Avaya Solution & Interoperability Test Lab

Application Notes for Talkaphone 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 Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Talkaphone 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 Talkaphone 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 Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Talkaphone 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 Talkaphone 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 Talkaphone 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 Talkaphone 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 Talkaphone IP Call Station with Session Manager.
- Inbound and outbound calls between Talkaphone IP Call Station and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Inbound and outbound calls between the Talkaphone 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 Talkaphone IP Call Station.
- Proper system recovery after a restart of the Talkaphone 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 Talkaphone 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 Talkaphone 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 Talkaphone IP Call Stations.

2.3. Support

For technical support and information on Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations, contact Talkaphone support at:

- Phone: 1-773-539-1100
- Website: http://www.talkaphone.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.

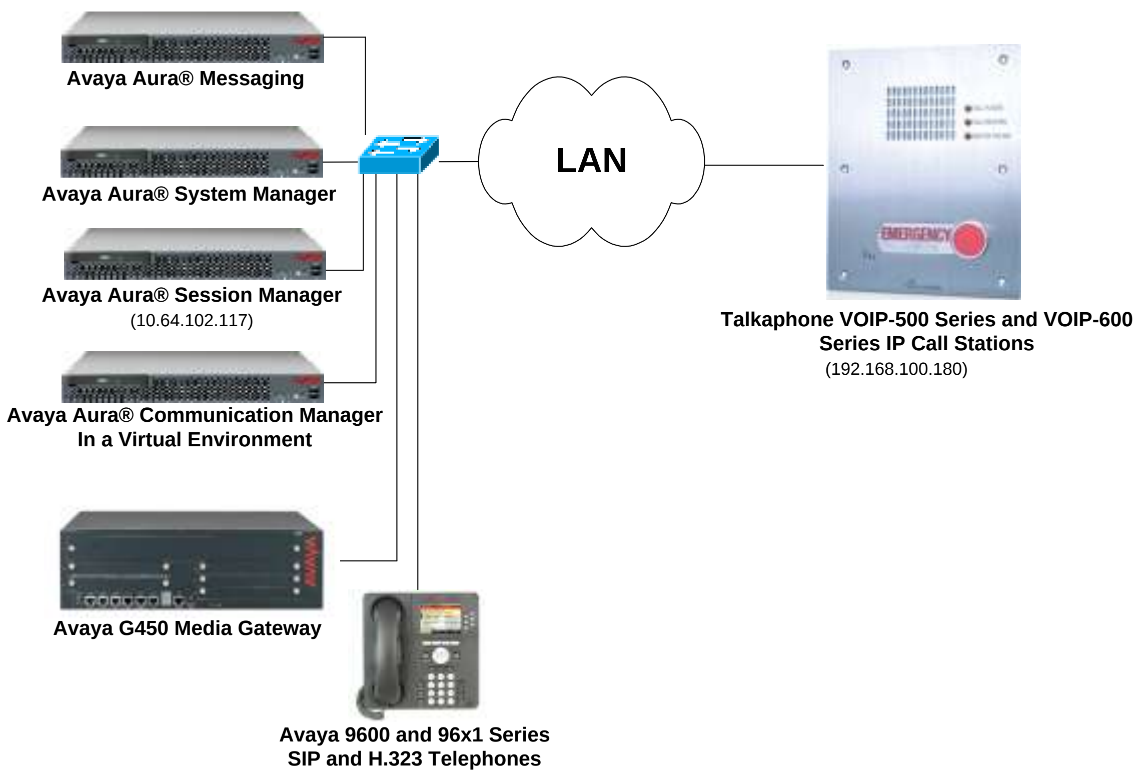


Figure 1: Avaya SIP Network with Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations

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)
Talkaphone VOIP-500 Series IP Call Stations	1.0.2.4
Talkaphone 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 Talkaphone 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 Talkaphone 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

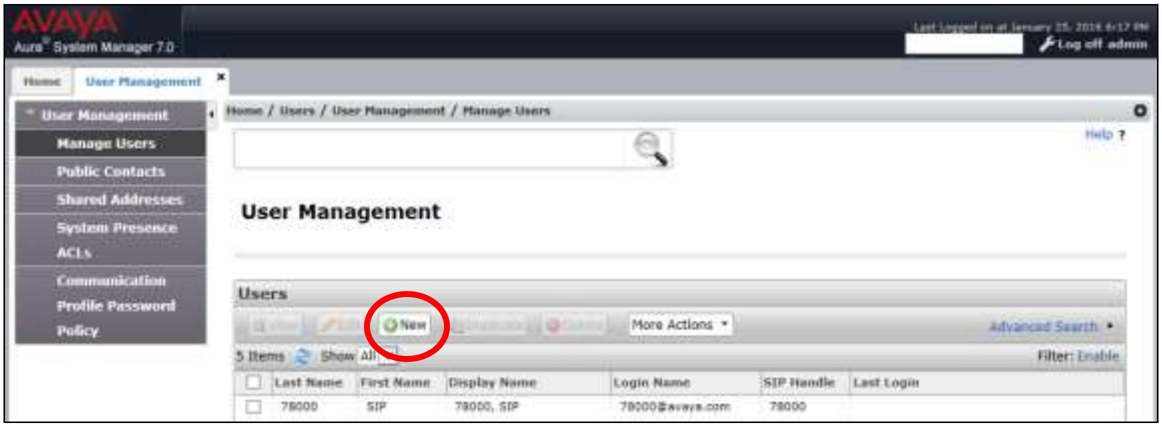
Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for the Talkaphone IP Call Station.

6.1. Launch System Manager

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

6.2. 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.

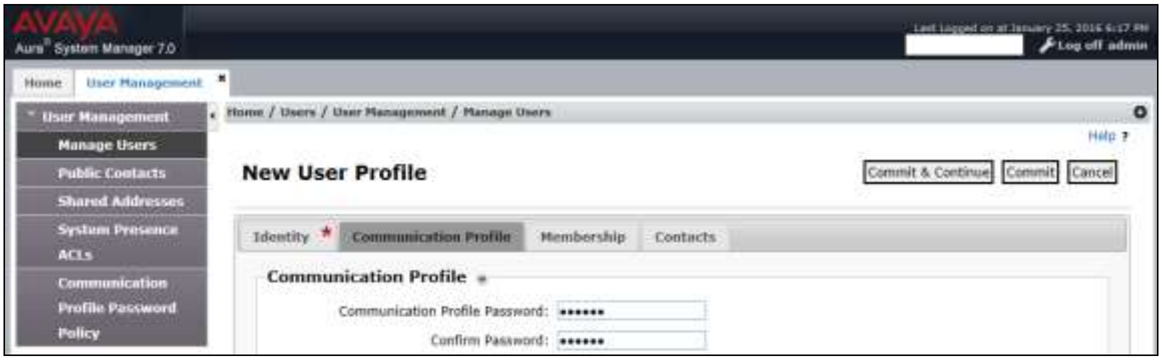


6.2.1. Identity

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 Talkaphone 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.

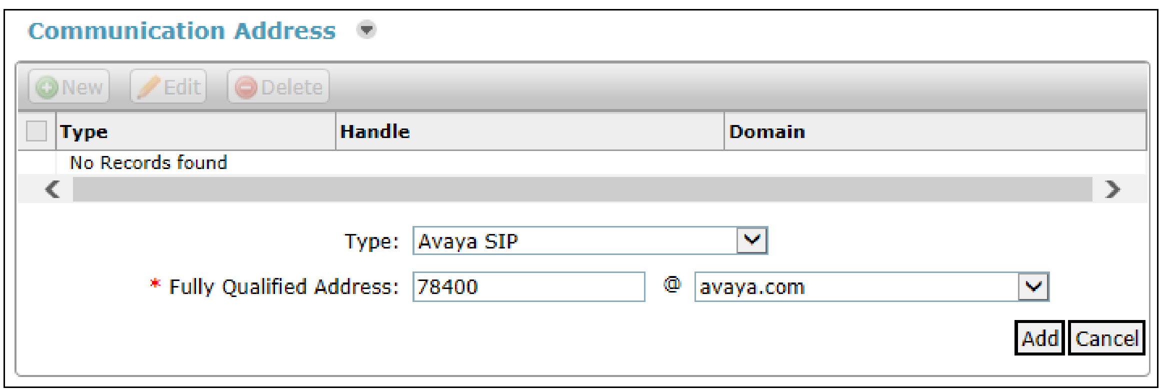
6.2.2. Communication Profile

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



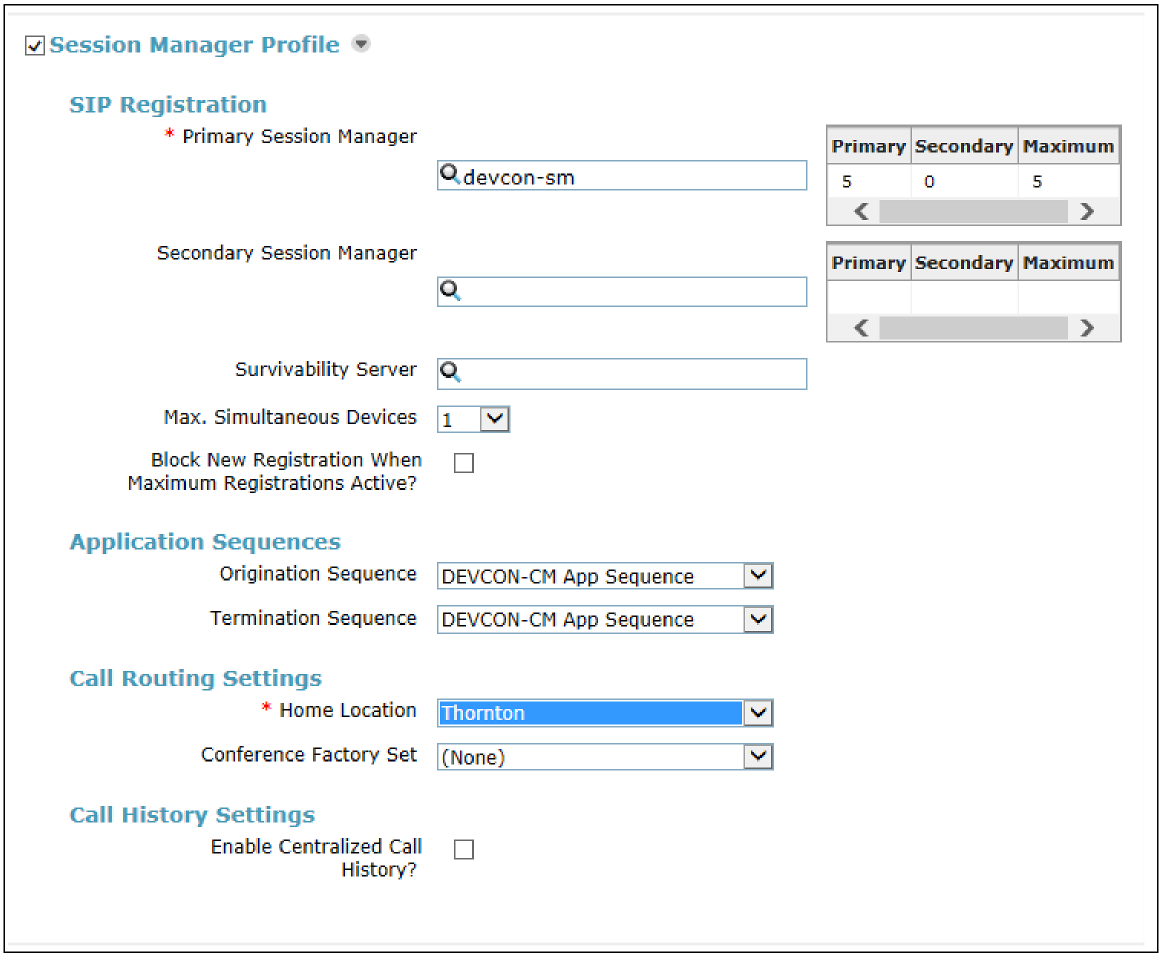
6.2.3. Communication Address

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**.



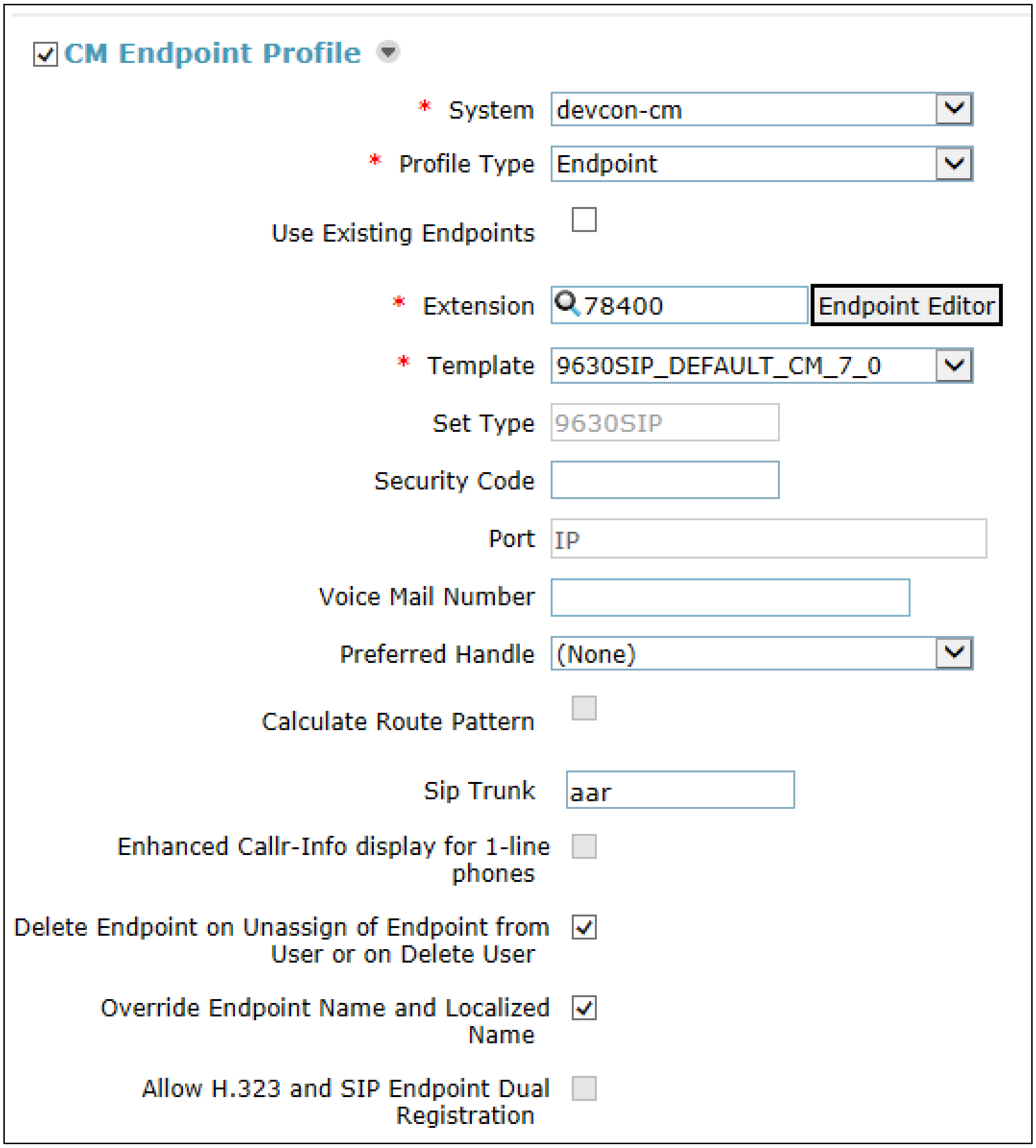
6.2.4. Session Manager Profile

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.



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).



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

This section covers the configuration of the Talkaphone 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

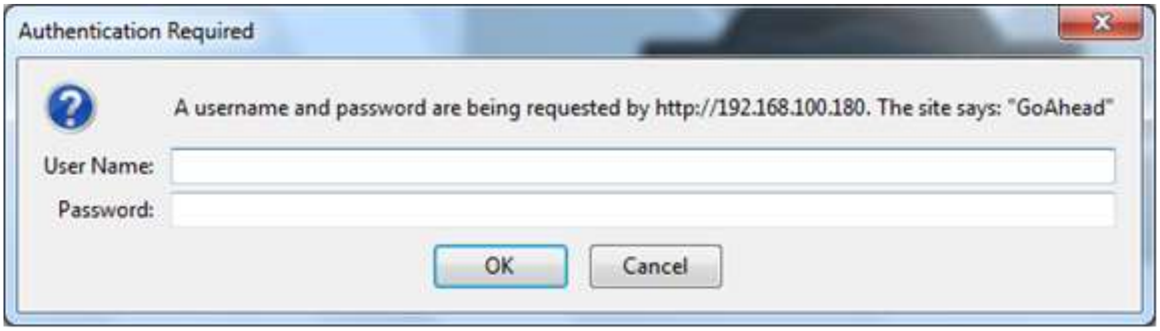
For more information on configuring other features of the Talkaphone IP Call Stations, refer to [**3, 4**].

7.1. Launching the Web Administration Interface

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

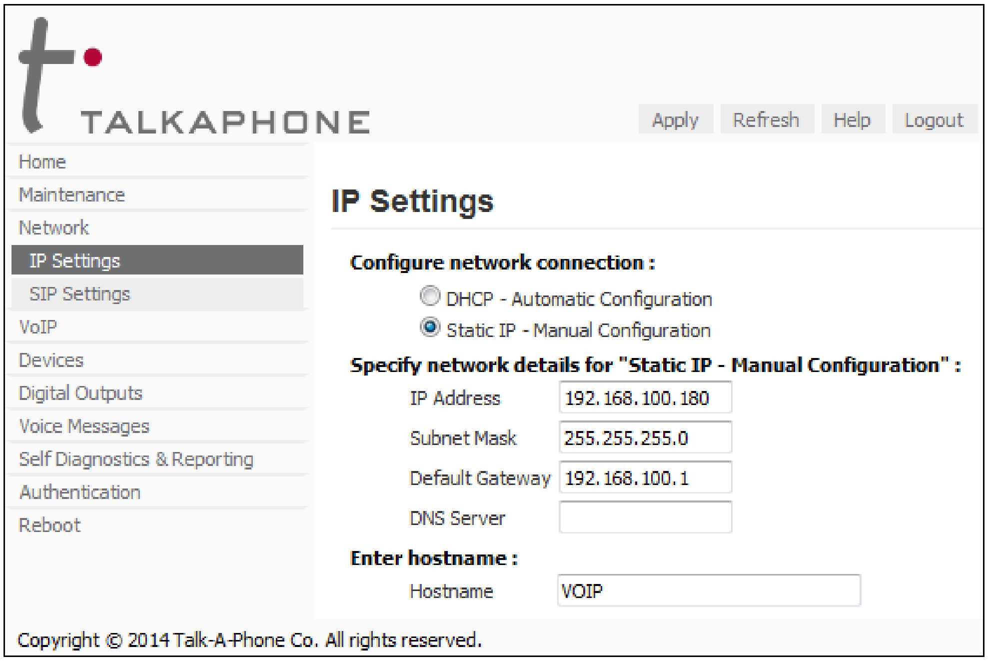
- **IP Address:** 192.168.100.180
- **Username:** admin
- **Password:** admin@123

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



7.2. Network Configuration

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



7.3. SIP Configuration

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

Under **Assign a phone number**:

- **Phone Number:** Specify the SIP number (e.g., *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.

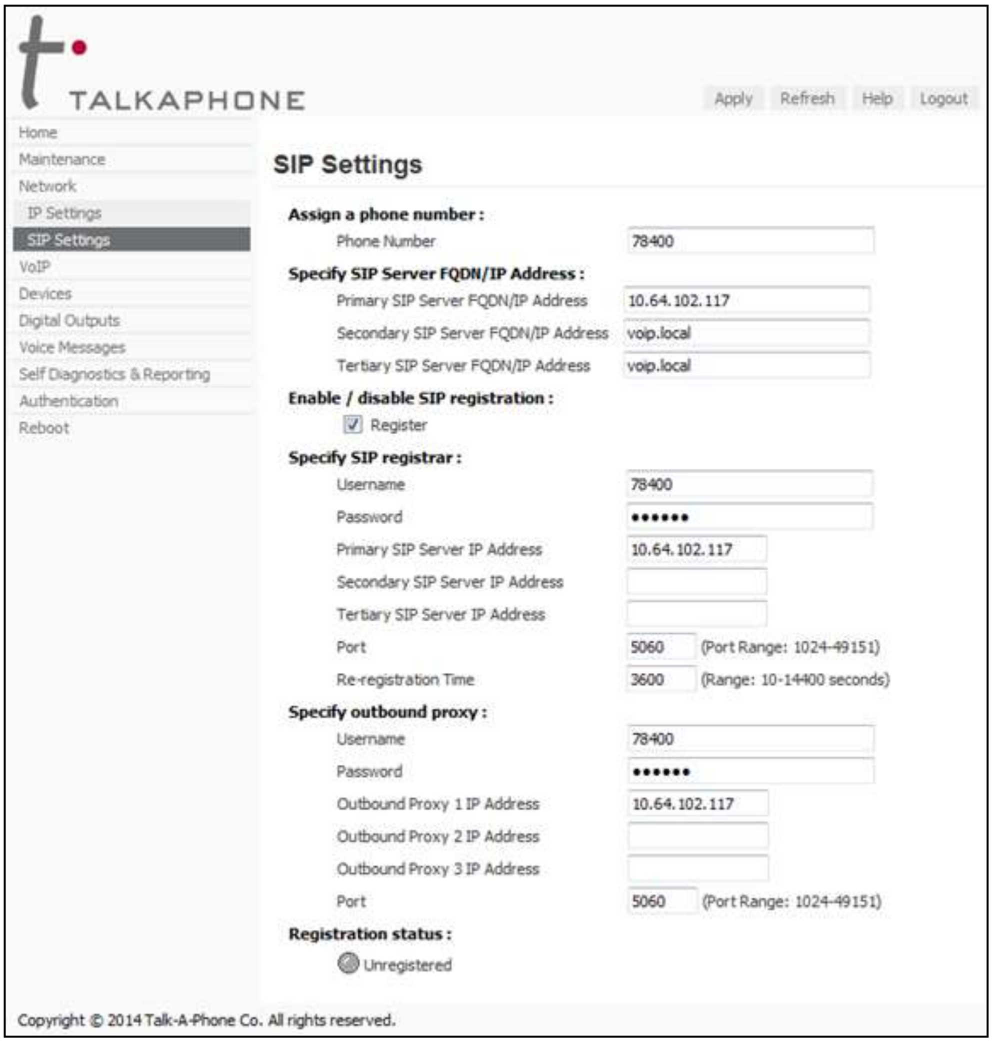
Under **Enable / disable SIP registration**:

- **Register:** Select the checkbox.

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

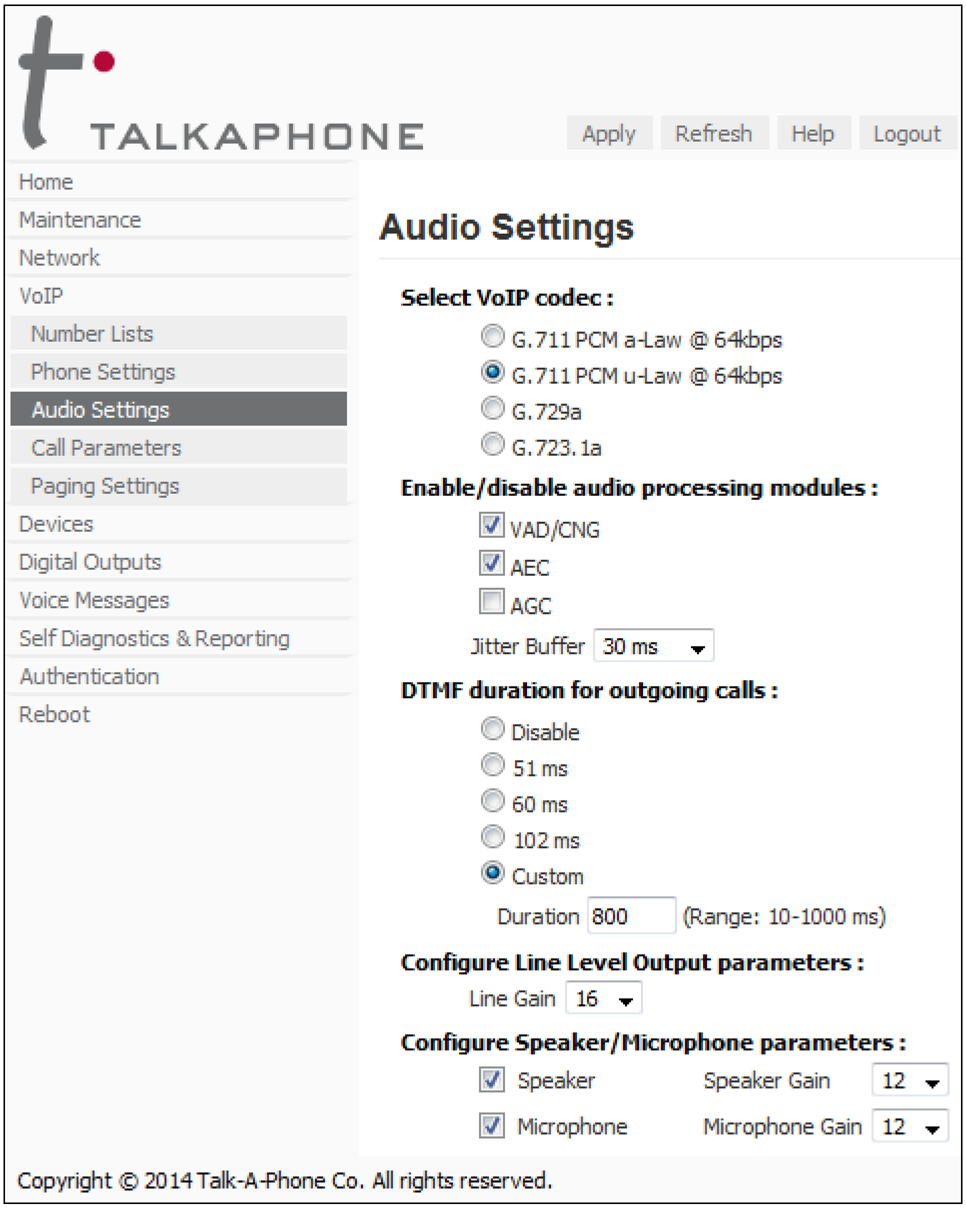
- **Username:** Specify the SIP number of the Talkaphone IP Call Station (e.g., *78400*).
- **Password:** Specify the SIP password configured in **Section 6.2.2**.
- **Primary SIP Server IP Address:** Specify the IP address of the Session Manager signaling interface (e.g., *10.64.102.117*).
- **Port:** Specify the SIP port (e.g., *5060*).

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



7.4. Configure Audio Settings

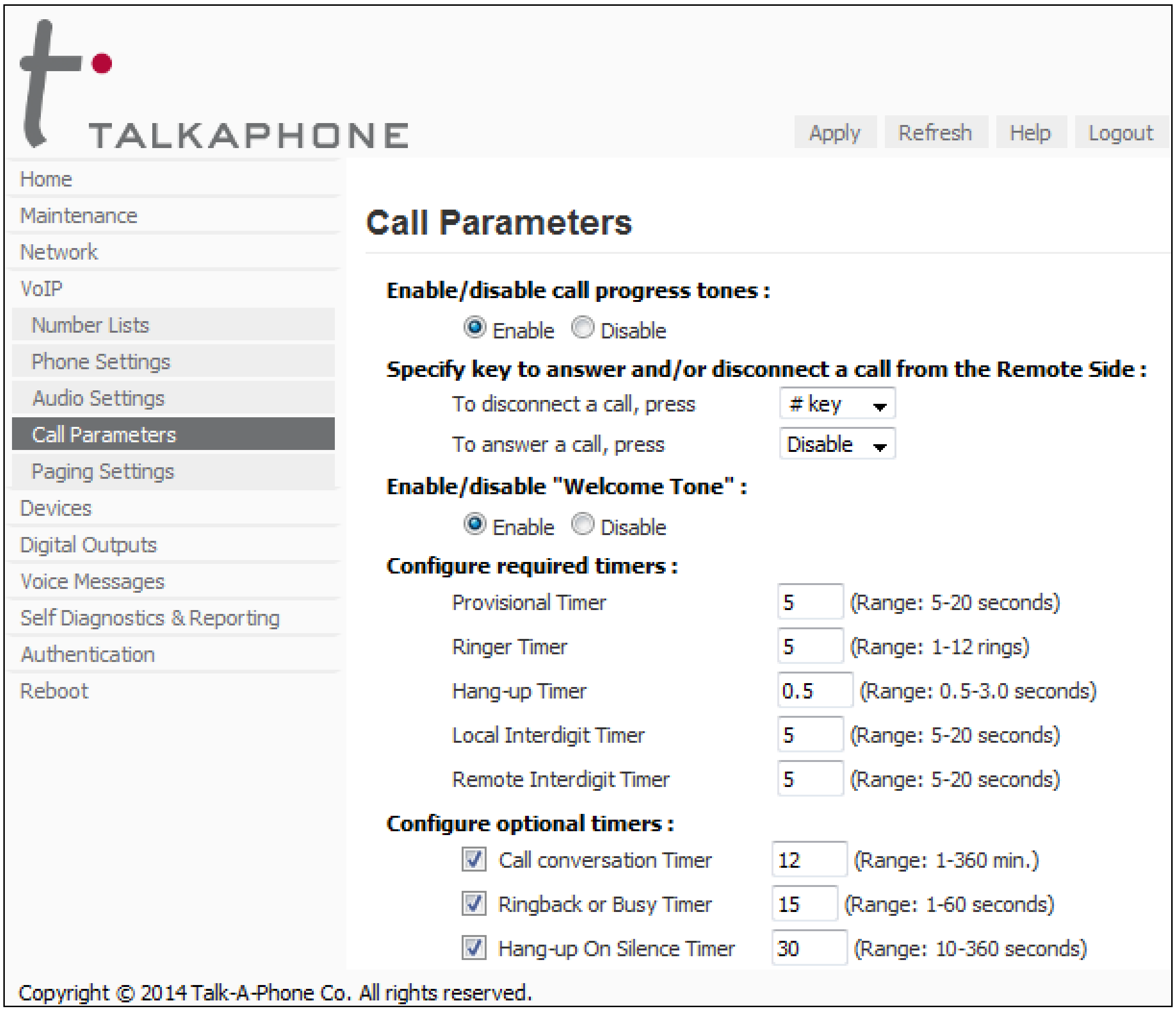
Navigate to **VoIP → Audio Settings** to configure the preferred codec, outbound DTMF duration, and microphone and speaker parameters. For the compliance test, the **DTMF duration for outgoing calls** was set to **Custom** with **Duration** of *800 ms*. This is required so that a user can navigate through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



7.5. Configure Call Parameters

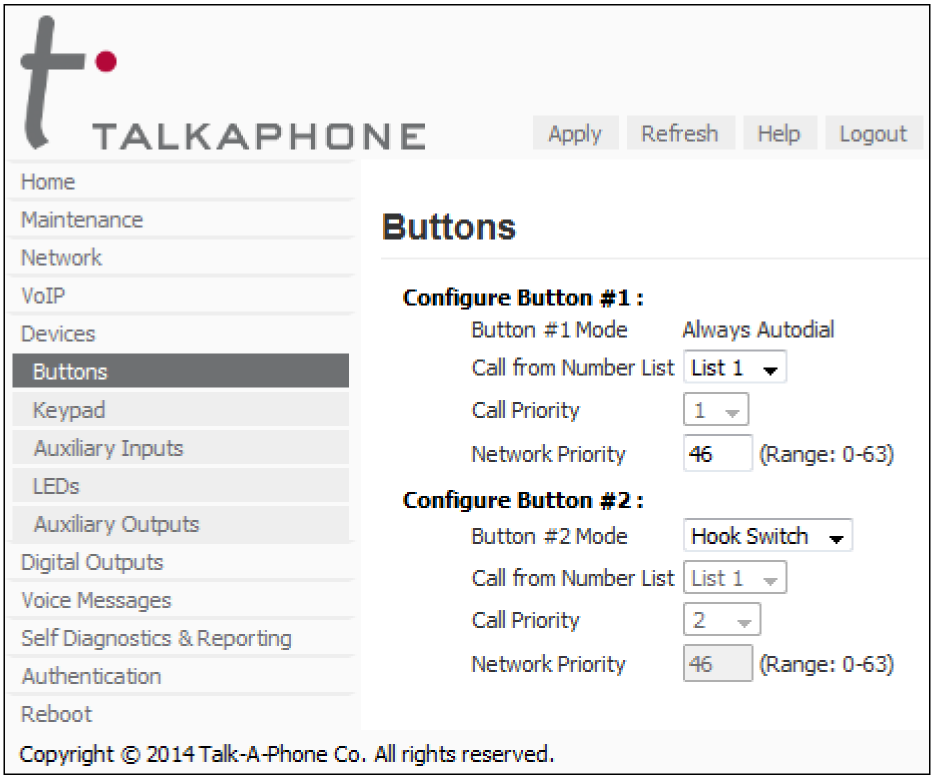
Navigate to **VoIP → Call Parameters** to view and customize any of the call parameters, such as **Local Interdigit Timer**, which dictates how long to wait before initiating a call after the user dials the digits, or the **Call conversation Timer**, which specifies how long an emergency call should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

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



7.6. Configure Buttons

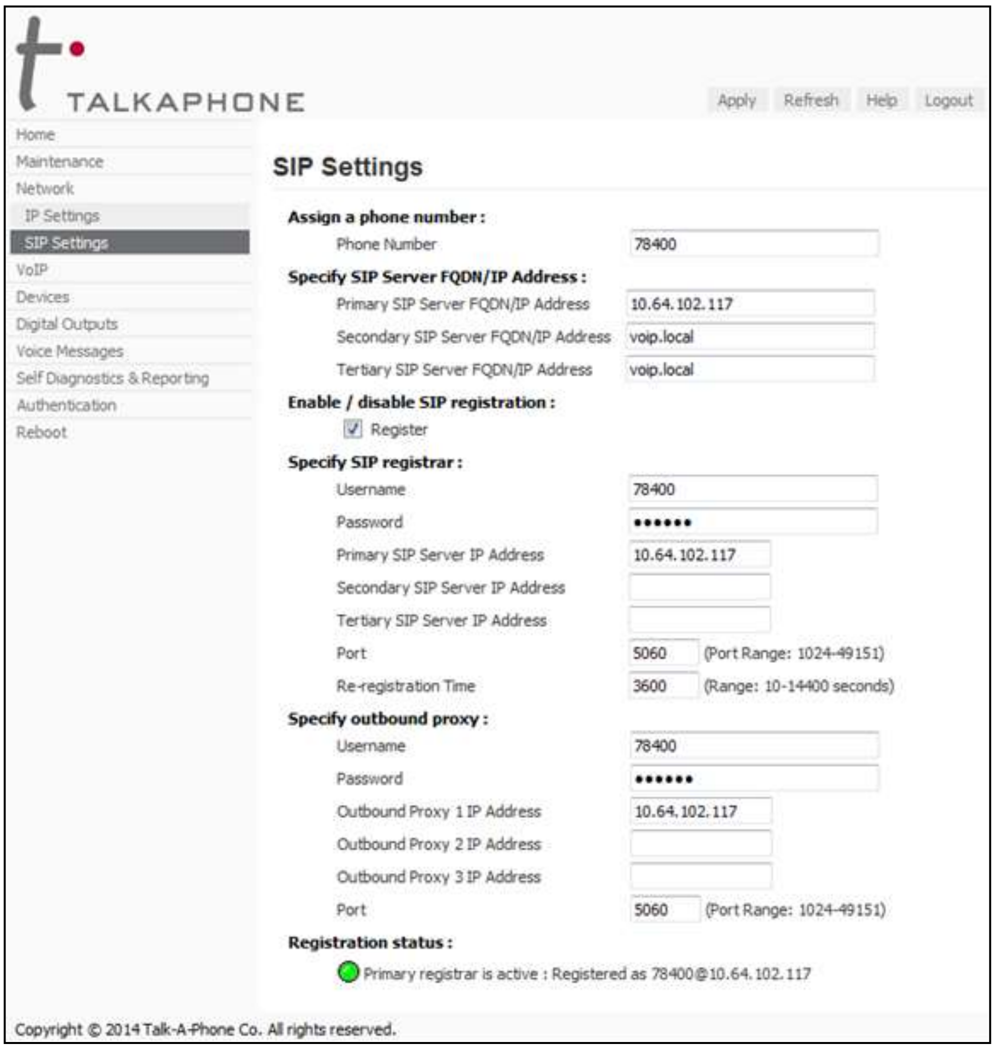
Navigate to **Devices → Buttons** to verify the appropriate settings. For the compliance test, the **Buttons** were configured as shown below.



8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the Talkaphone IP Call Station has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status. Alternatively, the SIP Settings screen on the Talkaphone IP Call Station also shows the registration status as shown below.



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

9. Conclusion

These Application Notes have described the administration steps required to integrate the Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Talkaphone 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 Talksphone 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 Talkaphone documentation may be found at www.talkaphone.com.

[3] *Talkaphone VOIP-500 Series Phone Configuration and Operation Manual v3.0.2*, Rev 7/31/2012.

[4] *Talkaphone VOIP-600 Series Configuration and Operation Manual v1.0.1*, Rev 9/17/2014.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.