Avaya Solution & Interoperability Test Lab

Application Notes for Talkaphone VOIP-500 Series IP Call Stations with Avaya Aura® Session Manager 8.1 and Avaya Aura® Communication Manager 8.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Talkaphone VOIP-500 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Talkaphone VOIP-500 Series IP Call Stations are a family of indoor- and outdoor-rated (ruggedized) VoIP emergency/information phones for use in locations such as parking facilities, college campuses, medical centers and industrial parks. Talkaphone VOIP-500 Series IP Call Stations support SIP (RFC 3261) and can operate as a paging/mass notification device via a standard SIP-based inbound call. Talkaphone VOIP-500 Series IP Call Stations register with Avaya Aura® Session Manager as a SIP endpoint. For the compliance test, a Talkaphone VOIP-500E IP Call Station was used, which provides an emergency call button only. There is no handset or keypad.

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 Talkaphone VOIP-500 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Talkaphone VOIP-500 Series IP Call Stations are a family of indoor- and outdoor-rated (ruggedized) VoIP emergency/information phones for use in locations such as parking facilities, college campuses, medical centers and industrial parks. Talkaphone VOIP-500 Series IP Call Stations support SIP (RFC 3261) and can operate as a paging/mass notification device via a standard SIP-based inbound call. Talkaphone VOIP-500 Series IP Call Stations register with Avaya Aura® Session Manager as a SIP endpoint. For the compliance test, a Talkaphone VOIP-500E IP Call Station was used, which provides an emergency call button only. There is no handset or keypad.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Talkaphone VOIP-500E IP Call Station, Avaya SIP / H.323 Deskphones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer, and conference, from the perspective of the Avaya IP Deskphones. Additional telephony features, such as call forward and call coverage, were also verified.

The serviceability testing focused on verifying that the Talkaphone VOIP-500E IP Call Station came back into service after re-connecting the Ethernet cable or rebooting the IP Call Station.

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 these DevConnect Application Notes 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 these Application Notes, the interface between Avaya systems and Talkaphone VOIP-500 Series IP Call Stations did not include use of any specific encryption features as requested by Talkaphone.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of VOIP-500E with Session Manager.
- Inbound and outbound calls between VOIP-500E and Avaya SIP / H.323 Deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Inbound and outbound calls between the VOIP-500E and the PSTN.
- G.711 and G.729 codec support.
- Basic telephony features, including hold, mute, redial, transfer, and 3-way conference, initiated from the perspective of Avaya IP Deskphones.
- Use of paging, recorded messages, emergency calls, and number lists on the VOIP-500E.
- Proper system recovery after a restart of the VOIP-500E and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observation(s):

- Emergency calls cannot be terminated from the Talkaphone VOIP-500E IP Call Station. The calls can only be disconnected by the called party or upon expiration of the Call Conversation Timer. The destination phone of an emergency call shouldn't cover to voicemail. The Talkaphone VOIP-500E Series IP Call Station dials a list of programmed numbers in a round-robin fashion. If the first number in the list does not answer (i.e., Busy, Out of Order, Invalid number), it will call the next number in line and will keep doing so until the destination answers the call or until the 'Call Conversation Timer' expires.

- Avaya H.323 / SIP Deskphones hear welcome tone when calls are made to Talkaphone VOIP-500E IP Call Station and Direct IP Media (shuffling) is disabled. When shuffling is enabled, the welcome tone is not heard.

2.3. Support

For technical support and information on Talkaphone VOIP-500 Series IP Call Stations, contact Talkaphone support at:

- Phone: + 1-773-539-1100
- Email: customerservice@talkaphone.com
- Website: http://www.talkaphone.com/contact-support

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 virtualized environment with a 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 telephones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya 96x1 Series SIP and H.323 Deskphones.
- Avaya J100 Series SIP Deskphones.
- Talkaphone VOIP-500E IP Call Station.

Talkaphone VOIP-500E IP Call Station registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

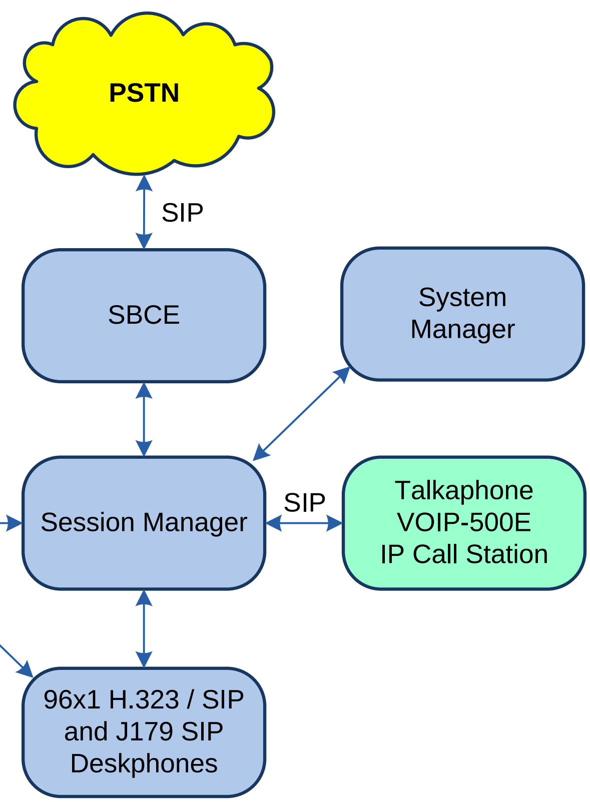


Figure 1: Avaya SIP Network with Talkaphone VOIP-500E IP Call Station

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	8.1.2.0.0-FP2
Avaya G450 Media Gateway	FW 41.24.0
Avaya Aura® Media-Server	v.8.0.2.93
Avaya Aura® System Manager	8.1.2.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.2.0.0611167 Feature Pack 2
Avaya Aura® Session Manager	8.1.2.0.812039
Avaya 96x1 Series IP Deskphones	6.8304 (H.323) 7.1.9.0.8 (SIP)
Avaya J100 Series IP Deskphones	4.0.5.0.10 (SIP)
Talkaphone VOIP-500E IP Call Station	1.0.3.1

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 the appropriate credentials.

Note: The SIP station configuration for the Talkaphone IP Call Stations are configured through Avaya Aura® System Manager in **Section 6.3**.

5.1. Verify 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    89
                                      Maximum Stations: 36000    28
                              Maximum XMOBILE Stations: 36000     0
                    Maximum Off-PBX Telephones - EC500: 41000     0
                    Maximum Off-PBX Telephones -   OPS: 41000    16
                    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*). The 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 and IP Codec Set

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 traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

```
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
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the Teo IP phones. 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. The Talkaphone VOIP-500E IP Call Station was tested using G.711 and G.729 codecs.

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

                          IP Codec Set
    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:
```

5.4. 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*.

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? n                                   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? n
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 Talkaphone IP Call Stations, Avaya SIP Deskphones, and Avaya Aura® Messaging. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, 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  5
                                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: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 10
                                                     Number of Members: 10
```

5.5. Administer 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. Talkaphone VOIP-500E IP Call Station was assigned extension 78005.

```
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

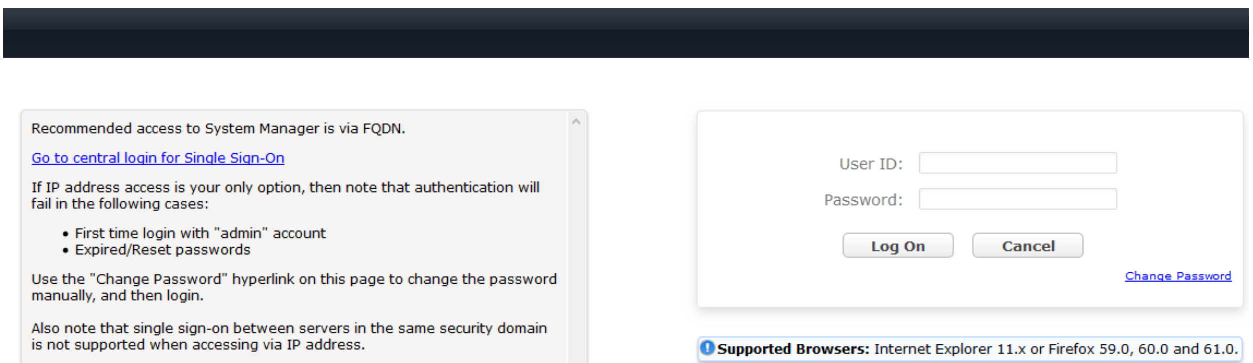
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol
- Administer SIP User

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 the Talkaphone IP Call Station.

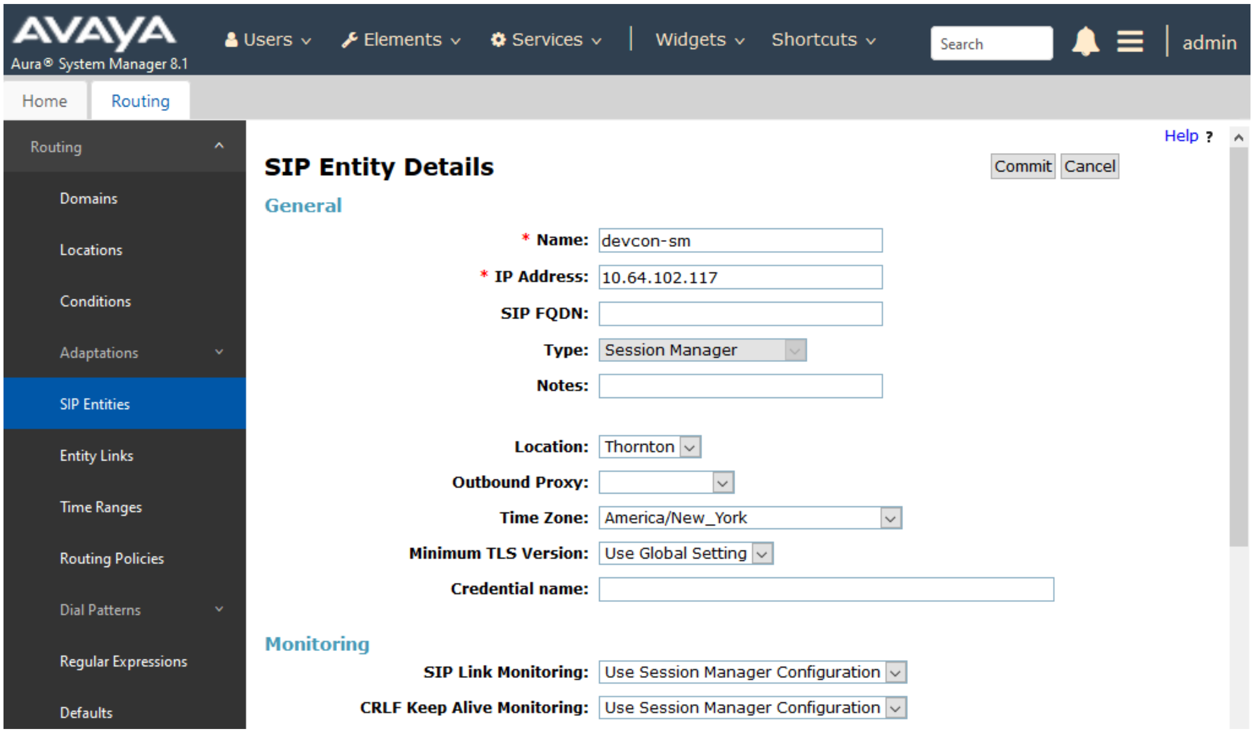
6.1. Launch System Manager

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.

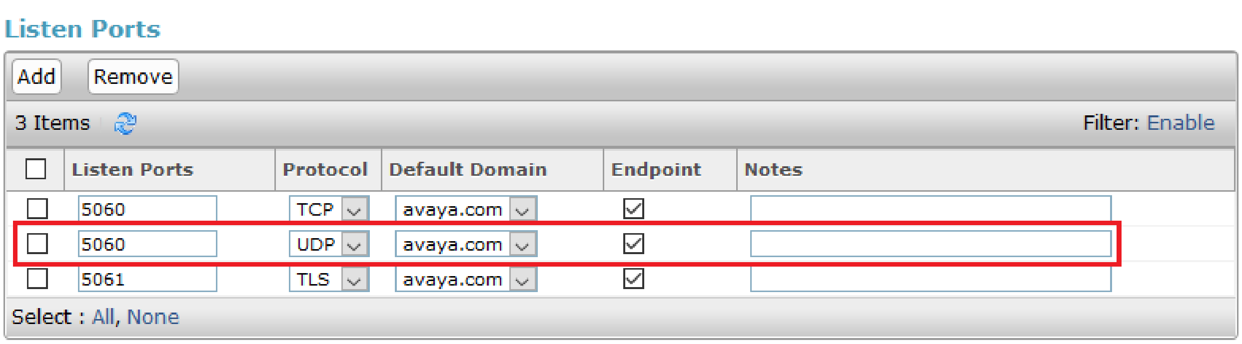


6.2. Set Network Transport Protocol

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 Talkaphone IP Call Station is specified in the list below. For the compliance test, the solution used UDP network transport.


Listen Ports

Add | Remove

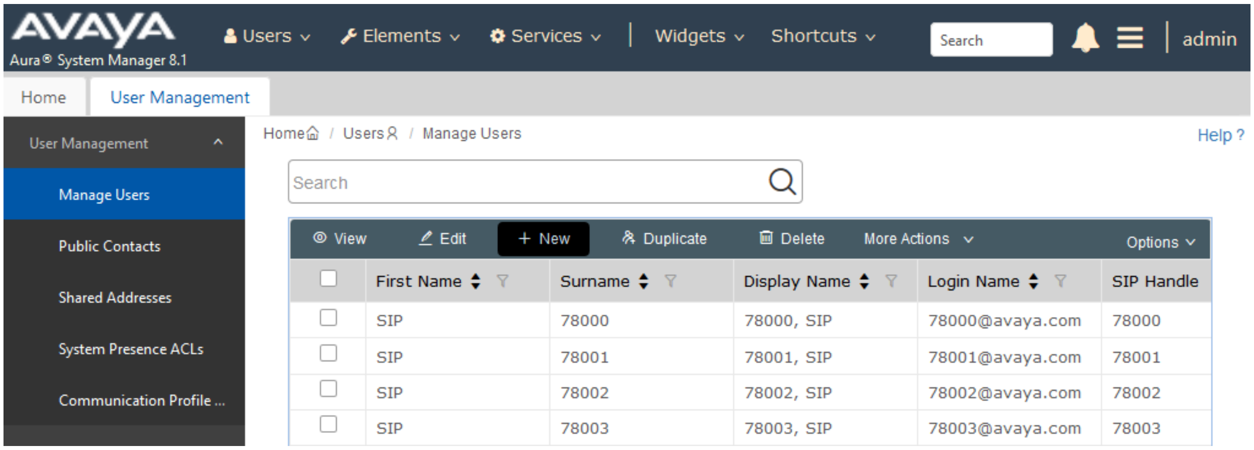
3 Items | Filter: Enable

	Listen Ports	Protocol	Default Domain	Endpoint	Notes
☐	5060	TCP	avaya.com	☑	
☐	5060	UDP	avaya.com	☑	
☐	5061	TLS	avaya.com	☑	

Select : All, None

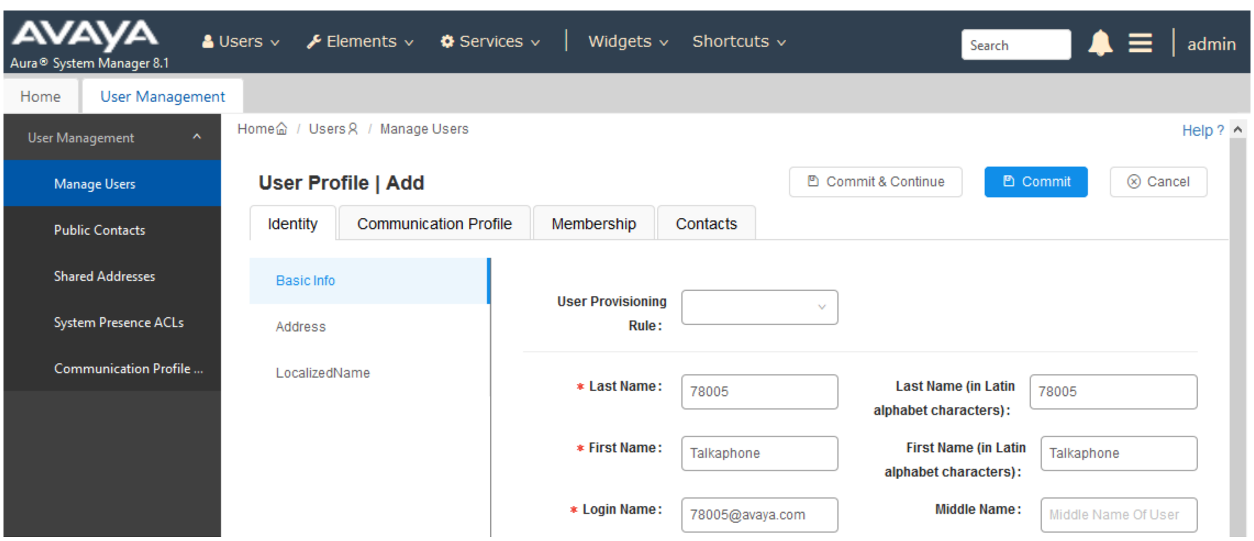
6.3. Administer SIP User

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.



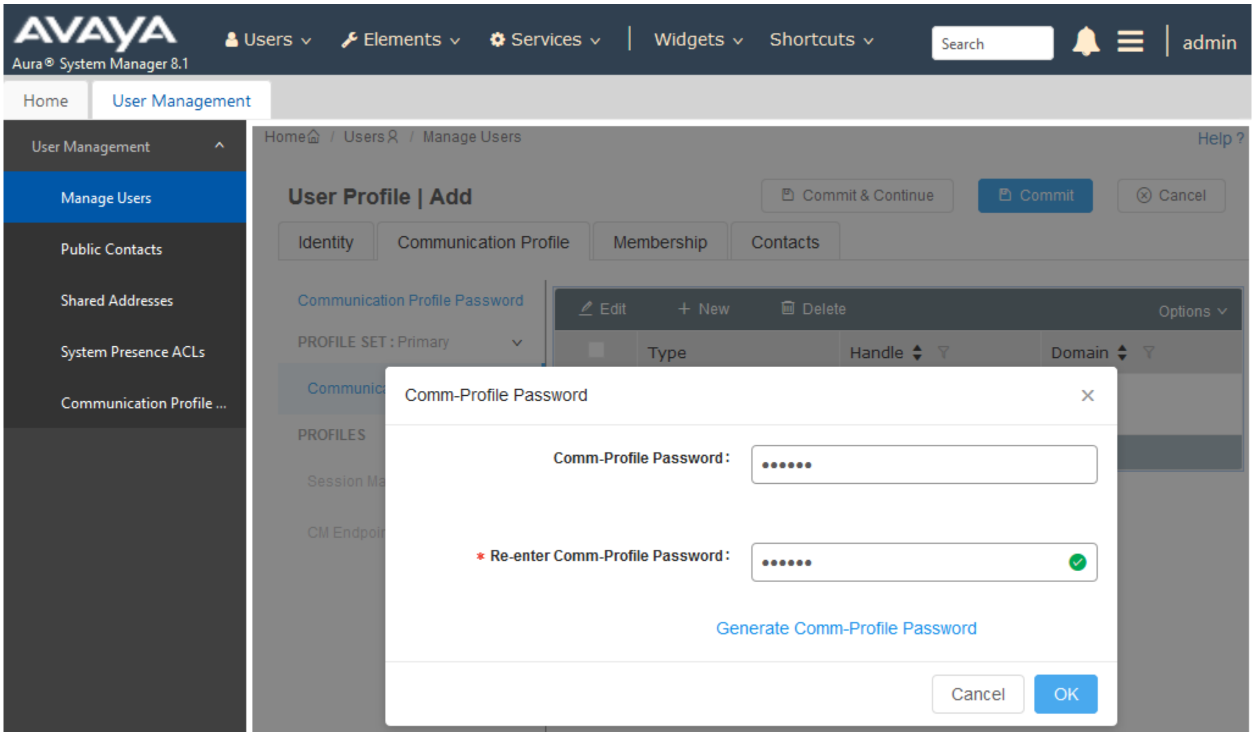
6.3.1. Identity

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 Talkaphone IP Call Station SIP extension and "*<domain>*" is the applicable SIP domain name from **Section 5.3**. Retain the default values in the remaining fields.



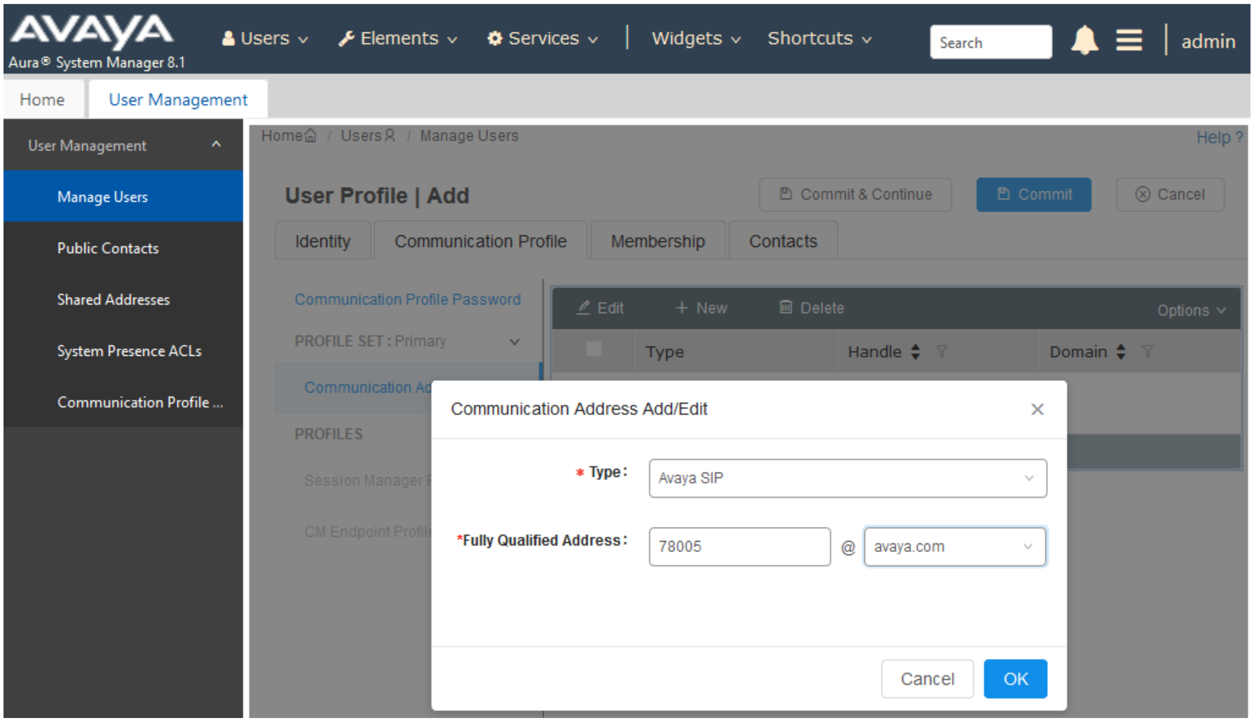
6.3.2. Communication Profile

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**.



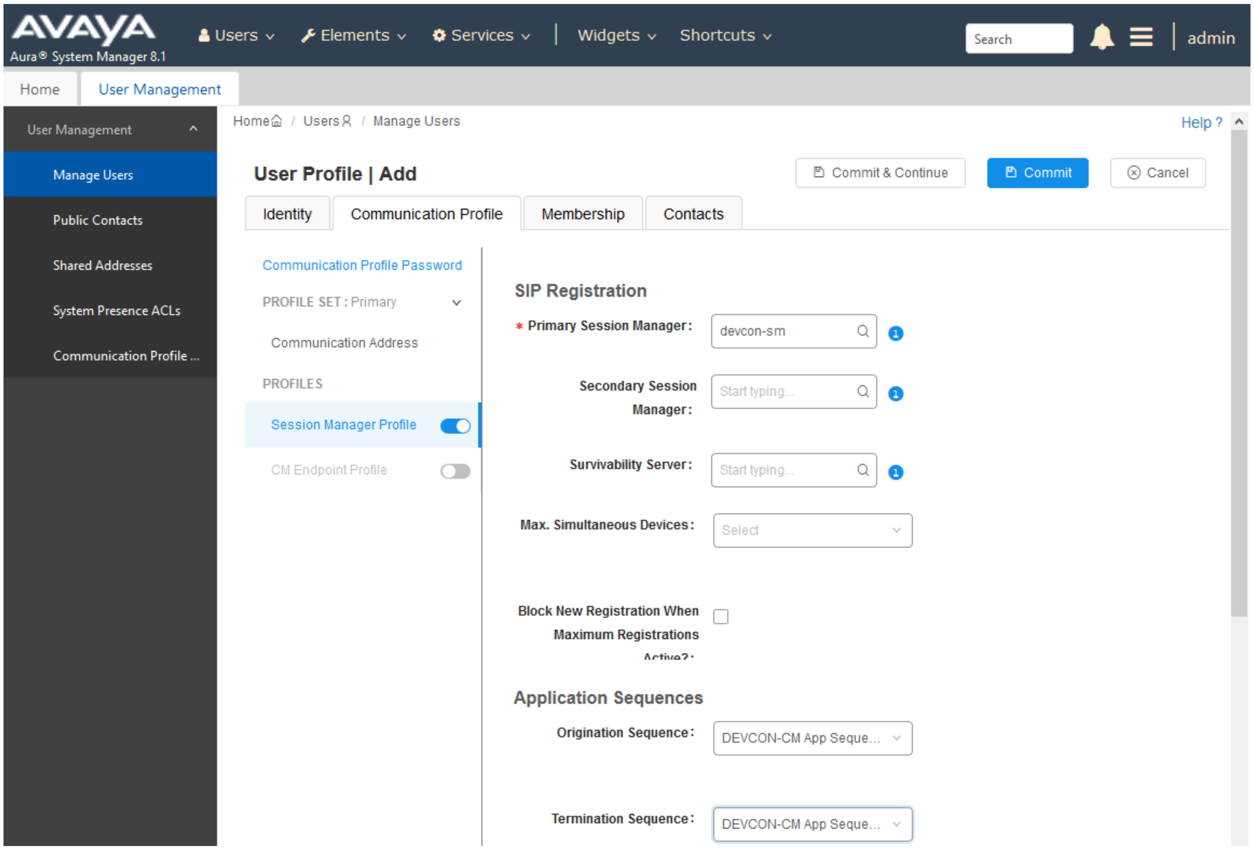
6.3.3. Communication Address

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**.

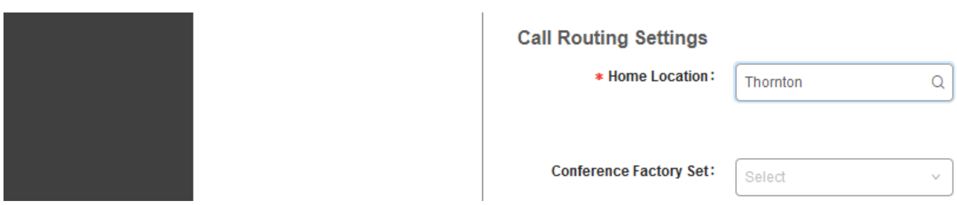


6.3.4. Session Manager Profile

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.

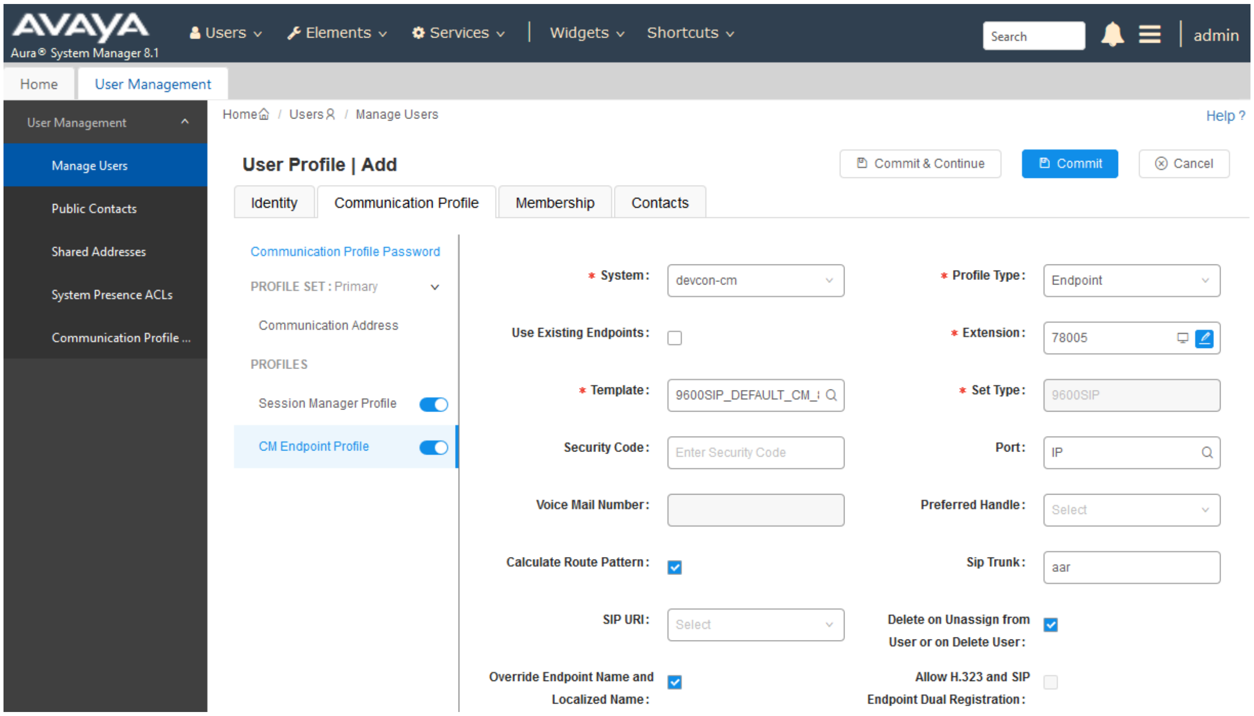


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.



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 *9600SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields.



7. Configure Talkaphone VOIP-500E IP Call Station

This section covers the configuration of the Talkaphone VOIP-500E IP Call Station. The following procedures are covered:

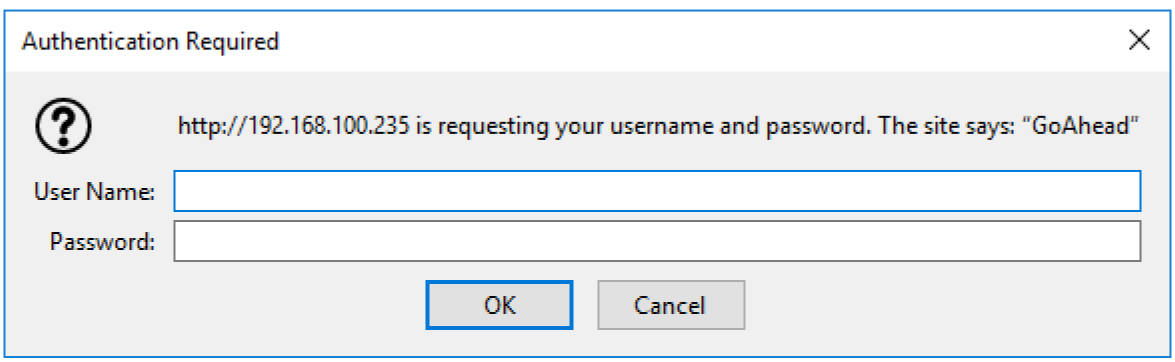
1. Launching the Web Administration Interface
2. Network Configuration
3. SIP Configuration
4. Configure Audio Settings
5. Configure Call Parameters
6. Configure Buttons
7. Configure Number Lists

7.1. Launching the Web Administration Interface

Talkaphone IP Call Stations are pre-configured with the following default values:

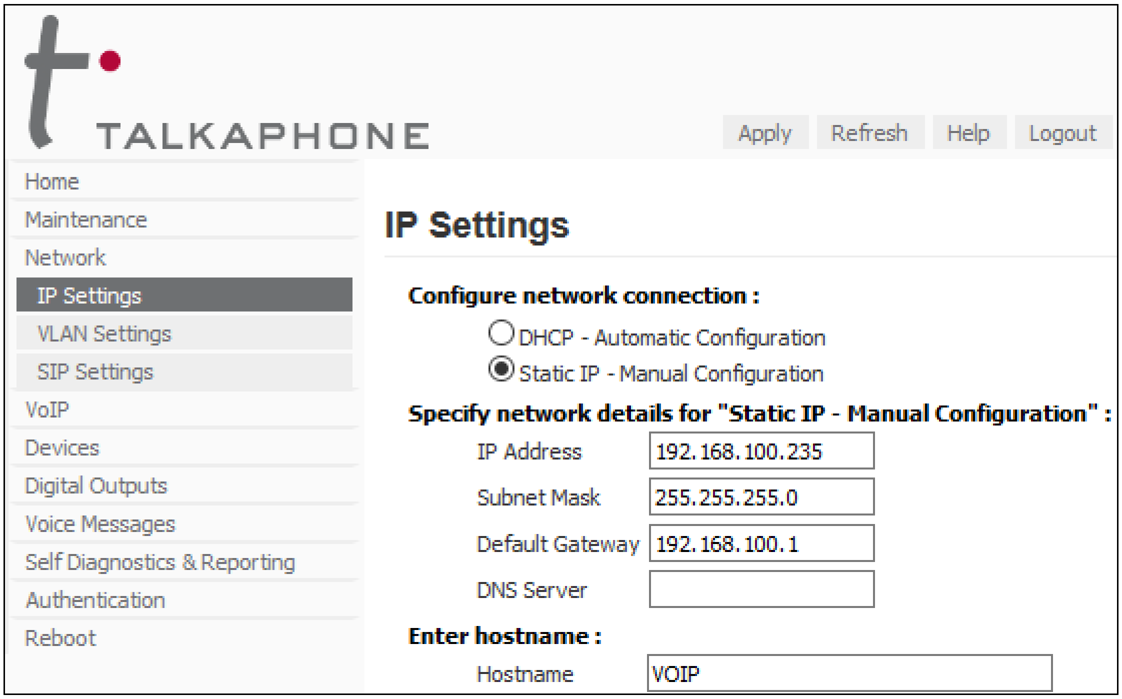
- **IP Address:** 192.168.1.10
- **Username:** admin
- **Password:** admin@123

Ensure that the administration PC and Talkaphone IP Call Station are connected to the LAN. Open a web browser and enter the default IP address of the Talkaphone IP Call Station in the URL field. The browser prompts for authentication. Log in with the appropriate credentials.



7.2. Network Configuration

To modify the IP network configuration of the Talkaphone IP Call Station, navigate to the **Network → IP Settings** page. Configure the IP settings so that it conforms to the customer network requirements. Click **Apply** when done.



7.3. SIP Configuration

Navigate to **Network → SIP Settings** to configure the SIP setting of the Talkaphone IP Call Station. Configure the following parameters.

Under **Assign a phone number**:

- **Phone Number:** Specify the SIP number (e.g., *78005*) configured in **Section 6.3**.

Under **Specify SIP Server FQDN/IP Address**:

- **Primary SIP Server FQDN/IP Address:** Specify the IP address of the Session Manager signaling interface (e.g., *10.64.102.117*) or the SIP domain (e.g., *avaya.com*). For the compliance test, the Session Manager IP address was used.

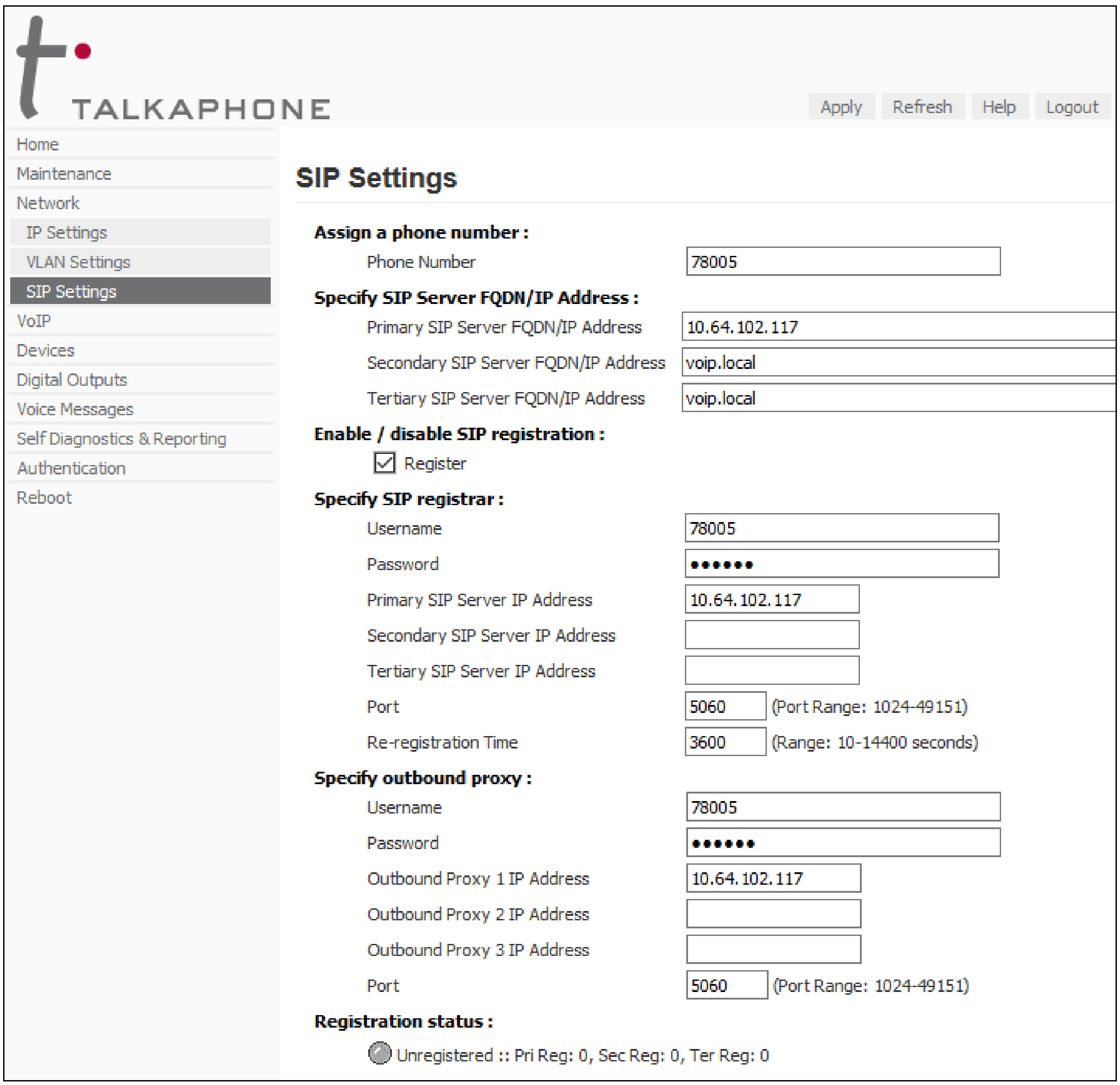
Under **Enable / disable SIP registration**:

- **Register:** Select the checkbox.

Under **Specify SIP registrar** and **Specify outbound proxy**:

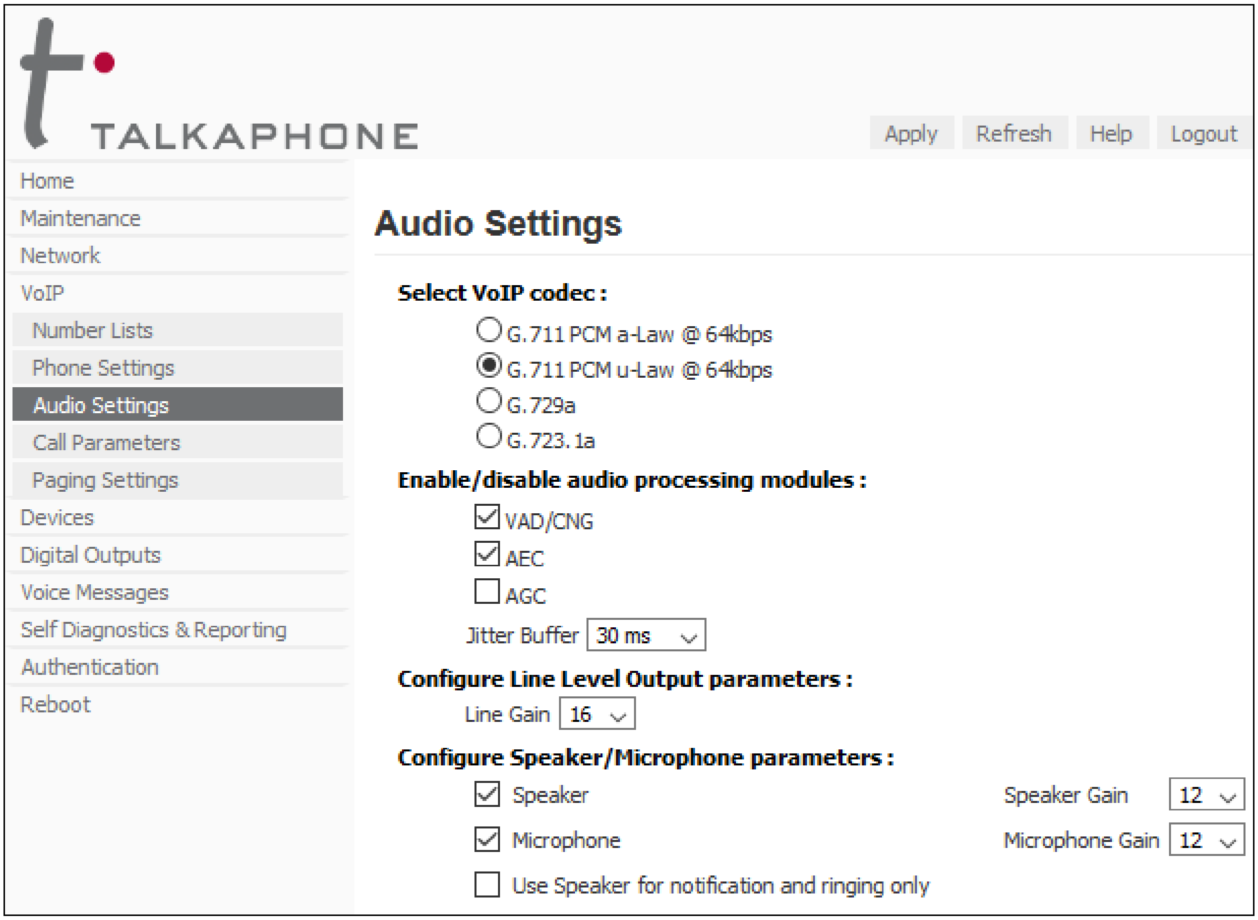
- **Username:** Specify the SIP number of the Talkaphone IP Call Station (e.g., *78005*).
- **Password:** Specify the SIP password configured in **Section 6.3.2**.
- **Primary SIP Server IP Address:** Specify the IP address of the Session Manager signaling interface (e.g., *10.64.102.117*).
- **Port:** Specify the SIP port (e.g., *5060*).

Accept the default values for the remaining fields and click **Apply** when done.



7.4. Configure Audio Settings

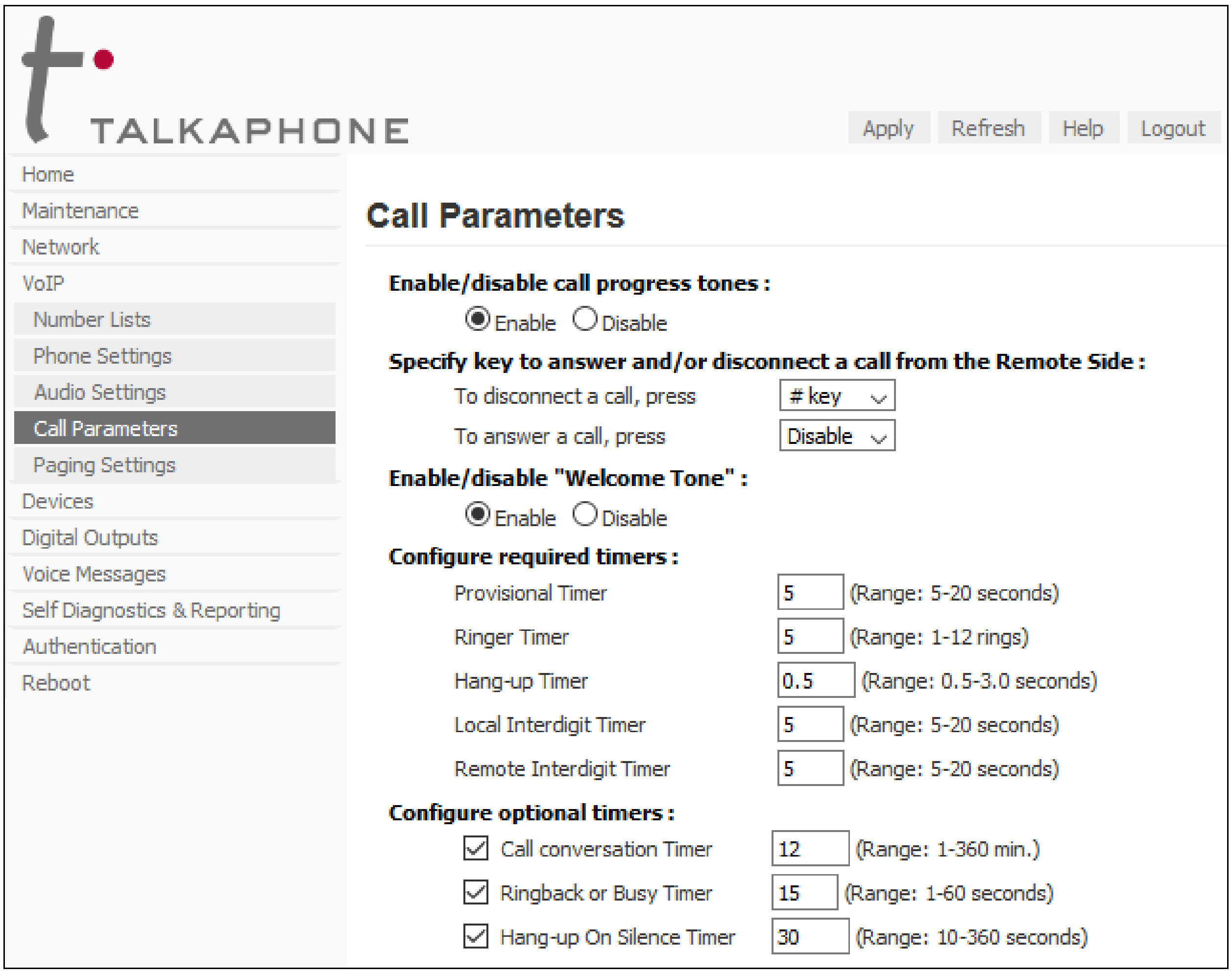
Navigate to **VoIP → Audio Settings** to configure the preferred codec and microphone and speaker parameters. In addition, the Speaker Gain can be adjusted to control the volume. All other fields were left at the default values. Click **Apply** when done.



7.5. Configure Call Parameters

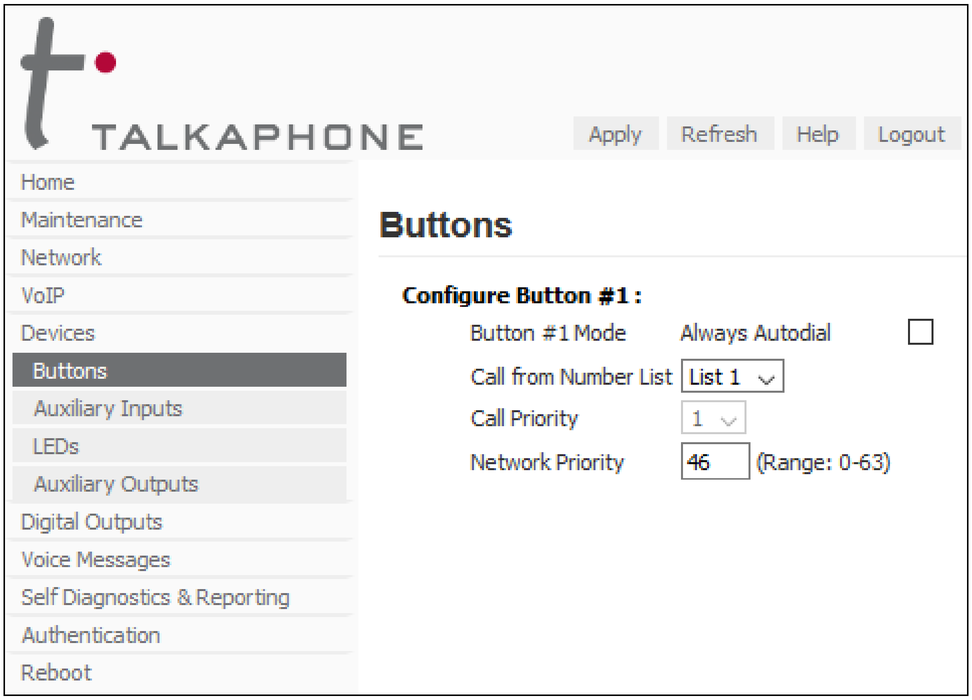
Navigate to **VoIP → Call Parameters** to view and customize any of the call parameters, such as **Local Interdigit Timer**, which dictates how long to wait before initiating a call after the user dials the digits, or the **Call Conversation Timer**, which specifies how long an emergency call should remain active, unless the far-end drops the call. The following screen shows the default values for the call parameters.

Note: After a number is dialed on the Talkaphone IP Call Station, the **Local Interdigit Timer** must expire before the call is initiated. The minimum value for the **Local Interdigit Timer** is 5 secs.



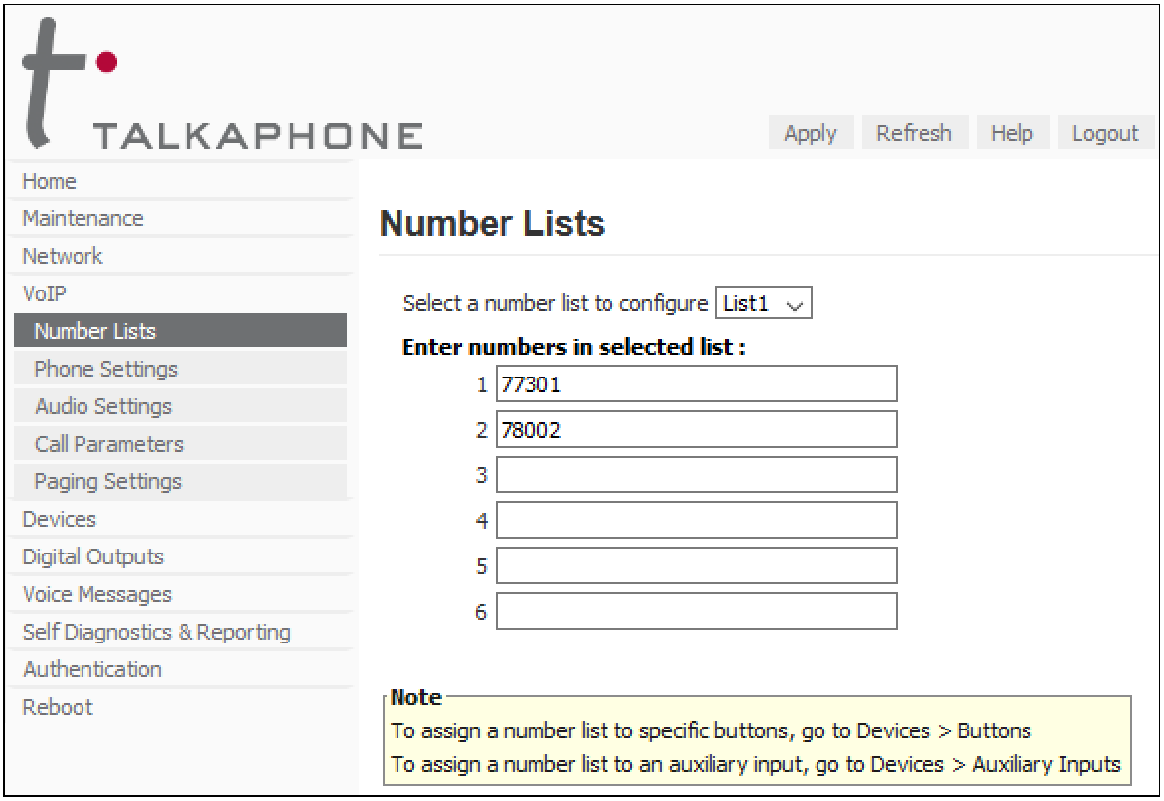
7.6. Configure Buttons

Navigate to **Devices → Buttons** to verify the appropriate settings. For the compliance test, the **Buttons** were configured as shown below. Note that the **Call from Number List** specifies the extension list that the Talkaphone IP Call Station will call when the emergency button is pressed. In this case, *List 1* is used.



7.7. Configure Number Lists

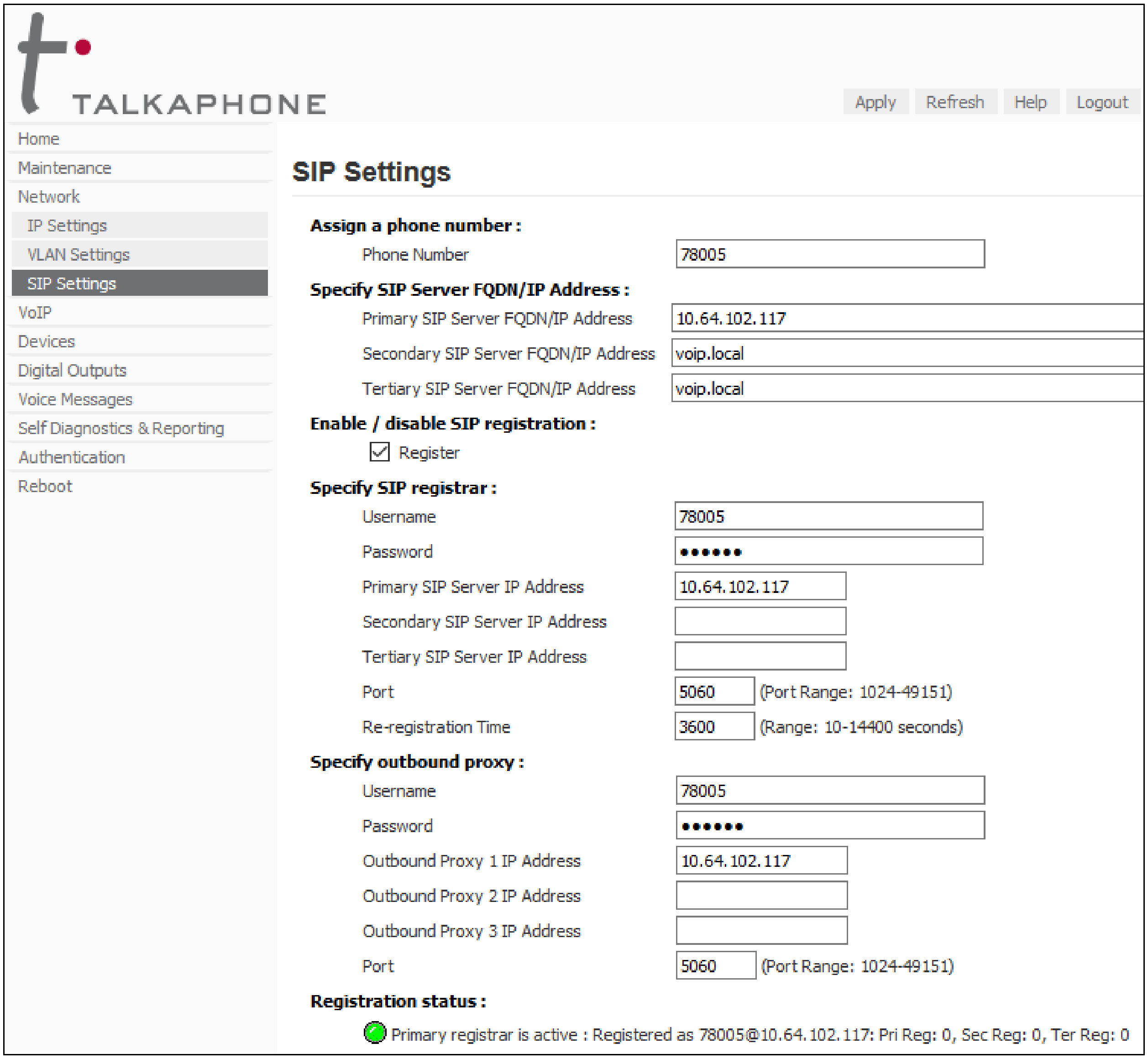
Navigate to **VoIP → Number Lists** to specify the numbers to be called when the emergency button is pressed. In this case, extension 77301 is dialed first, if no answer, then extension 78002 is dialed. Talkaphone IP Call Station will continue calling these numbers in round robin fashion until the call is answered or until the Conversation Call Timer expires.



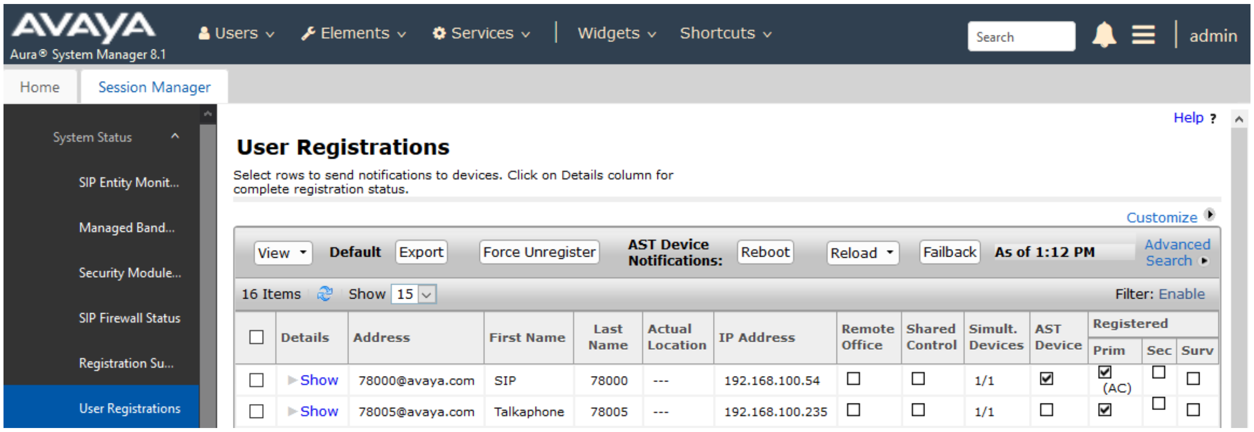
8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Talkaphone VOIP-500 Series IP Call Stations with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the Talkaphone IP Call Station has successfully registered with Session Manager. Alternatively, the SIP Settings screen on the Talkaphone IP Call Station shows the registration status as shown below.



2. Alternatively, verify that Talkaphone IP Call Station has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status.



3. Place incoming/outgoing calls to/from Talkaphone IP Call Station, verify 2-way audio and proper call termination.

9. Conclusion

These Application Notes have described the administration steps required to integrate the Talkaphone VOIP-500 Series IP Call Stations with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Talkaphone IP Call Station successfully registered with Session Manager and basic telephony features were verified from the perspective of Avaya IP Deskphones. All test cases passed with observations noted in **Section 2.2.**

10. Additional References

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

[1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 6, March 2020, available at http://support.avaya.com.

[2] *Administering Avaya Aura® System Manager for Release 8.1.x*, Release 8.1.x, April 2020, available at http://support.avaya.com.

[3] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 3, March 2020, available at http://support.avaya.com.

[4] *Talkaphone VOIP-500 Series Phone Configuration and Operation Manual*, v3.0.2, Rev 7/31/12, available at http://talkaphone.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.