



Avaya Solution & Interoperability Test Lab

Application Notes for Talkphone VOIP-200 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Talkphone VOIP-200 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Talkphone VOIP-200 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. Talkphone VOIP-200 Series IP Call Stations support SIP (RFC 3261) and can operate as a paging/mass notification device via a standard SIP-based inbound call. Talkphone VOIP-200 Series IP Call Stations register with Avaya Aura® Session Manager as a SIP endpoint. For the compliance test, a Talkphone VOIP-201C3 IP Call Station was used.

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 Talkphone VOIP-200 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Talkphone VOIP-200 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. Talkphone VOIP-200 Series IP Call Stations support SIP (RFC 3261) and can operate as a paging/mass notification device via a standard SIP-based inbound call. Talkphone VOIP-200 Series IP Call Stations register with Avaya Aura® Session Manager as a SIP endpoint. For the compliance test, a Talkphone VOIP-201C3 IP Call Station was used.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Talkphone VOIP-200 Series IP Call Stations, Avaya SIP / H.323 Deskphones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer, and conference, from 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 Talkphone VOIP-200 Series IP Call Stations come 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 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 Talkphone VOIP-200 Series IP Call Stations did not include use of any specific encryption features as requested by Talkphone.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Talkphone IP Call Station with Session Manager.
- Inbound and outbound calls between Talkphone IP Call Station and Avaya SIP / H.323 Deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Inbound and outbound calls between the Talkphone IP Call Station and the PSTN.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, transfer, and 3-way conference, initiated from the Avaya IP Deskphones.
- Use of recorded messages, speed-dial buttons, and number lists on the Talkphone IP Call Station.
- Proper system recovery after a restart of the Talkphone IP Call Station 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 Talkphone VOIP-200 Series IP Call Stations. This is by design. The calls can only be disconnected by the destination phone or upon expiration of the Call Conversation Timer. The destination phone of an emergency call shouldn't cover to voicemail. The Talkphone VOIP-200 Series IP Call Stations dial 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.
- Voice messages can only be recorded and played back when using G.711 codec. G.729 codec isn't supporting with recorded messages.
- Dialing Feature Access Codes (FACs) or Feature Name Extensions (FNEs) to activate telephony features are not applicable to Talkphone IP Call Stations.

2.3. Support

For technical support and information on Talkphone VOIP-200 Series IP Call Stations, contact Talkphone support at:

- Phone: 1-773-539-1100
- Email: customerservice@talkphone.com
- Website: <http://www.talkphone.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 Aura® Messaging serving as the voicemail system.
- Avaya 96x1 Series SIP and H.323 Deskphones.
- Talkaphone VOIP-200 Series IP Call Station.

Talkaphone VOIP-200 Series IP Call Stations registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

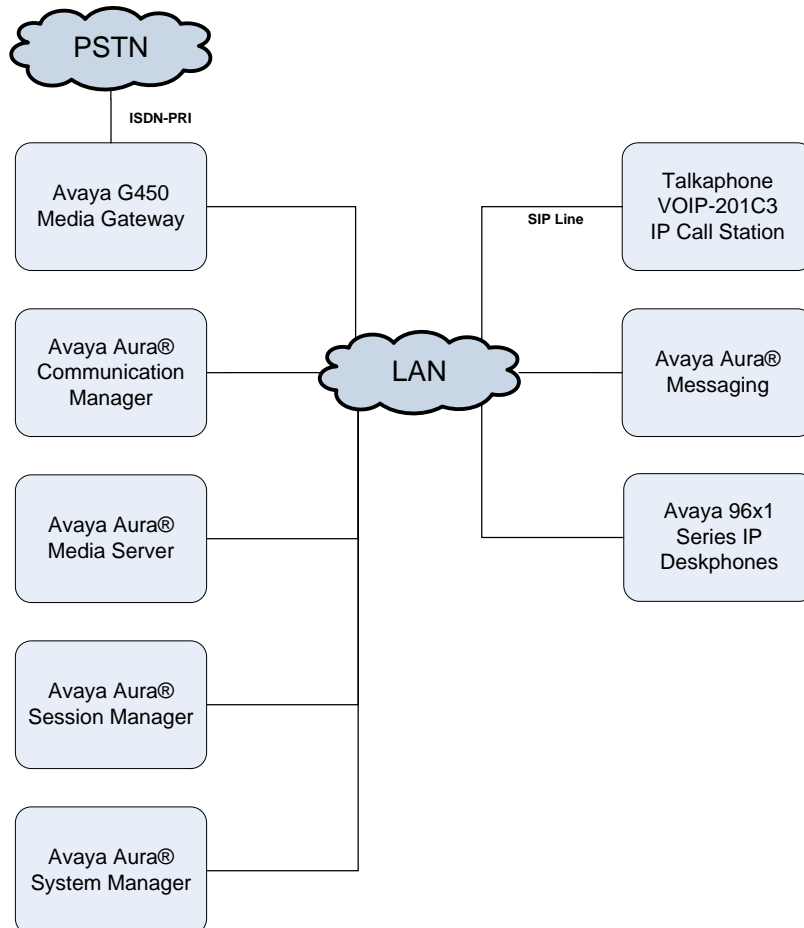


Figure 1: Avaya SIP Network with Talkaphone VOIP-200 Series IP Call Station

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	8.0.1.0.0-FP1 (R018x.00.0.822.0 with Patch 25031)
Avaya G450 Media Gateway	FW 38.21.1
Avaya Aura® Media-Server	v.8.0.0.173
Avaya Aura® System Manager	8.0.1.0 Build No. – 8.0.0.0.931077 Software Update Revision No: 8.0.1.0.038826 Feature Pack 1
Avaya Aura® Session Manager	8.0.1.0801007
Avaya Aura® Messaging	7.1.3.1.0-FP3SP1
Avaya 96x1 Series IP Deskphones	6.7104 (H.323) 7.1.4.0.11 (SIP)
Talkphone VOIP-200 Series IP Call Stations	2.4.1.40

Note: For the compliance test, a Talkphone VOIP-201C3 IP Call Station was used.

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 Talkphone 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      87
Maximum Stations: 36000 26
Maximum XMOBILE Stations: 36000 0
Maximum Off-PBX Telephones - EC500: 41000 0
Maximum Off-PBX Telephones - OPS: 41000 17
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
                     Inter-region IP-IP Direct Audio: yes
                     IP Audio Hairpinning? n
Codec Set: 1
UDP Port Min: 2048
UDP Port Max: 3329
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 Talkphone IP Call Stations were 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*.
- 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? n                    Enforce SIPS URI for SRTP? n
Peer Detection Enabled? y Peer Server: SM
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

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

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to/from Talkphone 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 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: 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. The Talkphone 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
2:
3:
4:
5:
6:
      DCS/ IXC
      n      user
      n      user
      n      user
      n      user
      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

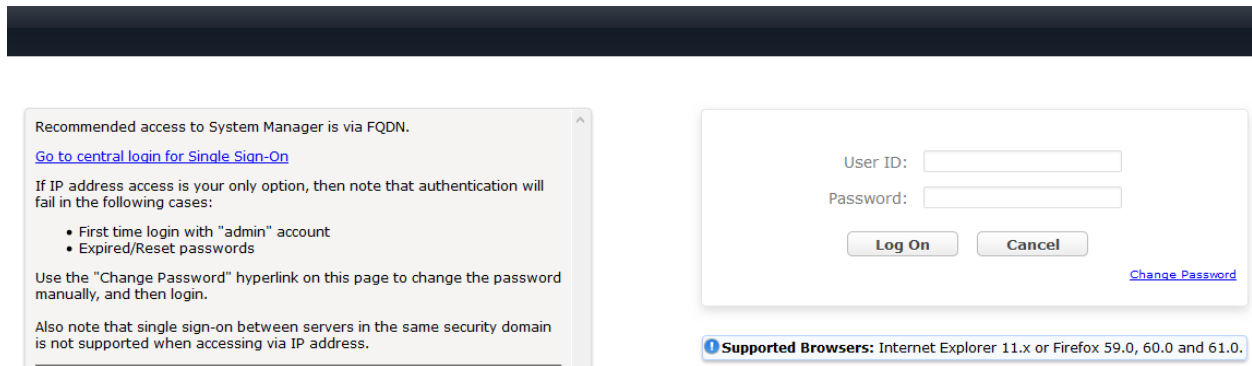
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- 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 Talkphone 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.

The screenshot shows the Avaya Aura System Manager 8.0 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The main content area is titled 'SIP Entity Details' and is divided into 'General' and 'Monitoring' sections. The 'General' section contains the following fields:

- Name:** devcon-sm
- IP Address:** 10.64.102.117
- SIP FQDN:** (empty)
- Type:** Session Manager
- Notes:** (empty)
- Location:** Thornton
- Outbound Proxy:** (empty)
- Time Zone:** America/New_York
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)
- SIP Link Monitoring:** Use Session Manager Configuration
- CRLF Keep Alive Monitoring:** Use Session Manager Configuration

Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Talkphone IP Call Station is specified in the list below. For the compliance test, the solution used UDP network transport.

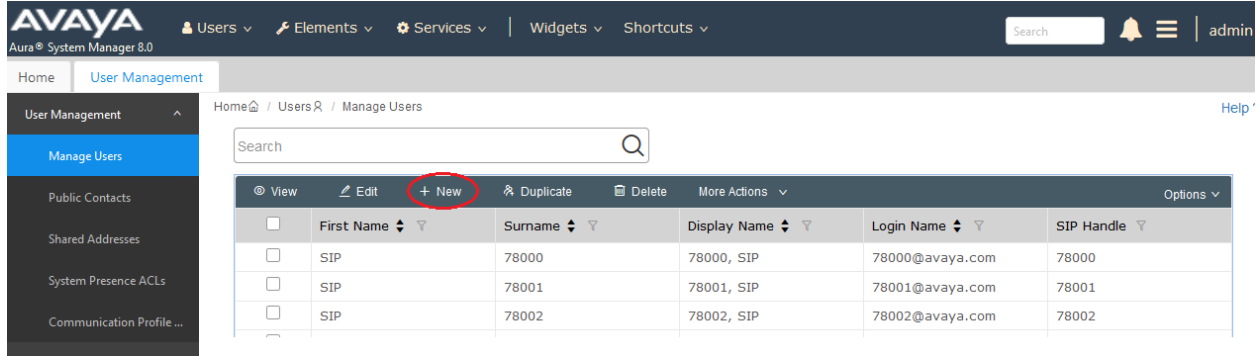
Listen Ports

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5060	UDP	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5061	TLS	avaya.com	<input type="checkbox"/>	

Select : All, None

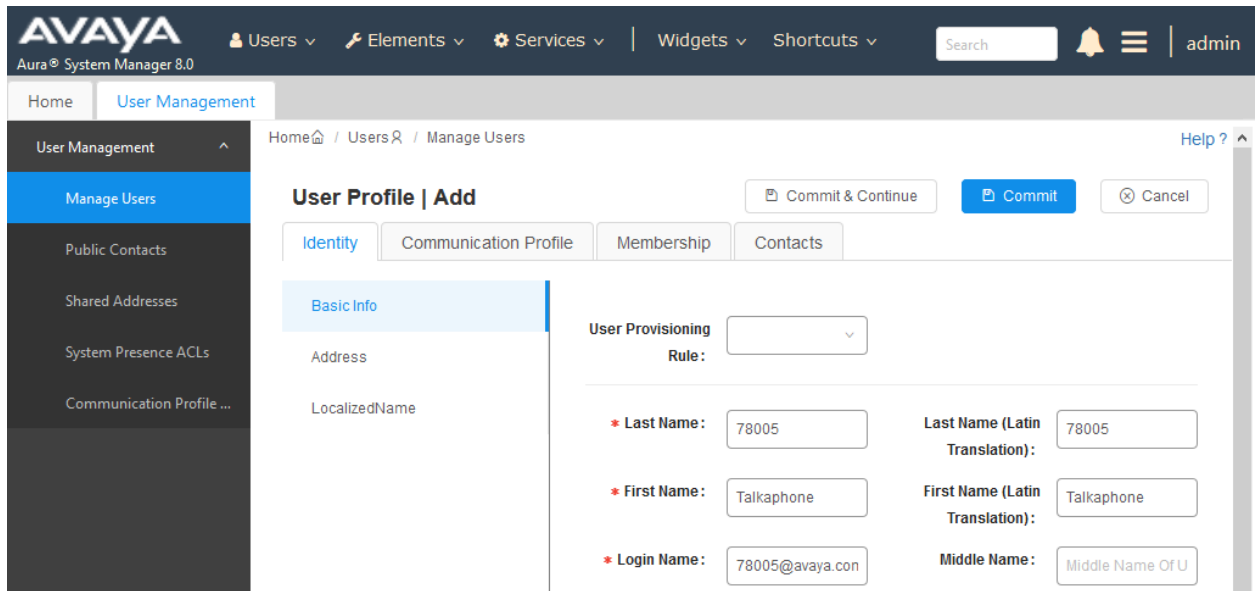
6.3. Administer SIP User

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.



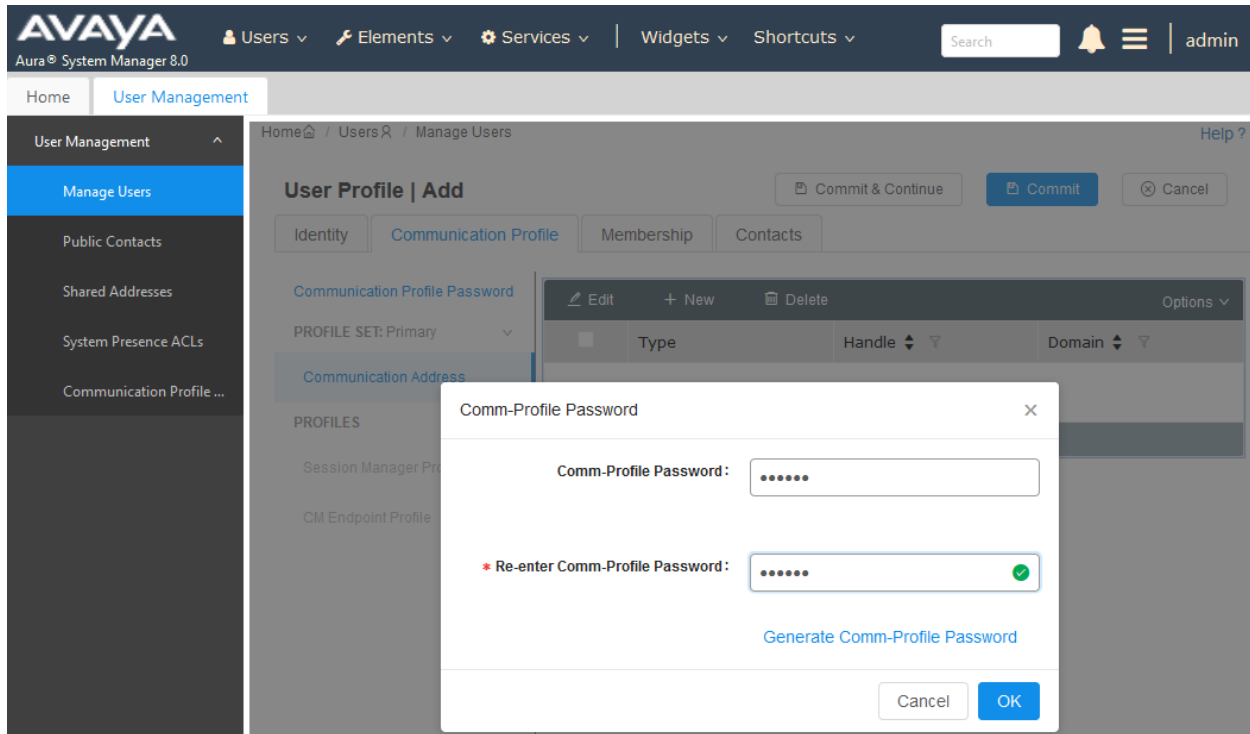
6.3.1. Identity

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 Talkphone 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.



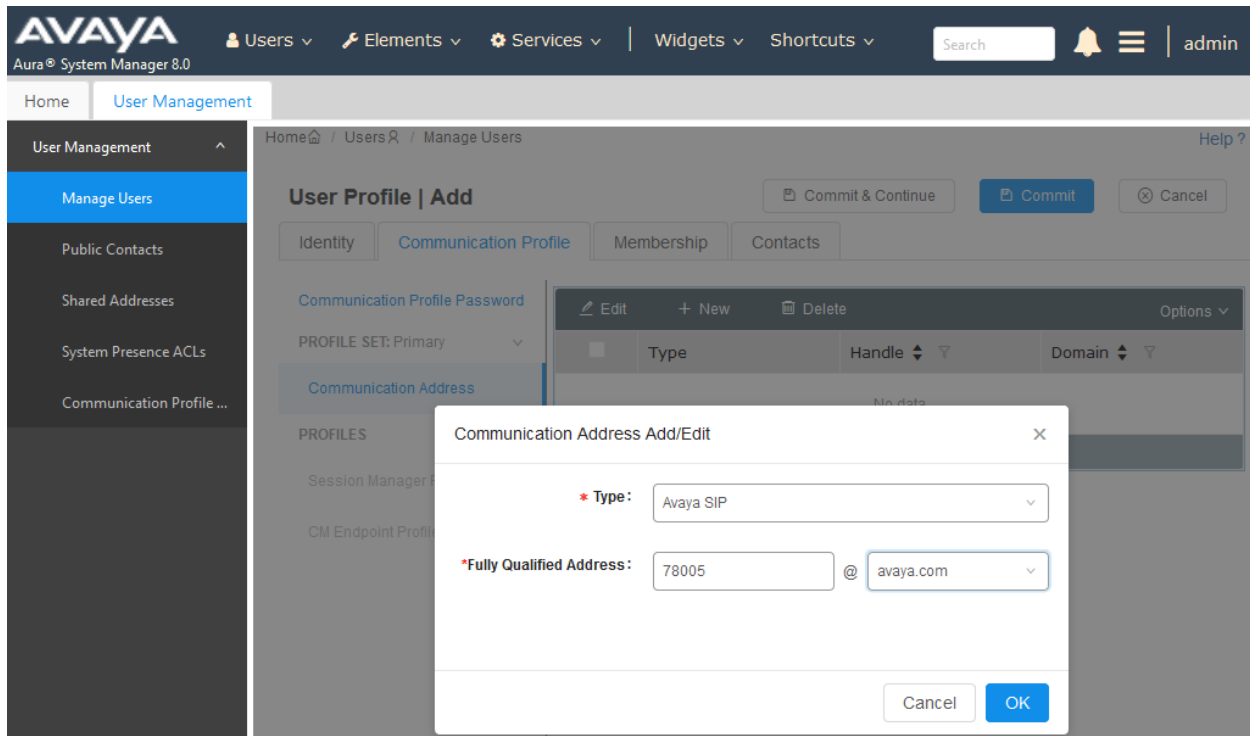
6.3.2. Communication Profile

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**.



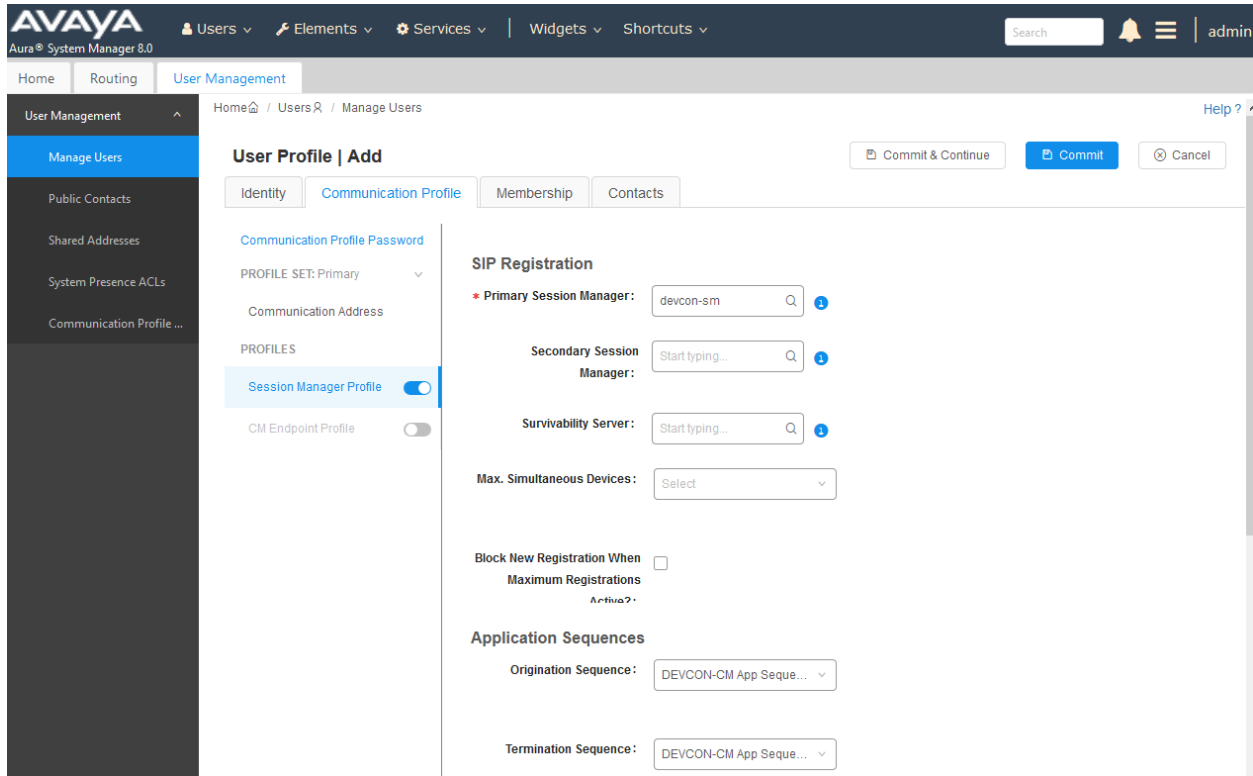
6.3.3. Communication Address

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_0*. For **Port**, click and select *IP*. Retain the default values in the remaining fields.

The screenshot displays the Avaya Aura System Manager 8.0 interface. The top navigation bar includes the Avaya logo, user profile (admin), and various menu items like Users, Elements, Services, Widgets, and Shortcuts. The main content area is titled 'User Profile | Add' and features several tabs: Identity, Communication Profile (selected), Membership, and Contacts. On the left, a sidebar lists navigation options such as 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence ACLs', and 'Communication Profile...'. The 'Communication Profile' section is active, showing a 'Communication Profile Password' dropdown set to 'Primary'. Below this, a 'PROFILES' section has two toggle switches: 'Session Manager Profile' (off) and 'CM Endpoint Profile' (on). The main form fields are as follows:

- * System:** devcon-cm
- * Profile Type:** Endpoint
- Use Existing Endpoints:**
- * Extension:** 78005
- * Template:** 9600SIP_DEFAU Q
- * Set Type:** 9600SIP
- Sub Type:** Select
- Terminal Number:** [][][][]
- System ID:** Enter System Id
- Security Code:** Enter Security Code
- Port:** IP
- Voice Mail Number:** []
- Preferred Handle:** Select
- Calculate Route Pattern:**
- Sip Trunk:** aar
- SIP URI:** Select
- Enhanced Call-Info display for 1-line:**
- Delete on Unassign from User or on:**
- Override Endpoint Name and Localized Name:**
- Allow H.323 and SIP Endpoint Dual Registration:**

7. Configure Talkphone VOIP-200 Series IP Call Station

This section covers the configuration of the Talkphone VOIP-200 Series IP Call Station. The following procedures are covered:

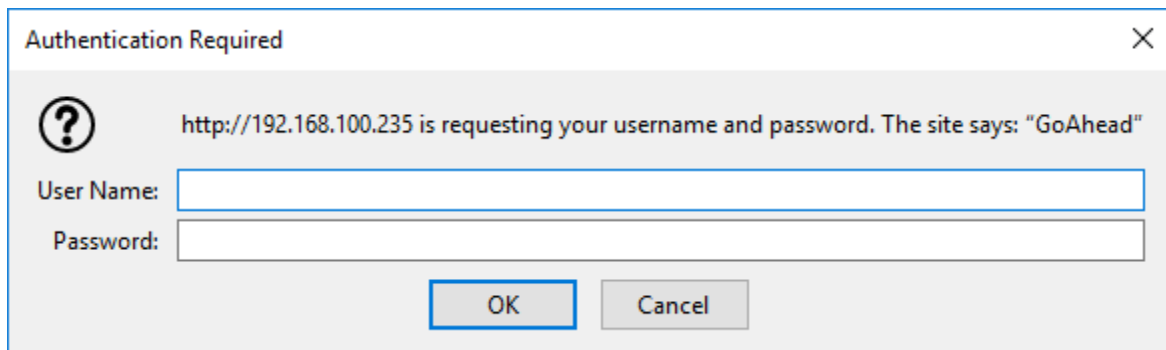
1. Launching the Web Administration Interface
2. Network Configuration
3. SIP Configuration
4. Configure Buttons

7.1. Launching the Web Administration Interface

The Talkphone 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 Talkphone IP Call Station are connected to the LAN. Open a web browser and enter the default IP address of the Talkphone 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 Talkphone IP Call Station, navigate to the **Configuration → IP Settings** page. Configure the IP settings so that it conforms to the customer network requirements. Click **Save** when done.

TALKAPHONE

Home Configuration Administration Diagnostics Network Security

IP Settings

DHCP Static IP

IP Address: 192 . 168 . 100 . 235

Subnet Mask: 255 . 255 . 255 . 0

Gateway: 192 . 168 . 100 . 1

DNS Server 1: 0 . 0 . 0 . 0

DNS Server 2: 0 . 0 . 0 . 0

Hostname: voip000A01

Use last IP address on DHCP failure:

IGMP Version: Default

Save

7.3. SIP Configuration

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

Under **Registration Settings**:

- **Display Name:** Specify a display name (e.g., 78005).
- **Directory Number (SIP ID):** Specify the SIP number (e.g., 78005) configured in **Section 6.3**.
- **Primary SIP Server:** Specify the IP address of the Session Manager signaling interface (e.g., 10.64.102.117).
- **Username:** Specify the SIP number of the Talkphone IP Call Station (e.g., 78005).
- **Password:** Specify the SIP password configured in **Section**

6.3.2.

- **Outbound Proxy 1 (optional):** 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 **Call Settings** section and the remaining fields and click **Save** when done.

The screenshot shows the TALKAPHONE configuration interface. The top navigation bar includes Home, Configuration, Administration, Diagnostics, and Network Security. The left sidebar contains a tree view with categories like IP Settings, SIP Settings, Audio Settings, Buttons, Auxiliary Output, Digital Outputs Scripts, Digital Outputs Events, Voice Messages Played to User, Voice Messages Played to Remote Side, and Time Settings. The main content area is divided into two sections: Registration Settings and Call Settings.

Registration Settings

Description	Configuration
Display Name:	78005
Directory Number (SIP ID):	78005
Primary SIP Server:	10.64.102.117
Secondary SIP Server:	
Tertiary SIP Server:	
Registration Method:	Parallel
Username:	78005
Password:	••••••
Re-registration Time:	3600 (Range: 60-14400 seconds)
Outbound Proxy 1 (optional):	10.64.102.117 Port: 5060
Outbound Proxy 2 (optional):	Port: 5060
Outbound Proxy 3 (optional):	Port: 5060

Call Settings

Description	Configuration
Enable auto-answer:	<input checked="" type="checkbox"/>
Auto-answer Delay:	0 seconds (Range: 0 to 30 seconds)
Provisional Timer:	0 seconds (Range: 0 to 60 seconds) Delays call setup using input buttons
Overlap dialing:	<input type="checkbox"/>
DTMF method:	RFC 2833
Call LED off during ringing:	<input type="checkbox"/>
Hang-up on Silence Timer:	0 seconds (0 = Hang-up on Silence disabled)
Codec G.711 PCM u-law:	High Priority
Codec G.711 PCM A-law:	Low Priority
Codec G.722:	Low Priority
Codec G.729:	Low Priority

Save

7.4. Configure Buttons

Navigate to **Configuration** → **Buttons** to verify the appropriate settings. For the compliance test, the **Buttons** were configured as shown below. The **Value** field for **Button 1** was set to a valid extension. This is the destination that will be dialed when the call button is pressed. In the **Buttons (Active Call)** section, set the **Button 1 Function** field to Disconnect. This will allow an active call to be disconnected when the call button is pressed. This is optional based on customer requirements.

The screenshot shows the TalkPhone configuration interface. The navigation menu includes Home, Configuration, Administration, Diagnostics, and Network Security. The left sidebar contains various settings categories, with 'Buttons' expanded. The main content area is divided into three sections: 'Buttons (Idle)', 'Buttons (Active Call)', and 'Numberlist Settings'.

Buttons (Idle)

	Function	Value	Option
Button 1	Call To	77301	None
Button 2	Call To		None
Button 3	Call To		None

Buttons (Active Call)

	Function	Activated	Deactivated
Button 1	Disconnect		
Button 2	Do Nothing		
Button 3	Do Nothing		

Numberlist Settings

	Ringlist 1	With Previous	Ringlist 2	With Previous	Ringlist 3	With Previous
Value 1		<input type="checkbox"/>		<input type="checkbox"/>		<input type="checkbox"/>
Value 2		<input type="checkbox"/>		<input type="checkbox"/>		<input type="checkbox"/>
Value 3		<input type="checkbox"/>		<input type="checkbox"/>		<input type="checkbox"/>
Value 4		<input type="checkbox"/>		<input type="checkbox"/>		<input type="checkbox"/>

Scroll down to the **Numberlist Settings** section to view and customize call parameters, such as **Ringer Time**, which may need to be increased to provide enough time for the destination to cover to an alternate destination, if necessary, **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. Click **Save** when done.

Note: After a number is dialed on the Talkphone 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.

Numberlist Settings						
	Ringlist 1	With Previous	Ringlist 2	With Previous	Ringlist 3	With Previous
Value 1	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 2	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 3	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 4	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 5	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 6	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 7	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 8	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 9	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 10	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 11	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 12	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 13	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 14	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Call Until Answer	<input checked="" type="checkbox"/>	(loops the numberlist)				
Ringer Time	<input type="text" value="10"/>	seconds, (0=unlimited)				
Call Conversation Timer	<input type="text" value="720"/>	seconds, (Range: 0 to 9999 seconds, 0 = unlimited)				
Local Interdigit Timer	<input type="text" value="5"/>	seconds, (Range: 5 to 20 seconds)				
<input type="button" value="Save"/>						

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Talkphone VOIP-200 Series IP Call Stations with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that Talkphone 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.

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered
<input type="checkbox"/>	Show	78005@avaya.com	Talkphone	78005	---	192.168.100.235	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>
<input type="checkbox"/>	Show	78000@avaya.com	SIP	78000	---	192.168.100.54	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	(AC) <input type="checkbox"/>

2. Place an incoming/outgoing call to to/from the Talkphone 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 Talkphone VOIP-200 Series IP Call Stations with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Talkphone IP Call Stations successfully registered with Session Manager and basic telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

10. Additional References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at support.avaya.com.

- [1] *Administering Avaya Aura® Communication Manager*, Release 8.0.1, Issue 3, December 2018, available at <http://support.avaya.com>.
- [2] *Administering Avaya Aura® System Manager for Release 8.0.1*, Release 8.0.x, Issue 7, January 2019, available at <http://support.avaya.com>.
- [3] *Administering Avaya Aura® Session Manager*, Release 8.0.1, Issue 3, December 2018, available at <http://support.avaya.com>.

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