



Avaya Solution & Interoperability Test Lab

Application Notes for Talkphone VOIP-600 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-600 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Talkphone VOIP-600 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-600 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-600 Series IP Call Stations register with Avaya Aura® Session Manager as a SIP endpoint. For the compliance test, a Talkphone VOIP-600ECK 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-600 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Talkphone VOIP-600 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-600 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-600 Series IP Call Stations register with Avaya Aura® Session Manager as a SIP endpoint. For the compliance test, a Talkphone VOIP-600ECK 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-600 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-600 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-600 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 paging, recorded messages, emergency calls, 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-600 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-600 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-600 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-600 Series IP Call Stations.

Talkaphone VOIP-600 Series IP Call Stations registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

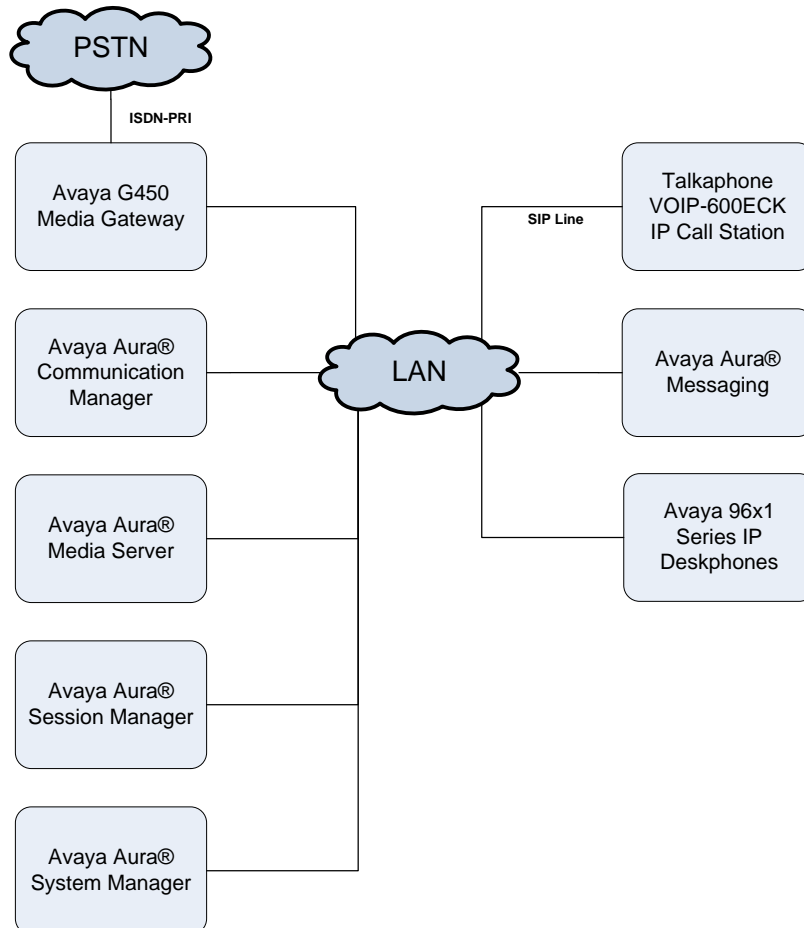


Figure 1: Avaya SIP Network with Talkaphone VOIP-600 Series IP Call Station

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-600 Series IP Call Stations	1.0.2.10

Note: For the compliance test, a Talkphone VOIP-600ECK 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
    Codec Set: 1                Inter-region IP-IP Direct Audio: yes
    UDP Port Min: 2048                IP Audio Hairpinning? n
    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:
                                     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

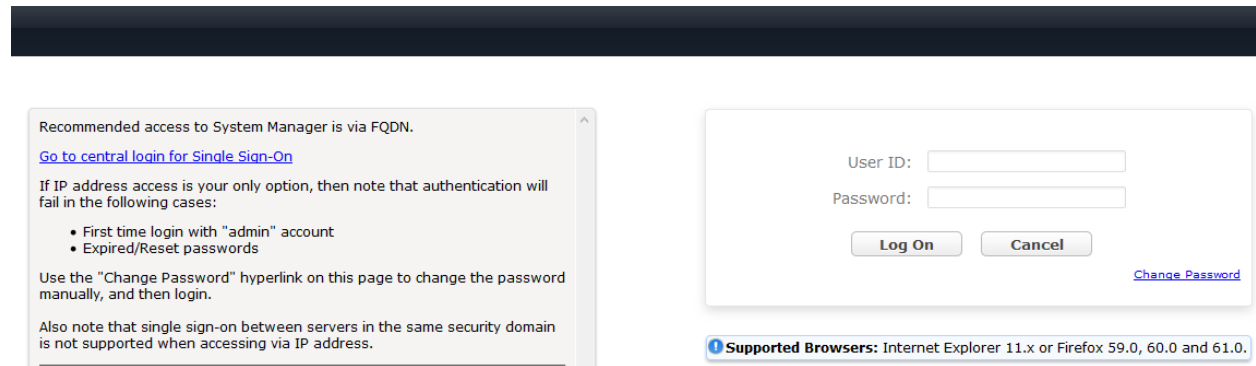
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- 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.

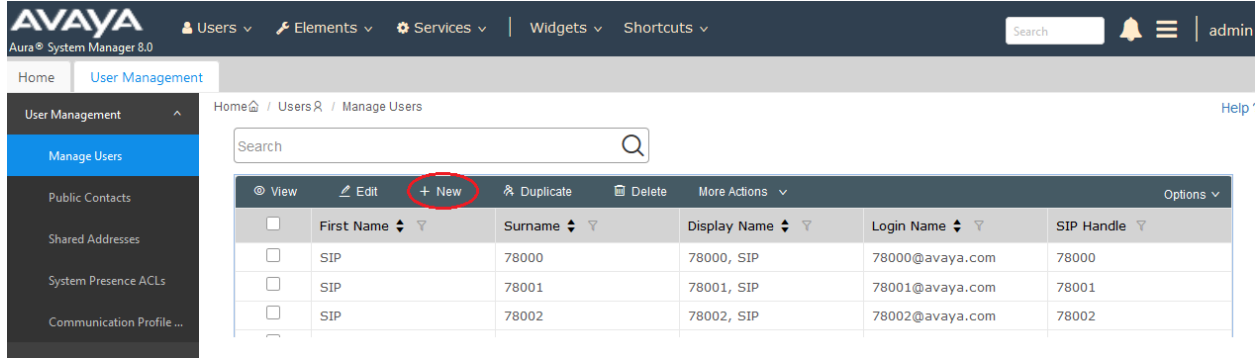
Listen Ports

<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

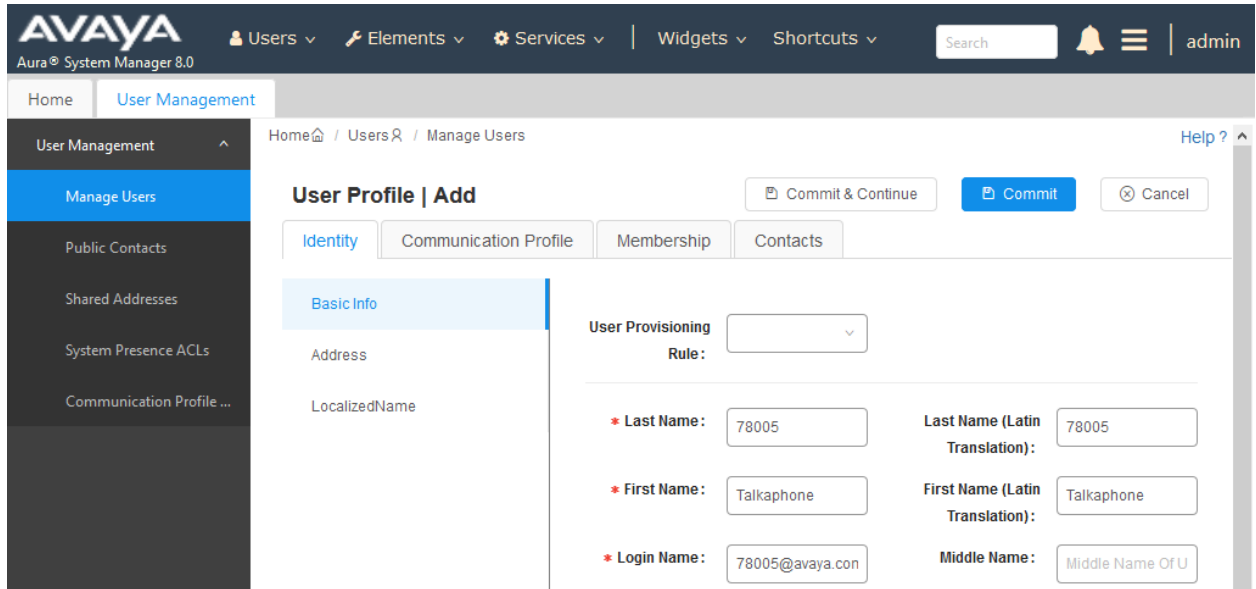
6.3. Administer SIP User

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.



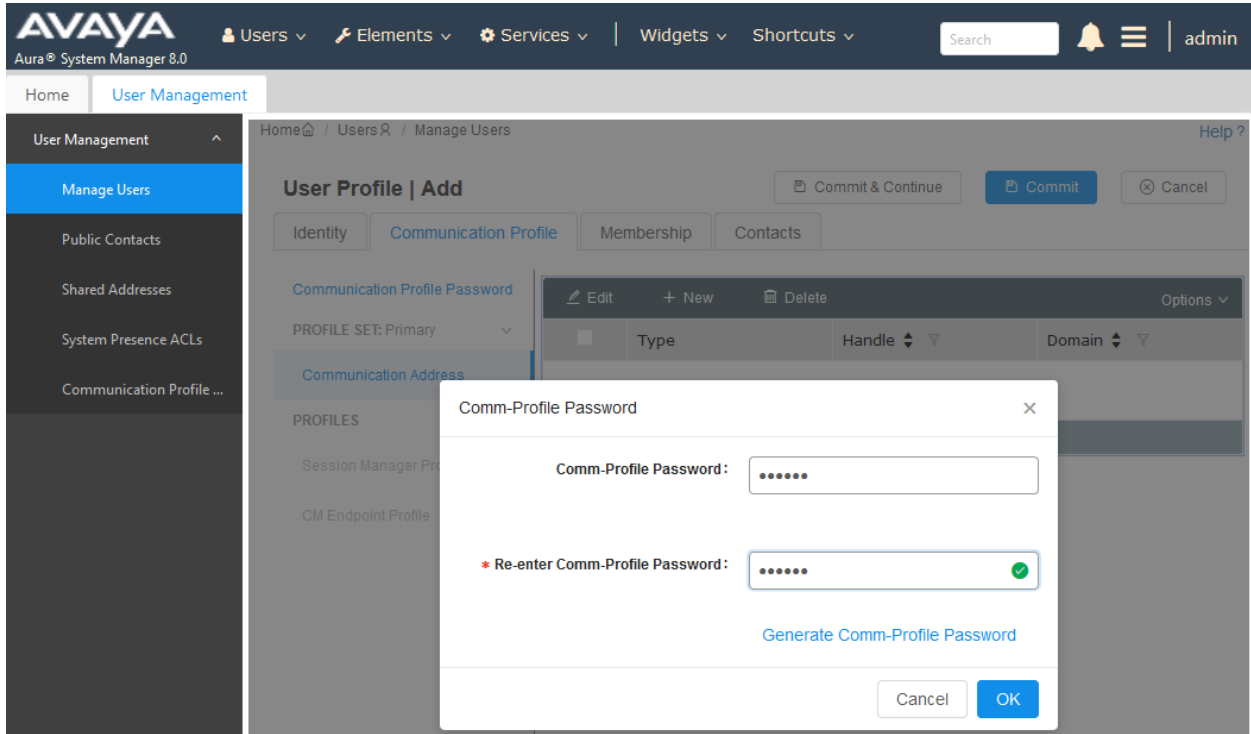
6.3.1. Identity

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.



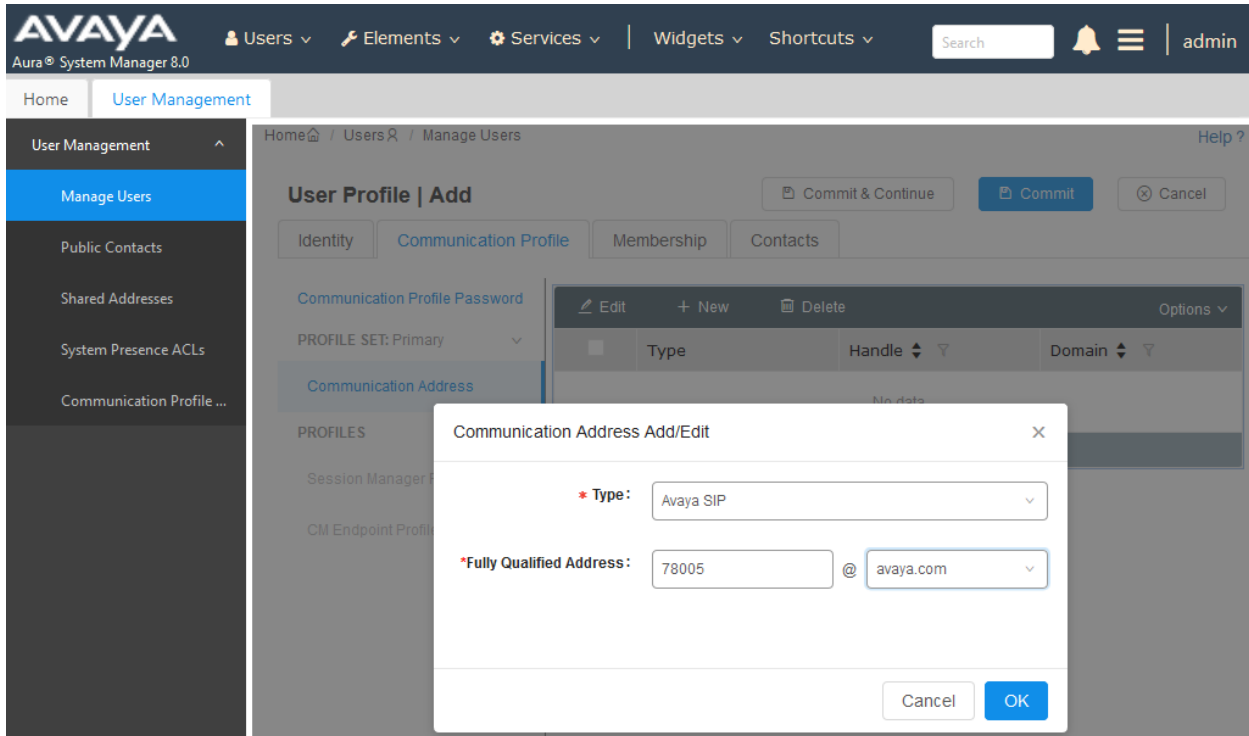
6.3.2. Communication Profile

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.

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 'User Profile | Add' and has tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active. On the left, there is a sidebar with 'Manage Users' selected and a 'Session Manager Profile' toggle switch turned on. The main configuration area includes:

- Communication Profile Password:** PROFILE SET: Primary, Communication Address.
- PROFILES:** Session Manager Profile (toggle on), CM Endpoint Profile (toggle off).
- SIP Registration:**
 - Primary Session Manager: devcon-sm
 - Secondary Session Manager: Start typing...
 - Survivability Server: Start typing...
 - Max. Simultaneous Devices: Select
 - Block New Registration When Maximum Registrations:
- Application Sequences:**
 - Origination Sequence: DEVCON-CM App Seque...
 - Termination Sequence: DEVCON-CM App Seque...

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

The screenshot shows the 'Call Routing Settings' section. It includes:

- Home Location:** Thornton
- Conference Factory Set:** Select

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, 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts' menus, a search bar, and a user profile icon labeled 'admin'. The main content area is titled 'User Profile | Add' and features tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, showing a 'Communication Profile Password' section with a dropdown for 'PROFILE SET: Primary' and a 'Communication Address' field. Below this is a 'PROFILES' section with two toggle switches: 'Session Manager Profile' (off) and 'CM Endpoint Profile' (on). The main form area contains various fields: 'System' (devcon-cm), 'Profile Type' (Endpoint), 'Use Existing Endpoints' (checkbox), 'Extension' (78005), 'Template' (9600SIP_DEFAULT_CM_8_0), 'Set Type' (9600SIP), 'Sub Type' (Select), 'Terminal Number' (four input boxes), 'System ID' (Enter System Id), 'Security Code' (Enter Security Code), 'Port' (IP), 'Voice Mail Number' (input box), 'Preferred Handle' (Select), 'Calculate Route Pattern' (checkbox), 'Sip Trunk' (aar), and 'SIP URI' (Select). At the bottom, there are checkboxes for 'Enhanced Call-Info display for 1-line' (off), 'Override Endpoint Name and Localized Name' (checked), 'Delete on Unassign from User or on' (checked), and 'Allow H.323 and SIP Endpoint Dual Registration' (off).

7. Configure Talkphone VOIP-600 Series IP Call Station

This section covers the configuration of the Talkphone VOIP-600 Series IP Call Station. The following procedures are covered:

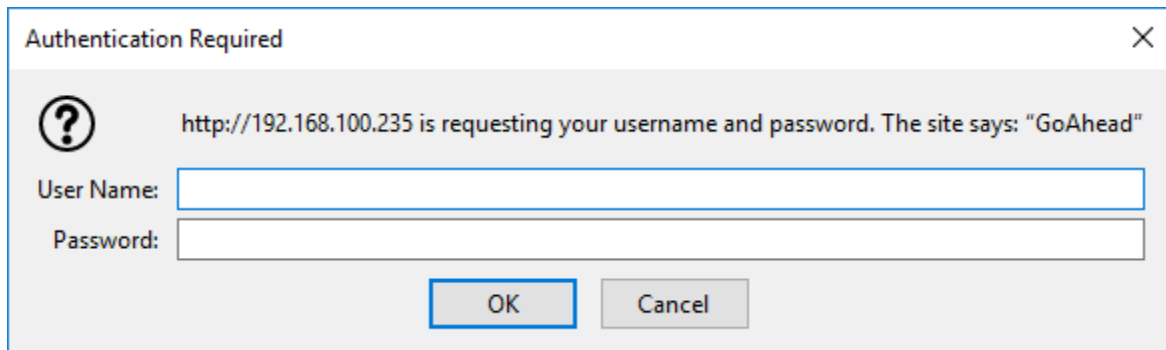
1. Launching the Web Administration Interface
2. Network Configuration
3. SIP Configuration
4. Configure Audio Settings
5. Configure Call Parameters
6. Configure Buttons

7.1. 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.



Authentication Required

http://192.168.100.235 is requesting your username and password. The site says: "GoAhead"

User Name:

Password:

OK Cancel

7.2. Network Configuration

To modify the IP network configuration of the Talkphone 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.

The screenshot shows the Talkphone web interface. The top left has the 't TALKPHONE' logo. The top right has buttons for 'Apply', 'Refresh', 'Help', and 'Logout'. A left-hand navigation menu includes 'Home', 'Maintenance', 'Network', 'IP Settings' (highlighted), 'VLAN Settings', 'SIP Settings', 'VoIP', 'Devices', 'Digital Outputs', 'Voice Messages', 'Self Diagnostics & Reporting', 'Authentication', and 'Reboot'. The main content area is titled 'IP Settings' and contains the following sections:

- Configure network connection :**
 - DHCP - Automatic Configuration
 - Static IP - Manual Configuration
- Specify network details for "Static IP - Manual Configuration" :**
 - IP Address:
 - Subnet Mask:
 - Default Gateway:
 - DNS Server:
- Enter hostname :**
 - Hostname:

7.3. SIP Configuration

Navigate to **Network → SIP Settings** to configure the SIP setting of the Talkphone 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 Talkphone 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.

The screenshot displays the 'SIP Settings' configuration page in the Talkphone web interface. The page includes a navigation menu on the left with 'SIP Settings' selected. The main content area is titled 'SIP Settings' and contains several sections:

- Assign a phone number :** Phone Number: 78005
- Specify SIP Server FQDN/IP Address :** Primary SIP Server FQDN/IP Address: 10.64.102.117; Secondary SIP Server FQDN/IP Address: voip.local; Tertiary SIP Server FQDN/IP Address: voip.local
- Enable / disable SIP registration :** Register
- Specify SIP registrar :** Username: 78005; Password: [masked]; Primary SIP Server IP Address: 10.64.102.117; Secondary SIP Server IP Address: [empty]; Tertiary SIP Server IP Address: [empty]; Port: 5060 (Port Range: 1024-49151); Re-registration Time: 3600 (Range: 10-14400 seconds)
- Specify outbound proxy :** Username: 78005; Password: [masked]; Outbound Proxy 1 IP Address: 10.64.102.117; Outbound Proxy 2 IP Address: [empty]; Outbound Proxy 3 IP Address: [empty]; Port: 5060 (Port Range: 1024-49151)
- Registration status :** Unregistered

At the top right of the page, there are buttons for 'Apply', 'Refresh', 'Help', and 'Logout'.

7.4. Configure Audio Settings

Navigate to **VoIP → Audio Settings** to configure the preferred codec, outbound DTMF duration, and microphone and speaker parameters. For the compliance test, the **DTMF duration for outgoing calls** was set to **Custom** with **Duration** of *800 ms*. This is required so that a user can navigate through Avaya Aura® Messaging using DTMF tones. 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.

The screenshot shows the 'Audio Settings' page in the TALKAPHONE interface. The left sidebar contains a navigation menu with items: Home, Maintenance, Network, VoIP (selected), Number Lists, Phone Settings, Audio Settings (highlighted), Call Parameters, Paging Settings, Devices, Digital Outputs, Voice Messages, Self Diagnostics & Reporting, Authentication, and Reboot. The main content area is titled 'Audio Settings' and includes the following sections:

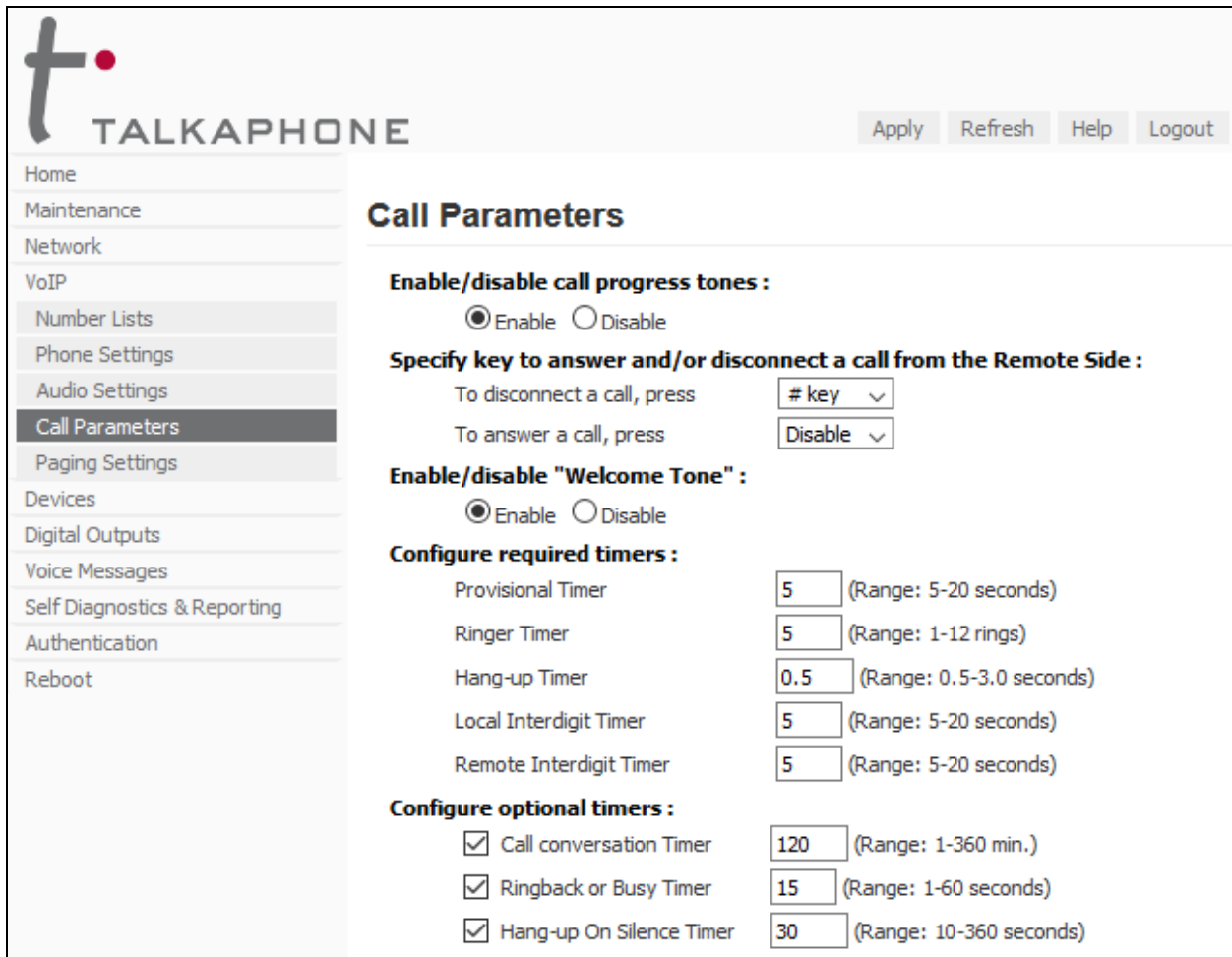
- Select VoIP codec :**
 - G.711 PCM a-Law @ 64kbps
 - G.711 PCM u-Law @ 64kbps
 - G.729a
 - G.723.1a
- Enable/disable audio processing modules :**
 - VAD/CNG
 - AEC
 - AGC
 - Jitter Buffer:
- DTMF duration for outgoing calls :**
 - Disable
 - 51 ms
 - 60 ms
 - 102 ms
 - Custom
 - Duration: (Range: 10-1000 ms)
- Configure Line Level Output parameters :**
 - Line Gain:
- Configure Speaker/Microphone parameters :**
 - Speaker Speaker Gain:
 - Microphone Microphone Gain:
 - Use Speaker for notification and ringing only

At the top right of the page, there are buttons for 'Apply', 'Refresh', 'Help', and 'Logout'.

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 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.



TALKPHONE [Apply] [Refresh] [Help] [Logout]

Home
Maintenance
Network
VoIP
Number Lists
Phone Settings
Audio Settings
Call Parameters
Paging Settings
Devices
Digital Outputs
Voice Messages
Self Diagnostics & Reporting
Authentication
Reboot

Call Parameters

Enable/disable call progress tones :
 Enable Disable

Specify key to answer and/or disconnect a call from the Remote Side :
To disconnect a call, press
To answer a call, press

Enable/disable "Welcome Tone" :
 Enable Disable

Configure required timers :

Provisional Timer	<input type="text" value="5"/>	(Range: 5-20 seconds)
Ringer Timer	<input type="text" value="5"/>	(Range: 1-12 rings)
Hang-up Timer	<input type="text" value="0.5"/>	(Range: 0.5-3.0 seconds)
Local Interdigit Timer	<input type="text" value="5"/>	(Range: 5-20 seconds)
Remote Interdigit Timer	<input type="text" value="5"/>	(Range: 5-20 seconds)

Configure optional timers :

<input checked="" type="checkbox"/> Call conversation Timer	<input type="text" value="120"/>	(Range: 1-360 min.)
<input checked="" type="checkbox"/> Ringback or Busy Timer	<input type="text" value="15"/>	(Range: 1-60 seconds)
<input checked="" type="checkbox"/> Hang-up On Silence Timer	<input type="text" value="30"/>	(Range: 10-360 seconds)

7.6. Configure Buttons

Navigate to **Devices** → **Buttons** to verify the appropriate settings. For the compliance test, the **Buttons** were configured as shown below.

The screenshot displays the TALKAPHONE web interface. The top left features the logo and the text 'TALKAPHONE'. To the right of the logo are buttons for 'Apply', 'Refresh', 'Help', and 'Logout'. A vertical navigation menu on the left lists various system components, with 'Buttons' highlighted. The main content area is titled 'Buttons' and contains two configuration sections: 'Configure Button #1' and 'Configure Button #2'. Each section includes settings for mode, call from number list, call priority, and network priority.

Section	Setting	Value
Configure Button #1	Button #1 Mode	Always Autodial <input type="checkbox"/>
	Call from Number List	List 1
	Call Priority	1
	Network Priority	46 (Range: 0-63)
Configure Button #2	Button #2 Mode	Hook Switch
	Call from Number List	List 1
	Call Priority	2
	Network Priority	46 (Range: 0-63)

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Talkphone VOIP-600 Series IP Call Stations with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the 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. Alternatively, the SIP Settings screen on the Talkphone IP Call Station also shows the registration status as shown below.

TALKPHONE Apply Refresh Help Logout

Home
Maintenance
Network
IP Settings
VLAN Settings
SIP Settings
VoIP
Devices
Digital Outputs
Voice Messages
Self Diagnostics & Reporting
Authentication
Reboot

SIP Settings

Assign a phone number :
Phone Number: 78005

Specify SIP Server FQDN/IP Address :
Primary SIP Server FQDN/IP Address: 10.64.102.117
Secondary SIP Server FQDN/IP Address: voip.local
Tertiary SIP Server FQDN/IP Address: voip.local

Enable / disable SIP registration :
 Register

Specify SIP registrar :
Username: 78005
Password: ●●●●●●
Primary SIP Server IP Address: 10.64.102.117
Secondary SIP Server IP Address:
Tertiary SIP Server IP Address:
Port: 5060 (Port Range: 1024-49151)
Re-registration Time: 3600 (Range: 10-14400 seconds)

Specify outbound proxy :
Username: 78005
Password:
Outbound Proxy 1 IP Address: 10.64.102.117
Outbound Proxy 2 IP Address:
Outbound Proxy 3 IP Address:
Port: 5060 (Port Range: 1024-49151)

Registration status :
 Primary registrar is active : Registered as 78005@10.64.102.117

- Alternatively, 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.

The screenshot shows the Avaya Aura System Manager 8.0 interface. The main content area is titled "User Registrations" and includes a table of registered devices. The table has the following columns: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, and Registered (Prim, Sec, Surv). Two rows are visible, both with a "Show" button and a checked "Registered" status.

Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
										Prim	Sec	Surv
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"/>	<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"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>

- 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-600 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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