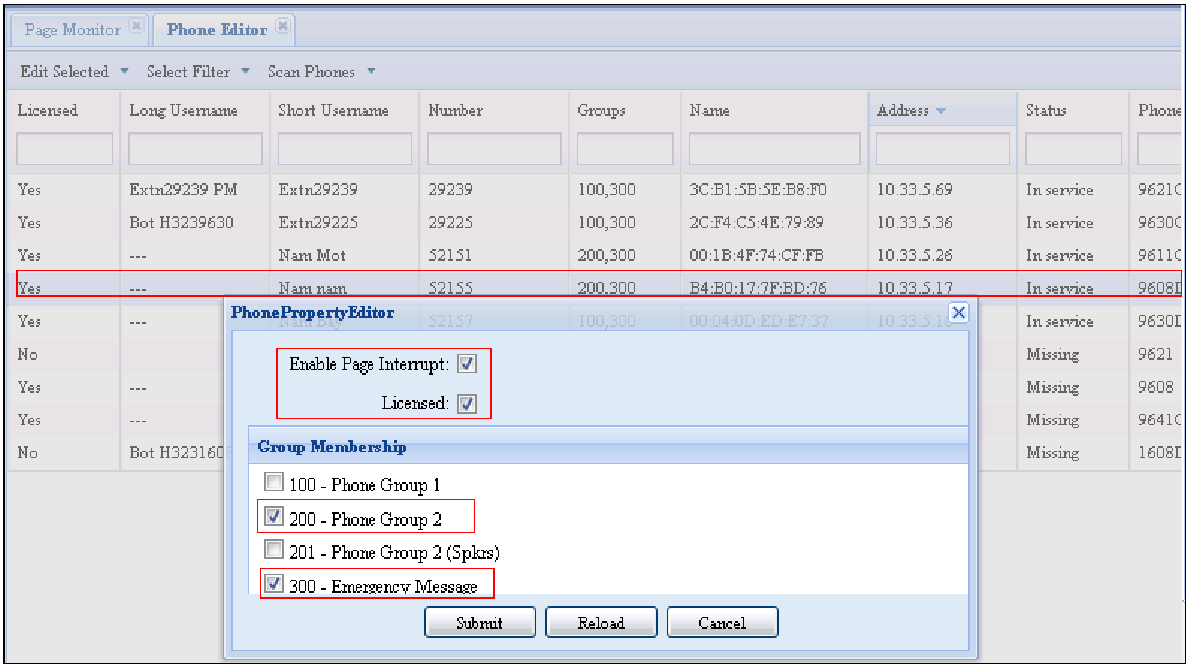


7.5.2. Configure Avaya Phone:

Select **Editors → Avaya Phones,** the Phone Editor provides a list of all the phones the VE6023 was able to discover in the network.



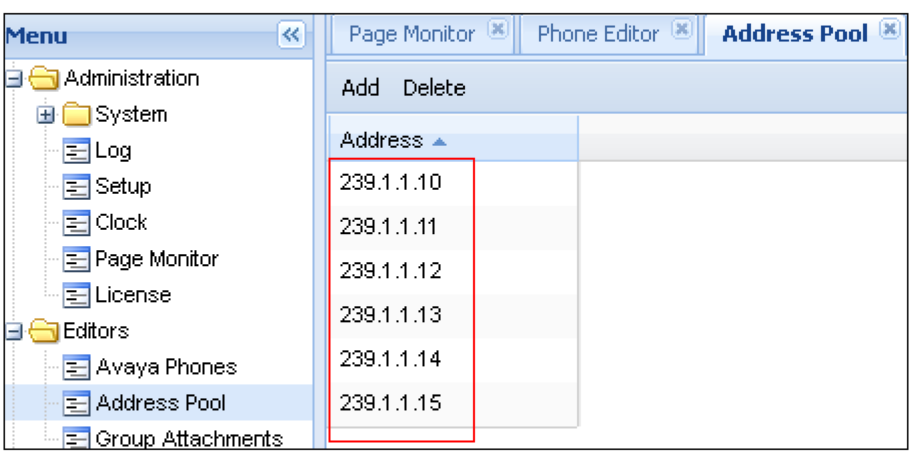
Double click on a phone to add phones to page groups, manage which phones are licensed, and control page interrupt settings for each phone. The following figure shows that device with extension 52155 which belongs to group 200, 300, is licensed and Page Interrupt is Enabled.



7.5.3. Configure Address Pool

The VE6023 streams audio to the IP phones using multicast. By default, the multicast address 239.1.1.10 is the only address used. This can be changed using the Address Pool editor. By default, only a single address is in this pool. To support multiple simultaneous pages, several addresses should be added to the pool.

Select **Editors → Address Pool,** click **Add** to add more addresses. Following is the list of address used during the compliance test.



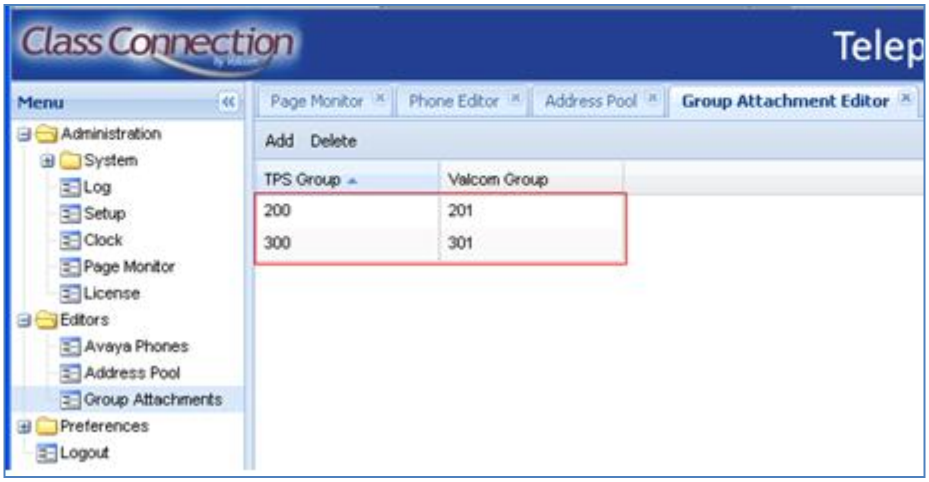
7.5.4. Configure Group Attachment

The VE6023 detects a page has started and begins setting up the IP Phones. During this setup time, the page audio is buffered. As a result, if Valcom speakers and IP Phones are both playing the same page they may be out of sync. To solve this problem, the VE6023 can source the audio to both IP Phones and Valcom Speakers and ensure they stay synchronized. This is done via the Group Attachment Editor.

Select **Editors → Group Attachment,** click **Add.**

- **TPS Group Code**: Select a selected TPS group from the list.
- **Valcom group Code**: Select a Valcom group.

Click Submit to save changes. Following is the list of Group Attachments used during the compliance test.



8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and the VE6023 Valcom Telephone Page Server. These steps verify the functionality in conjunction with the components used in the compliance test (VIP-201 and VIP-401).

8.1. Verify User Registrations

On Session Manager, verify the registration status of the Valcom SIP devices by navigating to **Elements → Session Manager → System Status → User Registrations**. Verify that all the users administered in **Section 0** are listed as registered users.

8.2. Verify Valcom Telephone Page Server

Generate a page to one of the Valcom page groups assigned to one or more telephones. Verify that the page audio is connected to the appropriate phone group with a one-way talk path.

9. Conclusion

These Application Notes describe the configuration steps required for Valcom Telephone Page Server to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

10. Additional References

This section references the product documentation relevant to these Application Notes.

1. *Administering Avaya Aura® Communication Manager*, Document 03-300509, Issue 7.0, Release 6.2, February 2012, available at http://support.avaya.com

2. *Administering Avaya Aura® Session Manager* available at http://support.avaya.com

3. Telephone Page Server SIP Based Paging Server documentation is available at http://www.valcom.com

4. Valcom Talkback IP Speaker documentation is available at http://www.valcom.com

5. Valcom VIP-102B IP Solutions Setup Tool Reference Manual is available at http://www.valcom.com

6. Telephone Page Server VE6023 User Manual is available at http://www.valcom.com

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.