



Avaya Solution & Interoperability Test Lab

Application Notes for Teo 7810 and 7810 TSG-6 Series IP Phones 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 Teo 7810 and 7810 TSG-6 Series IP Phones with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The Teo IP Phones registered with Avaya Aura® Session Manager via SIP.

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 Teo 7810 and 7810 TSG-6 Series IP Phones with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The Teo IP Phones register with Avaya Aura® Session Manager via SIP.

Both the Teo 7810 and the 7810 TSG-6 Series IP Phones are executive level SIP deskphones designed for government, military and commercial users. There are three models currently available in the Teo 7810 TSG-6 Series: 7810-TSG, 7810PoE-TSGA and 7810PoE-TSGB. During the compliance test, 7810-TSG and 7810PoE-TSGB sets were used. Since all models share the same core SIP firmware, it is expected that the results of the compliance test extends to all the models in the IP phone series.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Teo 7810 Series IP Phones, Avaya SIP and H.323 telephones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer and conference. Additional telephony features, such as call forward, call coverage, call park/unpark, and call pickup were also verified using Communication Manager Feature Access Codes (FACs) and Feature Name Extensions (FNEs).

The serviceability testing focused on verifying that the Teo 7810 Series IP Phones come back into service after re-connecting the Ethernet cable or rebooting the phone.

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's formal testing and Declaration of Conformity is provided only on the headsets/handsets that carry the Avaya brand or logo. Avaya may conduct testing of non-Avaya headset/handset to determine interoperability with Avaya phones. However, Avaya does not conduct the testing of non-Avaya headsets/handsets for: Acoustic Pressure, Safety, Hearing Aid Compliance, EMC regulations, or any other tests to ensure conformity with safety, audio quality, long-term reliability or any regulation requirements. As a result, Avaya makes no representations whether a particular non-Avaya headset will work with Avaya's telephones or with a different generation of the same Avaya telephone.

Since there is no industry standard for handset interfaces, different manufacturers utilize different handset/headset interfaces with their telephones. Therefore, any claim made by a headset vendor that its product is compatible with Avaya telephones does not equate to a guarantee that the headset will provide adequate safety protection or audio quality.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Teo 7810 Series IP Phones with Session Manager.
- Calls between Teo IP phones and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Teo IP phones and the PSTN.
- G.711, G.729 and G.722 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, multiple calls, and 3-party conference.
- Extended telephony features using Communication Manager FACs and FNEs for Call Forward, Call Park/Unpark, and Call Pickup.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve voice messages.
- Bridged Line Appearances on Teo IP phones for other Avaya and Teo SIP phones.
- Use of programmable buttons on the Teo IP phones.
- Proper system recovery after a restart of the Teo IP phones and loss of IP connectivity.

2.2. Test Results

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

- Teo IP phones do not support blind conference. Attended conference is supported. When a Teo IP phone initiates a 3-way conference, two call appearances are required.
- Teo TSG-6 Series IP Phones support their own implementation of bridge line appearances. They do not support bridge line appearances using Avaya AST. Therefore, only basic bridged scenarios covered during the compliance test are supported. Bridge line appearances for Avaya non-SIP Deskphones are not supported.
- Ensure that the Initial IP-IP Direct Media option on the Communication Manager signaling group form is disabled so that all bridged line appearance scenarios work properly.
- In order for Teo TSG-6 Series IP Phones to bridge onto a call, the Avaya or Teo SIP phone assigned to the bridge line appearance must first place the call on hold.

2.3. Support

For technical support and information on Teo 7810 and 7810 TSG-6 Series IP Phones, contact Teo customer support at:

- Phone: 1-800-524-0024
- Website: <http://www.teotech.com/teosupport/support.php>
- Email: helpdesk@teotech.com

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 virtual environment with an Avaya G450 Media Gateway.
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server (not shown in figure).
- 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 H.323 and SIP Deskphones.
- Teo 7810 and 7810 TSG-6 Series IP Phones.

Teo 7810 and 7810 TSG-6 Series IP Phones registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

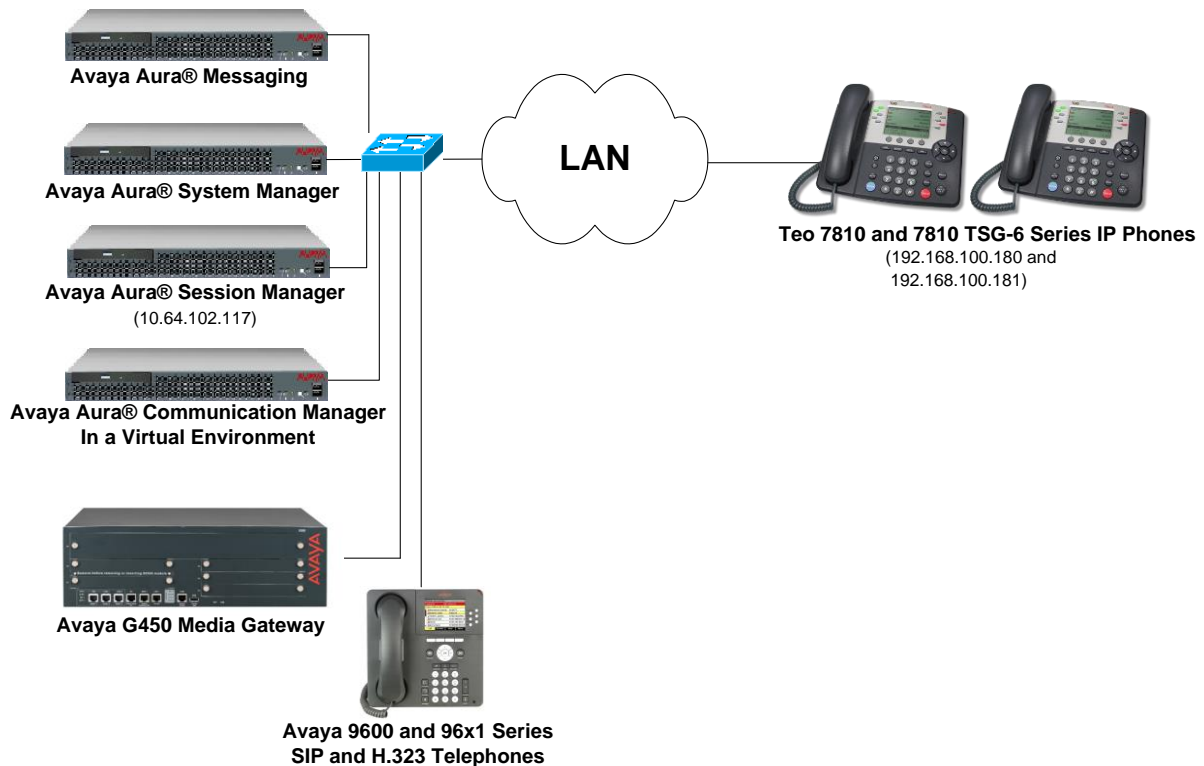


Figure 1: Avaya SIP Network with Teo 7810 and 7810 TSG-6 Series IP Phones

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 with an Avaya G450 Media Gateway	7.0.1.1 FP1 SP1 (R017x.00.0.441.0 with Patch 23169)
Avaya Aura® Media Server	7.7.0.226
Avaya Aura® Session Manager	7.0.1.1 (7.0.1.1.701114)
Avaya Aura® System Manager	7.0.1.1 (Build No. 7.0.0.016266 Software Update Revision No: 7.0.1.1.065378 Service Pack 1)
Avaya Aura® Messaging	6.3.2 SP 2 Patch 3
Avaya 9600 Series IP Deskphones	3.260A (H.323) 2.6.16.1 (SIP)
Avaya 96x1 Series IP Deskphones	6.6115 (H.323) 7.0.1.1.5 (SIP)
Teo 7810 and 7810 TSG-6 Series IP Phones	05.04.23

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
- Enable Bridged Calls (optional)

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 Teo IP Phones 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.

```
display 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 30
                                Maximum Stations: 2400 15
                                Maximum XMOBILE Stations: 2400 0
Maximum Off-PBX Telephones - EC500: 9600 0
Maximum Off-PBX Telephones - OPS: 9600 7
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 or Avaya Aura® Media Server. 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 Teo IP phones were tested using G.711, G.729, and G.722 codecs.

```

change ip-codec-set 1                                       Page 1 of 2
                                                           IP CODEC SET
  Codec Set: 1
  

| Audio Codec | Silence Suppression | Frames Per Pkt | Packet Size (ms) |
|-------------|---------------------|----------------|------------------|
| 1: G.711MU  | n                   | 2              | 20               |
| 2:          |                     |                |                  |
| 3:          |                     |                |                  |
| 4:          |                     |                |                  |
| 5:          |                     |                |                  |
| 6:          |                     |                |                  |
| 7:          |                     |                |                  |


```

5.3. Enable Bridged Calls

In the **Stations with Off-PBX Telephone Integration** form, find the entry for the Teo IP phone, configured via System Manager, and set the **Bridged Calls** field to *both*, if a bridged line appearance will be configured on the phone.

change off-pbx-telephone station-mapping 78010					Page	2 of	3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION							
Station	Appl	Call	Mapping	Calls	Bridged	Location	
Extension	Name	Limit	Mode	Allowed	Calls		
78010	OPS	4	both	all	both		

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 Teo IP phone.

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:

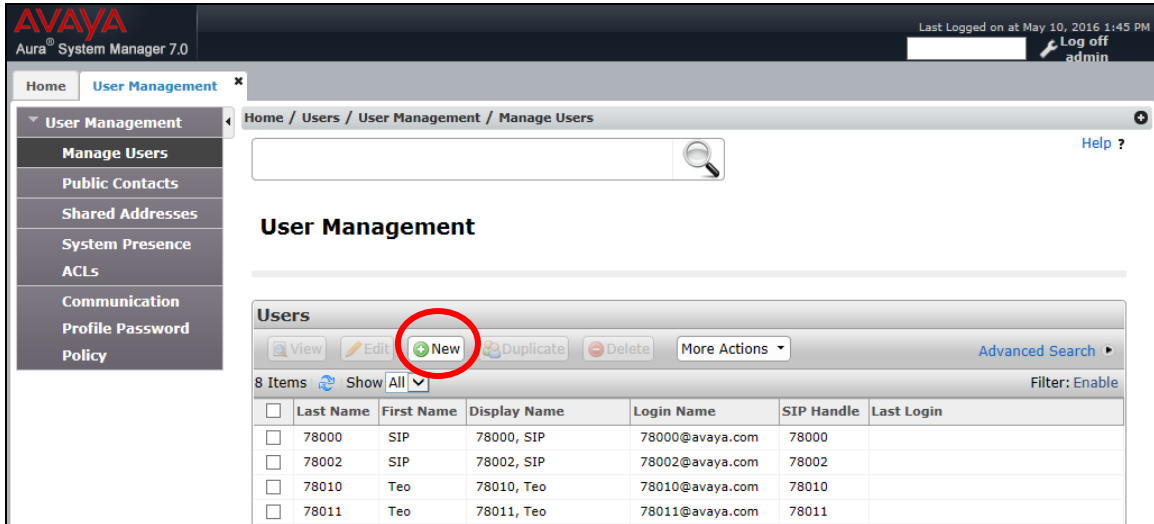
Password:

[Change Password](#)

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

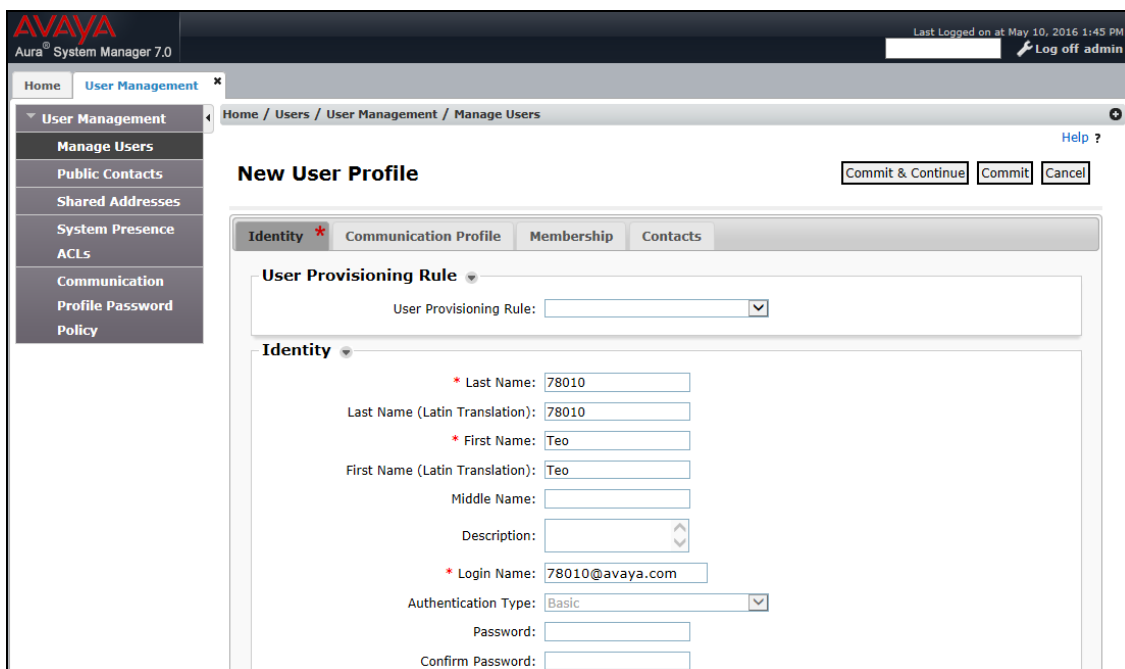
6.2. Administer SIP User

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



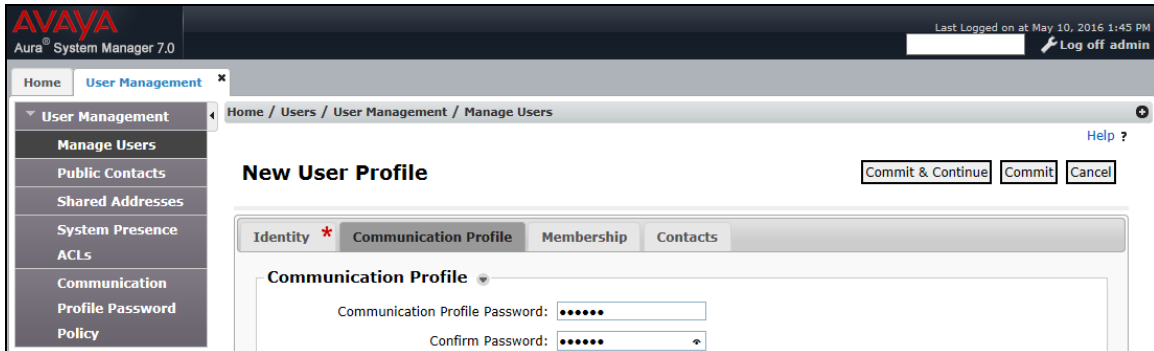
6.2.1. Identity

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 Teo IP phone 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

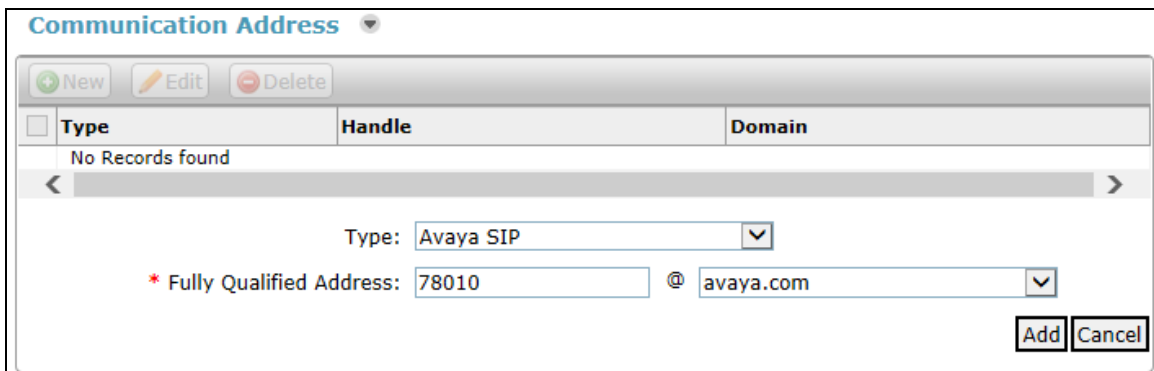
Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration.



The screenshot shows the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 7.0', and a user session summary: 'Last Logged on at May 10, 2016 1:45 PM' and a 'Log off admin' button. The left sidebar contains a 'User Management' menu with options: 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence', 'ACLs', 'Communication', 'Profile Password', and 'Policy'. 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. Action buttons '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 currently empty, displaying 'No Records found'. Below the table, there are input fields for 'Type' (set to 'Avaya SIP'), and '* Fully Qualified Address' (with '78010' in the extension field and 'avaya.com' in the domain field). 'Add' and 'Cancel' buttons are at the bottom right.

Type	Handle	Domain
No Records found		

Type: Avaya SIP

* Fully Qualified Address: 78010 @ avaya.com

6.2.4. Session Manager Profile

Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Sequence**, **Termination 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
8	0	8

Secondary Session Manager

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 *9600SIP_DEFAULT_CM_7_0*. For **Port**, ensure that it is automatically set to *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

In the **CM Endpoint Profile** sub-section, click the **Endpoint Editor** button to display the page below. In the **General Options** tab, specify the coverage path that points to the voicemail system in the **Coverage Path 1** field. This provides voicemail coverage for the SIP user. In this example, coverage path 10 was used.

The screenshot shows the 'Edit Endpoint' configuration page in Avaya Aura System Manager 7.0. The 'General Options' tab is selected, displaying various configuration fields. The 'Coverage Path 1' field is set to '10'. Other fields include System (devcon-cm), Extension (78010), Template (9600SIP_DEFAULT_CM_7_0), Set Type (9600SIP), Port (IP), Security Code, and Name (78010, Teo). The 'Class of Restriction (COR)' is set to '1', and 'Class Of Service (COS)' is also set to '1'. The 'Emergency Location Ext' is '78010', and 'Message Lamp Ext.' is '78010'. The 'SIP Trunk' is 'aar', and 'Type of 3PCC Enabled' is 'None'. The 'Localized Display Name' is '78010, Teo'. The 'Enable Reachability for Station Domain Control' checkbox is checked. The 'Done' and 'Cancel' buttons are visible at the bottom right.

Scroll down to the **Features** section and enable **Bridged Call Alerting**.

The screenshot shows the 'Features' section of the configuration page. The 'Bridged Call Alerting' checkbox is checked. Other features listed include 'Always Use', 'Audible Message Waiting', 'Auto Select Any Idle Appearance', 'Bridged Idle Line Preference', 'CDR Privacy', 'Conf/Trans On Primary Appearance', 'Coverage Message Retrieval', 'Customizable Labels', 'Data Restriction', 'Direct IP-IP Audio Connections', 'Display Client Redirection', 'IP Video', 'EMU Login Allowed', 'Bridged Appearance Origination Restriction', 'H.320 Conversion', 'Idle Appearance Preference', 'IP Audio Hairpinning', 'IP SoftPhone', 'LWC Activation', 'LWC Log External Calls', 'Multimedia Early Answer', 'Mute Button Enabled', 'Per Button Ring Control', 'Redirect Notification', 'Restrict Last Appearance', 'Select Last Used Appearance', and 'Survivable Trunk Dest'. The 'Done' and 'Cancel' buttons are visible at the bottom right.

In the **Feature Options** tab, set the **MWI Served User Type** field to *sip-adjunct*. This allows MWI to be enabled for the SIP user. The voicemail system was connected via SIP to Session Manager.

The screenshot shows the 'Edit Endpoint' configuration page in Avaya Aura System Manager 7.0. The 'Feature Options' tab is active, displaying the following settings:

- System: devcon-cm
- Extension: 78010
- Template: 9600SIP_DEFAULT_CM_7_0
- Set Type: 9600SIP
- Port: IP
- Security Code: (empty)
- Name: 78010, Teo

Under the 'Feature Options' tab, the following settings are visible:

- Active Station Ringing: single
- Auto Answer: none
- Coverage After Forwarding: system
- Multimedia Mode: enhanced
- MWI Served User Type: sip-adjunct
- Per Station CPN - Send Calling Number: None

In the **Button Assignment** tab, configure a button as a Bridged Line Appearance (BLA) if desired. Configure the button as *brdg-appr* and specify the call appearance button number and extension of the station for which the BLA is assigned. Click **Done** (not shown) when complete, followed by **Commit** on the previous page.

The screenshot shows the 'Button Assignment' tab in the 'Feature Options' section. The 'Main Buttons' sub-tab is active, displaying a table for configuring buttons:

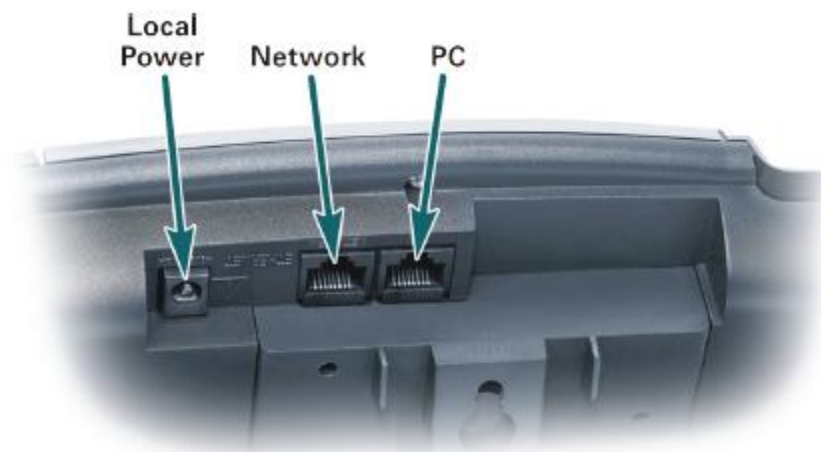
Button Number	Main Buttons	Feature Buttons	Expansion Module	Button	Ext
1	call-appr				
2	call-appr				
3	call-appr				
4	None				
5	None				
6	brdg-appr			1	78011
7	None				
8	None				

7. Configure Teo 7810 and 7810 TSG-6 Series IP Phones

This section covers the SIP configuration of the Teo 7810 and 7810 TSG-6 Series IP Phones. For more information on configuring the Teo IP Phones, including speed dial buttons for voicemail and FACs, refer to [3].

7.1. Power and Network Connection

Connect power to the Teo telephone using the **Local Power** option, or in models where Power over Ethernet (PoE) is supported, connecting a CAT 5 or better cable to the **Network** jack of the telephone. Both the Teo IP Phones 7810 and the 7810PoE-TSGB models used in the compliance test support PoE. During the compliance test, IP addresses were configured manually as described in [3]. Alternatively, DHCP is also an option.



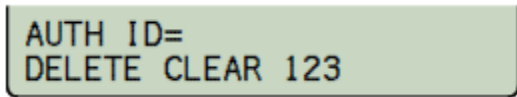
7.2. SIP Configuration

The configuration steps in this section represent the basic steps required to achieve the registration of the Teo IP phones with Session Manager. A detailed description of the different menu options on the telephones is beyond the scope of these Application Notes. For detailed information on the installation and configuration of the Teo IP Phones, see [6] and [7].

Once power is applied and the telephone initializes, the display shows different progress messages and it will then prompt for the **LINE ID**. This is the extension number assigned to the user, previously in **Section 6.2.1**. Extension **78010** was used in the example and it was entered at the prompt below. Press the **OK** on the phone when the entry is completed.

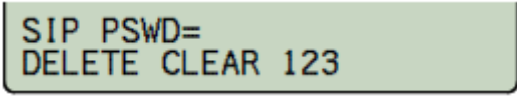
```
LINE ID=  
DELETE CLEAR 123
```

The telephone prompts next for the Authentication ID. Enter the same extension number that was assigned to the **LINE ID**. Press the **OK** key on the phone when the entry is completed.



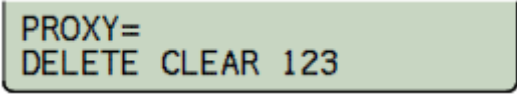
AUTH ID=
DELETED CLEAR 123

Next, the phone will prompt for the SIP Authentication Password. This is the **Communication Profile Password** assigned to the user previously in **Section 6.2.2**. Press the **OK** on the phone when the entry is completed.



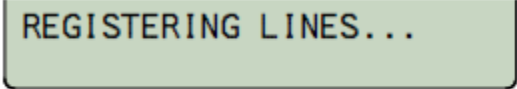
SIP PSWD=
DELETED CLEAR 123

The next prompt will be for the proxy server IP address, if this address has not been previously provided by DHCP. This is the IP address of the Session Manager signaling interface, *10.64.102.117* in the reference configuration. Press the **OK** on the phone when the entry is completed.

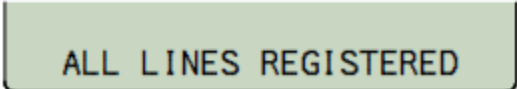


PROXY=
DELETED CLEAR 123

At this point, the telephone will register with Session Manager. Upon successful registration, all affected line key indicators will turn solid green momentarily, and then will go out. The following status messages will be displayed.



REGISTERING LINES...

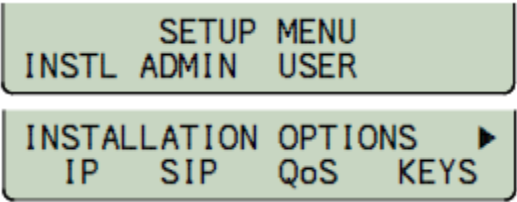


ALL LINES REGISTERED

The idle display will then appear indicating that the phone is ready for use (not shown).

To ensure that MWI is enabled, press **SETUP** button and then navigate to **INSTL → SIP → MWI** and verify that the option is enabled (not shown).

Once the telephone is registered, the SIP Configuration menu can be accessed by pressing the **SETUP** key on the phone and selecting **INSTL → SIP**.



SETUP MENU
INSTL ADMIN USER

INSTALLATION OPTIONS ▶
IP SIP QoS KEYS

Once on the SIP Configuration screen, selecting one of the available submenus allows to make changes or to review the following parameters, if needed:

SIP CONFIGURATION PHONE PROXY REGSTR

- **PHONE** – Domain name, port, and RTP start port for the telephone.
- **PROXY** – SIP Proxy server IP address and port.
- **REGSTR** – Registration enable, SIP Registrar server IP address and port.

7.3. Configure Bridged Line Appearance (BLA)

This section covers the programming of BLAs, if desired. Note that only SIP extensions may be assigned to BLAs. To configure the phone to support BLAs, first program the primary number and the number that is going to appear as a BLA as **SHARED**.

Press the **SETUP** key and select **INSTL** → **KEYS**, registered keys will light green. Select the key that is to be programmed as **SHARED**. The selected key and any other key associated with it will start to flash red. Select **EDIT** and press the right arrow key once or until the word **SHARE** appears in the menu. Select **SHARE** and then **ENABLE**. Press **OK** twice and press **SETUP** to exit.

Next, configure BLAs to non-registered keys. In this example, line key 6 is unassigned and not registered. To configure a BLA on line key 6, press the **SETUP** key. From the displayed setup menu, select **INSTL** then **KEYS**, and press line key 6. Line key 6 will start flashing red and green. From the menu select **LINE** and enter the **LINE ID** for the key and press **OK**. In this example, **LINE ID** was set to SIP extension 78011. Next, select **AUTHID** and enter the same information that was entered for the **LINE ID** and press **OK**. Lastly, select **PSWD** and enter the password for the specified line ID. Press the right arrow key and select **SHARE** followed by **ENABLE**. Press **OK** If an additional line key with the same information is required, press the **OK** key one more time and select **ADD**. Button 7 will be automatically programmed with the information that was entered on button 6. Continue to press the **ADD** key to program additional keys with the same information. Press the **OK** key and then the **SETUP** key to exit and save. Note that if multiple BLAs are programmed, they need to also be added to the SIP user's button assignments tab configured in System Manager described in **Section 6.2.5**.

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Teo 7810 and 7810 TSG-6 Series IP Phones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the Teo IP phone has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status.
2. Verify basic telephony features by establishing calls between a Teo IP phone and another phone.

9. Conclusion

These Application Notes have described the administration steps required to integrate the Teo 7810 and 7810 TSG-6 Series IP Phones with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Teo IP phones successfully registered with Session Manager and basic and supplementary telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

10. Additional References

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

- [1] *Administering Avaya Aura® Communication Manager*, Release 7.0.1, Issue 2, May 2016, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, Release 7.0.1, Issue 2, May 2016.

The following Teo documentation may be found at www.teotech.com.

- [3] *IP Phone 7810 TSG Series Installations Instructions*, March 2013, Document ID 13-280138.
- [4] *IP Telephone Network Administration Guide*, March 2013, Document ID 13-280132.
- [5] *IP Phone 7810 Installation Instructions*, Document ID 13-280124, March 2013.
- [6] *IP Phone 7810 TSG Series User Guide*, March 2013, Document ID 14-280211.
- [7] *IP Phone 7810 User Guide*, March 2013, Document ID 14.

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