



## **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.

## 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-way conference.
- Extended telephony features using Communication Manager FACs and FNEs for Do Not Disturb, Call Forward, Call Park/Unpark, and Call Pickup.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve messages.
- Bridged Line Appearances on Teo IP phones for other SIP stations.
- 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):

- Do not use the Call Forward on Busy feature supported locally on the Teo IP phone. If the Teo IP phone is busy, Communication Manager will not deliver the call to the Teo IP phone; therefore, the call will not be redirected. Use the Call Forward on Busy feature from Communication Manager using the appropriate FAC or FNE.
- Teo IP phones do not support blind conference. Attended conference is 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.

## 2.3. Support

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

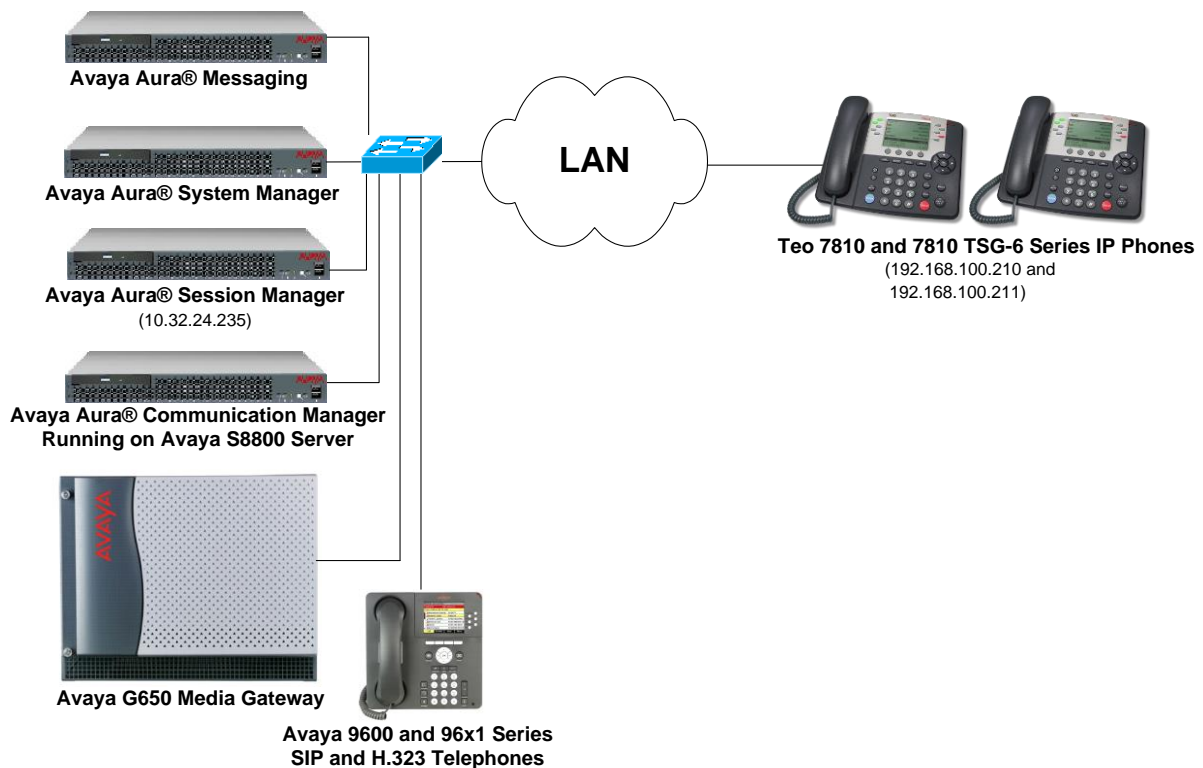
- Phone: 1-800-524-0024
- Website: <http://www.teotech.com/teosupport/support.php>
- Email: [helpdesk@teotech.com](mailto:helpdesk@teotech.com)

### 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 on an Avaya S8800 Server with a G650 Media Gateway.
- 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.
- 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 running on an Avaya S8800 Server with a G650 Media Gateway	6.3 SP 8 (R016x.03.0.124.0 w/Patch 21588)
Avaya Aura® Session Manager	6.3 SP 7 (6.3.7.0.637008)
Avaya Aura® System Manager	6.3.7 Build No. – 6.3.0.8.5682-6.3.8.3204 Software Update Revision No: 6.3.7.7.2275
Avaya Aura® Messaging	6.2 SP 2
Avaya 9600 Series IP Deskphones	S3.230A (H.323) 2.6.13.1 (SIP)
Avaya 96x1 Series IP Deskphones	6.4014 (H.323) 6.4.1.25 (SIP)
Teo 7810 and 7810 TSG-6 Series IP Phones	05.04.20

## 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 11
                                OPTIONAL FEATURES

G3 Version: V16                                     Software Package: Enterprise
Location: 2                                           System ID (SID): 1
Platform: 28                                          Module ID (MID): 1

                                USED
Platform Maximum Ports: 65000 128
Maximum Stations: 41000 31
Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 41000 0
Maximum Off-PBX Telephones - OPS: 41000 18
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 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 G650 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? y
      UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 34
      Audio PHB Value: 46
      Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 7
      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      Silence      Frames      Packet
      Codec      Suppression  Per Pkt    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	<b>Bridged</b>	Location
Extension	Name	Limit	Mode	Allowed	<b>Calls</b>	
78010	OPS	3	both	all	<b>both</b>	

## 6. Configure Avaya Aura® Session Manager

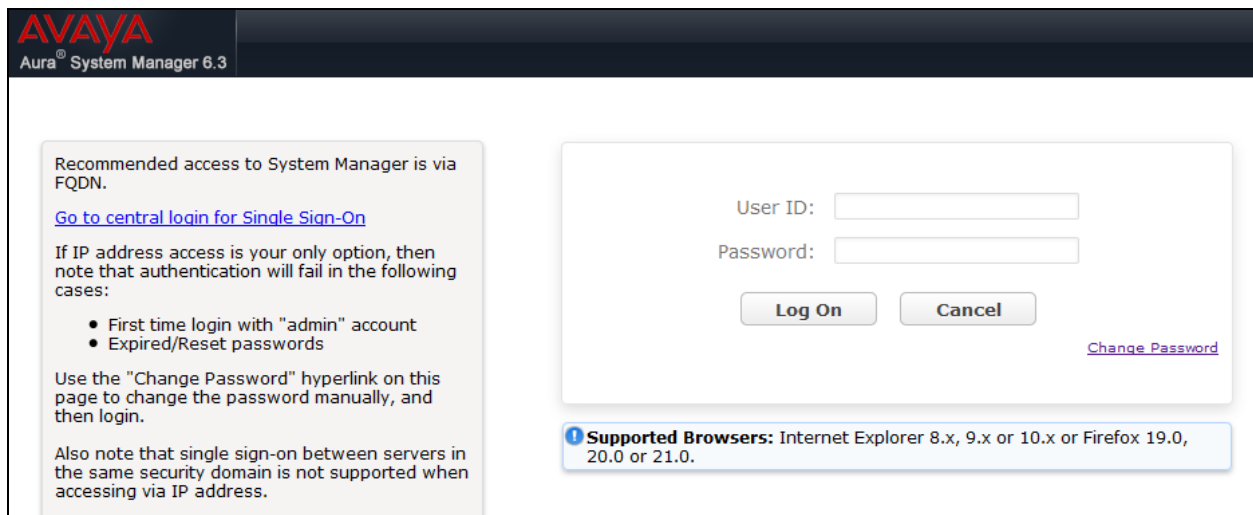
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- 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

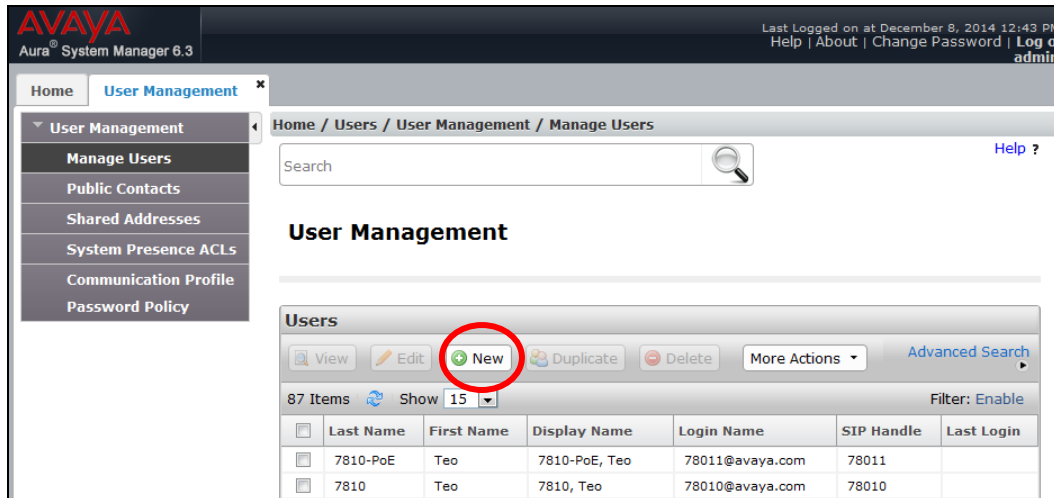
Access the System Manager Web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of the System Manager server. Log in using the appropriate credentials.



The screenshot shows the Avaya Aura System Manager 6.3 login interface. The header includes the Avaya logo and the text "Aura® System Manager 6.3". The main content area is divided into two columns. The left column contains instructional text: "Recommended access to System Manager is via FQDN." followed by a link "Go to central login for Single Sign-On". Below this, it states "If IP address access is your only option, then note that authentication will fail in the following cases:" and lists two bullet points: "First time login with 'admin' account" and "Expired/Reset passwords". It then instructs to use the "Change Password" hyperlink to change the password manually. At the bottom of this column, it notes that single sign-on between servers in the same security domain is not supported when accessing via IP address. The right column features a login form with fields for "User ID:" and "Password:", "Log On" and "Cancel" buttons, and a "Change Password" link. At the bottom of the right column, a blue box lists "Supported Browsers: Internet Explorer 8.x, 9.x or 10.x or Firefox 19.0, 20.0 or 21.0."

## 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.

The screenshot shows the 'New User Profile' screen in the Avaya Aura System Manager 6.3 interface. The 'Identity' tab is selected. The form includes fields for 'Last Name', 'First Name', 'Login Name', 'Password', and 'Confirm Password'. The 'Login Name' field is pre-filled with '78010@avaya.com'. The 'Authentication Type' is set to 'Basic'. The 'User Provisioning Rule' is set to 'User Provisioning Rule:'. The 'Identity' section is expanded, showing the 'Last Name' and 'First Name' fields. The 'Last Name' field is pre-filled with '7810' and the 'First Name' field is pre-filled with 'Teo'. The 'Login Name' field is pre-filled with '78010@avaya.com'. The 'Password' and 'Confirm Password' fields are masked with dots.

**New User Profile**

Commit & Continue | Commit | Cancel

**Identity** \* | Communication Profile | Membership | Contacts

**User Provisioning Rule**

User Provisioning Rule: [User Provisioning Rule]

**Identity**

\* Last Name: 7810

Last Name (Latin Translation): 7810

\* First Name: Teo

First Name (Latin Translation): Teo

Middle Name: [Middle Name]

Description: [Description]

\* Login Name: 78010@avaya.com

\* Authentication Type: Basic

Password: [Password]

Confirm Password: [Confirm Password]

### 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 6.3 interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 6.3', and a user status bar indicating 'Last Logged on at December 8, 2014 12:43 PM' with links for 'Help', 'About', 'Change Password', and 'Log off admin'. The left sidebar contains a 'User Management' menu with options: 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence ACLs', 'Communication Profile', and 'Password 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 fields for 'Communication Profile Password' and 'Confirm Password', both masked with dots. Action buttons 'Commit & Continue', 'Commit', and 'Cancel' are at the top right. A breadcrumb trail reads 'Home / Users / User Management / Manage Users'.

### 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' form. At the top, there are 'New', 'Edit', and 'Delete' buttons. Below is a table with columns 'Type', 'Handle', and 'Domain'. The table is empty, displaying 'No Records found'. Below the table, the 'Type' is set to 'Avaya SIP' in a dropdown menu. The 'Fully Qualified Address' field is populated with '78010' and '@ avaya.com' in a dropdown menu. '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 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

devcon-asm

Primary	Secondary	Maximum
25	0	25

Secondary Session Manager

(None)

Survivability Server

(None)

Max. Simultaneous Devices

1

Block New Registration When Maximum Registrations Active?

☐

**Application Sequences**

Origination Sequence

DEVCON13 App Sequence

Termination Sequence

DEVCON13 App Sequence

**Call Routing Settings**

\* Home Location

BR-DevConnect

Conference Factory Set

(None)

## 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\_6\_3*. 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 devcon13-CM-ES ▼

\* Profile Type Endpoint ▼

Use Existing Endpoints ☐

\* Extension 78010 Endpoint Editor

\* Template 9600SIP\_DEFAULT\_CM\_6\_3 ▼

Set Type 9600SIP

Security Code

Port IP

Voice Mail Number

Preferred Handle (None) ▼

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 ☒

In the **CM Endpoint Profile** sub-section, click the **Endpoint Editor** button to display the page below. In the **General Options** tab, specify that 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 20 was used.

The screenshot shows the 'Edit Endpoint' page in the Avaya Aura System Manager 6.3. The 'General Options' tab is selected. The form contains the following fields:

System	devcon13-CM-ES	Extension	78010
Template	9600SIP_DEFAULT_CM_6_3	Set Type	9600SIP
Port	IP	Security Code	
Name	7810,Teo		

* Class of Restriction (COR)	1	* Class Of Service (COS)	1
* Emergency Location Ext	78010	* Message Lamp Ext.	78010
* Tenant Number	1	Type of 3PCC Enabled	None
* SIP Trunk	Q.aar	Coverage Path 2	
Coverage Path 1	20	Localized Display Name	7810,Teo
Lock Message	<input type="checkbox"/>		

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' page in the Avaya Aura System Manager 6.3, with the 'Feature Options' tab selected. The form contains the following fields:

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

In the **Button Assignments** tab, configure a button as a Bridged Line Appearance (BLA) if desired. Configure the button as *brdge-appr* and specify the call appearance button number and extension of the station for which the BLA is assigned. Click **Done** when complete, followed by **Commit** on the previous page.

General Options (G) *		Feature Options (F)		Site Data (S)		Abbreviated Call Dialing (A)		Enhanced Call Fwd (E)	
Button Assignment (B)		Group Membership (M)							

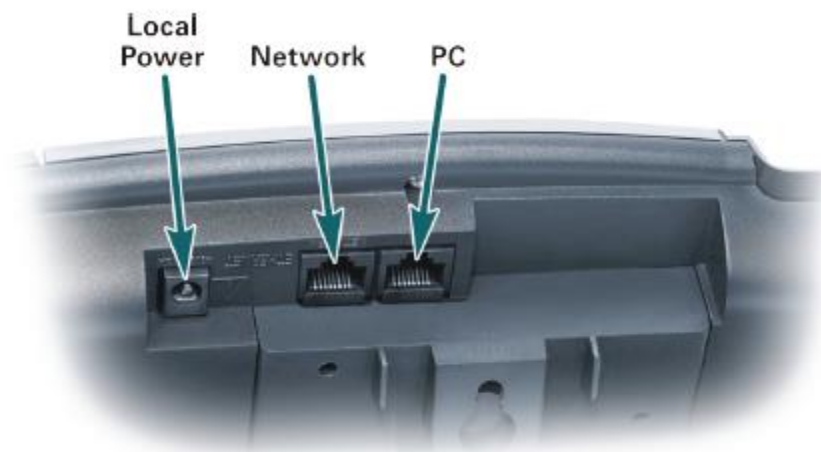
Main Buttons		Feature Buttons		Expansion Module	
1	call-appr ▼				
2	call-appr ▼				
3	call-appr ▼				
4	None ▼				
5	None ▼				
6	brdg-appr ▼	Button	1	Ext	78011
7	None ▼				
8	None ▼				

## 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.



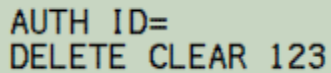
### 7.2. SIP Configuration

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.

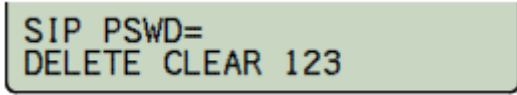
LINE ID=  
DELETE CLEAR 123

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.



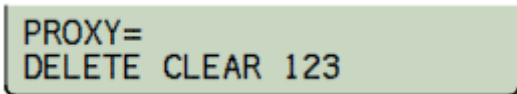
AUTH ID=  
DELETE CLEAR 123

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.



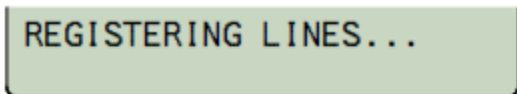
SIP PSWD=  
DELETE CLEAR 123

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, *10.32.24.235* in the reference configuration. Press the **OK** on the phone when the entry is completed.

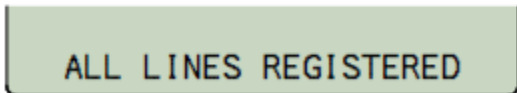


PROXY=  
DELETE CLEAR 123

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...

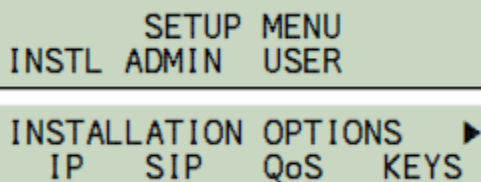


ALL LINES REGISTERED

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**. 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](http://support.avaya.com).

- [1] *Administering Avaya Aura® Communication Manager*, Release 6.3, Issue 9, October 2013, Document Number 03-300509.
- [2] *Administering Avaya Aura® Session Manager*, Release 6.3, Issue 3, October 2013.

The following Teo documentation may be found at [www.teotech.com](http://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.

---

**©2015 Avaya Inc. All Rights Reserved.**

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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](mailto:devconnect@avaya.com).