Avaya Solution & Interoperability Test Lab

# Application Notes for Valcom V-9972 Universal Paging Interface with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Endpoint - Issue 1.0

## Abstract


These Application Notes describe the configuration steps required to integrate the Valcom V-9972 Universal Paging Interface with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Valcom V-9972 Universal Paging Interface provides access to paging systems, such as Valcom VIP-430A IP Wall Speakers, which was used in the compliance test. For this compliance test, Valcom V-9972 Universal Paging Interface registered with Avaya Aura® Session Manager as a SIP endpoint. In addition, Valcom V-9972 Universal Paging Interface also registered with Avaya Aura® Session Manager through Avaya Session Border Controller for Enterprise as a remote worker. The Valcom V-9972 Universal Paging Interface supports two-way audio intercom (talkback) calls and one-way audio group paging calls.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 to integrate the Valcom V-9972 Universal Paging Interface with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Valcom V-9972 Universal Paging Interface provides access to paging systems, such as Valcom VIP-430A IP Wall Speakers, which was used in the compliance test. For this compliance test, Valcom V-9972 Universal Paging Interface registered with Avaya Aura® Session Manager as a SIP endpoint. In addition, Valcom V-9972 Universal Paging Interface also registered with Avaya Aura® Session Manager through Avaya Session Border Controller for Enterprise as a remote worker. The Valcom V-9972 Universal Paging Interface supports two-way audio intercom (talkback) calls and one-way audio group paging calls.

When a call is placed to the Valcom V-9972 Universal Paging Interface using its direct dial SIP extension, the V-9972 plays dial tone back to the caller. The caller can then dial a Valcom speaker Dial Code or Group Code to establish an intercom call (two-way audio) with a single Valcom speaker or a group paging call (one-way audio) to one or more Valcom speakers.

Alternatively, the Valcom VIP-430A IP Wall Speaker can establish intercom calls by pressing its call button. Pressing the call button would place a call to the specified destination in the V-9972 configuration. Pressing the call button during an active call, terminates the call.

All calls to/from the VIP-430A IP Wall Speaker go through the V-9972. Communication between V-9972 and VIP-430A IP Wall Speaker uses unicast for intercom (talkback) calls and multicast for paging calls.

Valcom offers Universal Paging Adapters as different products/models to accommodate different environments. They share the same SIP stack and firmware version, therefore, this testing also applies to those products, as detailed in **Attachment 1**. **Section 4** of this document shows the actual products/models and SIP Stack and software versions that were tested. For additional details, contact Valcom Support, as noted in **Section 2.3**.

# 2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between the Valcom V-9972 Universal Paging Interface with the Valcom VIP-430A IP Wall Speaker, Avaya SIP / H.323 IP Deskphones, and the PSTN. Two-way audio intercom calls and one-way audio group paging calls were exercised. In addition, basic telephony features were exercised from Avaya SIP / H.323 IP Deskphones, such as hold/resume, call transfer, and conference.

The serviceability testing focused on verifying that the Valcom V-9972 Universal Paging Interface came back into service after reconnecting the network connection or a reboot.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Valcom V-9972 Universal Paging Interface used TLS/SRTP encryption features.

## 2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of V-9972 directly with Session Manager as a SIP endpoint.
- SIP registration of V-9972 with Session Manager through Avaya Session Border Controller for Enterprise as a remote worker.
- Calls between V-9972 and Avaya H.323/SIP endpoints with Direct IP Media (Shuffling) enabled and disabled. Shuffling allows IP endpoints to send audio RTP packets directly to each other without using media resources on Avaya Media Gateway or Avaya Aura® Media Server.
- Establishing two-way audio intercom calls between VIP-430A IP Wall Speaker, via V-9972, Avaya H.323 / SIP Deskphones, and PSTN in both directions.
- Establishing one-way paging calls from Avaya H.323 / SIP Deskphones to VIP-430A IP Wall Speaker via V-9972.
- Verifying that higher priority paging calls take precedence over existing lower priority intercom calls.
- Terminating calls by pressing the call button on the VIP-430A IP Wall Speaker.
- Support of G.711 mu-law codec.
- Support of TLS/SRTP using mutual TLS authentication.
- Since the VIP-430A IP Wall Speaker does not provide a keypad or feature buttons, basic telephony features, such as hold/resume, call transfer, and conference were performed from Avaya H.323/SIP Deskphones.
- Long duration calls and outbound calls from V-9972 that were rejected due to dialing an invalid number or a busy station.
- Proper system recovery after re-establishing network connectivity to the V-9972 or restarting the V-9972.

## 2.2. Test Results

All test cases passed.

## 2.3. Support

For technical support and information on Valcom V-9972 Universal Paging Interface, contact Valcom Technical Support at:

- Phone: +1 (800) 825-2661 or +1 (540) 563-2000
- Website: https://www.valcom.com/Support/techsupport.html
- Email: support@valcom.com

# 3. Reference Configuration

**Figure 1** illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Media Gateway.
- Media resources in Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP endpoints, including the V-9972.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Session Border Controller for Enterprise to provide connectivity to a simulated SIP service provider or to register the V-9972 as a remote worker.
- Avaya 96x1 Series H.323 and SIP Deskphones.
- Valcom V-9972 Universal Paging Interface and Valcom VIP-430A IP Wall Speaker.

V-9972 Universal Paging Interface registered with Session Manager as a SIP endpoint and was configured as Off-PBX Stations (OPS) on Communication Manager.

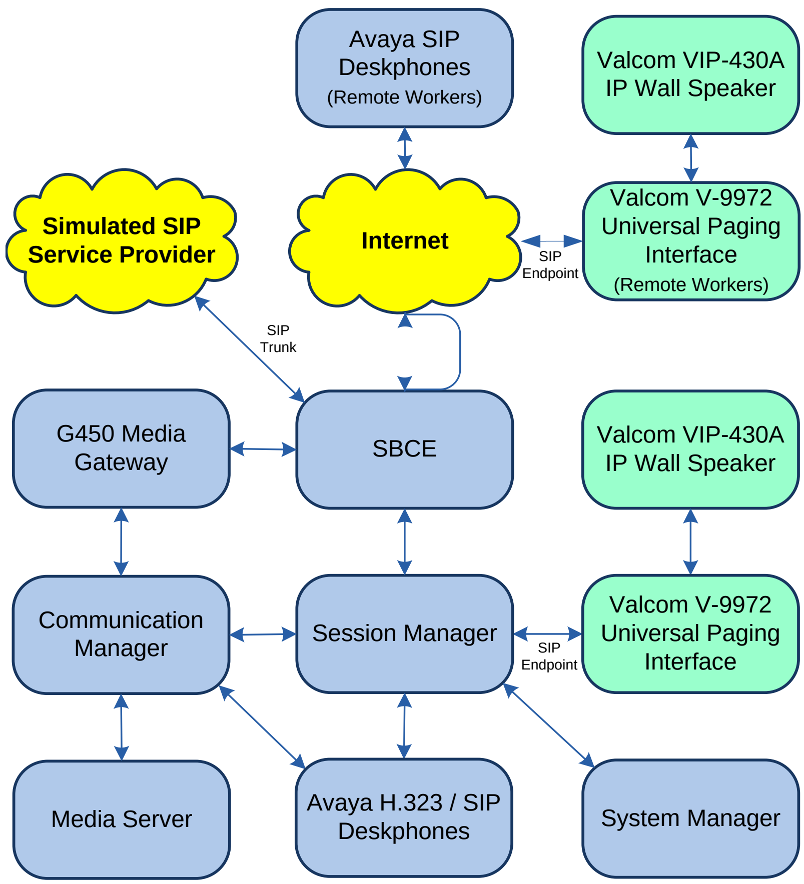



**Figure 1: Avaya SIP Network with Valcom V-9972 Universal Paging Interface and Valcom VIP-430A IP Wall Speakers**

# 4. Equipment and Software Validated

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

| Equipment/Software | Release/Version |
|---|---|
| Avaya Aura® Communication Manager | 8.1.3.4.0-FP3SP4 |
| Avaya G450 Media Gateway | 41.34.4 |
| Avaya Aura® Media Server | 8.0.2.138 |
| Avaya Aura® System Manager | 8.1.3.4<br>Build No. – 8.1.0.0.733078<br>Software Update Revision No: 8.1.3.4-1014185 |
| Avaya Aura® Session Manager | 8.1.3.4.813401 |
| Avaya Session Border Controller for Enterprise | 8.1.2.0-19794 |
| Avaya 96x1 Series IP Deskphones | 6.8511 (H.323) |
| Avaya J100 Series IP Deskphones | 4.0.10.3.2 (SIP) |
| Valcom V-9972 Universal Paging Interface, including optional L9972-2 feature license | 3.00.14 |
| Valcom VIP-430A IP Wall Speaker | 3.23.7 |
| Valcom VIP-102B IP Solutions Setup Tool | 8.4.0.0 |

# 5. Configure Avaya Aura® Communication Manager

This section provides the procedure for configuring Communication Manager. The procedure includes the following areas:

- Verify Communication Manager license
- Administer IP Node Names
- Administer IP Network Region and IP Codec Set
- Administer SIP Trunk Group to Session Manager
- Administer AAR Call Routing

Use the System Access Terminal (SAT) to configure Communication Manager and log in with appropriate credentials.

**Note:** The SIP station configuration for Valcom V-9972 Universal Paging Interface is configured through Avaya Aura® System Manager in **Section 6.3**.

## 5.1. Verify Communication Manager License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                      Page   1 of  12
                                OPTIONAL FEATURES

     G3 Version: V18                                 Software Package: Enterprise
       Location: 2                                    System ID (SID): 1
       Platform: 28                                   Module ID (MID): 1

                                                               USED
                            Platform Maximum Ports:  48000    131
                                  Maximum Stations:  36000     37
                          Maximum XMOBILE Stations:  36000      0
                Maximum Off-PBX Telephones - EC500:  41000      0
                Maximum Off-PBX Telephones -   OPS:  41000     23
                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        0




        (NOTE: You must logoff & login to effect the permission changes.)
```

## 5.2. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). These host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                                Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
devcon-aes          10.64.102.119
devcon-ams          10.64.102.118
devcon-sm           10.64.102.117
procr               10.64.102.115
procr6              ::

( 6  of 6    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

## 5.3. Administer IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio RTP traffic to be sent directly between IP endpoints or between Communication Manager and SBCE for remote workers without using media resources in Avaya Aura® Media Servers after the call is established. Note that for remote workers, media is anchored at the SBCE so remote workers will always send/receive audio to/from the SBCE, not directly between each other. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. The UDP port range is also specified in this form.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name:                       Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 50999
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                         RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

## 5.4. Administer IP Codec Set

In the **IP Codec Set** form, the audio codec type supported for calls routed over the SIP trunk to V-9972 is specified. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set 1 was specified in IP Network Region 1 shown above. The default settings of the **IP Codec Set** form are shown below. V-9972 supports G.711 codecs with the VIP-430A IP Wall Speaker.

To enable SRTP, **Media Encryption** was set to *1-srtp-aescm128-hmac80* and **Encrypted SRTCP** was left at the default value of *best-effort*. Note that RTP, which would be indicated by *none* under **Media Encryption**, must not be included.

```
change ip-codec-set 1                                          Page   1 of   2

                          IP MEDIA PARAMETERS
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n          2        20
 2:
 3:
 4:
 5:
 6:
 7:

     Media Encryption                  Encrypted SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: 2-srtp-aescm128-hmac32
 3:
 4:
 5:
```

## 5.5. Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Set the **Enforce SIPS URI for SRTP** field to *n*.
- Specify Communication Manager (*procr*) and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Enable **Initial IP-IP Direct Media**.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 10                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 10              Group Type: sip
  IMS Enabled? n         Transport Method: tls
        Q-SIP? n
     IP Video? y                                 Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                      Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                 Far-end Node Name: devcon-sm
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                    Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to/from V-9972, Avaya SIP Deskphones, and the PSTN. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie* or *public-ntwrk*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 10                                                Page   1 of  22
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: To devcon-sm               COR: 1        TN: 1        TAC: 1010
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: public-ntwrk          Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 10
                                                     Number of Members: 10
```

**Page 5** of the SIP trunk group was configured as follows.

```
add trunk-group 10                                                Page   5 of   5
                              PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                       Send Transferring Party Information? n
                                  Network Call Redirection? n

                                     Send Diversion Header? n
                                   Support Request History? y
                              Telephone Event Payload Type: 101


                        Convert 180 to 183 for Early Media? n
                  Always Use Re-INVITE for Display Updates? n
     Resend Display UPDATE Once on Receipt of 481 Response? n
                        Identity for Calling Party Display: P-Asserted-Identity
            Block Sending Calling Party Location in INVITE? n
                 Accept Redirect to Blank User Destination? n
          Enable Q-SIP? n
          Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
                                Request URI Contents: may-have-extra-digits
```

## 5.6. AAR Call Routing

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and enter add an entry that routes digits beginning with "78" to route pattern 10 as shown below.

```
change aar analysis 78                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 1

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    78                    5    5     10        lev0         n
```

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 as shown below.

```
change route-pattern 10                                         Page   1 of   3
                    Pattern Number: 10     Pattern Name: To devcon-sm
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                                DCS/ IXC
    No          Mrk Lmt List Del  Digits                                  QSIG
                             Dgts                                         Intw
 1: 10   0                                                                 n   user
 2:                                                                        n   user
 3:                                                                        n   user
 4:                                                                        n   user
 5:                                                                        n   user
 6:                                                                        n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               unk-unk   none
 2: y y y y y n  n            rest                                         none
```

# 6. Configure Avaya Aura® Session Manager

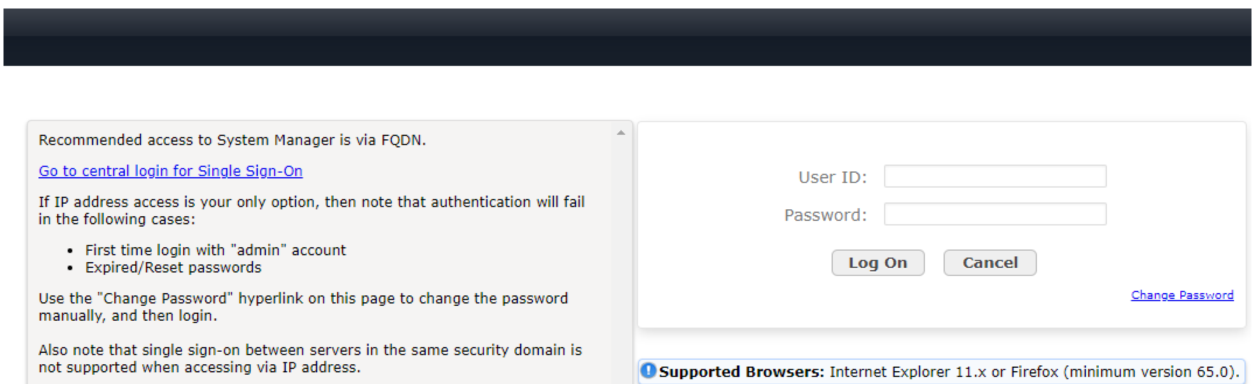
This section provides the procedure for configuring Session Manager, which is required whether V-9972 registers directly with Session Manager or through SBCE as a remote worker. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol
- Administer SIP User
- Install Valcom V-9972 Universal Paging Interface TLS Certificate

**Note:** It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for Valcom V-9972 Universal Paging Interface.

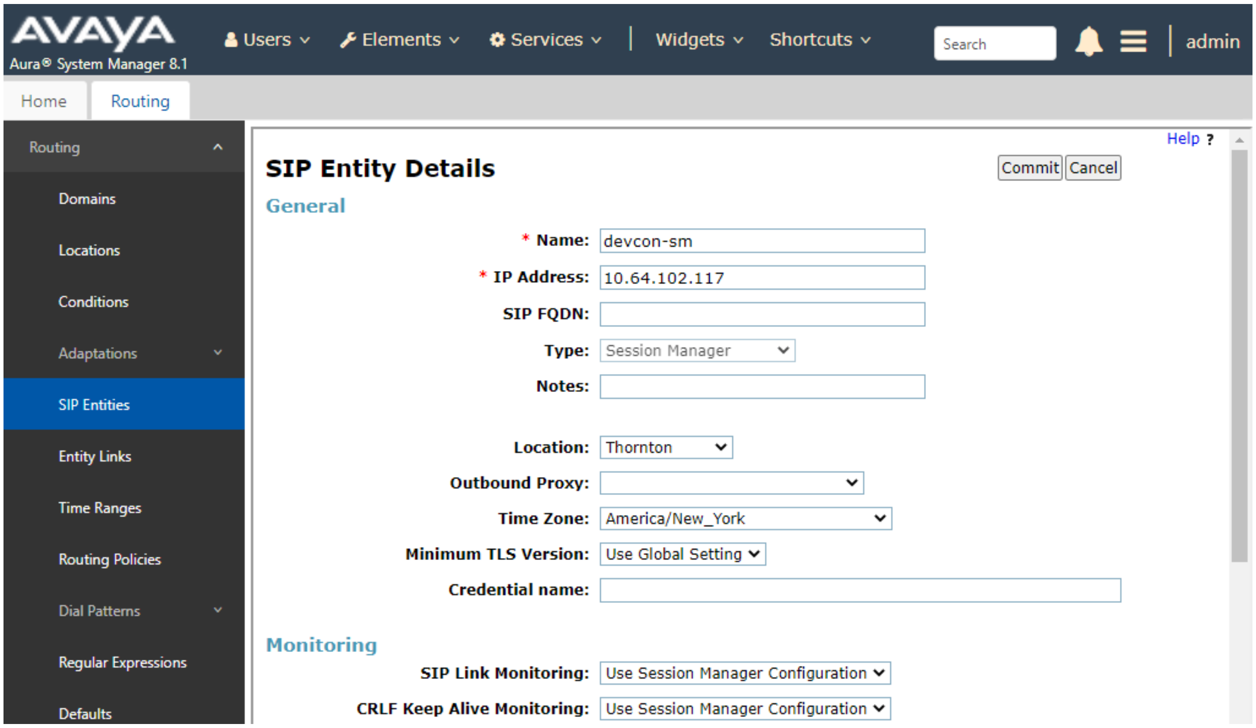
## 6.1. Launch System Manager

Access the System Manager Web interface by using the URL *https://<ip-address>* in an Internet browser window, where *<ip-address>* is the IP address of the System Manager server. Log in using the appropriate credentials.

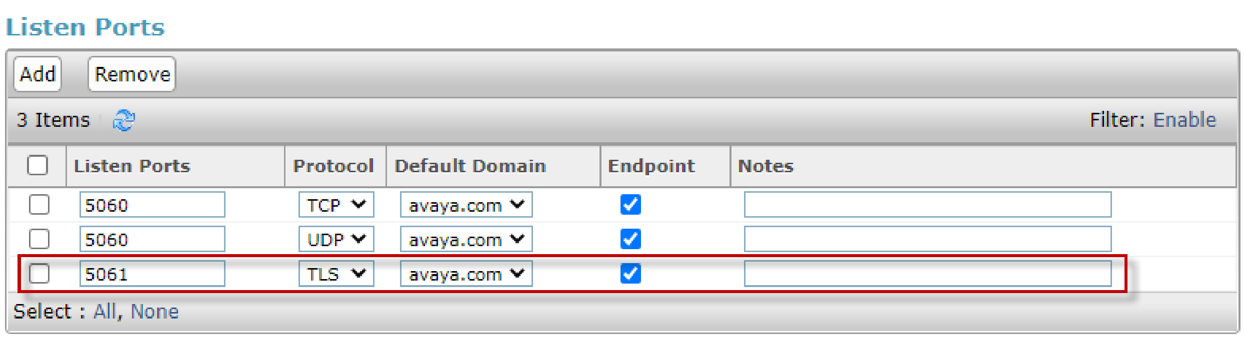


## 6.2. Set Network Transport Protocol

From the System Manager **Home** screen, select **Elements → Routing → SIP Entities** and edit the SIP Entity for Session Manager shown below.

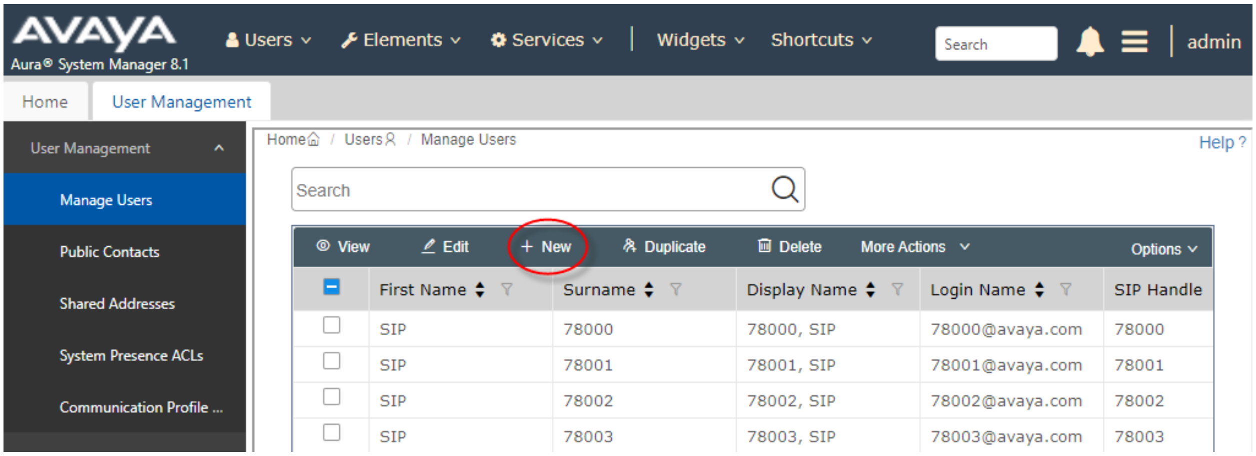



Scroll down to the **Listen Ports** section and verify that the transport network protocol used by V-9972 is specified in the list below. For the compliance test, the solution used TLS network transport.



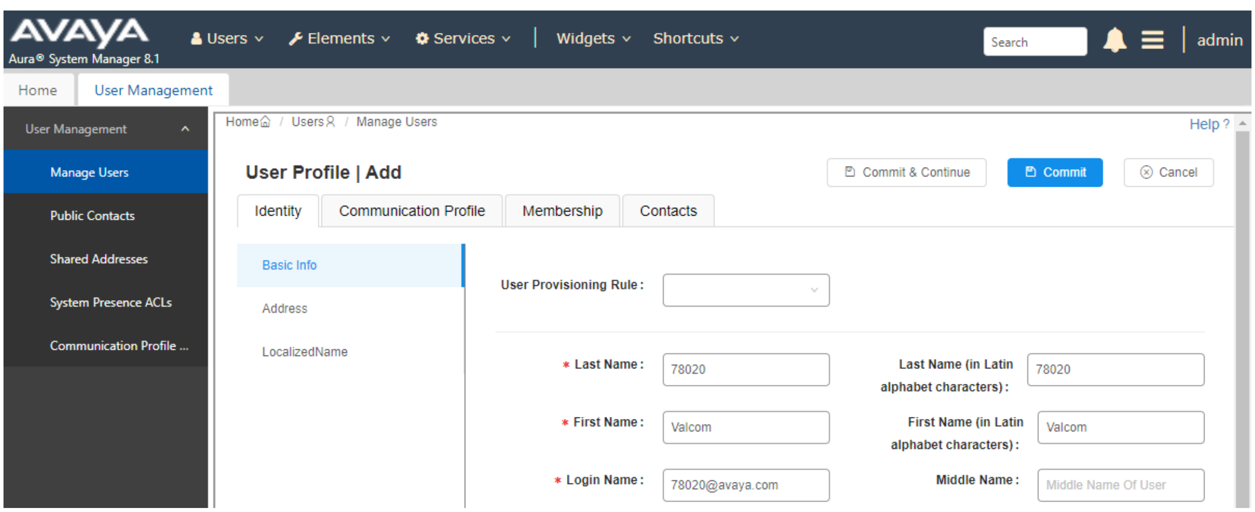
## 6.3. Administer SIP User

In the **Home** screen (not shown), select **Users → User Management → Manage Users** to display the **User Management** screen below.  Click **New** to add a user.



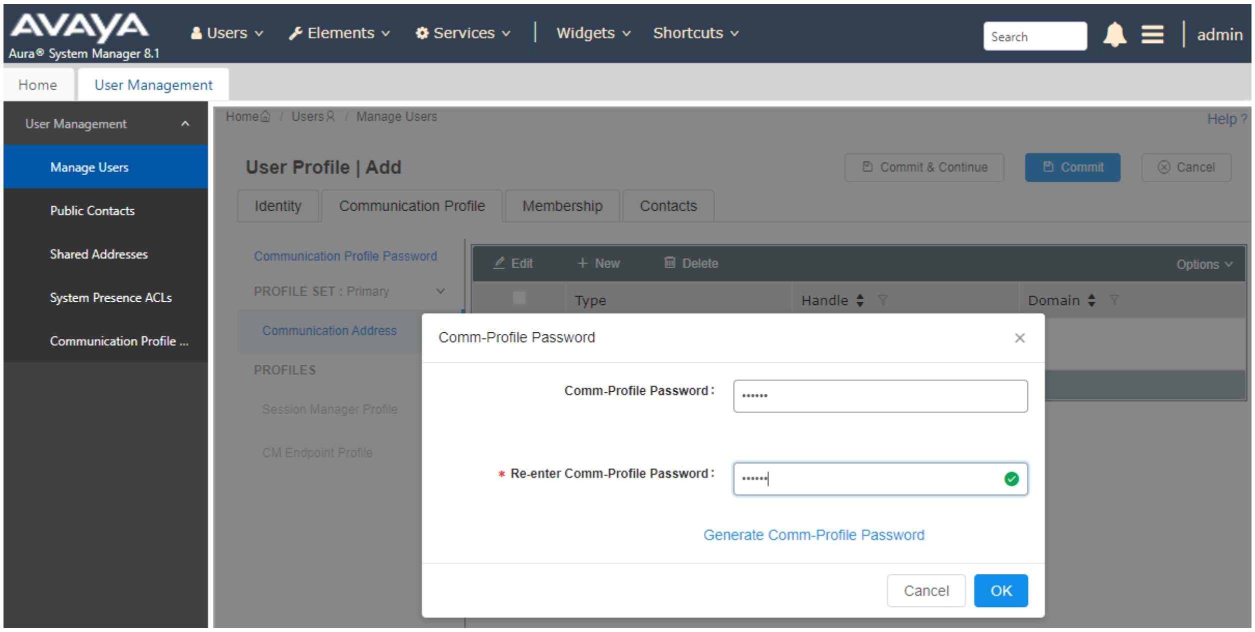

### 6.3.1. Identity

The **New User Profile** screen is displayed.  Enter desired **Last Name** and **First Name**.  For **Login Name**, enter "*<ext>@<domain>*", where "*<ext>*" is the desired V-9972 SIP extension and "*<domain>*" is the applicable SIP domain name from **Section 5.3**.  Retain the default values in the remaining fields.



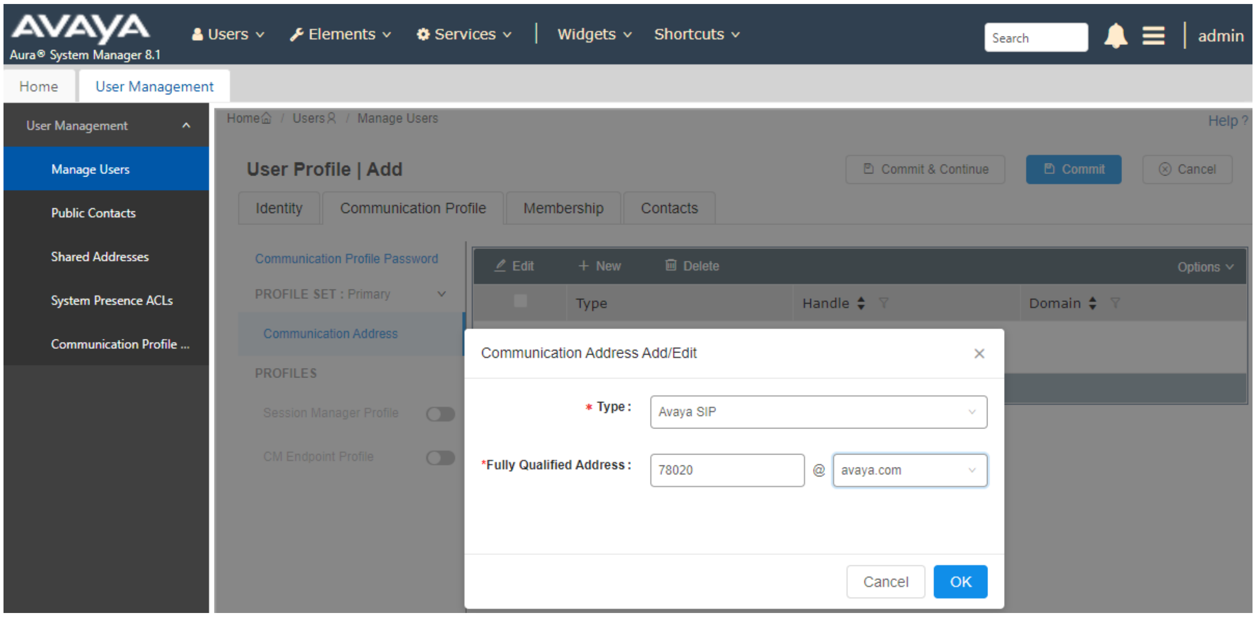
## 6.3.2. Communication Profile

Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration. Click **OK**.



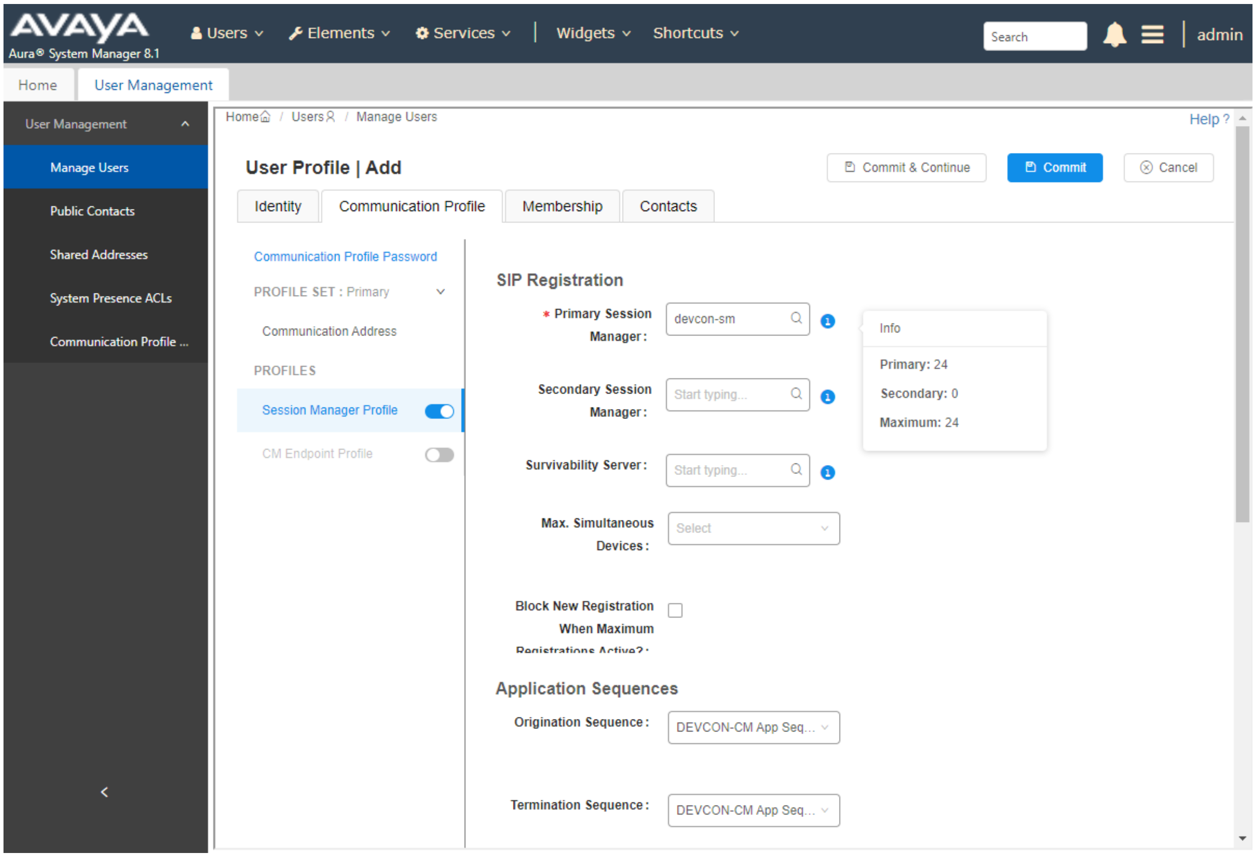
### 6.3.3. Communication Address

Click on **Communication Address** and then click **New** to add a new entry. The **Communication Address Add/Edit** dialog box is displayed as shown below. For **Type**, select *Avaya SIP*. For **Fully Qualified Address**, enter the SIP user extension and select the domain name to match the login name from **Section 6.3.1**. Click **OK**.

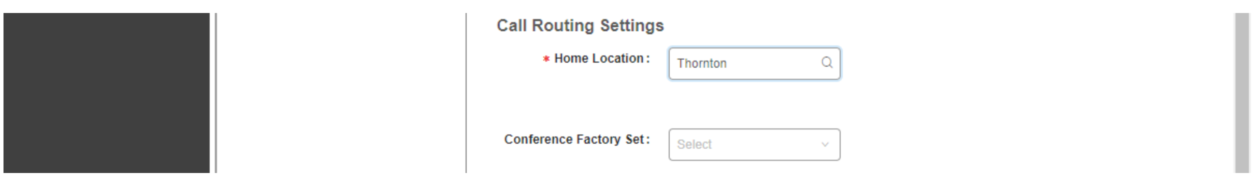


## 6.3.4. Session Manager Profile

Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

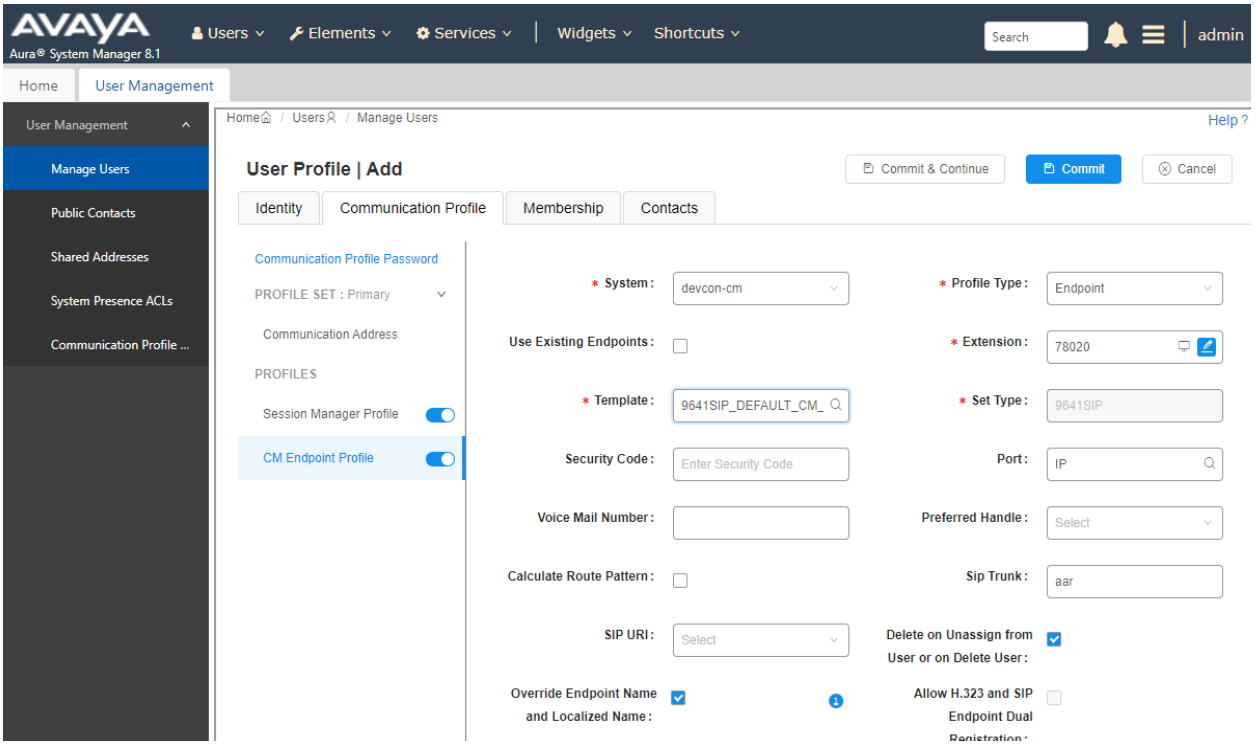



Scroll down to the **Call Routing Settings** section to configure the **Home Location**.



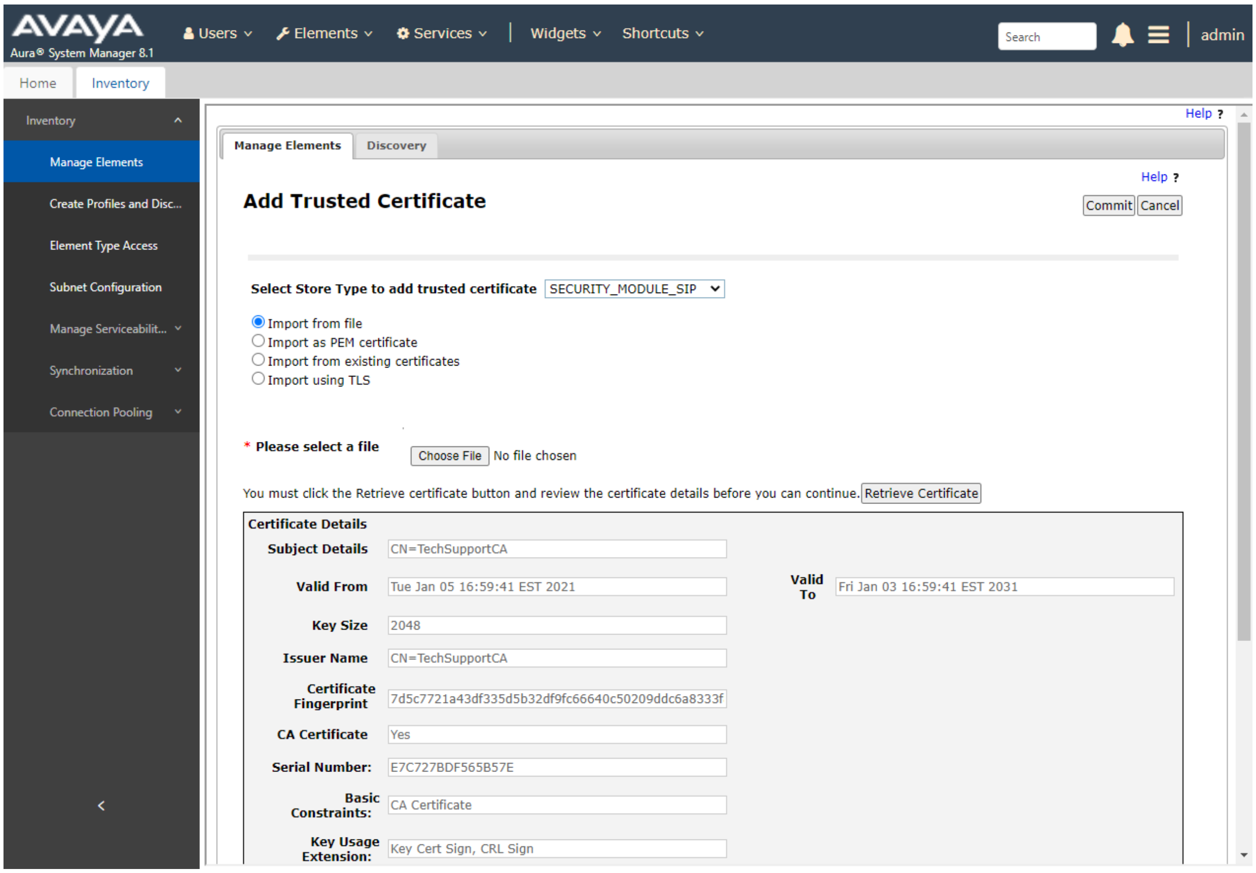
### 6.3.5. CM Endpoint Profile

Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9641SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields.

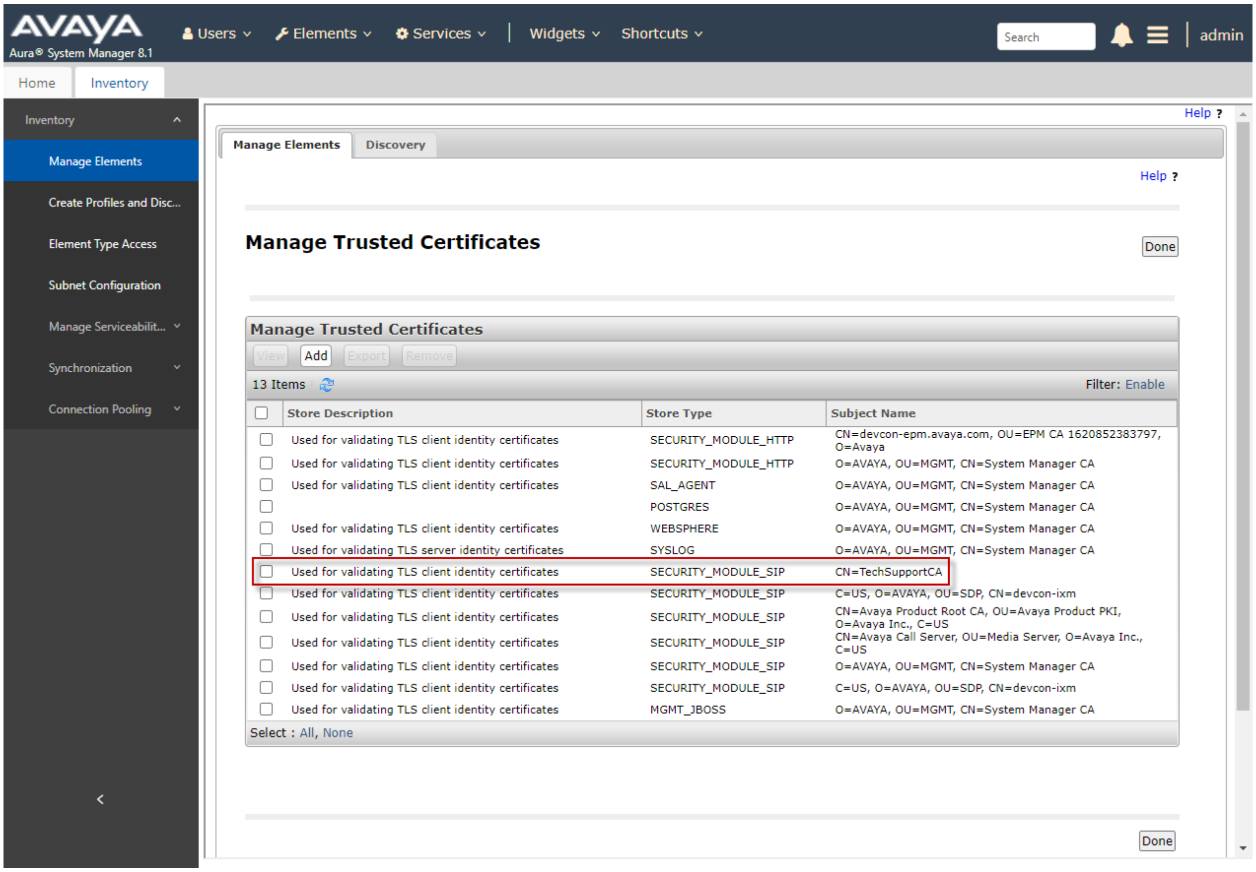


## 6.4. Install Valcom V-9972 Universal Paging Interface TLS Certificate

To support mutual TLS authentication, the V-9972 TLS certificate must be installed on Session Manager. From System Manager Web interface, navigate to **Services → Inventory → Manage Elements** and select checkbox for the Session Manager. From the **More Actions** drop-down box, select **Manage Trusted Certificate** (not shown). In **Manage Trusted Certificates**, click **Add**. In Add Trusted Certificate, select *SECURITY_MODULE_SIP* in the **Select Store Type to add trusted Certificate** field. Click the **Import from file** radio button and select the certificate file (e.g., *technicalsupportca.crt*). Next, click on **Retrieve Certificate** and then **Commit**.

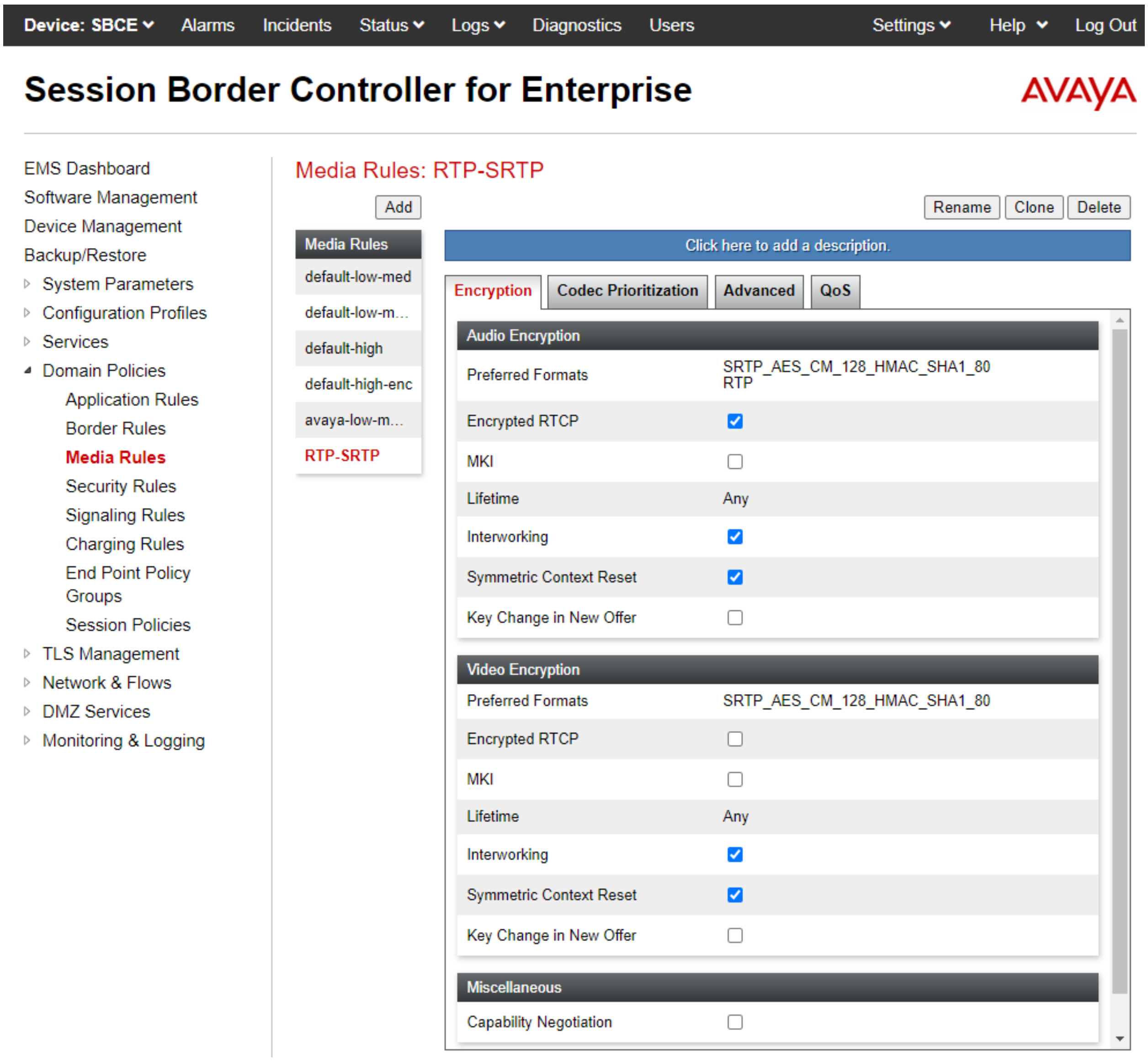


After the certificate has been imported, it should be listed in **Manage Trusted Certificates** as shown below.



# 7. Configure Avaya Session Border Controller

These Application Notes assume that the SBCE is already configured to support remote workers. No additional configuration is required to support V-9972 as a remote worker. However, it would be instructive to show how the **Media Rules** were configured to support SRTP for calls to V-9972 as a remote worker. This media rule is assigned to an **End Point Policy Group**, which in turn is assigned to **Subscriber Flows** and **Server Flows.**



# 8. Configure Valcom V-9972 Universal Paging Interface

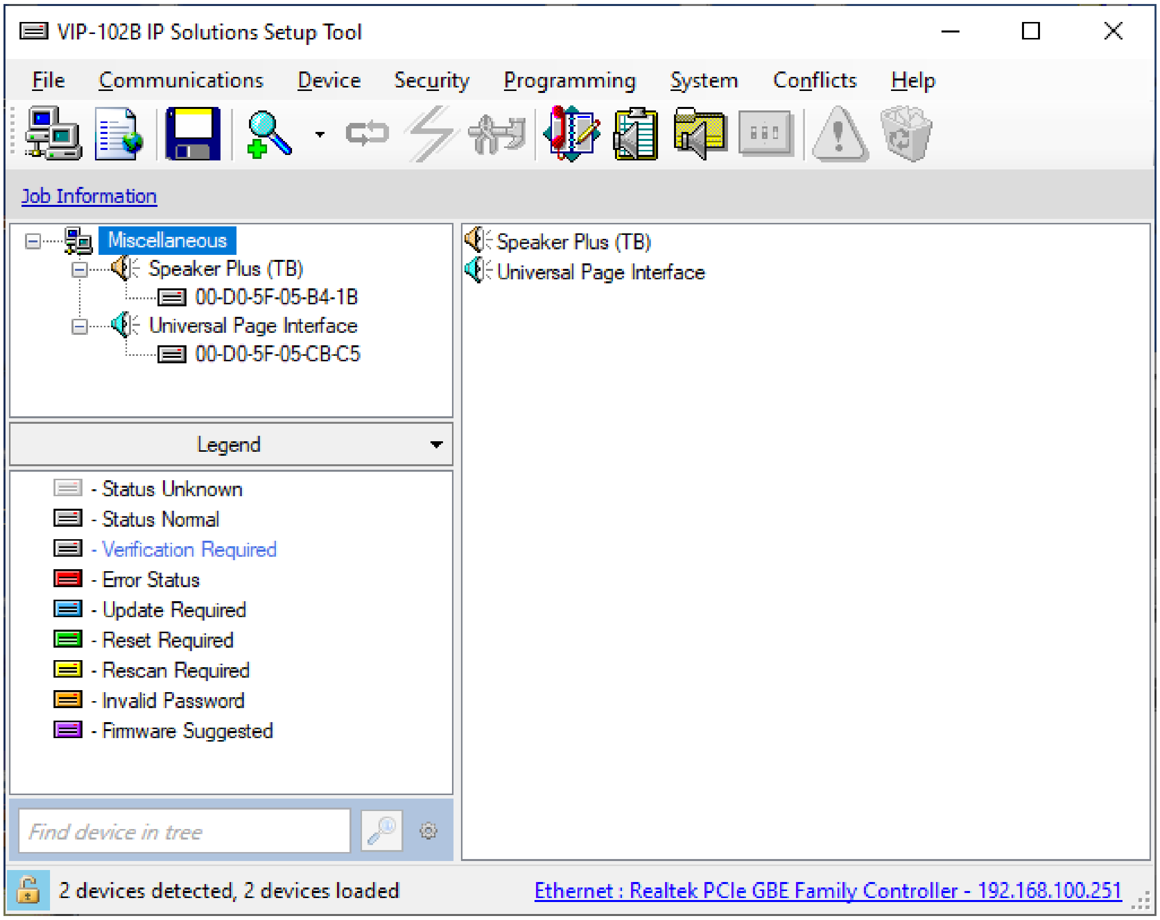
This section covers the configuration of Valcom V-9972 Universal Paging Interface using the Valcom VIP-102B IP Solutions Setup Tool. The configuration covers the following areas:

- Launch the Valcom VIP-102B IP Solutions Setup Tool
- Configure the Network Settings
- Configure Time
- Install System Manager CA TLS Certificate
- Configure SIP Parameters
- Verify Codec Settings
- Update Universal Paging Interface with the New Configuration

**Note:** These Application Notes do not cover the configuration of the Valcom VIP-430A IP Wall Speakers, Audio Groups, or the assignment of Dial Codes to Valcom speakers. Refer to **[5]** and **[6]** for details.

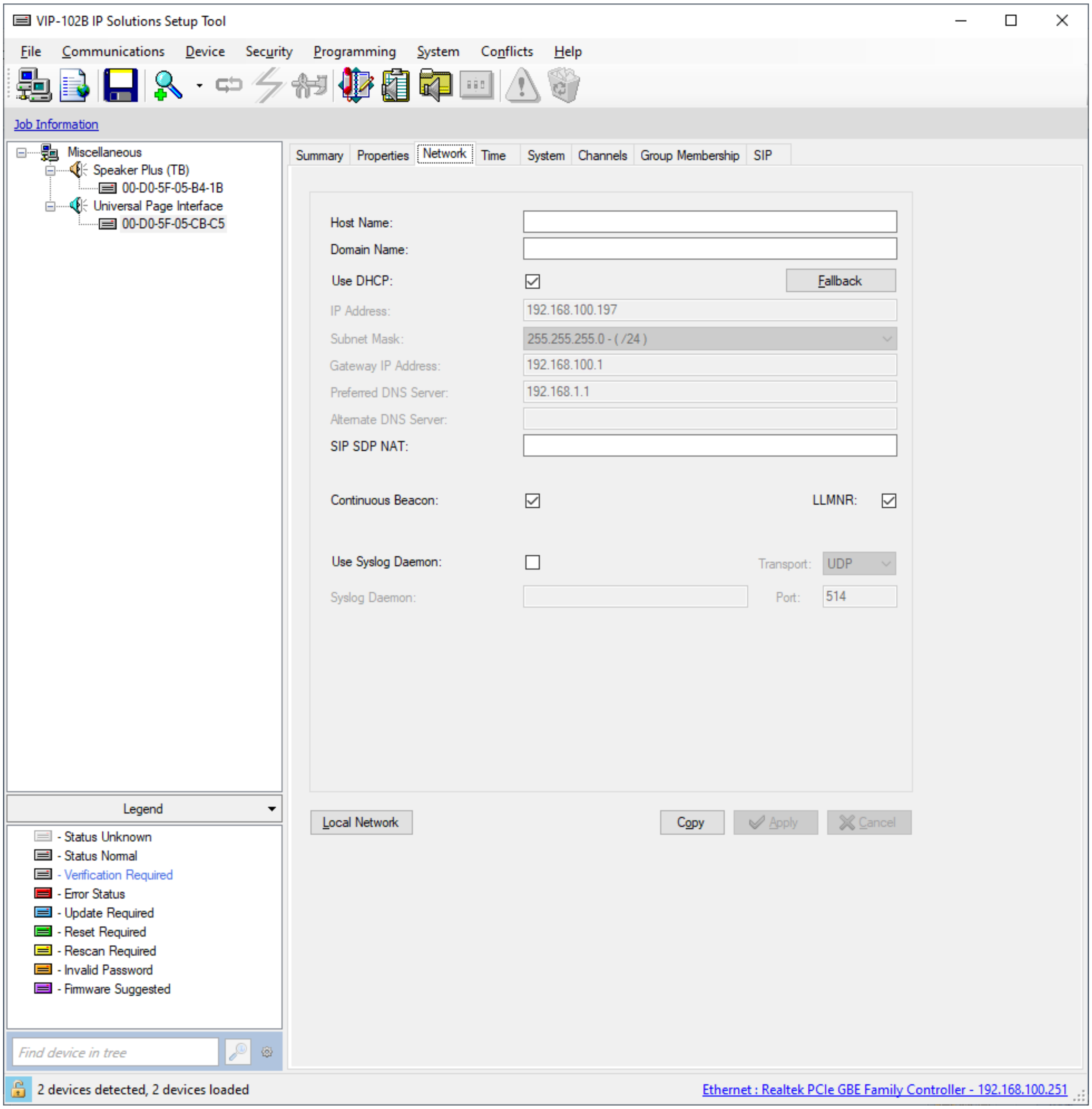
## 8.1. Launch Valcom VIP-102B IP Solutions Setup Tool

Launch the **VIP-102B IP Solutions Setup Tool** and follow the prompts. The main window is displayed as shown below.



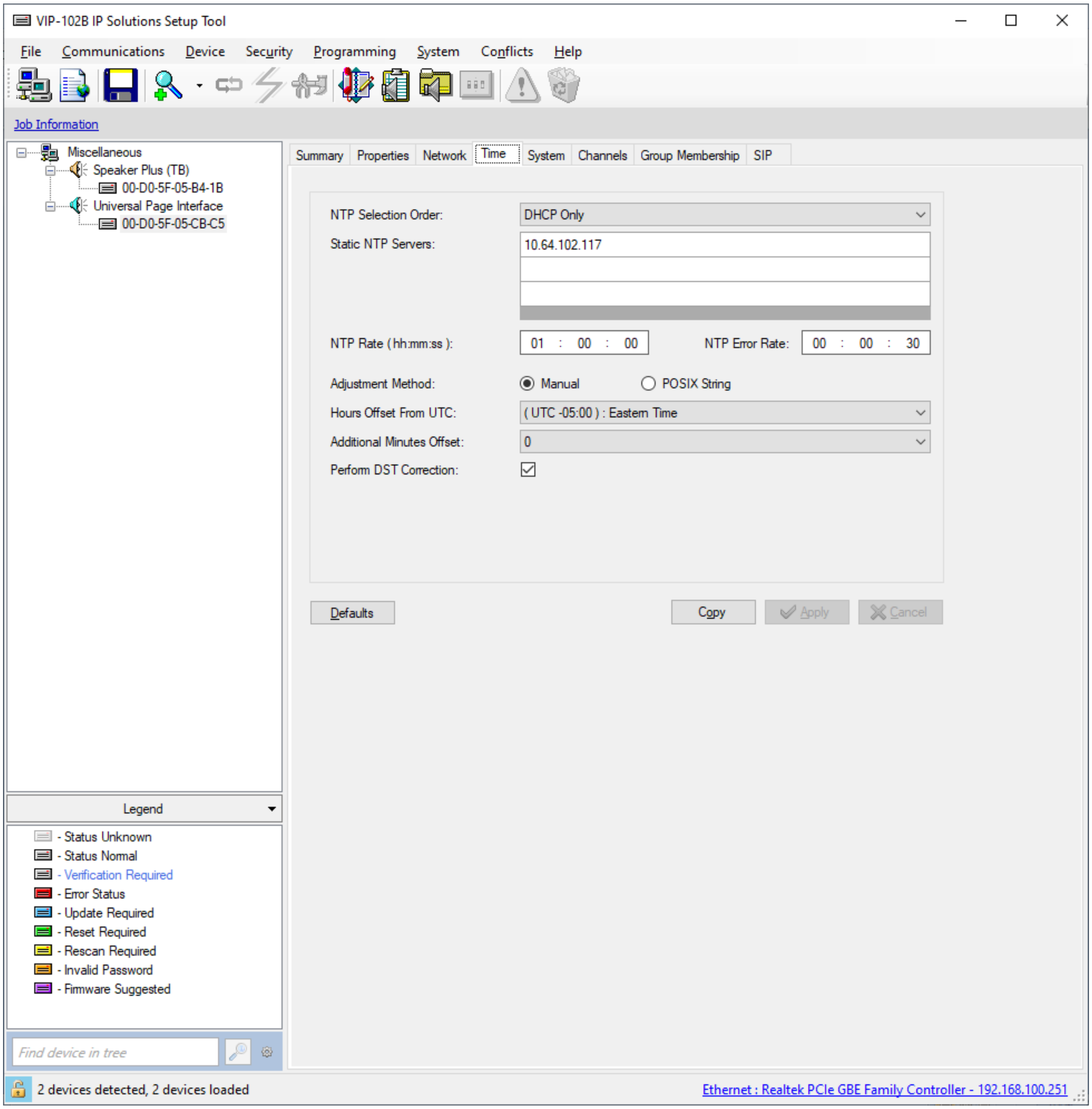
## 8.2. Configure the Network Settings

Click the MAC/hardware address under Universal Page Interface in the left pane and select the **Network** tab. V-9972 must first acquire IP network settings before proceeding with provisioning. These network settings were automatically obtained from a DHCP server as shown below. Alternatively, V-9972 could be configured with static IP addresses, but for the compliance test, DHCP was used.



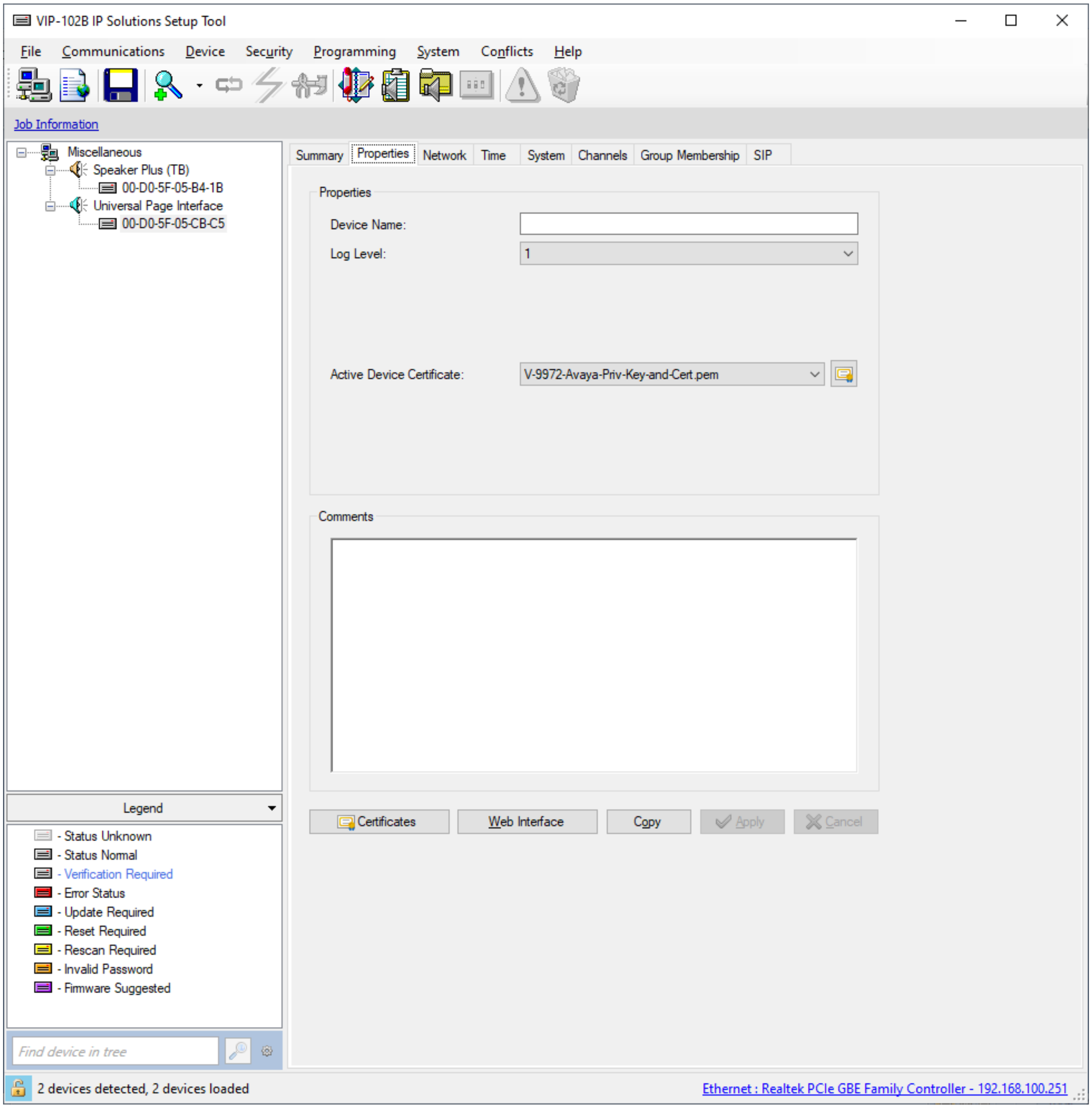
## 8.3. Configure the Time

Navigate to the **Time** tab and set the Static NTP Servers to ensure the proper date/time on the device.

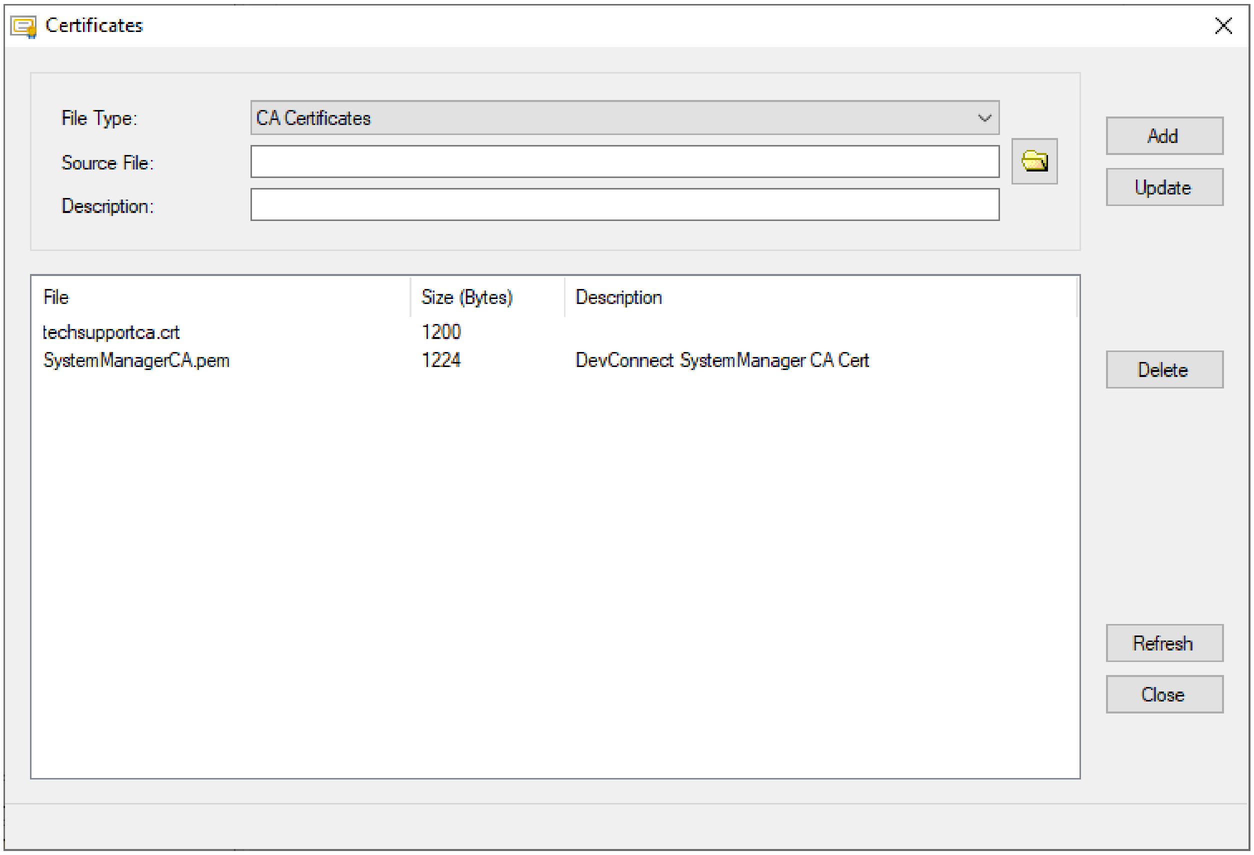


## 8.4. Install the System Manager CA TLS Certificate

Navigate to the **Properties** tab to install the System Manager CA certificate. Note that the V-9972 has a device certificate (*V-9972-Avaya-Priv-Key-and-Cert.pem*) signed by a different CA other than the System Manager. Click on **Certificates**.



In the **Certificate** dialog box, add the System Manager CA TLS certificate.  Note that the certificate has already been imported as shown below.  In addition, the V-9972 root certificate (*techsupportca.crt*) is also installed.  This certificate must be installed on Session Manager to support mutual TLS authentication.

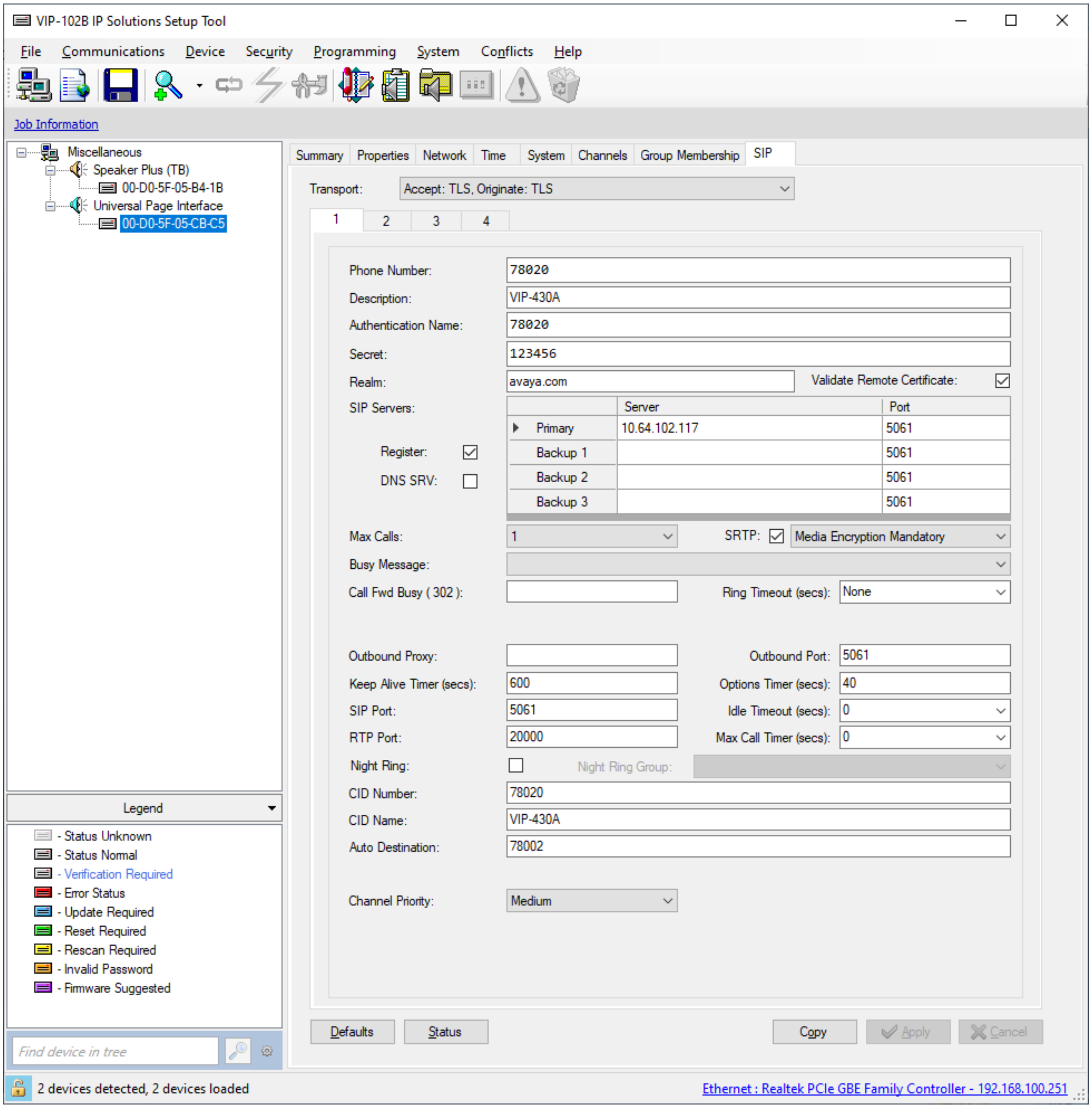


## 8.5. Configure SIP Parameters

From the **VIP-102B IP Solutions Setup Tool**, navigate to the **SIP** tab of the Universal Page Interface and configure the parameters as follows.

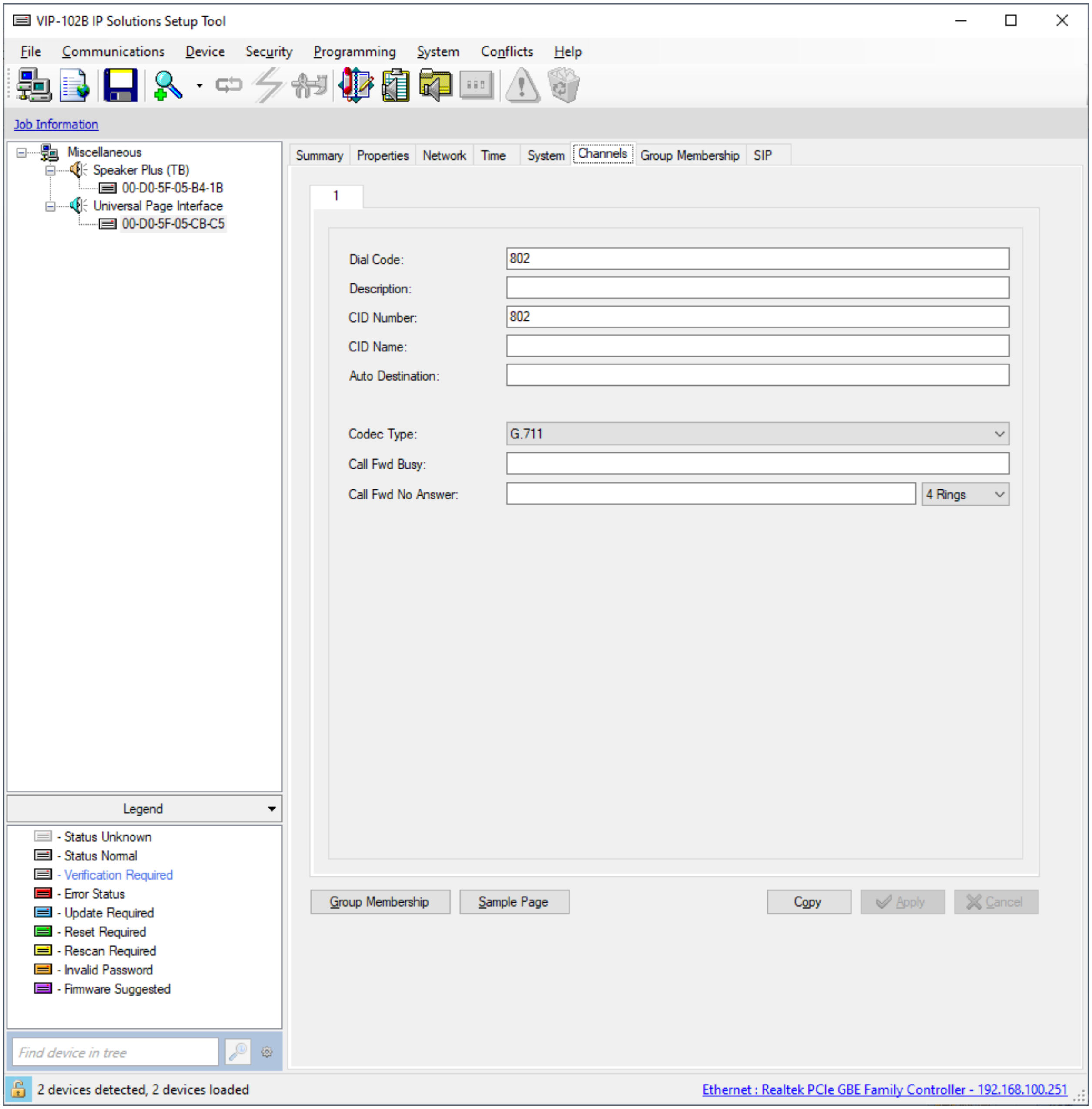
- **Transport:** Set to *Accept: TLS, Originate: TLS*.
- **Phone Number:** Set to SIP extension (e.g., *78020*).
- **Description:** Provide optional description.
- **Authentication Name:** Set to SIP extension configured in Session Manager in **Section 6.3**.
- **Secret:** Set to SIP password confgured in **Section 6.3.2**.
- **Realm:** Set to SIP domain (e.g., *avaya.com*).
- **Validate Remote Certificate:** Enable this option so that V-9972 validates the remote TLS certificate installed in **Section 8.4.**
- **Primary Server:** Set to Session Manager IP address (i.e., *10.64.102.117*), if V-9972 will register directly to Session Manager, or set to the IP address of the SBCE public interface, if V-9972 will register to Session Manager through SBCE as a remote worker.
- **Port:** Set to TLS port (e.g., *5061*).
- **Register:** Enable this option to allow V-9972 to register as a SIP endpoint.
- **Max Calls:** Specify maximum number of calls (e.g., *4*). For example, V-9972 could establish an intercom call to the IP speaker and then a higher priority paging call to the same IP speaker. In addition, V-9972 could establish up to four calls to four different IP speakers (not tested).
- **SRTP:** Enable SRTP and then select *Media Encryption Mandatory*.
- **Auto Destination:** Set to the number that should be dialed when the call button on the VIP-430A IP Wall Speaker is pressed.

Accept the values in the remaining fields and click **Apply**.



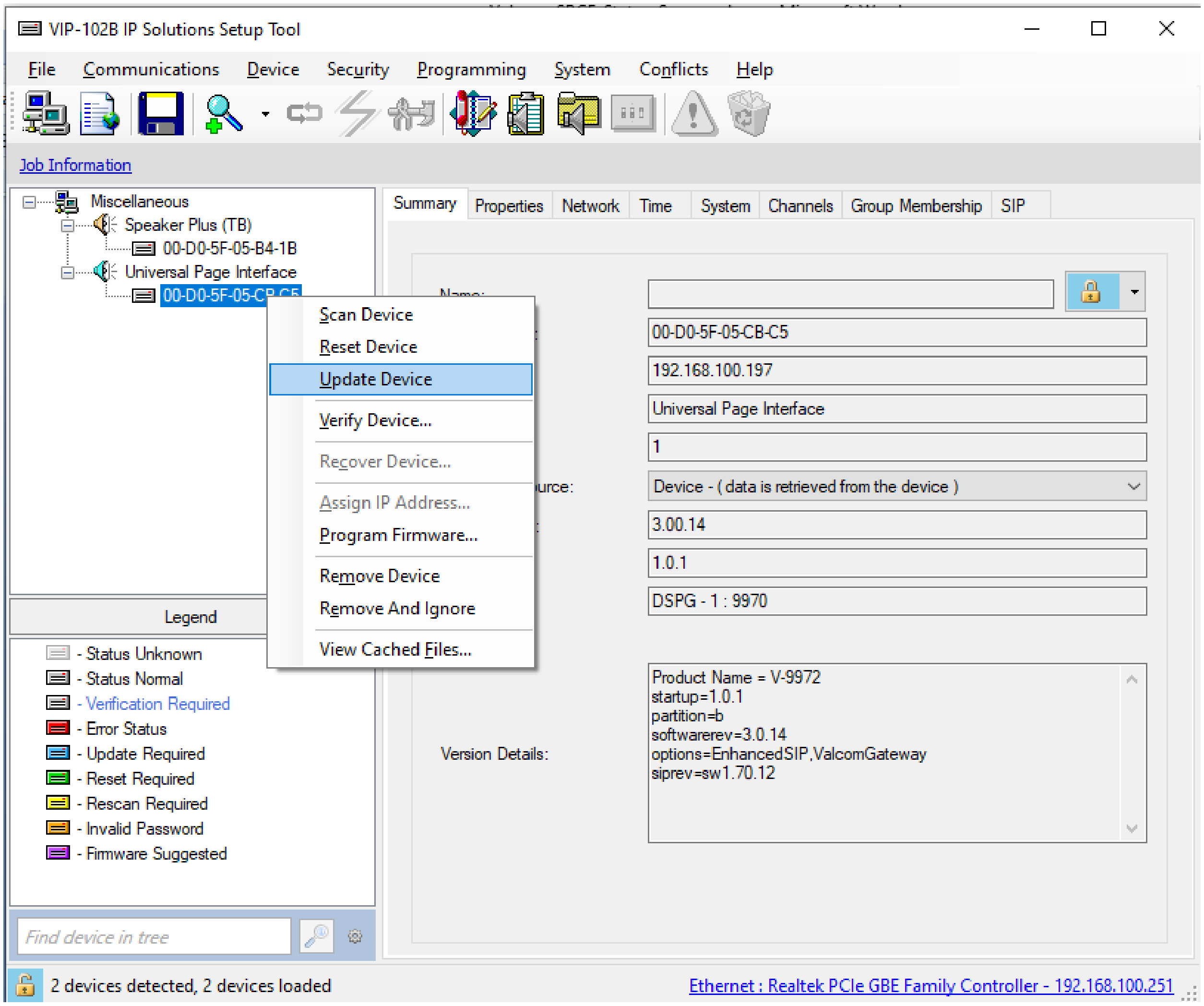
## 8.6. Verify Codec Settings

Navigate to the **Channels** tab shown below. The Codec Type should be set G.711, currently the only option supported with VIP-430A IP Wall Speaker.

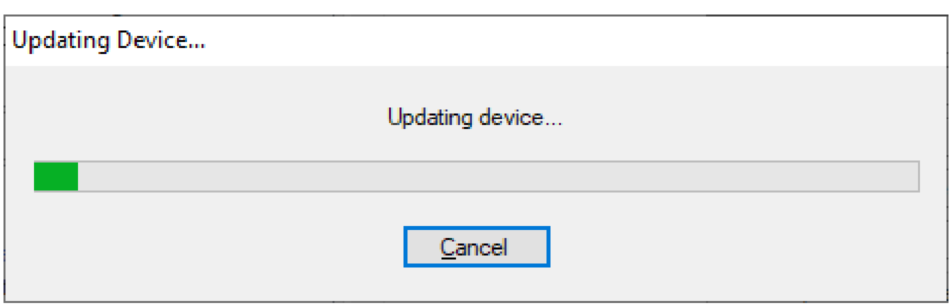


## 8.7. Update Universal Page Interface with the New Configuration

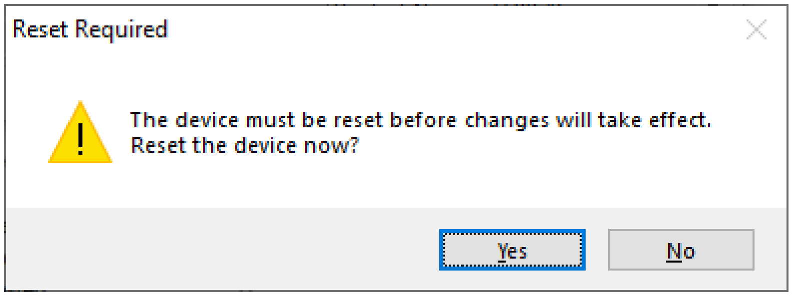
From the **VIP-102B IP Solutions Setup Tool**, right-mouse click on the MAC/hardware address of the Universal Page Interface and select **Update Device** from the pop-up menu as shown below.



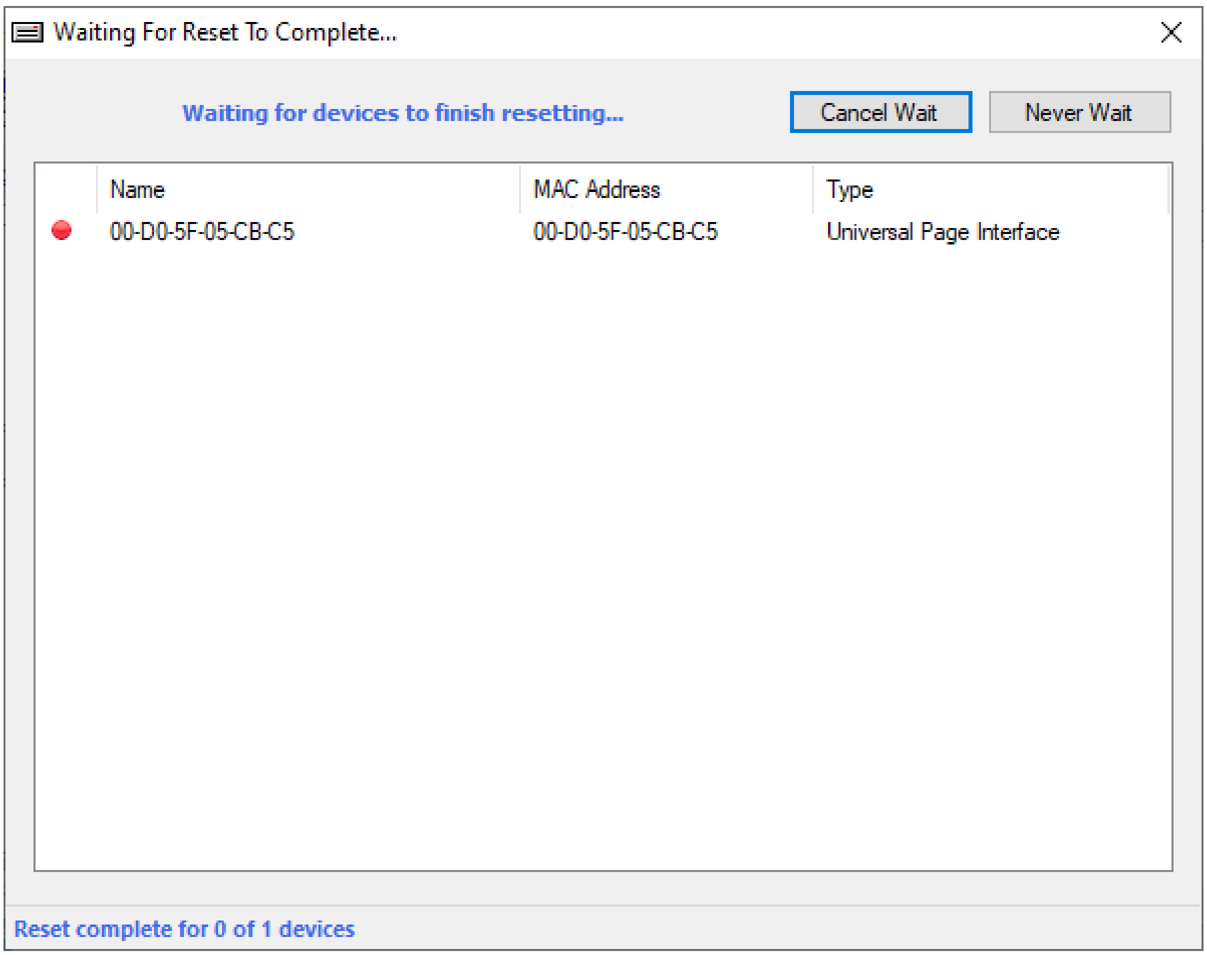

The following window is displayed indicating that the device is being updated.



A device reset is required so respond with **Yes** when prompted.



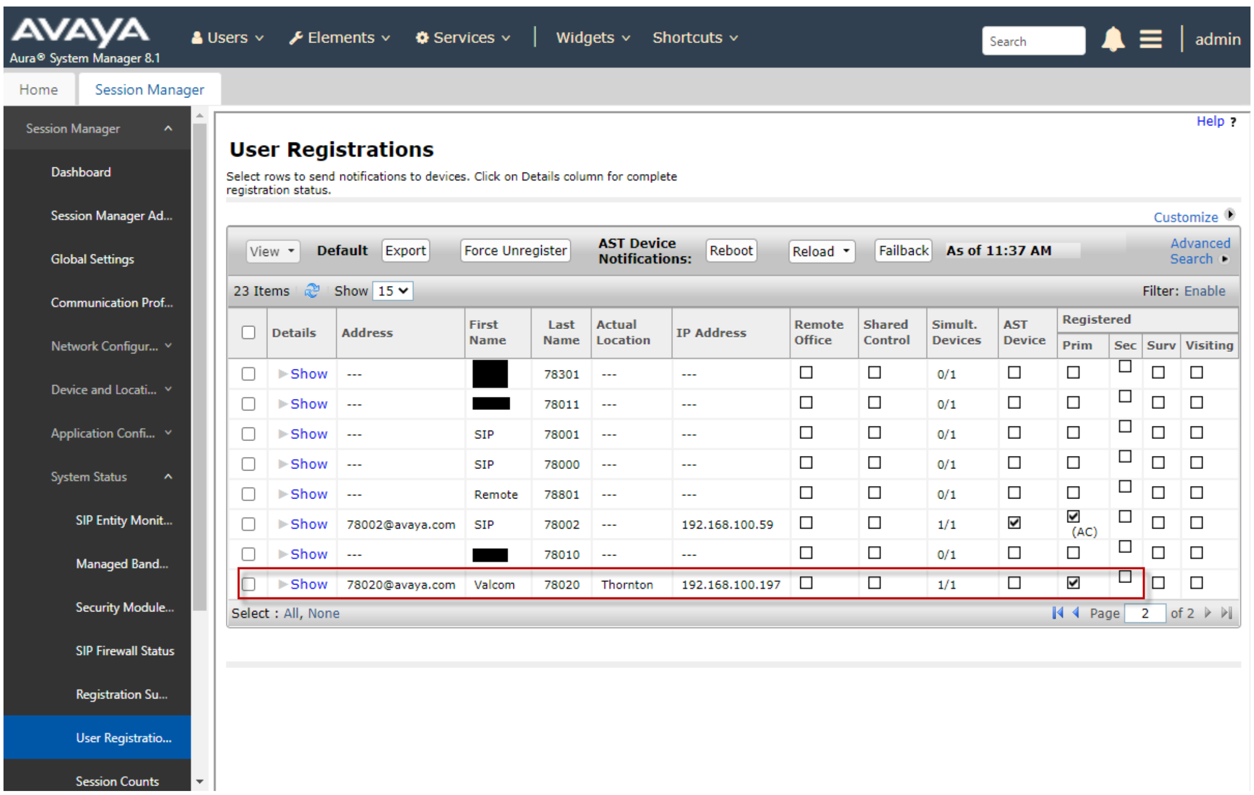

The following window will be displayed while the device is being reset.  When the reset is completed, the window will disappear.



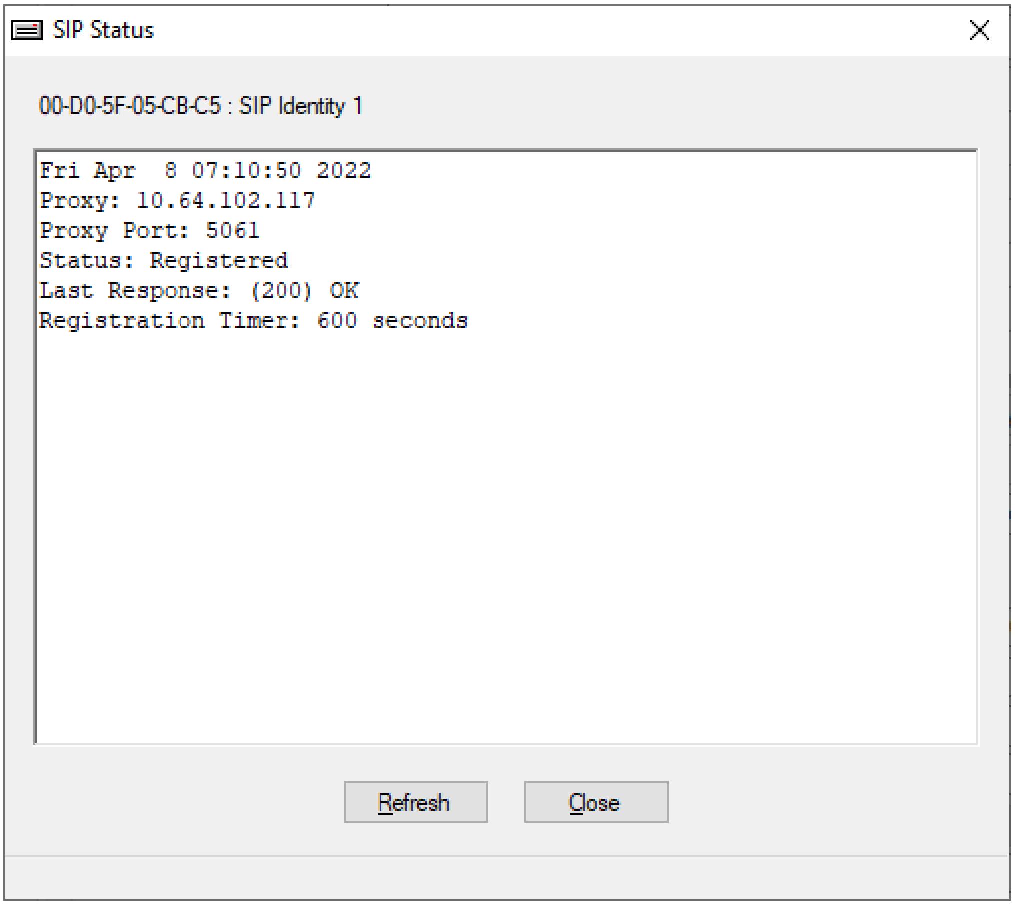
# 9. Verification Steps

This section provides the tests that may be performed to verify proper configuration of Valcom V-9972 Universal Paging Interface with Avaya Aura® Session Manager, Avaya Aura® Communication Manager, and Avaya Session Border Controller for Enterprise.

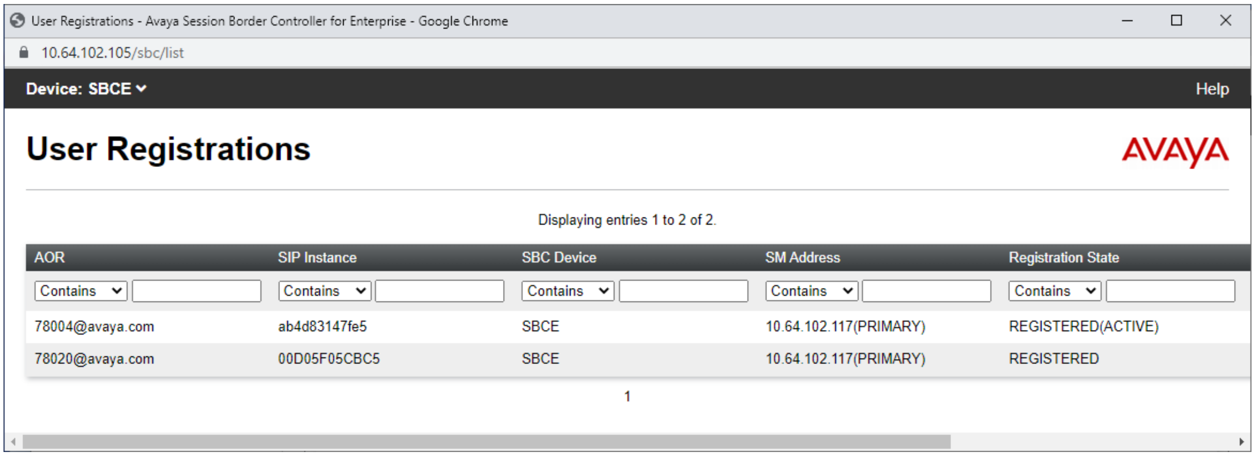
1. Verify that V-9972 has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status. Note that when V-9972 is registered as a remote worker, the **Remote Office** checkbox would be selected.



2. Alternatively, the registration state may be verified the Valcom VIP-102B IP Solutions Setup Tool. Navigate to the **SIP** tab of the Universal Page Interface and click **Status** button. The **Status** should be *Registered*. Note that the **Proxy** would be the Session Manager IP address, if V-9972 is registered directly to Session Manager. The **Proxy** would be the IP address of the SBCE public interface if V-9972 is registered through SBCE as a remote worker.




3. If the V-9972 is registered as a remote worker, the SBCE would also provide a registration status by navigating to **Status → User Registrations**.




4. Place a call to the V-9972 and at the dial tone, enter the dial code for the IP speaker to establish an intercom call from an Avaya IP deskphone to a Valcom speaker. Verify two-way audio. Terminate the call from the Avaya IP deskphone or by pressing the call button on the IP speaker.

5. Place a call to the V-9972 and at the dial tone, enter the dial code a group page code to establish a one-way paging call from an Avaya IP deskphone to IP speaker(s). Verify one-way audio. Terminate the call from the Avaya IP deskphone.

6. Place an intercom call by pressing the call button on the IP speaker. Verify two-way audio to the call destination. Terminate the call.

# 10. Conclusion

These Application Notes described the configuration steps required to integrate Valcom V-9972 Universal Paging Interface with Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and Avaya Session Border Controller for Enterprise. Intercom and paging calls were established with Valcom V-9972 Universal Paging Interface, Valcom VIP-430A IP Wall Speaker, Avaya H.323 / SIP Deskphones, and the PSTN. All feature and serviceability test cases were completed successfully.

# 11. References

This section references the Avaya and Valcom documentation relevant to these Application Notes.


[1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 12, July 2021, available at http://support.avaya.com.

[2] *Administering Avaya Aura® System Manager for Release 8.1.x*, Release 8.1.x, Issue 19, April 2022, available at http://support.avaya.com.

[3] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 11, March 2022, available at http://support.avaya.com.

[4] *Administering Avaya Session Border Controller for Enterprise*, Release 8.1.x, Issue 5, August 2021, available at http://support.avaya.com.

[5] *Valcom VIP-102B IP Solutions Setup Tool Version 8.4.0.0 Reference Manual*, Revision 17 – 3/16/22, available at https://www.valcom.com/resources/documents-manuals.

[6] *Valcom V-9972 Universal Page Interface Configuration Guide*, Rev. 3.1, available at https://www.valcom.com/resources/documents-manuals.

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.

# Declaration of Conformance

**May 20, 2022**

Jeff Gartner
Senior Manager
DevConnect Program
Avaya

**Dear Jeff Gartner:**

We, Valcom Inc, declare under sole responsibility that product series named Universal Paging Adapter, including product models V-9972, V-9972-2 or VRCPA share the same hardware circuitry, software, SIP stack and firmware version. Therefore, the products are expected to behave in the same manner. The differences between the different models in each series are generally cosmetic in nature, such as enclosure shape or color, mounting arrangement, etc.

Sincerely,

**/s/ David Ellison**

**David Ellison**
**Technical Support Manager**
**Valcom Inc**
**dellison@valcom.com**