



Avaya Solution & Interoperability Test Lab

Application Notes for Talkphone VOIP-500 Series and 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-500 Series and VOIP-600 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations registered with Avaya Aura® Session Manager via SIP. Although not explicitly tested, these Application Notes would also apply to the Talkphone Wide-Area Emergency Broadcast System (WEBS®) Series Devices, which leverage the same electronics and firmware with a similar subset of features (e.g. paging only with one-way communication) as the VOIP-500 Series and VOIP-600 Series Phones but differ in form factor and packaging.

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-500 Series and VOIP-600 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations registered with Avaya Aura® Session Manager via SIP. Although not explicitly tested, these Application Notes would also apply to the Talkphone Wide-Area Emergency Broadcast System (WEBS®) Series Devices, which leverage the same electronics and firmware with a similar subset of features (e.g. paging only with one-way communication) as the VOIP-500 Series and VOIP-600 Series Phones but differ in form factor and packaging.

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-500 Series and VOIP-600 Series IP Call Stations, Avaya SIP and H.323 telephones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer, and conference, from the Avaya IP phones. Additional telephony features, such as call forward and call coverage, were also verified.

The serviceability testing focused on verifying that the Talkphone VOIP-500 Series and 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.

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 and H.323 telephones 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 phone.
- Use of paging, 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-500 Series and VOIP-600 Series IP Call Stations. The calls can only be disconnected by the destination phone or upon expiration of the Call Conversation Timer. The Talkphone VOIP-500 Series and 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.
- 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-500 Series and VOIP-600 Series IP Call Stations, contact Talkphone support at:

- Phone: 1-773-539-1100
- Website: <http://www.talkphone.com/content/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 (not shown).
- 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 9600 and 96x1 Series SIP and H.323 Telephones.
- Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations.

Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

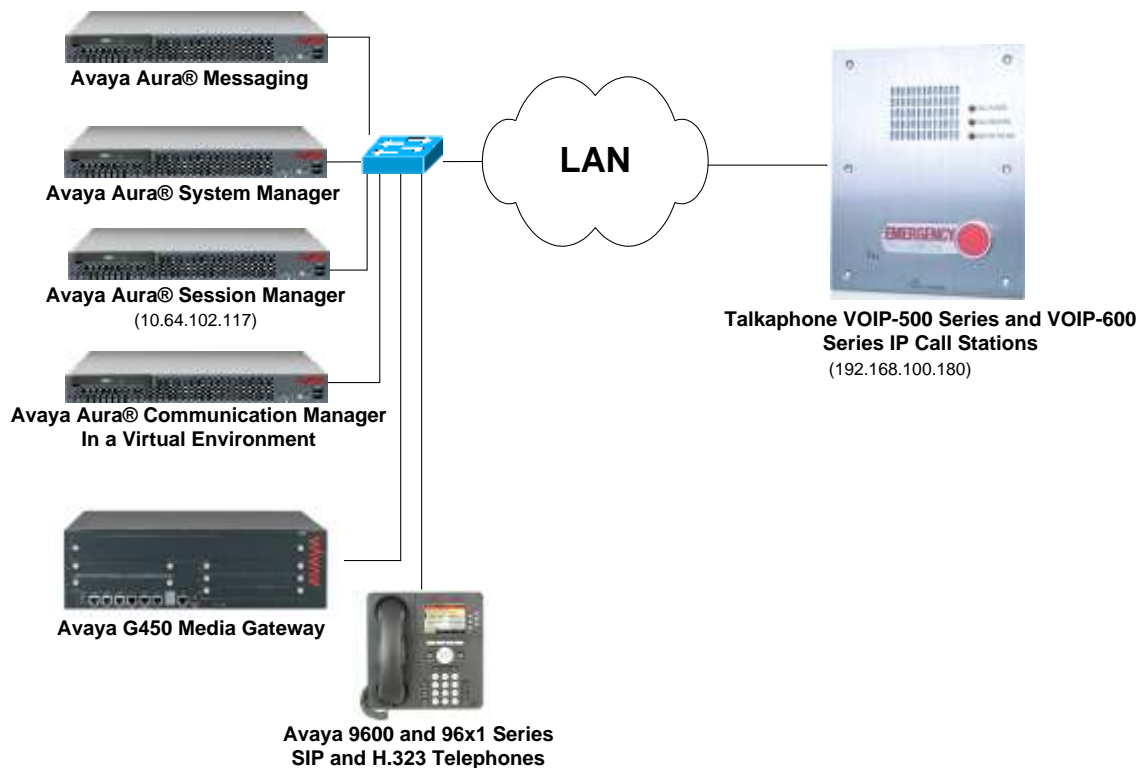


Figure 1: Avaya SIP Network with Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations

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 in a Virtual Environment with an Avaya G450 Media Gateway	7.0 SP 1 (R017x.00.0.441.0 with Patch 22477)
Avaya Aura® Media-Server	7.7.0.226
Avaya Aura® System Manager	7.0 (Build No. 7.0.0.016266-7.0.9.912 Software Update Revision No: 7.0.0.0.3929)
Avaya Aura® Session Manager	7.0.0.0.700007
Avaya Aura® Messaging	6.3.2 SP 2 Patch 3
Avaya 9600 Series IP Deskphones	S3.250A (H.323)
Avaya 96x1 Series IP Deskphones	7.0.0.39 (SIP)
Talkphone VOIP-500 Series IP Call Stations	1.0.2.4
Talkphone VOIP-600 Series IP Call Stations	1.0.2.4

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 Network Region and IP Codec Set

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

Note: It is assumed that basic configuration of the Communication Manager has already been completed, such as the SIP trunk to Session Manager. The SIP station configuration for the Talkphone IP Call Stations are configured through Avaya Aura® System Manager in **Section 6.2**.

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.

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

G3 Version: V17                                                         Software Package: Enterprise
Location: 2                                                             System ID (SID): 1
Platform: 28                                                            Module ID (MID): 1

                                USED
                                Platform Maximum Ports: 6400 22
                                Maximum Stations: 2400 7
                                Maximum XMOBILE Stations: 2400 0
Maximum Off-PBX Telephones - EC500: 9600 0
Maximum Off-PBX Telephones - OPS: 9600 4
Maximum Off-PBX Telephones - PBFMC: 9600 0
Maximum Off-PBX Telephones - PVFMC: 9600 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 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:
  
```

6. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- 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.

AVAYA
Aura System Manager 7.0

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

User ID:

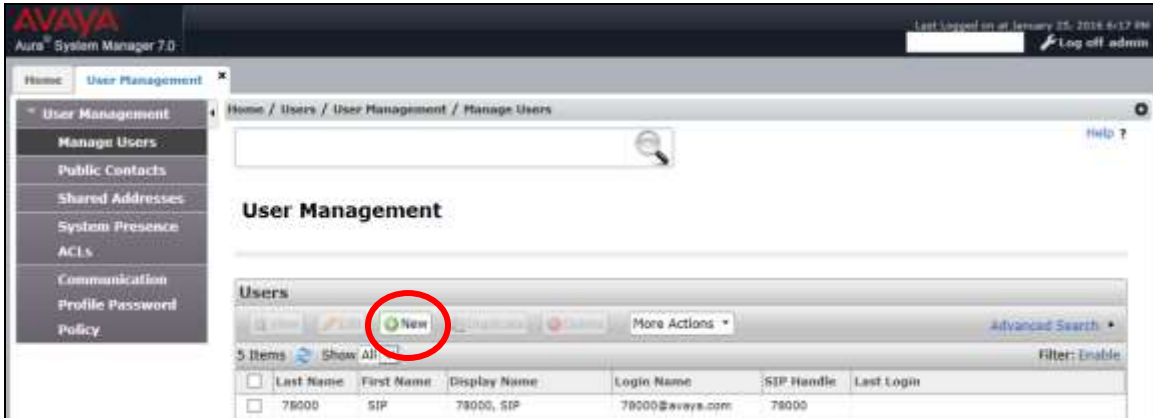
Password:

[Change Password](#)

Supported Browsers: Internet Explorer 9.x, 10.x or 11.x or Firefox 36.0, 37.0 and 38.0.

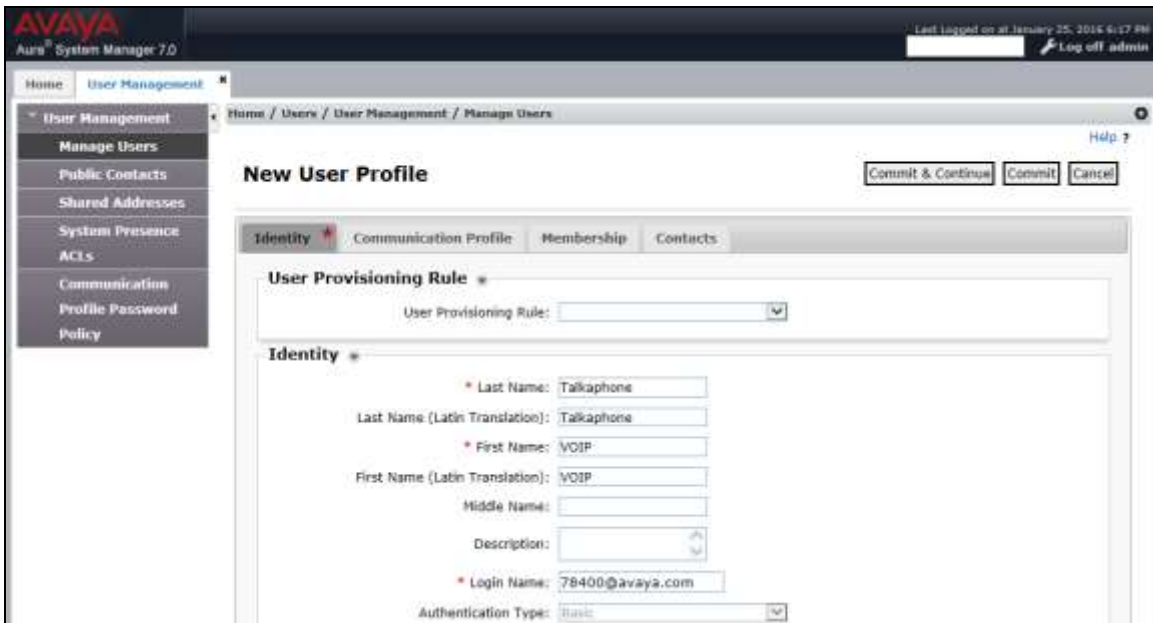
6.2. Administer SIP User

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



6.2.1. 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 5.2**. For **Password** and **Confirm Password**, enter the appropriate credentials for System Manager. Retain the default values in the remaining fields.



6.2.2. Communication Profile

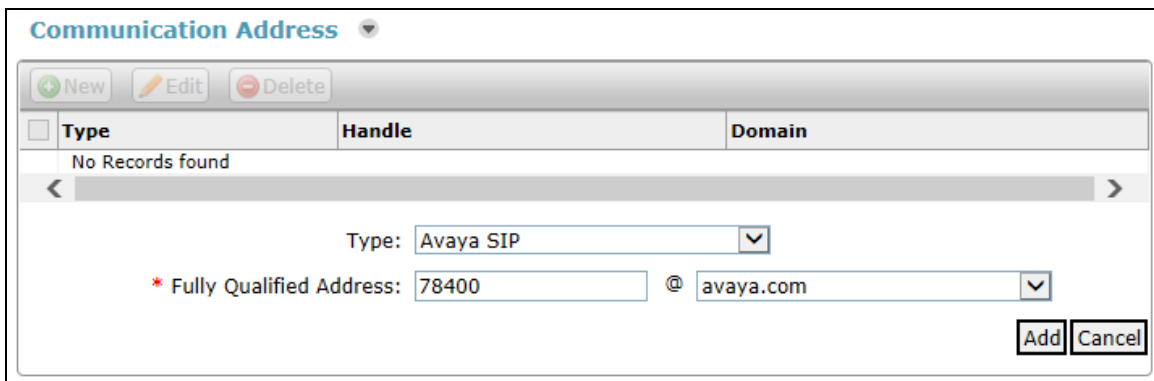
Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration.



The screenshot shows the Avaya System Manager 7.0 interface. The main content area is titled "New User Profile" and has tabs for "Identity", "Communication Profile", "Membership", and "Contacts". The "Communication Profile" tab is active, showing two password fields: "Communication Profile Password" and "Confirm Password", both with masked characters (*****). Buttons for "Commit & Continue", "Commit", and "Cancel" are visible at the top right of the form area.

6.2.3. Communication Address

In the **Communication Address** sub-section, click **New** to add a new entry. The **Communication Address** sub-section is updated with additional fields as shown below. For **Type**, retain "Avaya SIP". For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.2.1**. Click **Add**.



The screenshot shows the "Communication Address" sub-section. At the top, there are "New", "Edit", and "Delete" buttons. Below is a table with columns "Type", "Handle", and "Domain". The table is empty, with the text "No Records found" displayed. Below the table, there are input fields: "Type" is set to "Avaya SIP", and "Fully Qualified Address" is set to "78400" with a domain dropdown set to "avaya.com". "Add" and "Cancel" buttons are at the bottom right.

Type	Handle	Domain
No Records found		

Type: Avaya SIP
* Fully Qualified Address: 78400 @ avaya.com

6.2.4. Session Manager Profile

Scroll down to check and expand **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.

Session Manager Profile ▾

SIP Registration

* Primary Session Manager

Primary	Secondary	Maximum
5	0	5
< [Progress Bar] >		

Secondary Session Manager

Primary	Secondary	Maximum
< [Progress Bar] >		

Survivability Server

Max. Simultaneous Devices ▾

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence ▾

Termination Sequence ▾

Call Routing Settings

* Home Location ▾

Conference Factory Set ▾

Call History Settings

Enable Centralized Call History?

6.2.5. CM Endpoint Profile

Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.2.1**. For **Template**, select *9630SIP_DEFAULT_CM_7_0*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click **Commit** to save the configuration (not shown).

CM Endpoint Profile ▾

* System ▾

* Profile Type ▾

Use Existing Endpoints

* Extension **Endpoint Editor**

* Template ▾

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle ▾

Calculate Route Pattern

Sip Trunk

Enhanced Callr-Info display for 1-line phones

Delete Endpoint on Unassign of Endpoint from User or on Delete User

Override Endpoint Name and Localized Name

Allow H.323 and SIP Endpoint Dual Registration

7. Configure Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations

This section covers the configuration of the Talkphone VOIP-500 Series and VOIP-600 Series IP Call Stations. 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

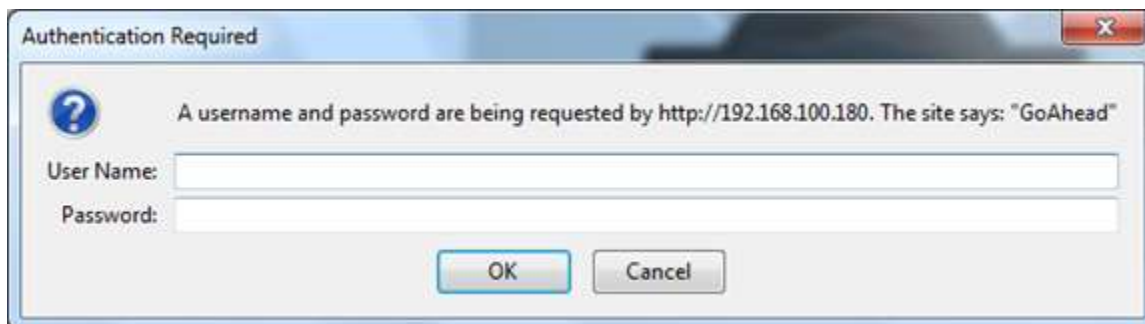
For more information on configuring other features of the Talkphone IP Call Stations, refer to [3, 4].

7.1. Launching the Web Administration Interface

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

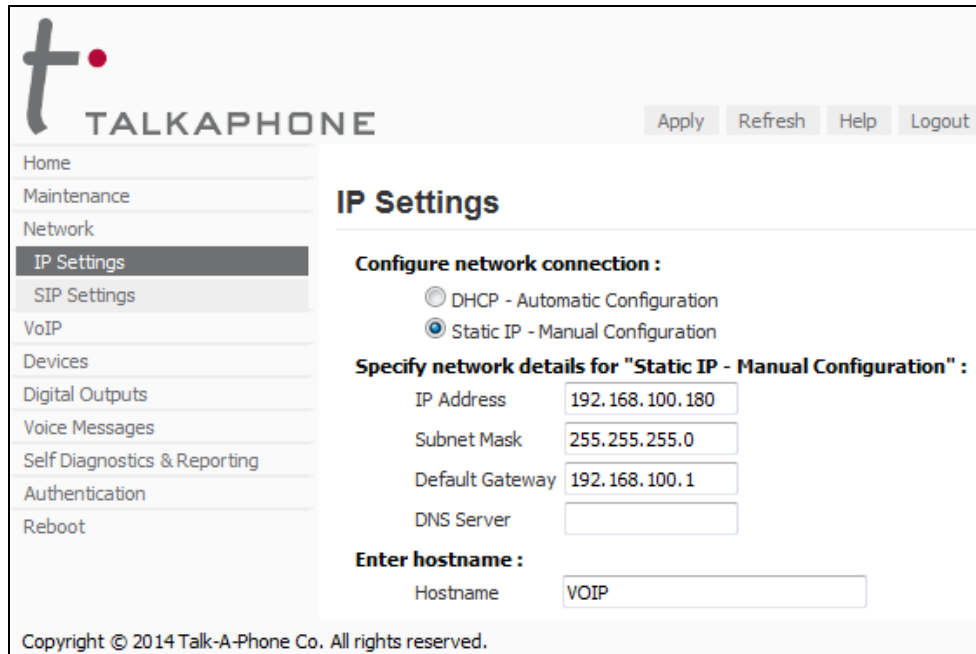
- **IP Address:** 192.168.100.180
- **Username:** admin
- **Password:** admin@123

Ensure that the administration PC and Talkphone IP Call Station are connected to the LAN. Open a web browser and enter the 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 **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 for IP Settings. The top left features the Talkphone logo and a navigation menu with options: Home, Maintenance, Network (selected), IP Settings (highlighted), SIP Settings, VoIP, Devices, Digital Outputs, Voice Messages, Self Diagnostics & Reporting, Authentication, and Reboot. The top right has buttons for Apply, Refresh, Help, and Logout. The main content area is titled 'IP Settings' and contains the following configuration options:

- Configure network connection :**
 - DHCP - Automatic Configuration
 - Static IP - Manual Configuration
- Specify network details for "Static IP - Manual Configuration" :**
 - IP Address: 192.168.100.180
 - Subnet Mask: 255.255.255.0
 - Default Gateway: 192.168.100.1
 - DNS Server: (empty field)
- Enter hostname :**
 - Hostname: VOIP

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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., 78400) configured in **Section 6.2.**

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., 78400).
- **Password:** Specify the SIP password configured in **Section 6.2.2**.
- **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 shows the 'SIP Settings' page in the Talkphone web interface. The page has a navigation menu on the left with options like Home, Maintenance, Network, IP Settings, SIP Settings (highlighted), VoIP, Devices, Digital Outputs, Voice Messages, Self Diagnostics & Reporting, Authentication, and Reboot. The main content area is titled 'SIP Settings' and contains several sections:

- Assign a phone number :** Phone Number: 78400
- 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: 78400; 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: 78400; 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 :** Unregistered

At the top right of the page are buttons for 'Apply', 'Refresh', 'Help', and 'Logout'. The footer of the page reads 'Copyright © 2014 Talk-A-Phone Co. All rights reserved.'

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 TALKAPHONE web interface. On the left is a navigation menu with items: Home, Maintenance, Network, VoIP (expanded), 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 contains several sections:

- Select VoIP codec :** Radio buttons for G.711 PCM a-Law @ 64kbps, G.711 PCM u-Law @ 64kbps (selected), G.729a, and G.723.1a.
- Enable/disable audio processing modules :** Checkboxes for VAD/CNG (checked), AEC (checked), and AGC (unchecked). A Jitter Buffer dropdown is set to 30 ms.
- DTMF duration for outgoing calls :** Radio buttons for Disable, 51 ms, 60 ms, 102 ms, and Custom (selected). A Duration input field is set to 800 (Range: 10-1000 ms).
- Configure Line Level Output parameters :** Line Gain dropdown is set to 16.
- Configure Speaker/Microphone parameters :** Checkboxes for Speaker (checked) and Microphone (checked). Speaker Gain dropdown is set to 12, and Microphone Gain dropdown is set to 12.

At the bottom of the page, there are buttons for "Apply", "Refresh", "Help", and "Logout". A copyright notice at the bottom reads: "Copyright © 2014 Talk-A-Phone Co. All rights reserved."

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.

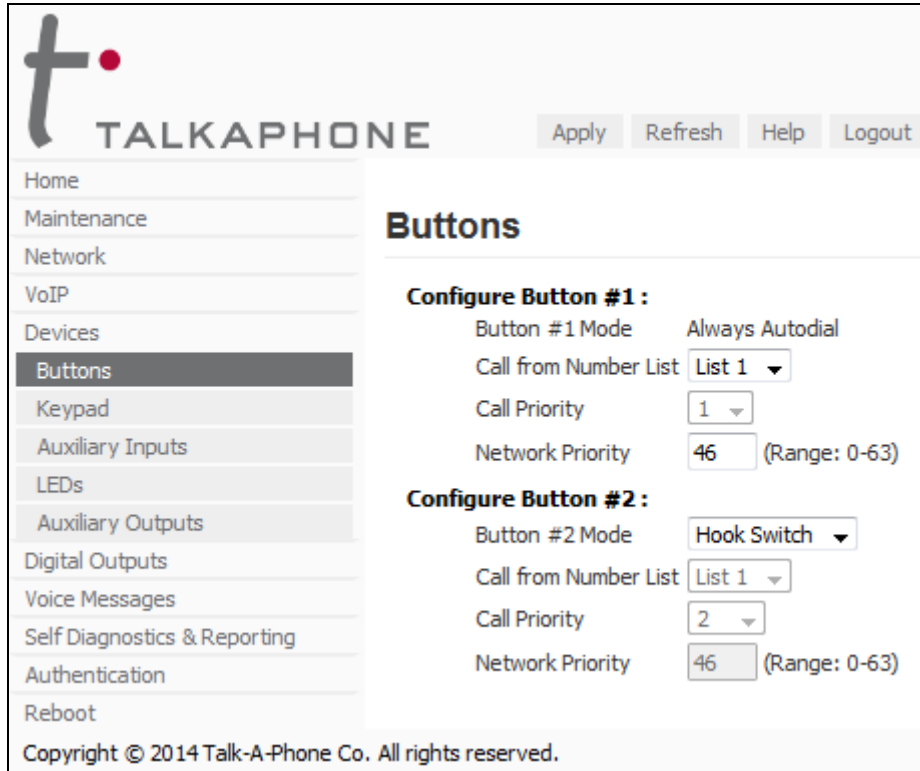
The screenshot displays the 'Call Parameters' configuration page in the Talkphone web interface. The page features a navigation menu on the left with options like Home, Maintenance, Network, VoIP, and Call Parameters (which is currently selected). The main content area is titled 'Call Parameters' and includes several sections for configuring call settings:

- Enable/disable call progress tones :** Radio buttons for 'Enable' (selected) and 'Disable'.
- Specify key to answer and/or disconnect a call from the Remote Side :** Two dropdown menus. The first is labeled 'To disconnect a call, press' with a '# key' dropdown. The second is labeled 'To answer a call, press' with a 'Disable' dropdown.
- Enable/disable "Welcome Tone" :** Radio buttons for 'Enable' (selected) and 'Disable'.
- Configure required timers :** A list of five timers, each with a text input field and a range in parentheses:
 - Provisional Timer: 5 (Range: 5-20 seconds)
 - Ringer Timer: 5 (Range: 1-12 rings)
 - Hang-up Timer: 0.5 (Range: 0.5-3.0 seconds)
 - Local Interdigit Timer: 5 (Range: 5-20 seconds)
 - Remote Interdigit Timer: 5 (Range: 5-20 seconds)
- Configure optional timers :** A list of three optional timers, each with a checked checkbox, a text input field, and a range in parentheses:
 - Call conversation Timer: 12 (Range: 1-360 min.)
 - Ringback or Busy Timer: 15 (Range: 1-60 seconds)
 - Hang-up On Silence Timer: 30 (Range: 10-360 seconds)

At the bottom of the page, there is a copyright notice: 'Copyright © 2014 Talk-A-Phone Co. All rights reserved.'

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 Talkphone web interface. At the top left is the logo 't TALKAPHONE'. To the right are buttons for 'Apply', 'Refresh', 'Help', and 'Logout'. A left-hand navigation menu 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 number list, call priority, and network priority.

Section	Setting	Value
Configure Button #1 :	Button #1 Mode	Always Autodial
	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)

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8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Talkphone VOIP-500 Series and 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.

The screenshot displays the 'SIP Settings' configuration page for a Talkphone device. The interface includes a navigation menu on the left with options like Home, Maintenance, Network, IP Settings, SIP Settings (highlighted), VoIP, Devices, Digital Outputs, Voice Messages, Self Diagnostics & Reporting, Authentication, and Reboot. The main content area is titled 'SIP Settings' and contains several sections:

- Assign a phone number :** Phone Number: 78400
- 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: 78400; 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: 78400; 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 78400@10.64.102.117

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2. Verify 2-way audio and proper call termination.

9. Conclusion

These Application Notes have described the administration steps required to integrate the Talkphone VOIP-500 Series and 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 and Talkphone documentation relevant to these Application Notes. The following Avaya product documentation is available at support.avaya.com.

- [1] *Administering Avaya Aura® Communication Manager*, Release 7.0, Issue 1, August 2015, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, Release 7.0, Issue 1, August 2015.

The following Talkphone documentation may be found at www.talkphone.com.

- [3] *Talkphone VOIP-500 Series Phone Configuration and Operation Manual v3.0.2*, Rev 7/31/2012.
- [4] *Talkphone VOIP-600 Series Configuration and Operation Manual v1.0.1*, Rev 9/17/2014.

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